DINSTAR

MTG200 Trunk Gateway User Manual V3.0

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Preface

Welcome

Thanks for choosing **MTG200 Trunk Gateway**! We hope you will make optimum use of this flexible, rich-feature trunk gateway. Please read this document carefully before install the gateway.

About This Manual

This manual provides information about the introduction of the gateway, and about how to install, configure or use the gateway.

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

This manual is written with reference to the default configurations of the **MTG200 Trunk Gateway**.

Intended Audience

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

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

Table of Contents

1 Product Description

1.1 Overview

MTG200 E1/T1 trunk gateway is a multi-functional trunk gateway designed for large enterprises, telecom operators, call centers and providers of value-added services. It supports a range of signaling protocols, realizing the interconversion between SIP and traditional signals like SS7, PRI and NO.1. It supports multiple codec methods such as G.711, G.723, G.729 and iLBC, offers signal encryption technology and smart voice recognition technology, and improves the utilizing efficiency of trucking resources while ensuring voice quality.

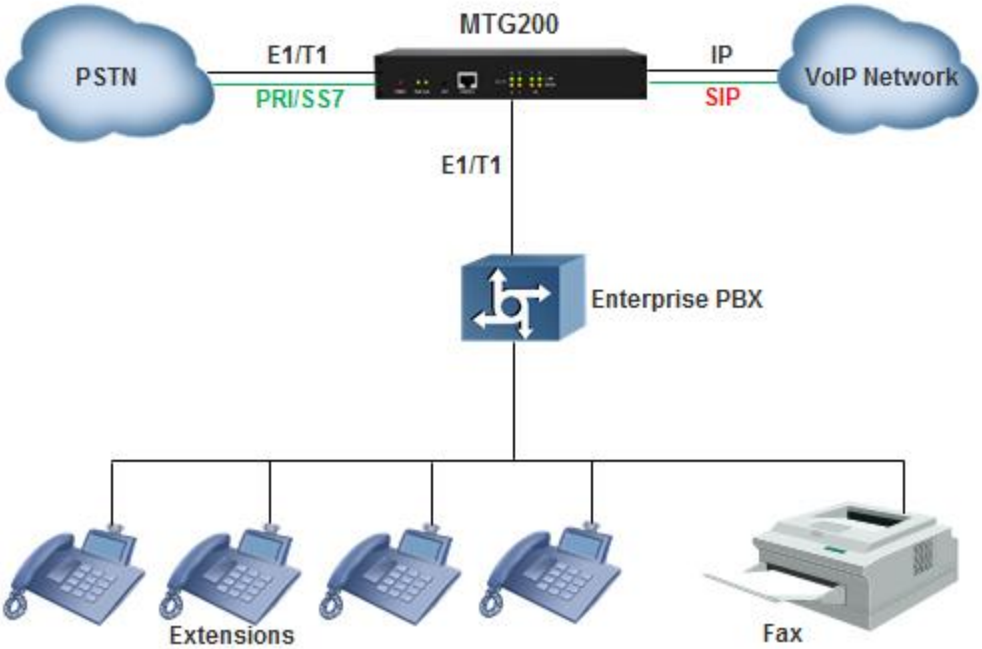
Compared with similar gateways, MTG200 trunk gateway has more advantages in terms of performance, system reliability and interoperability. Its high-efficient design and strong DSP processor ensure the interconversion of PCM voice signal and IP packets, although the gateway has been fully loaded. It can be connected with multiple devices such as softswitches, PBX and those servers equipped with digital trunk boards.

MTG200 trunk gateway has two models:

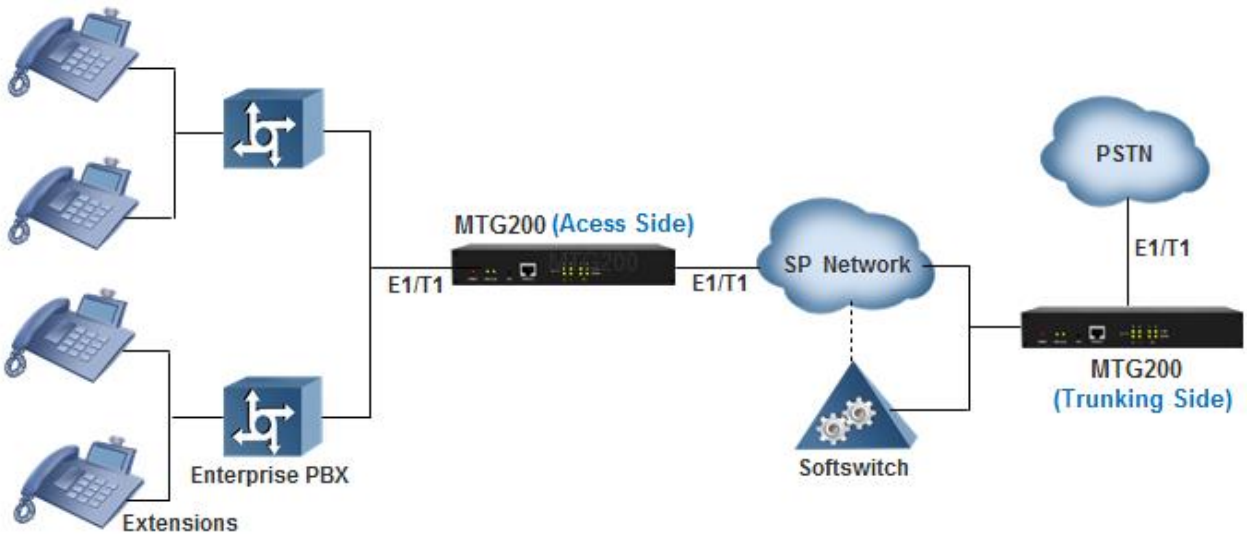
- ✓ MTG200-1E1/T1
- ✓ MTG200-2E1/T1

1.2 Application Scenario

The application scenario for enterprises is shown as follows:



The application scenario for services providers is shown as follows:

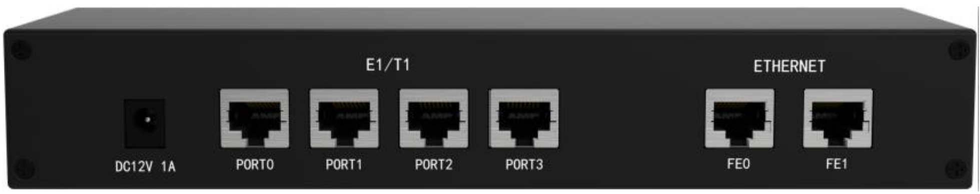


1.3 Product Appearance

1.3.1 Front View of MTG200

Indicator	Function	Color	Status
POWER	Power indicator	Green	Off: Power is off
			On: Power is on
RUN	Running indicator	Green	Slow flashing: the gateway is running normally.
			No flashing/Fast flashing: the gateway is running abnormally.
ALM	Alarm indicator	Yellow	Off: the gateway functions well.
			On: the gateway malfunctions.
RST	Reset button (it is used to reset the gateway)		
CONSOLE	RJ45, RS232, 115200bps, it is used to carry out maintenance-related configurations.		
E1/T1	Indicating the connection state of device E1/T1.	Green	Off：E1/T1 port is not connected
			On: E1/T1 port connection and sending/ receiving message are normal
			Flash:E1/T1 port connection fails
LINK	Indicator for network link	Green	Off: the gateway is normally connected to network.
			On: the network is not connected to network or the connection is improper. 0 refers to FE0 while 1 refers to FE1.
SPEED	Indicator for network speed	Yellow	Off: network bandwidth is 10Mbps.
			On: network bandwidth is 100Mbps.

1.3.2 Rear View of MTG200



Interface	Description
POWER	Connected to the power adapter, power supply: 110~240VAC, 50~60HZ, Output (12VDC, 1.0A)
Port0-Port1	E1/T1 ports
FE0	**Ethernet Interface for Services**, standard 10/100BASE-TX Ethernet interface. Default IP address is 192.168.1.111, default subnet mask is 255.255.255.0
FE1	**Ethernet Interface for Management**, standard 10/100BASE-TX Ethernet interfaces, Default IP address is 192.168.11.1, default subnet mask is 255.255.255.0

1.4 Functions and Features

1.4.1 Key Features

- 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 /SIP-T/R2/PRI/SS7 protocol
- NAT Traversing (STUN)
- Hypertext Transfer Protocol (HTTP)
- ITU-T G.711A-Law/U-Law, G.723.1, G.729AB, iLBC(optional)
- Domain Name System (DNS)
- Dynamic host configuration protocol (DHCP)

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: 1/2
- Interface Type: RJ48(Impedance 120Ω）
- Ethernet Interface:

  FE1: standard 10/100BASE-TX Ethernet interface, FE0: standard 10/100BASE-TX Ethernet interface
- Console Port : 1* RS232, 115200bps

1.4.5 Software Features

- Local/Transparent Ringback 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

- Power Supply: 100-240 VAC, 50-60Hz, Output (12VDC, 1.0A)
- Maximum Power Consumption: 10W
- Operating Temperature: 0 ℃ ～45 ℃
- Storage Temperature: -20 ℃ ~80 ℃
- Humidity: 10%-90%, Non-Condensing
- Dimensions (W/D/H): 225×150×38mm
- Net Weight: 0.75kg

2 Quick Installation

2.1 Preparations before Installation

2.1.1 Attentions for Installation

The attentions for installing MTG200 include:

- To guarantee MTG200 works normally and to lengthen the service life of the device, the humidity of the equipment room where MTG200 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 MTG200 should be 100 ~240V AC, and its socket is a three-pin socket which should be

grounded well;

- Please wear ESD wrist strap when installing MTG200;

2.1.2 Preparations about Installation Site

- Equipment Cabinet

  Ensure the cabinet is well-ventilated and strong enough to bear the weight of MTG200. It's required that the width of the cabinet should be 482.6mm (19 inches).

- IP Network

  Ensure Ethernet PBX or router under IP network has been prepared, since MTG200 is connected to the IP network through the standard 10/100 network port.

- Socket

  Ensure the socket of MTG200 is a three-pin socket and power supply is grounded well.

2.1.3 Installation Tools

- Screwdriver
- Anti-static wrist strap
- Network cables, power cable, telephone wires
- Hub, telephone, fax, and PBX
- Terminal (it can be a PC)

2.1.4 Unpacking

Open the packing container to check whether the MTG200 device and all accessories have been in it:

- One MTG200 device
- One Power Adapter
- One network cable
- E1/T1 cables (the number of the cables is the same with that of E1/T1 ports)
- Serial console cable

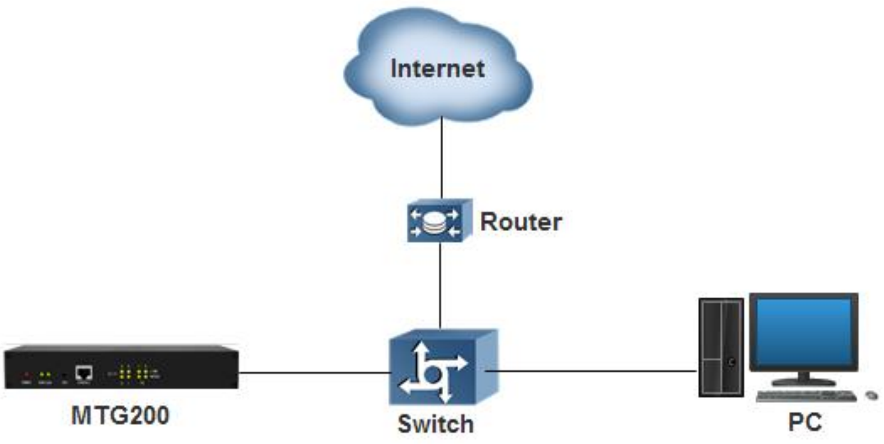
2.2 Installation of MTG200

2.2.1 Connect MTG200 to Network

MTG200 has two network ports, namely the Ethernet port for services (FE0) and the Ethernet port for management (FE1). It is advised to connect FE0 to the IP network.

Both FE1 and FE0 can be used to carry out management on MTG200, but only FE0 is put in use generally. FE1 is used when there is a need to separate the management on MTG200 from service processing.

Connect MTG200's FE0 port to the network according to the following figure:

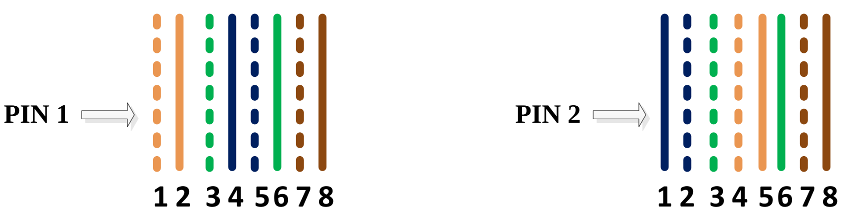


2.2.2 Connect MTG200 to PSTN

Connect one end of E1/T1 cable to one of the E1/T1 ports of MTG200, and then connect the other end an exchanger or a PBX under PSTN.

2.3 Wire Sequence of RJ48 (E1/T1) Cable

The E1/T1 ports of MTG200 trunk gateway are connected with RJ48 cables. A RJ48 cable has two PINs, and the wire sequence of each PIN is shown as follows:

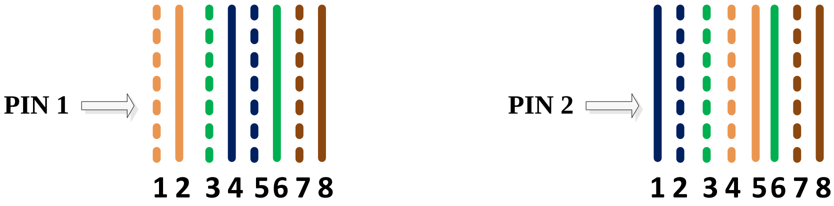


Wire sequence of PIN1: orange & white, orange, green & white, blue, blue & white, green, brown & white, brown.

Wire sequence of PIN2: blue, blue & white, green & white, orange & white, orange, green, brown & white, brown.

2.3.1 How to make RJ48 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.
2. Sequence the wires of the cable according to the following figure.



3. Put the wires into two PINs of RJ－48 joint according to the abovementioned sequence of the wires.
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 MTG200 device, and users have no need to make RJ－48 joints by themselves.

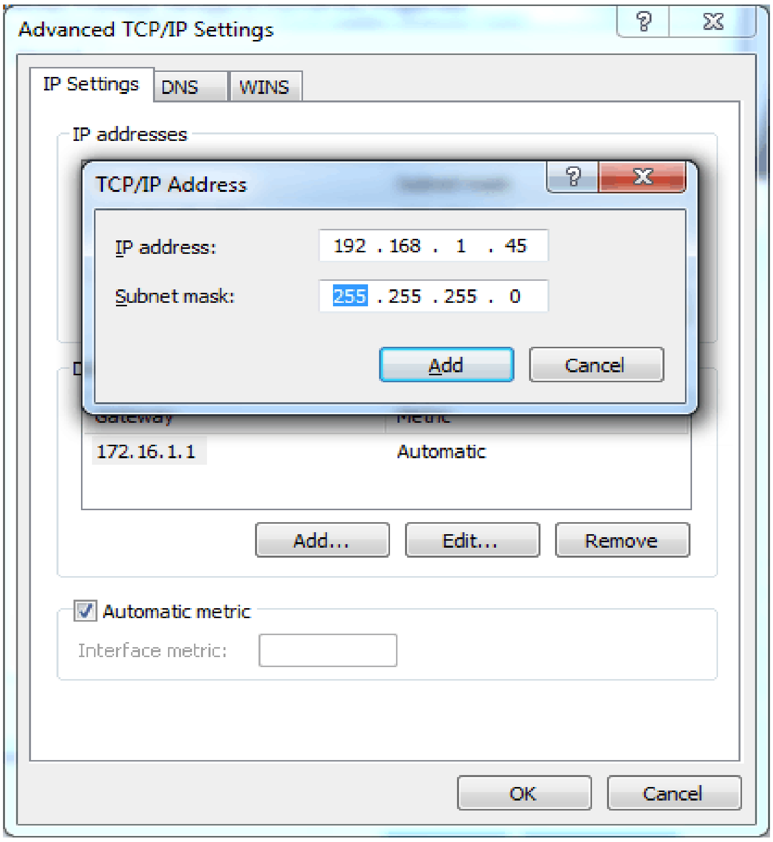
3 Basic Operation

3.1 Configuration of IP Address

The default IP address of FE1 (network port for services) is 192.168.11.1, while that of FE0 (network port for management) is 192.168.1.111.When FE0 is in use, it's required that the IP address of FE0 and the IP address of PC are at the same network segment.

1. Connect the FE0 port of MTG200 to a PC by using a network cable.

2. Open the **TCP/IPv4 Settings** interface on the PC, 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 MTG200 trunk gateway provides a standard RJ48 console port. Users can log in the MTG200 to carry out maintenance-related configurations through the console port.

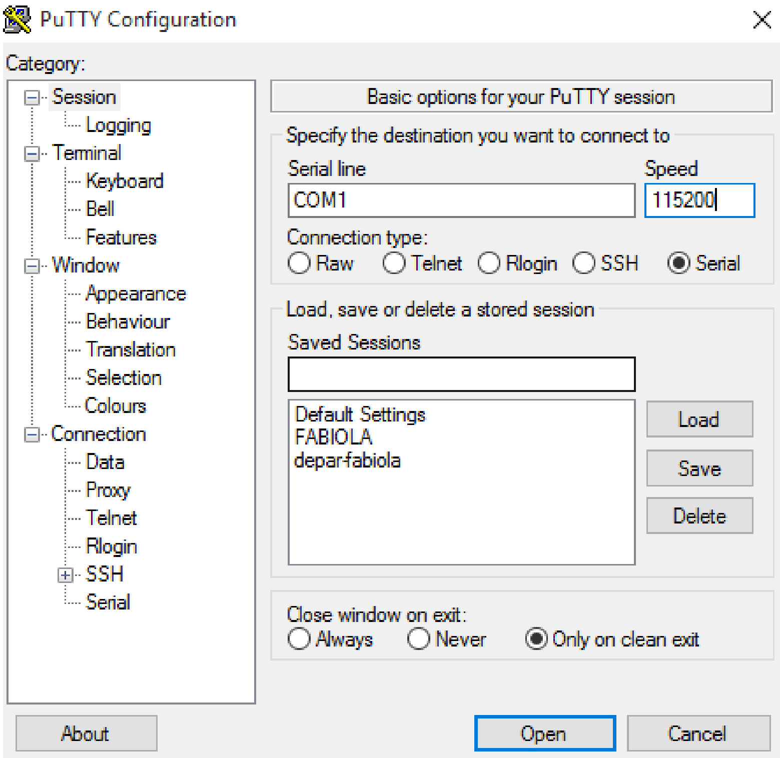
3.2.1 Example: Log in MTG200 via Console Port

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 tool to connect the serial cable to the PC.

Step 3: Connect the M port of the serial cable to the console port of MTG200.



Step 4: Conduct configurations on a login software.

Herein we take the PuTTY sofeware as an example. Detailed configurations are as follows:

(COM1 is an example. Please enter correct name of serial line according to actual conditions.)

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

```
Welcome to Command System!
Username:admin
Password:*****
ROS>en
ROS#
```

Enter username and password, which are the same with the username and password to log in the Web interface of MTG200. And then you will see a linux platform where you can carry out maintanance-related configurations.

Note: For commands to query MTG200 information, make reference to Chapter 6.

3.3 Query IP

If you have changed the default IP address of FE1 or FE0 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 MTG200 with a PC;
2. Modify the baud rate to 115200;
3. Click **OK**, and then enter 'show int', and the IP address of FE1 or FE0 of MTG200 will be displayed.

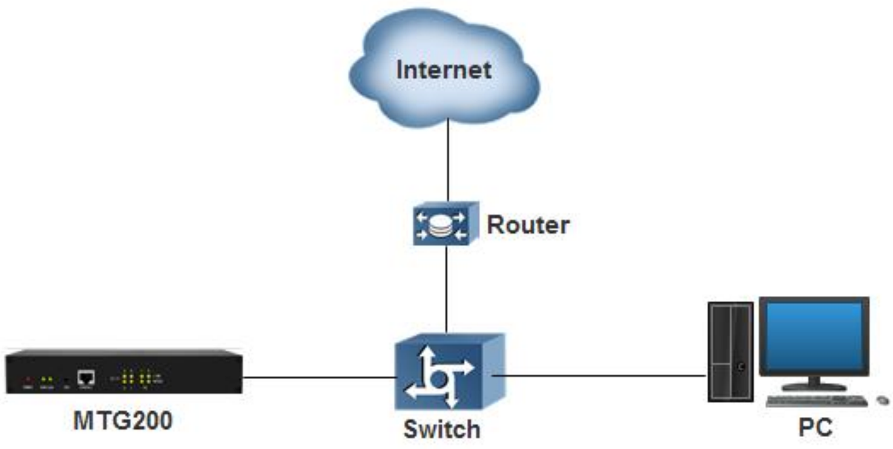
```
Welcome to Command System!
Username:admin
Password:*****
ROS>en
ROS#
ROS#sho int
Fast-ethernet eth0 is UP
Internet Address is owned: 172.16.51.76, Mask:255.255.0.0, MTU:1300
Hardware address is: 00:24:D5:B7:A3:10

Fast-ethernet eth2 is DOWN
Internet Address is owned: 192.168.11.1, Mask:255.255.255.0, MTU:1500
Hardware address is: 00:12:34:56:38:01

ROS#
```

4 Configurations on Web Interface

4.1 How to Log in Web Interface

4.1.1 Network Connection

Connect MTG200 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 FE0 port of MTG200 device. The format of PC IP is 192.168.1.XXX, since the default IP of FE0 port is 192.168.1.111.

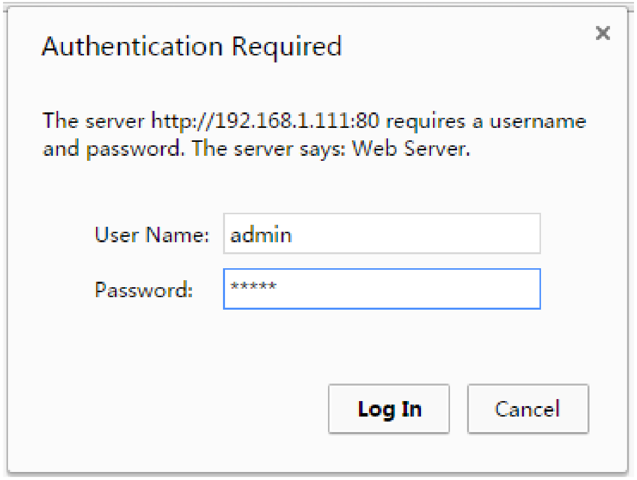
Check the connectivity between the PC and the MTG200. Click **Start → Run** of PC and enter cmd to execute 'ping 192.168.1.111' to check whether the IP address of the MTG200 runs normally.

4.1.3 Log in Web Interface

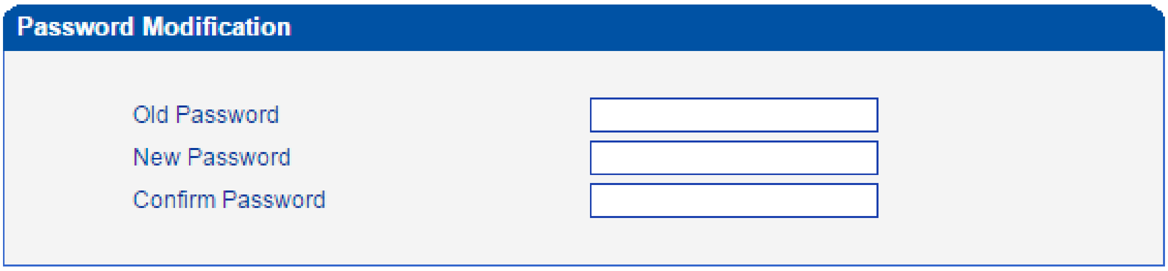
Open a web browser and enter the IP address of FE0 of MTG200 (the default IP is 192.168.1.111). Then the login GUI will be displayed. Both the default username and password are admin.

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

Login GUI:

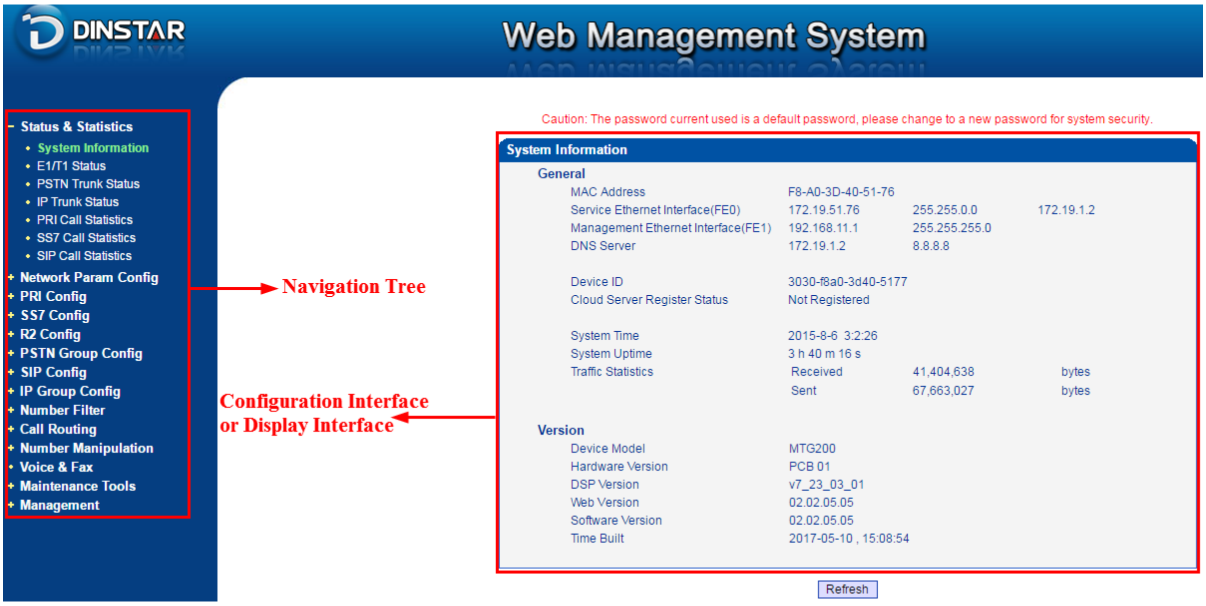


Password Modification Interface:



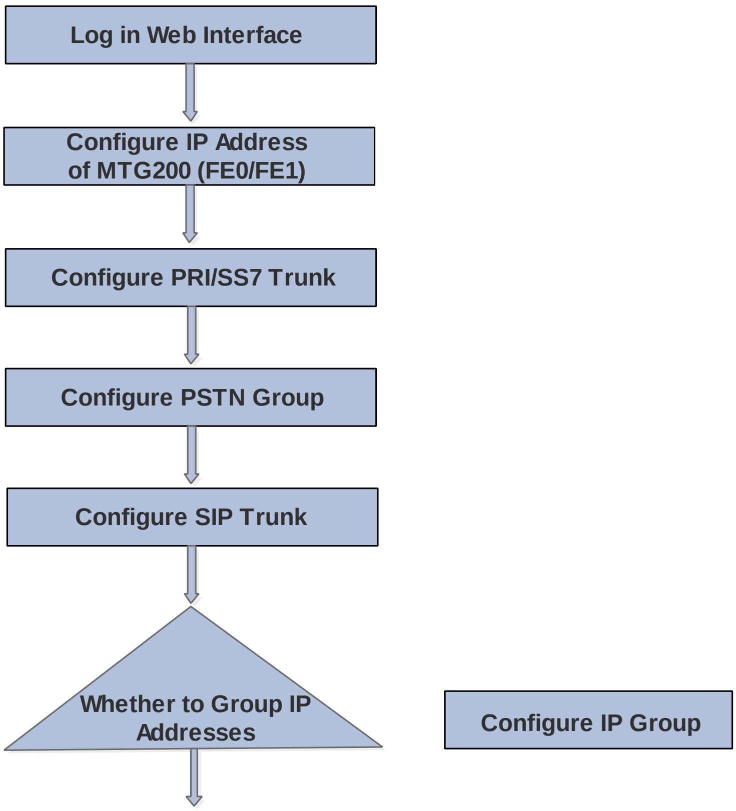
4.2 Introduction to Web Interface

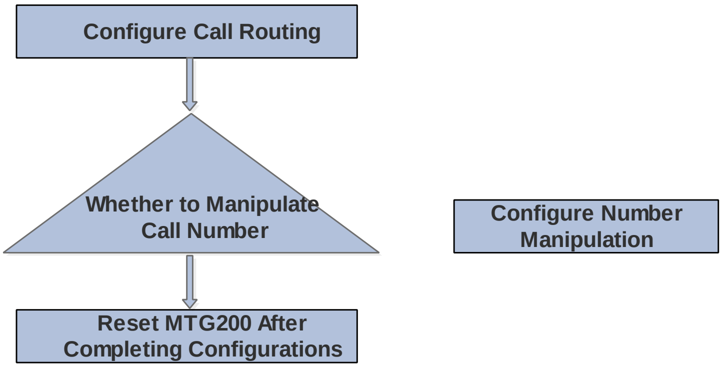
The Web Interface of the MTG200 consists of the navigation tree and detailed configuration interfaces. Click a node of the navigation tree, and you will see a detailed display interface or configuration interface:



4.3 Configuration Flows

The following is the configuration flows of MTG200:

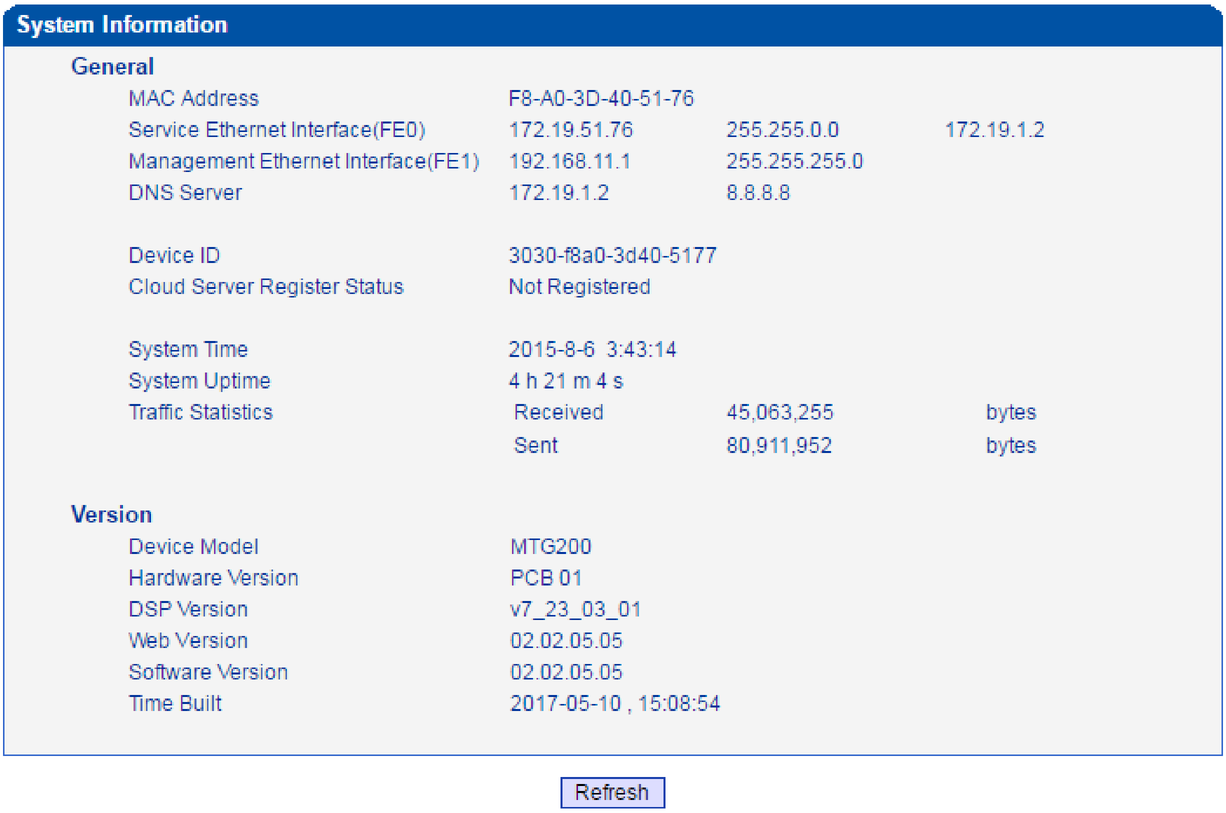




4.4 Status & Statistics

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, 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.

E1/T1 Port Status

Port No.	0	1	2	3
Physis Status				

NOTES: Actived, Disable, LOS Alarm, RAI Alarm, AIS Alarm, ISDN/SS7 Signal Alarm

E1/T1 Channel Status

Channel No.	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
Port 0																																
Port 1																																
Port 2																																
Port 3																																

Status	Frame-Sync	Idle	Signal	Busy	Fault	Disable	L-blocked	R-blocked	B-blocked	Looped
Color										
Totalize	0	0	0	0	32	0	0	0	0	0

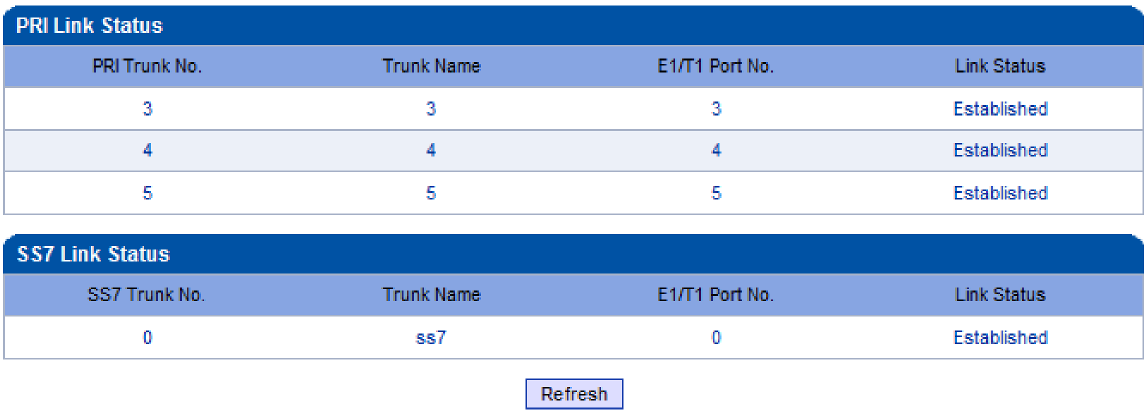
NOTES: L-Blocked -- Local Blocked, R-Blocked -- Remote Blocked, B-Blocked -- Both Sides Blocked

E1/T1 Port Status	Actived	Both physical connection and signal connection of the E1/T1 port are normal, and the port is activated.
	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 MTG200.
	AIS Alarm	AIS (Alarm Indication Signal) is an alarm raised by MTG200, 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.
	Frame-Sync	Frame synchronization

E1/T1 Channel Status	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.

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
3	3	3	Established
4	4	4	Established
5	5	5	Established

SS7 Link Status

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

1）PRI Link Status

PRI Trunk No.	The number of PRI trunk; each trunk corresponds to a PRI link
Trunk Name	Used to identify the trunk

E1/T1Port No	Indicate the E1/T1 port occupied by the PRI trunk.
Link Status	Indicate whether the PRI link is established or not.

2）SS7 Link Status

SS7 Trunk No.	The number of SS7 trunk; each trunk corresponds to a SS7 link
Trunk Name	Used to identify the name of the trunk
E1/T1 Port No	Indicate the E1/T1 line occupied by the SS7 trunk.
Link Status	Indicate whether the SS7 link is established or not

4.4.3 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
0	11.37	Peer	UDP	---	IP Address	Established
1	20.160	Peer	UDP	---	IP Address	Established
2	99.15	Peer	UDP	---	IP Address	Established
3	vos	Access	UDP	09902	IP Address	Established

Refresh

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.

4.4.4 PRI Call Statistics

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

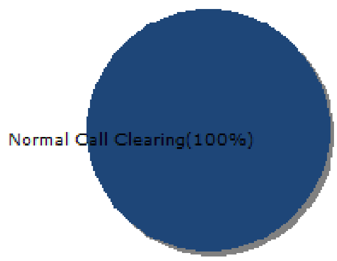
ASR (Answer-seizure Ratio): is a call success rate, which reflects the percentage of answered telephone calls with respect to the total call volume. ASR = answered call/total attempts of calls.

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
3	3	0	0	100%
4	4	0	0	100%
5	5	0	0	100%

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



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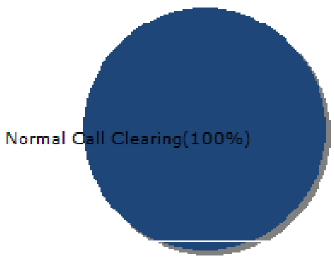
PRI Trunk No	The number of the PRI trunk
Trunk Name	The name used to identify the PRI trunk
Current Calls	The number of lines that are being called currently
Accumulated Calls	Total number of calls that have been gone through this PRI trunk since the gateway begins to run.
ASR	The percent of answered calls in total calls. ASR = answered call/total attempts of calls.

4.4.5 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
0	ss7	0	0	100%

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



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4.4.6 SIP Call Statistics

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

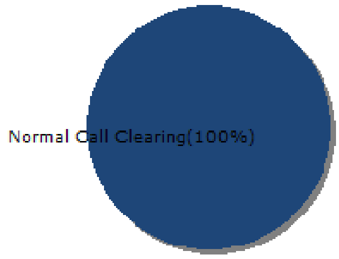
SIP Trunk No.	The number of the SIP trunk
Trunk Name	The name used to describe the SIP trunk
Current Calls	The number of lines that are being called currently

SIP Trunk Call Statistics

SIP Trunk No.	Trunk Name	Current Calls	Accumulated Calls	ASR
0	11.37	0	0	100%
1	20.160	0	0	100%
2	99.15	0	0	100%
3	vos	0	0	100%

Release Cause Statistics

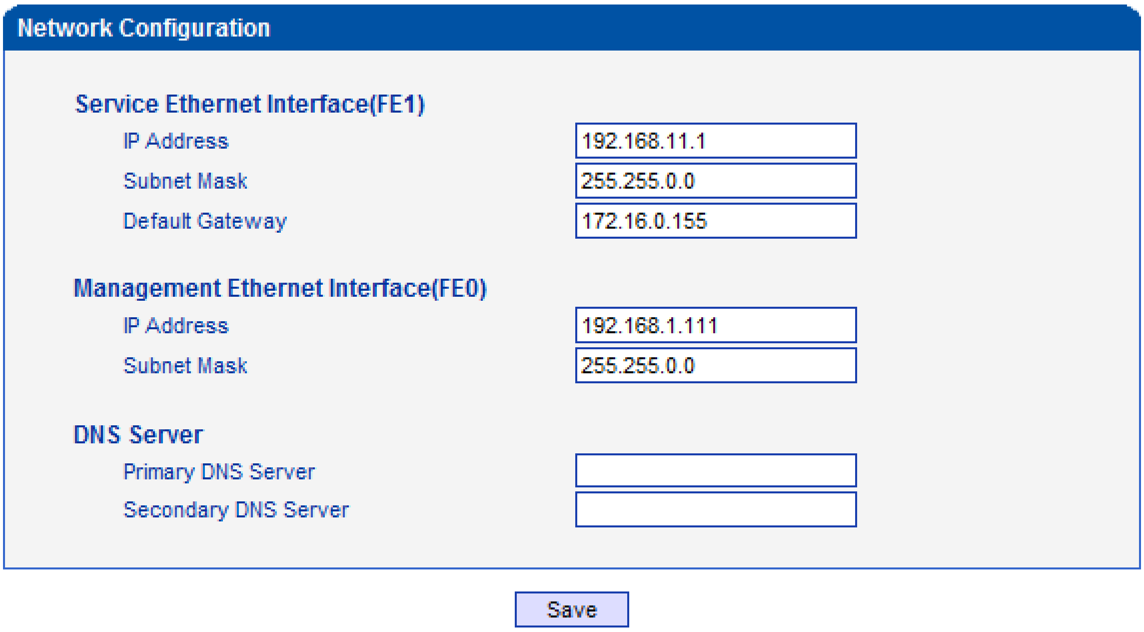
Normal Call Clearing	0
Temporarily Unavailable	0
Forbidden	0
Not Found	0
Busy Here	0
Internal Server Error	0
Server Time Out	0
Service Unavailable	0
Others	0



Refresh

4.5 Network

Generally, it's necessary to modify the default IP address of FE0 or FE1 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 FE0 or FE1. After completing the configurations, you need to restart the MTG200 device for the changes to take effect.



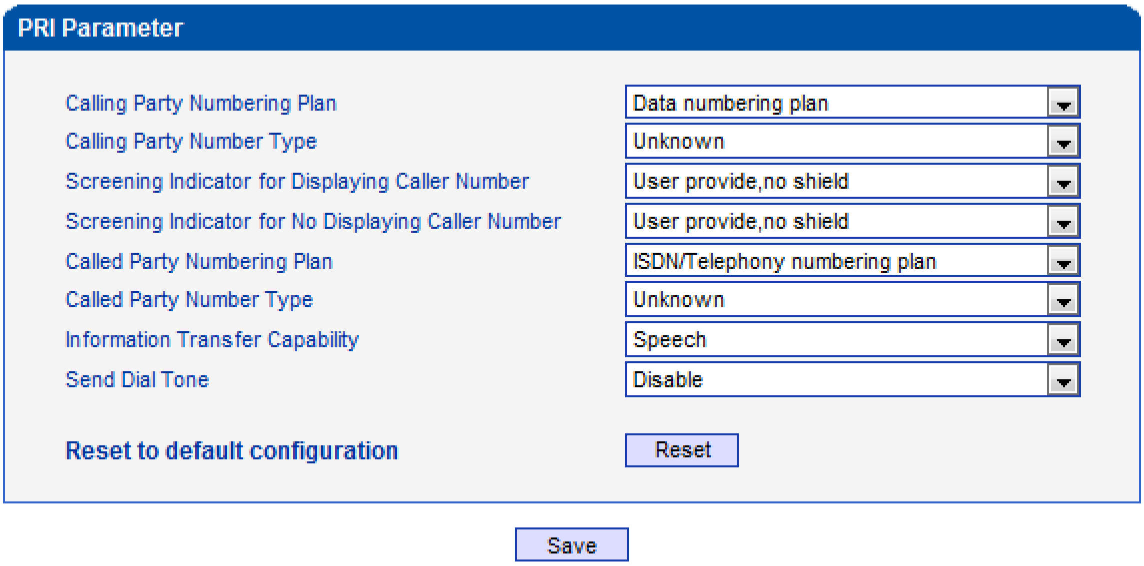
Belong to	Parameter	Explanation
FE1 Port	IP Address	The IP address of FE1, default value is 192.168.11.1
	Subnet Mask	Subnet mask of FE1, default: 255.255.0.0
	Default Gateway	The IP address of network gateway
FE1 Port	IP Address	The IP address of FE0, default value is 192.168.1.111
	Subnet Mask	Subnet mask of FE0, default: 255.255.0.0
DNS Server	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 FE1 and that of FE0 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.

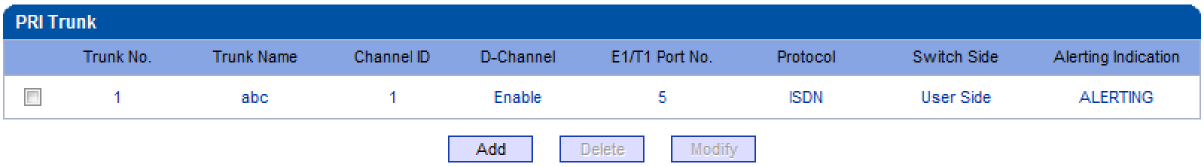


Parameter	Options
Calling Party Numbering Plan	Include 'ISDN/Telephony Numbering Plan', 'Data Numbering Plan', 'Telex Numbering Plan', 'National Standard Numbering Plan', 'Private Numbering Plan' and 'Unknown'.
Calling Party Number Type	Include 'International Number', 'National Number', 'Network Specific 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 on 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.

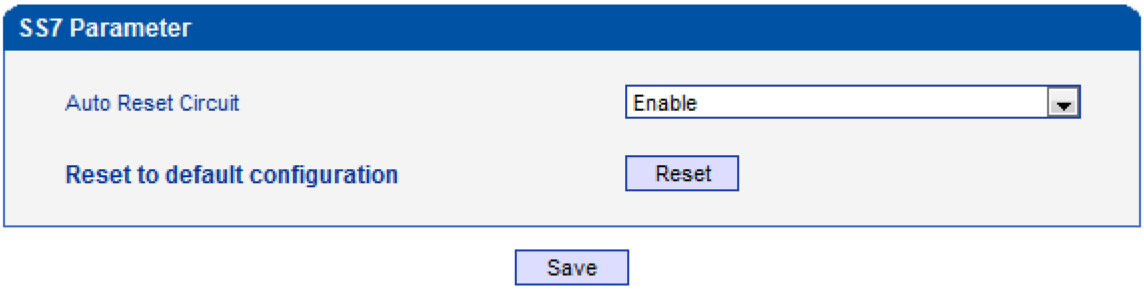


Parameter	Explanation
Trunk No.	Trunk No. starts from 1 to 7, it means you can establish 7 PRI trunks at most. The trunk No. is decided by the No. of the E1/T1 port linked to the trunk. But if D-channel is not enabled for a trunk, the No. of the trunk must be the same with a trunk under which D-channel 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

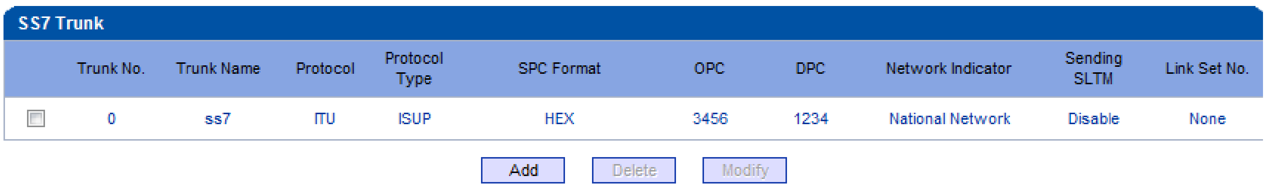
Whether the SS7 function is enabled or not is determined by the license.

4.7.1 SS7 Parameter



4.7.2 SS7 Trunk

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.

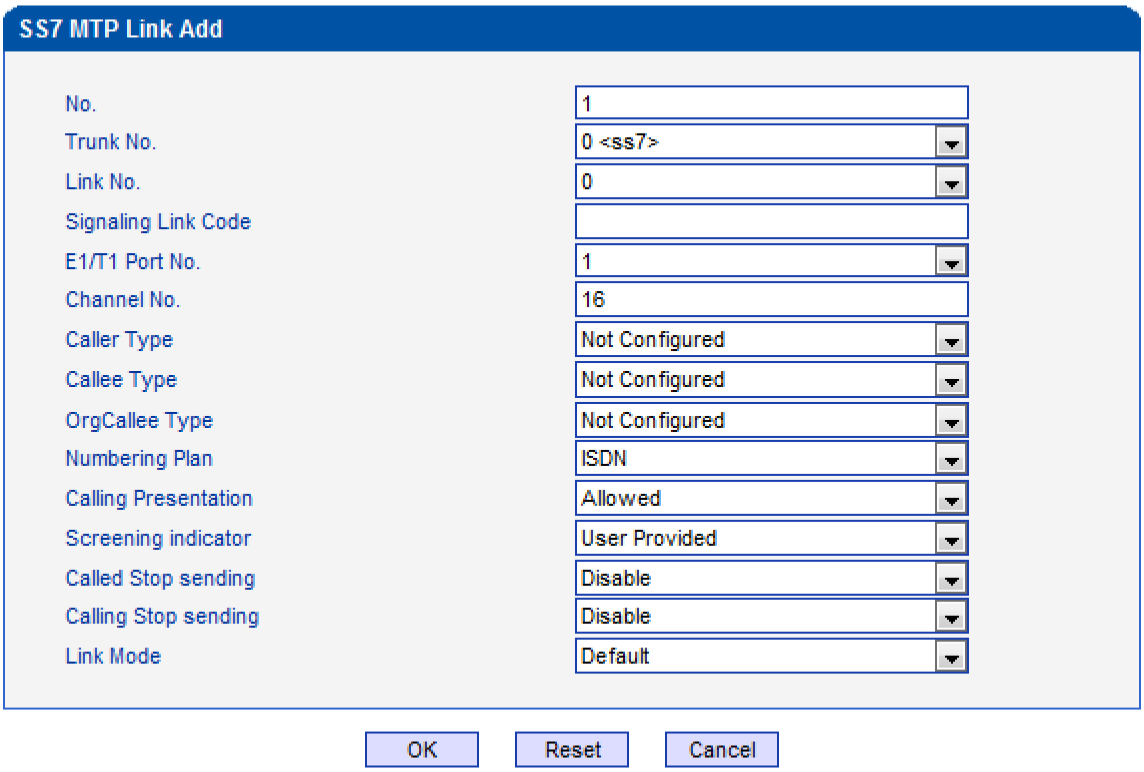


Parameter	**Explanation**
Trunk No.	The No. of the SS7 trunk. Generally, one SS7 trunk is for one DPC.
Trunk Name	The trunk name is used to distinguish the trunk from other trunks.
Protocol	SPC types: ITU-T (14 bit), ANSI (24 bit), ITU-CHINA (24 bit) SPC: Signaling Point Code
Protocol Type	ISUP (ISDN 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 MTG200, 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.

4.7.3 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 SS7 trunk. Two links can be established for an SS7 trunk at maximum.



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.
Called Stop Sending	'Stop Sending' is an end mark. If 'Yes' is selected for 'Called Stop Sending', it means there will be an end mark following the called number.

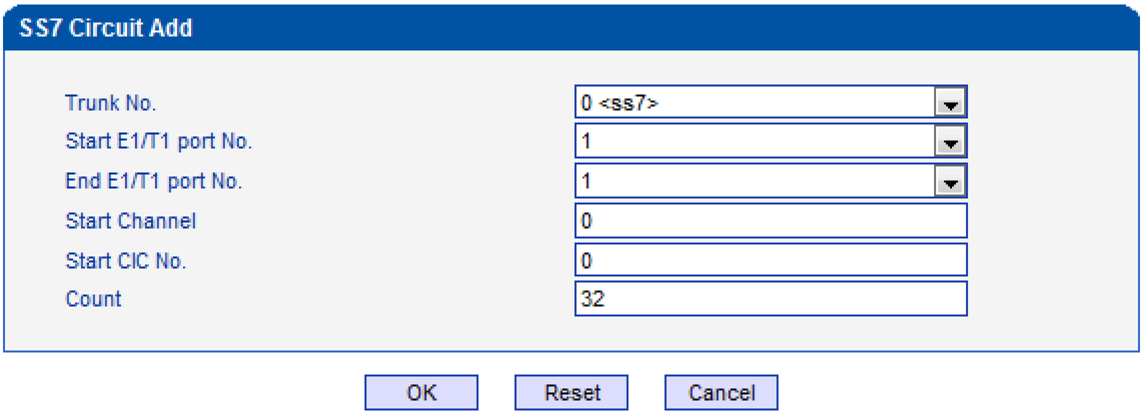
4.7.4 SS7 CIC

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.

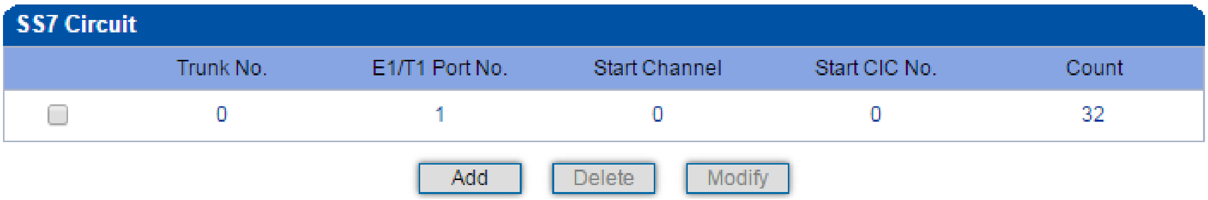
Step 2: Select a trunk and an E1/T1 port. (Trunk 0 and Port 1 are taken as example in the following figure



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.



SS7 Circuit

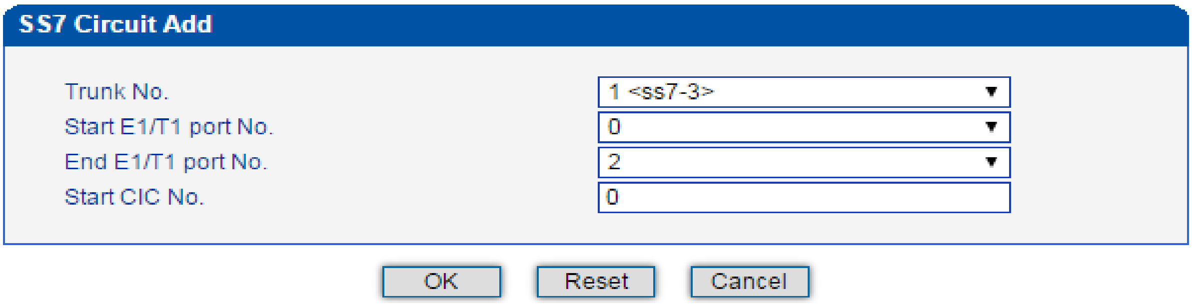
	Trunk No.	E1/T1 Port No.	Start Channel	Start CIC No.	Count
☐	0	1	0	0	32

Add Delete Modify

➢ 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.



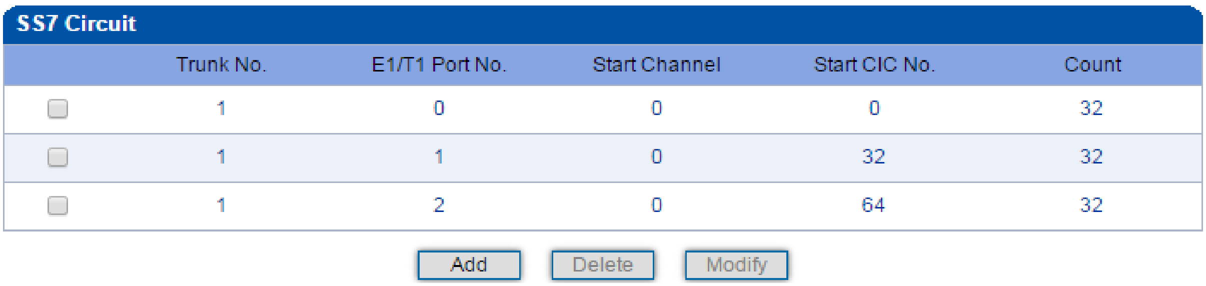
SS7 Circuit Add

Trunk No.	1 <ss7-3>
Start E1/T1 port No.	0
End E1/T1 port No.	2
Start CIC No.	0

OK Reset Cancel

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.



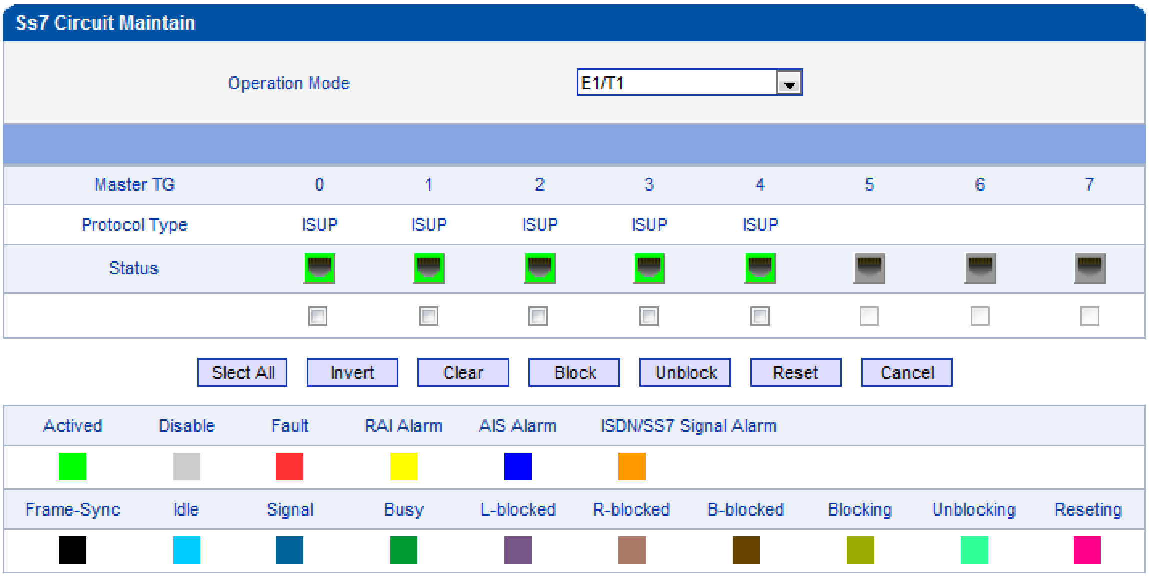
SS7 Circuit

	Trunk No.	E1/T1 Port No.	Start Channel	Start CIC No.	Count
☐	1	0	0	0	32
☐	1	1	0	32	32
☐	1	2	0	64	32

Add Delete Modify

4.7.5 SS7 CIC Maintain

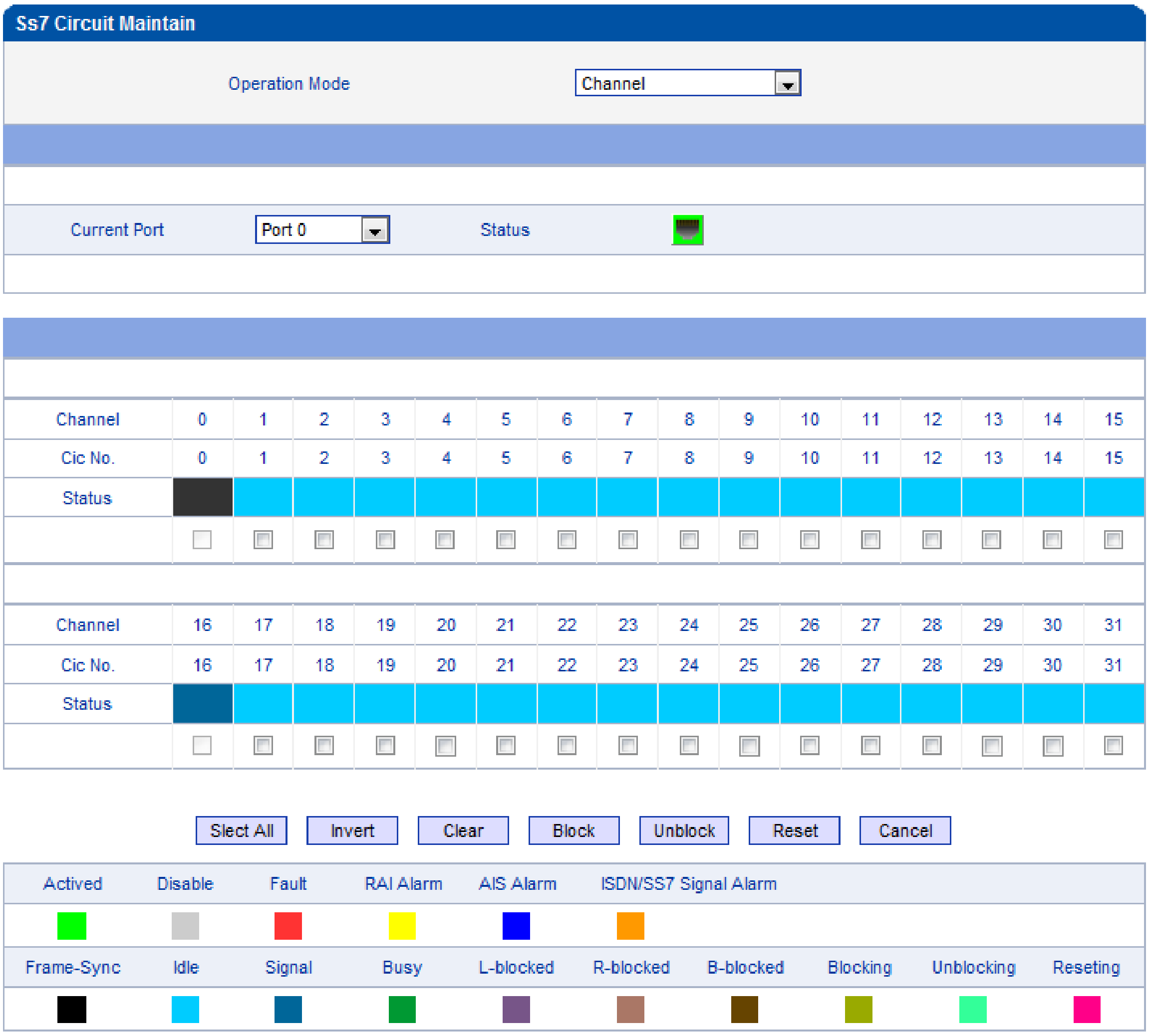
There are two objects to be maintained for SS7 CIC, namely E1/T1 ports and channels. Select **E1/T1** on the right of **Operation Mode**, and the following interface will be displayed.



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 CIC: Circuit Identification Code
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 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)

Modify

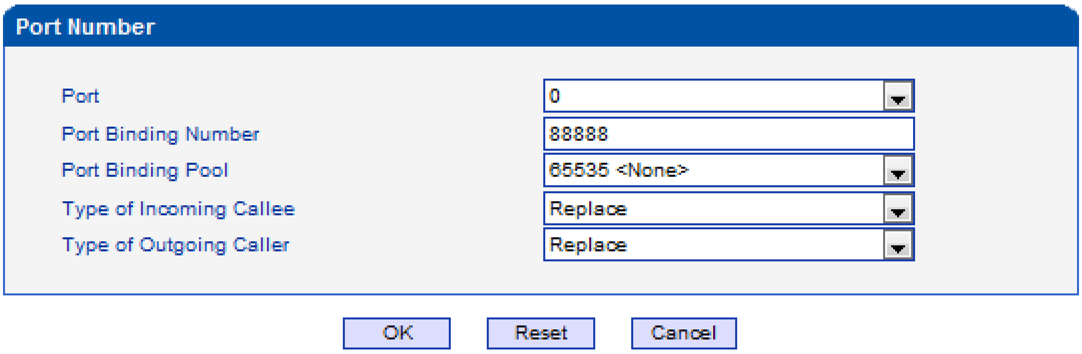
Parameter	Explanation
Port No.	The No. of each E1/T1 port
Work Mode	E1 or T1 If E1 is selected for one port, the work modes of all ports are E1.
PCM Mode	PCMA(A LAW) or PCMU(Mu LAW) If A LAW is selected for one port, the work modes of all ports are A LAW.
Frame Mode: DF CRC-4 CRC4_ITU	Frame modes of 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;

4.8.2 Port Number

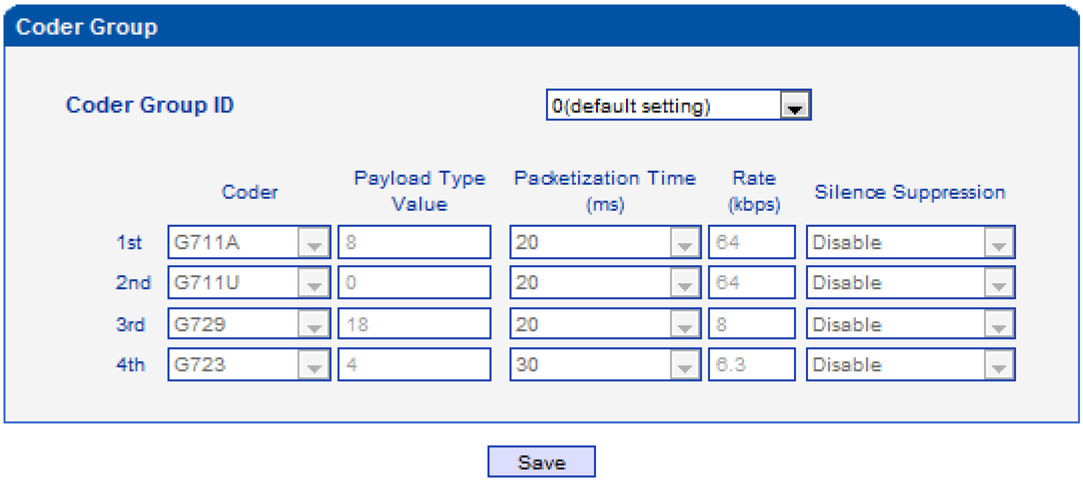
An E1/T1 port can be bound with a number. On the **Port Number** interface, you can do some configurations to make a preset 'port binding number' replace the called number or caller number. The 'port binding number' can be any number, and it can be from the binding pool.

When it is an incoming call, the caller number will be replaced. When it's an outgoing call, the called number will be replaced.



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.

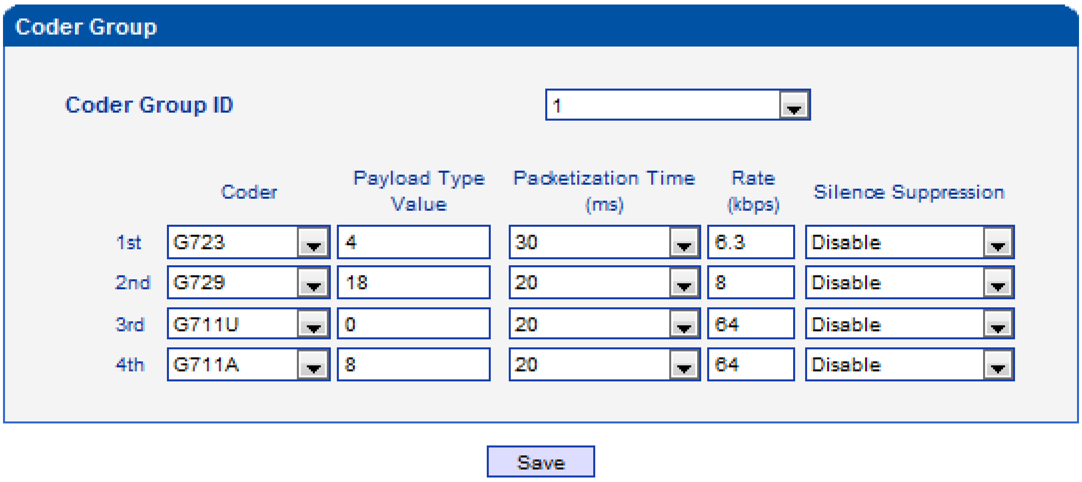


Parameter	**Explanation**
Codec Group ID	ID of each codec group for voice ability, from 0 to 7. The codec group 0 is default setting which cannot be modified.
Codec	MTG200 supports three kinds of voice codec: G711A, G711U, G729, G723, iLBC 13k and iLBC 15k.
Payload Type Value	Each codec has a unique 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 20ms.
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'.

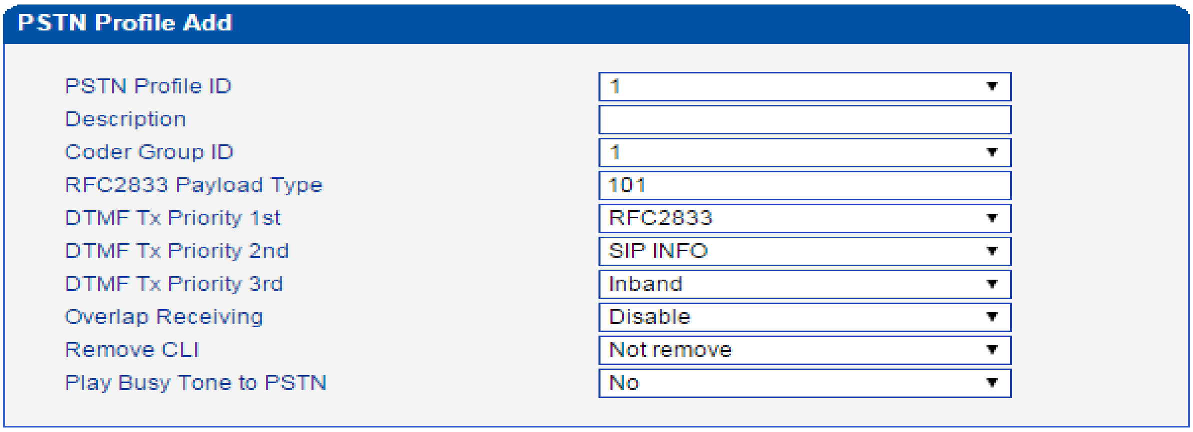
- **Example: How to configure preferred codec group**

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

Step2. Select preferred voice codec (G723, G729, G711U and G711A) in this example, as below:

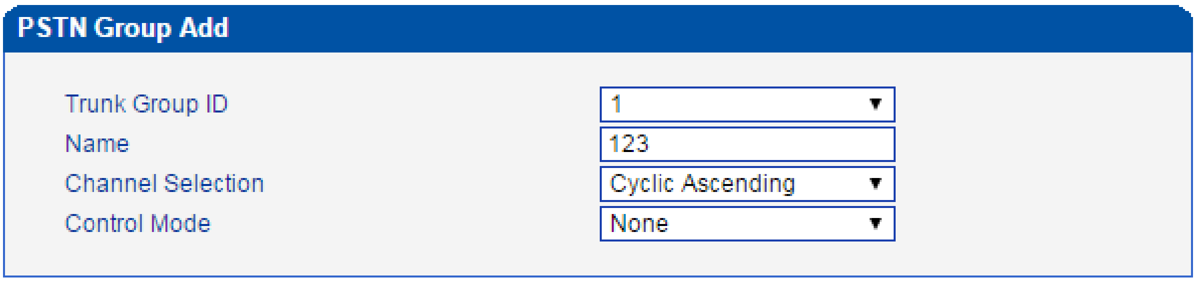


Step3. Enter into the **PSTN Profile** interface, click **Modify** to modify the default PSTN profile and change the codec group ID, or click **Add** to add a new PSTN profile.

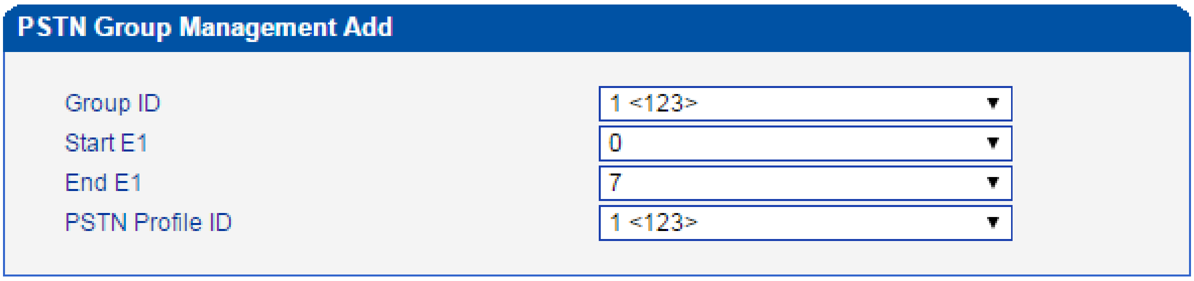


Step4. Click **OK** to save the above configuration.

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



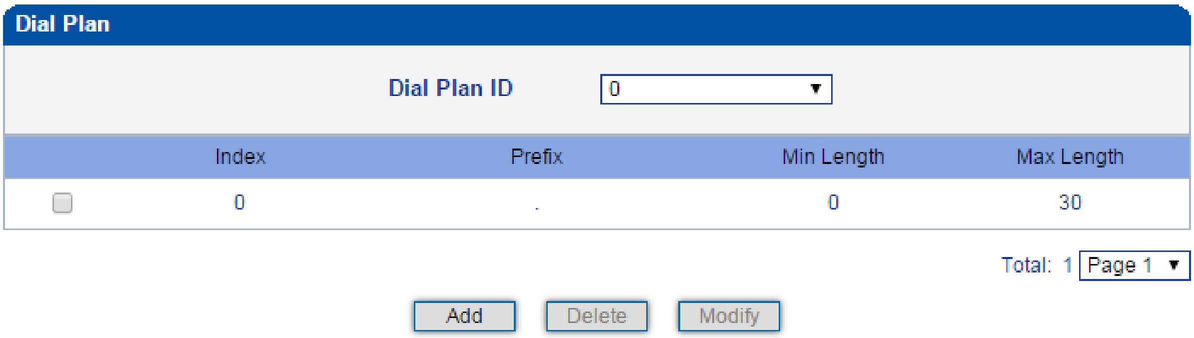
Step5. Enter into the **PSTN Group Management** interface to associate the PSTN profile and PSTN group to an E1/T1 port or multiple E1/T1 ports.



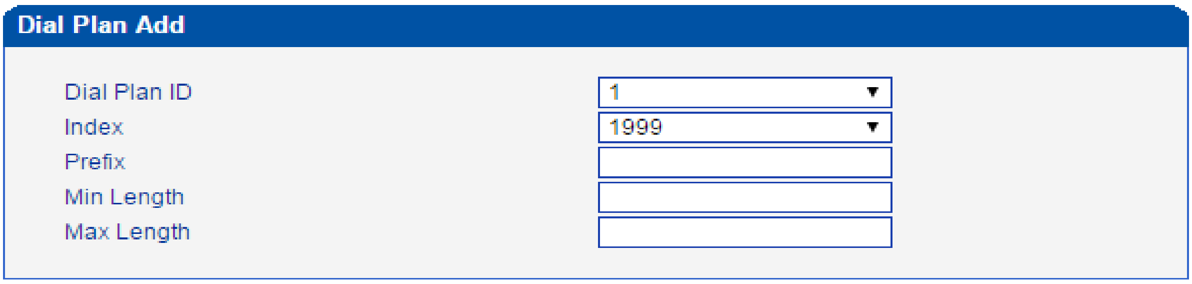
Step6. Click **OK** save the above configuration.

4.8.4 Dial Plan

Dial plan is used for the MTG200 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 MTG200 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, MTG200 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, MTG200 will not received them.

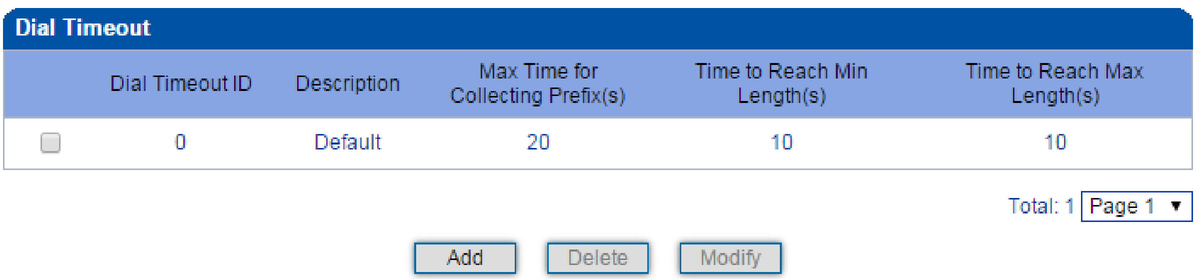
Note:

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.

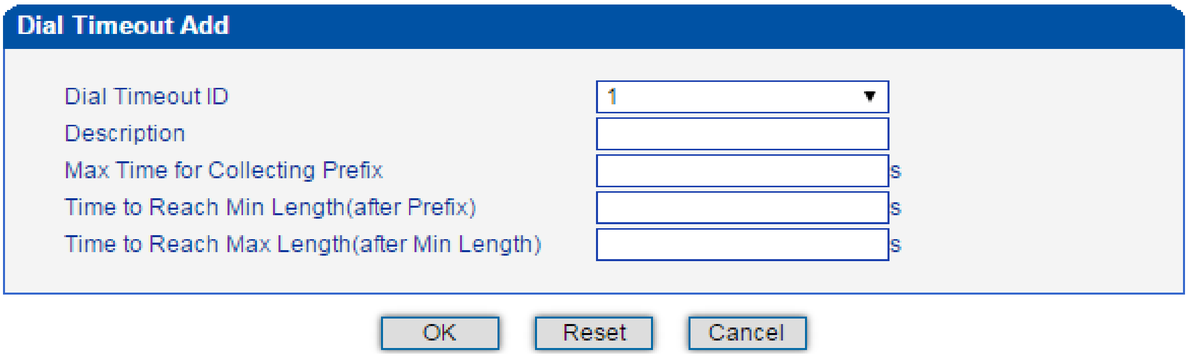
4.8.5 Dial Timeout

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.



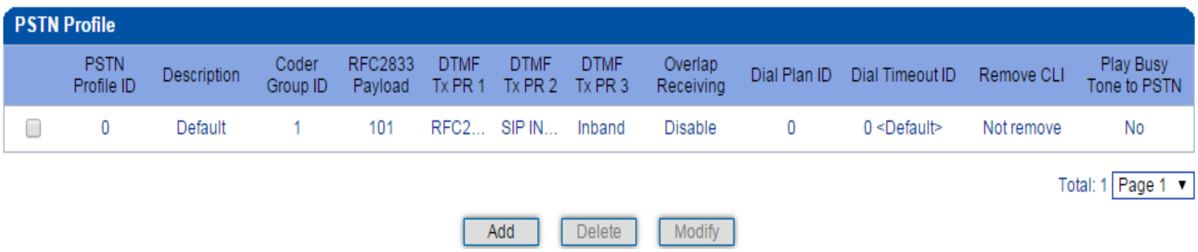
Click the **Add** button to add a new dial timeout rule.



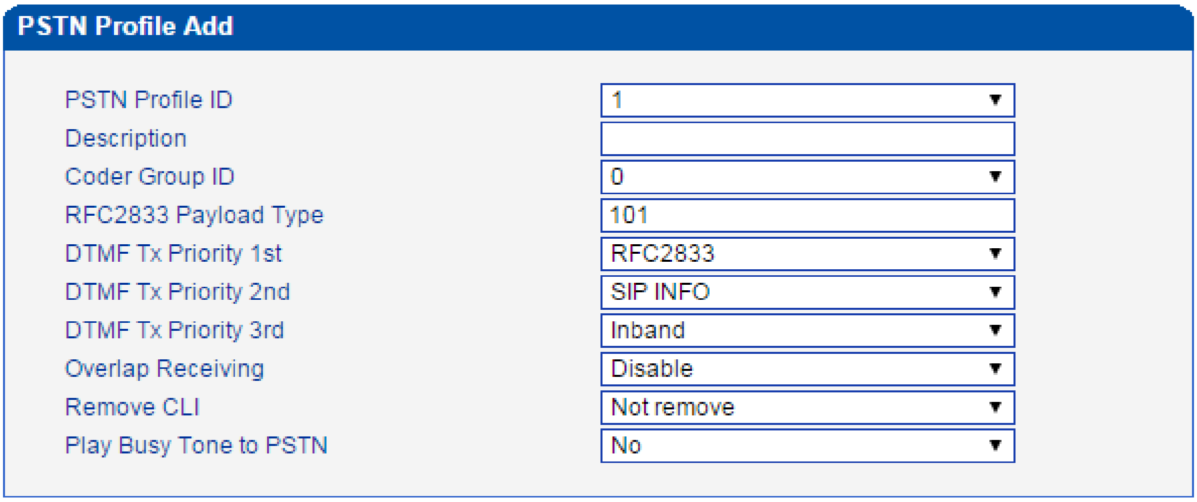
Parameter	Explanation
Dial Timeout ID	The ID of the dial timeout
Description	Description of the dial timeout
Max Time for Collecting Prefix	The maximum time for receiving all the digits of a prefix
Time to Reach Min Length (after Prefix)	After receiving the prefix, the maximum time before receiving the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



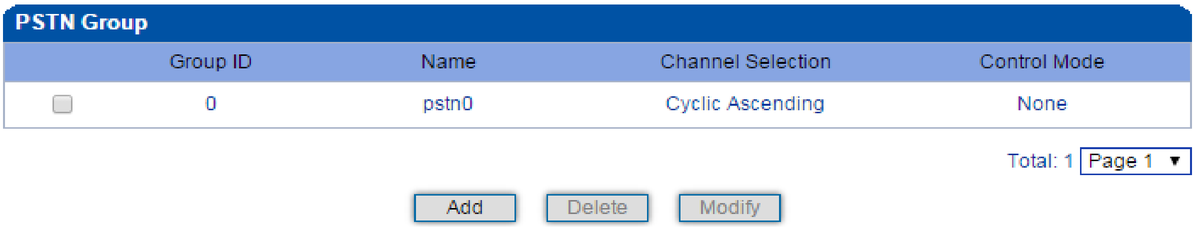
Click the **Add** button to add a new PSTN profile.



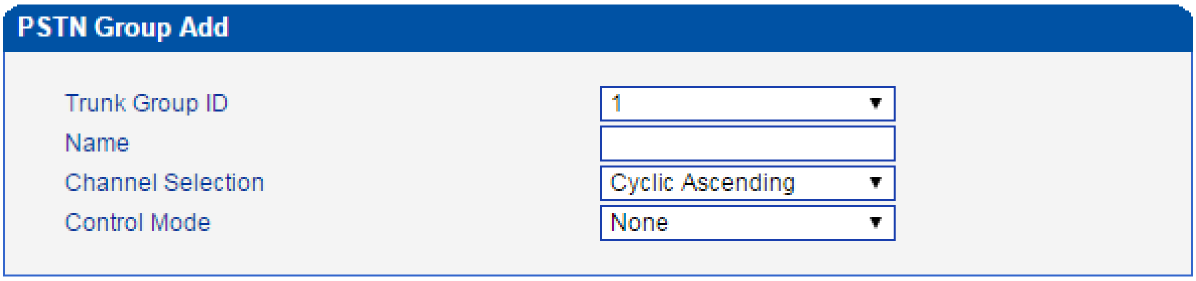
Parameter	Explanation
PSTN Profile ID	The ID of the PSTN profile
Description	The description of the PSTN profile
Coder Group ID	The ID of the coder group (the coder group needs to be created at the **Coder Group** interface first.)
RFC2833 Payload	Default value is 101.
DTMF Tx Priority 1st	There are three ways to send DTMF: RFC2833, SIP INFOR and Inband. You can set their priority. Priority 1st 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 2st 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 2st 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, MTG200 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.

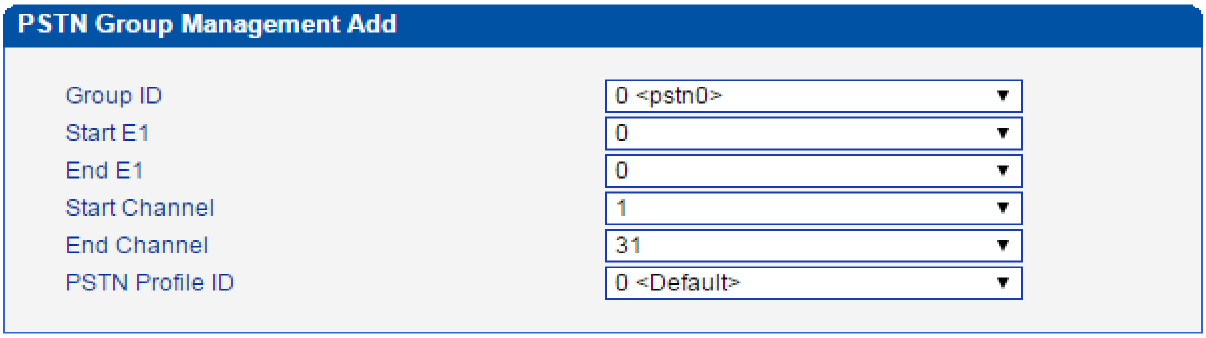


Parameter	**Explanation**
Trunk Group ID	The ID of the trunk group
Name	The name of the trunk group
Channel Selection	There are four selection strategies: Ascending, Descending, Cyclic Ascending and Cyclic Descending. Ascending: to search idle channels starting from channel 0 to channel 31; Cyclic ascending: 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

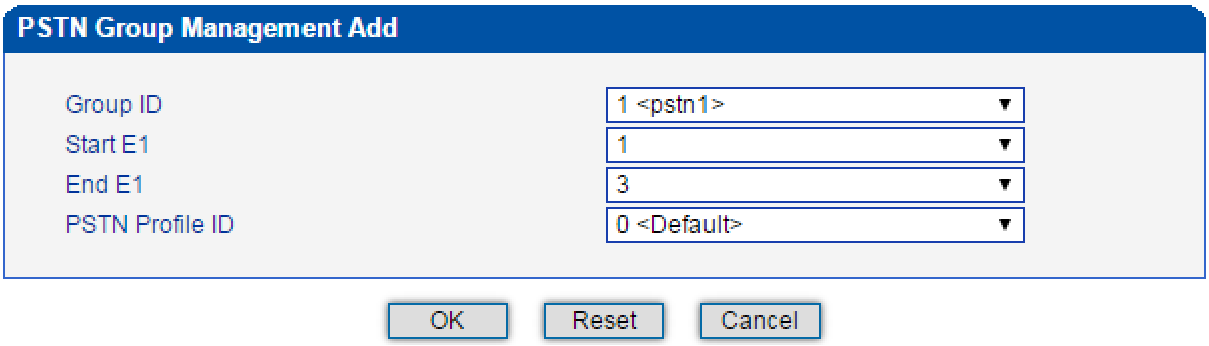
On the **PSTN Group Management** interface, you can add start E1/T1 port, end E1 /T1 port, start channel, end channel and PSTN profile to a PSTN group.

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.

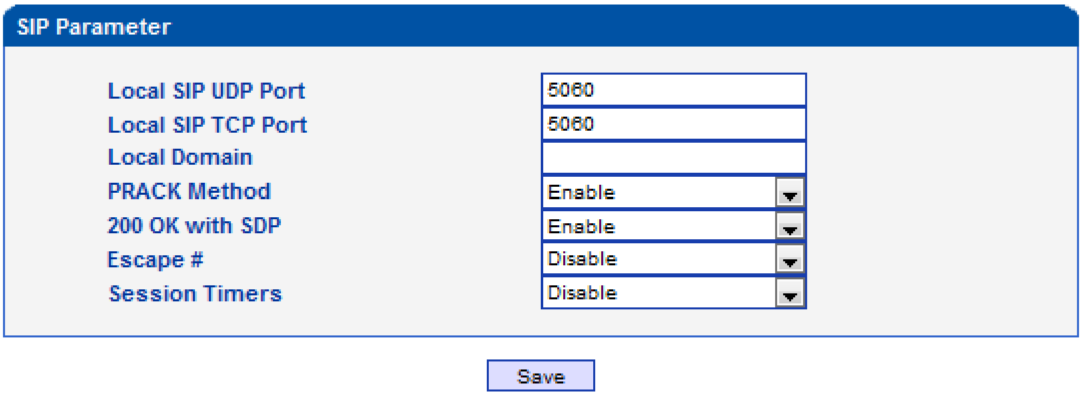


Parameter	Explanation
Group ID	The ID of the PSTN group
Start E1/T1	The start E1/T1 port in this PSTN group
End E1/T1	The end E1/T1 port in this PSTN group
Start Channel	The start channel in this PSTN group
End Channel	The end channel in this PSTN group
PSTN Profile ID	The ID of the PSTN profile in this 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

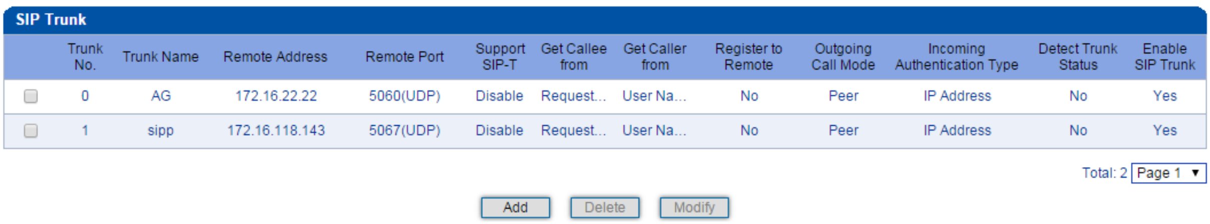
4.9.1 SIP Parameter



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 MTG200 and PBX or SIP servers under the IP network. It provides two modes to connect MTG200 and the IP network. One is Access (MTG200 registers to a softswitch), and the other is Peer (MTG200 connects to a peer device in the IP network via IP address).



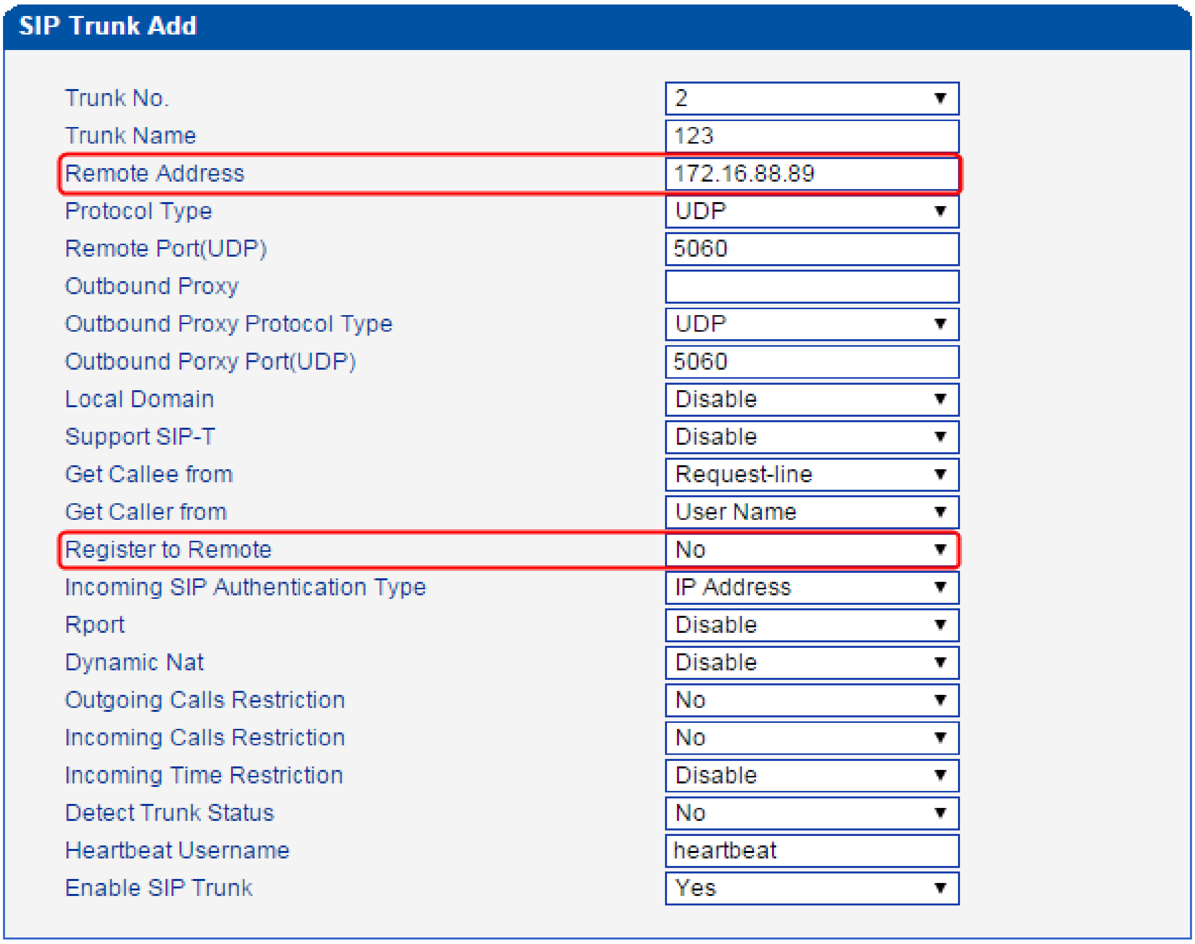
Configuration procedures for Peer Mode are as follows:

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 MTG200
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 MTG200; 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, MTG200 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.

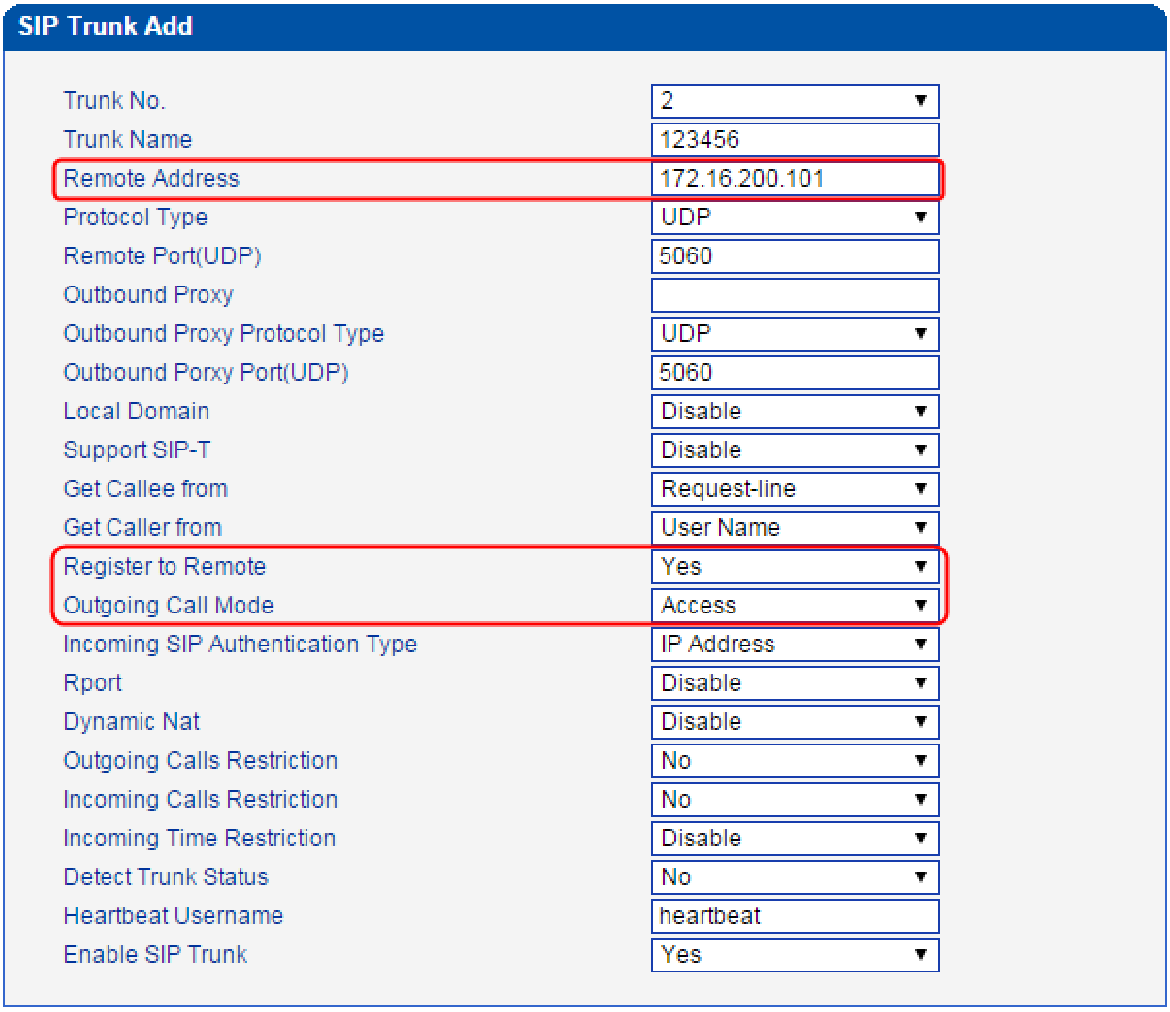
Outgoing Calls Registration	Whether to limit the number of the calls from PSTN to IP network. The default value is ‘No’. If ‘Yes’ is selected, then input the number of concurrent calls that are allowed to go out. The range is 0 to 65535.
Incoming Calls Registration	Whether to limit the number of the 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, MTG200 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.

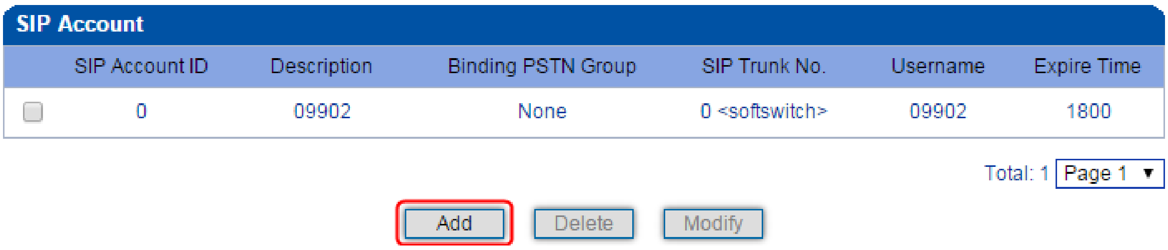
2. Configure parameters on the following interface according to related explanations.

As it is Access mode, you should select **Yes** for the **Register to Remote** parameter, and enter the IP address of a softswitch.

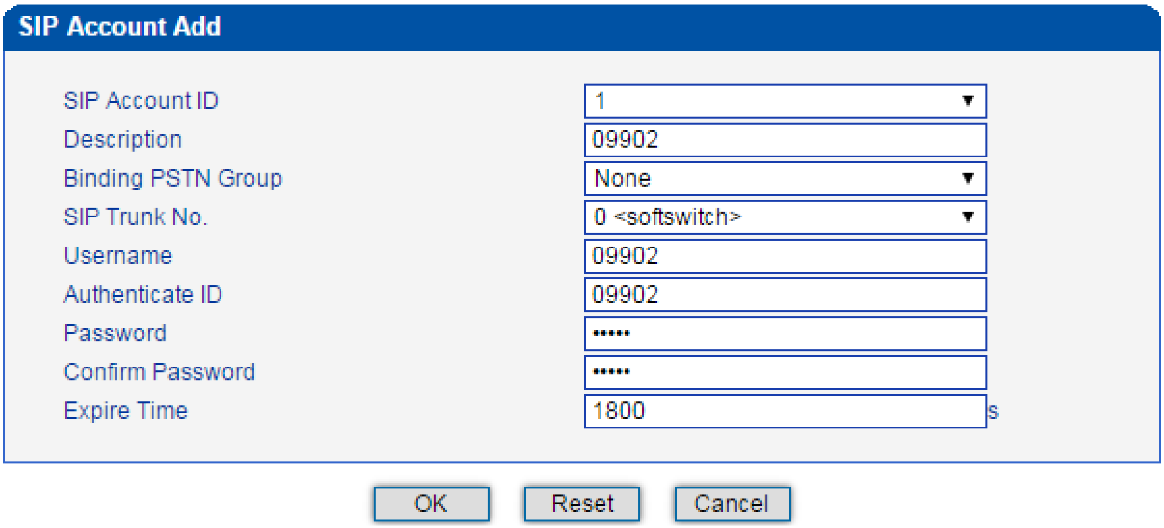


3. Click **OK.**

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.



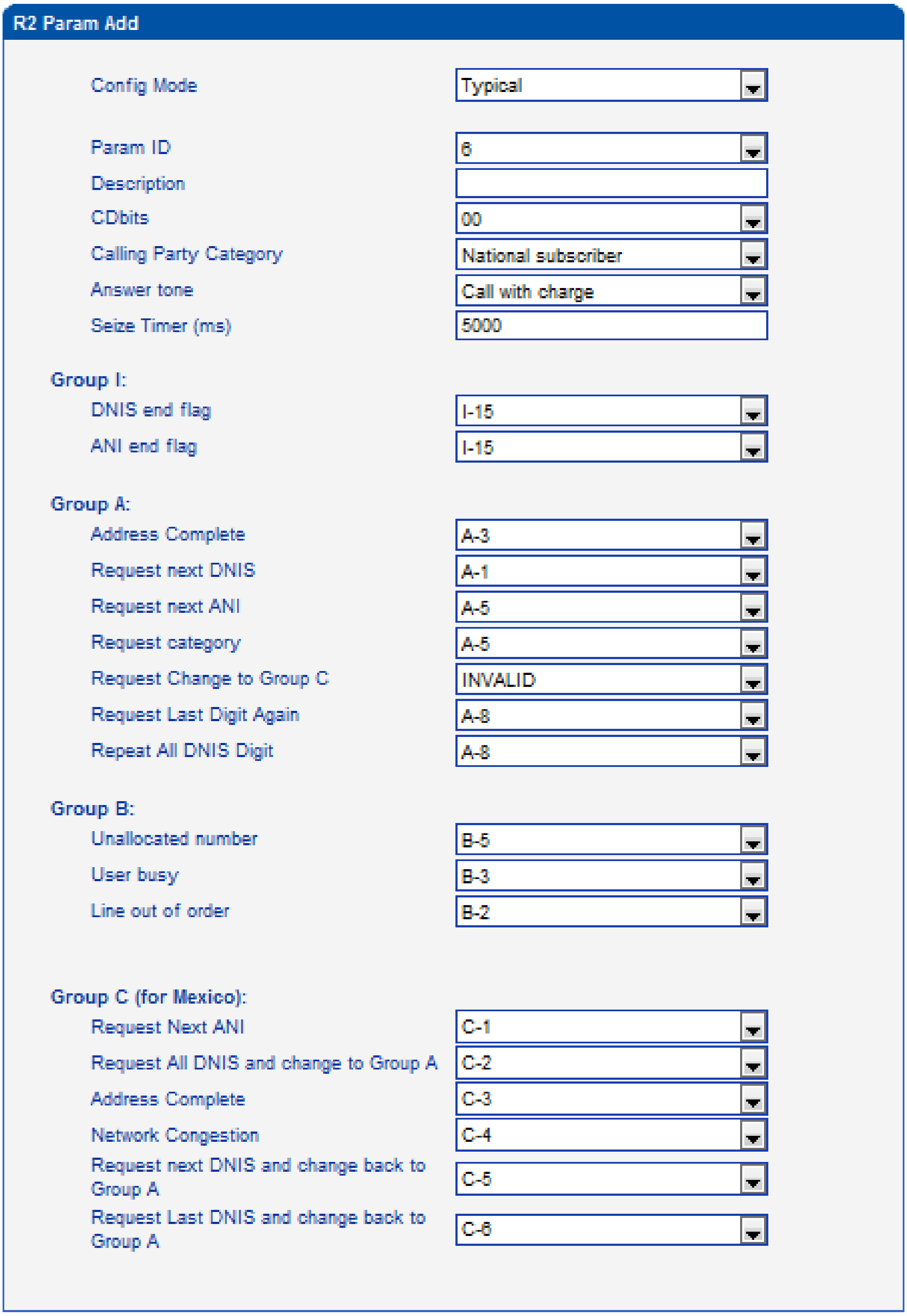
Parameter	Explanation
SIP Account ID	The ID of SIP Account, from 0 to 127
Description	Description of the SIP account
Binding PSTN Group	Choose a PSTN group that is bound to the SIP account
SIP Trunk No.	The No. of the SIP trunk bound to the SIP account
Username	The username of the SIP account, which is used to register the SIP account to softswitch
Authenticate ID	The authentication ID to authenticate the SIP account for the softswitch connected to MTG200
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.

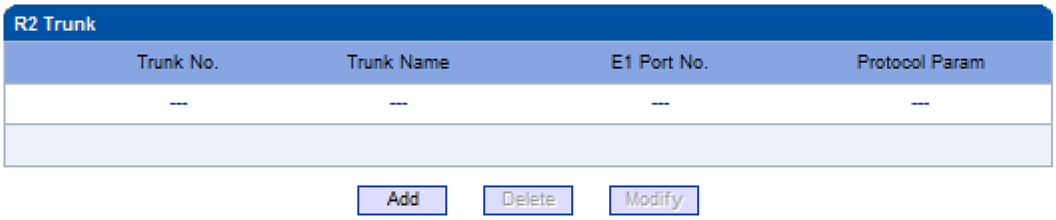
4.10 R2 Config

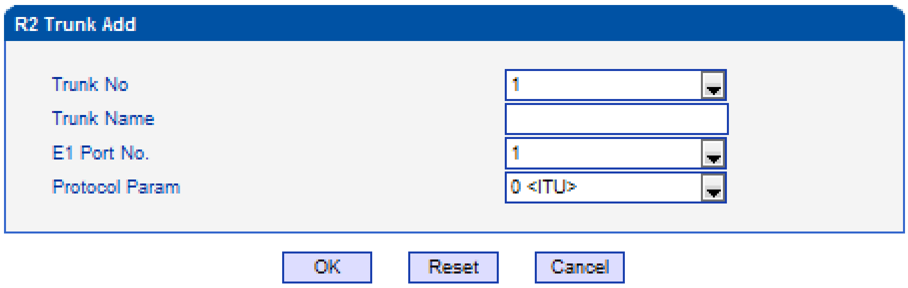
Whether the R2 function is enabled or not is determined by the license.

4.10.1 R2 Parameter



4.10.2 R2 Trunk





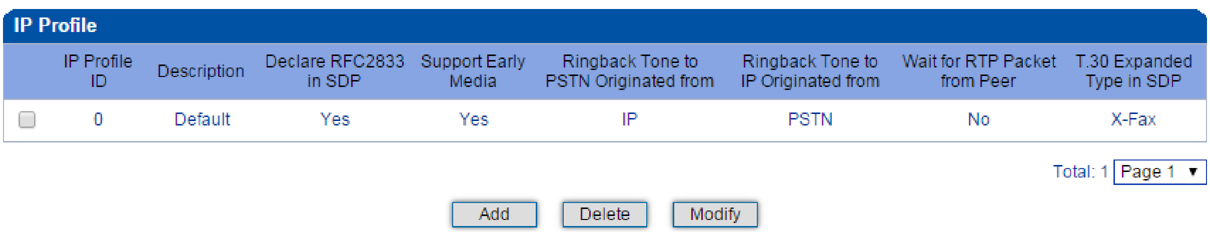
Trunk No.	The No. of this R2 trunk. User can add up to 4 R2 trunks at most.
Trunk Name	Name of this R2 trunk
E1 Port No.	The No. of the E1 port connected to this R2 trunk.
Protocol Parameter	Which country the protocol conforms to. ITU: International Telecommunications Union.

4.11 IP Group Config

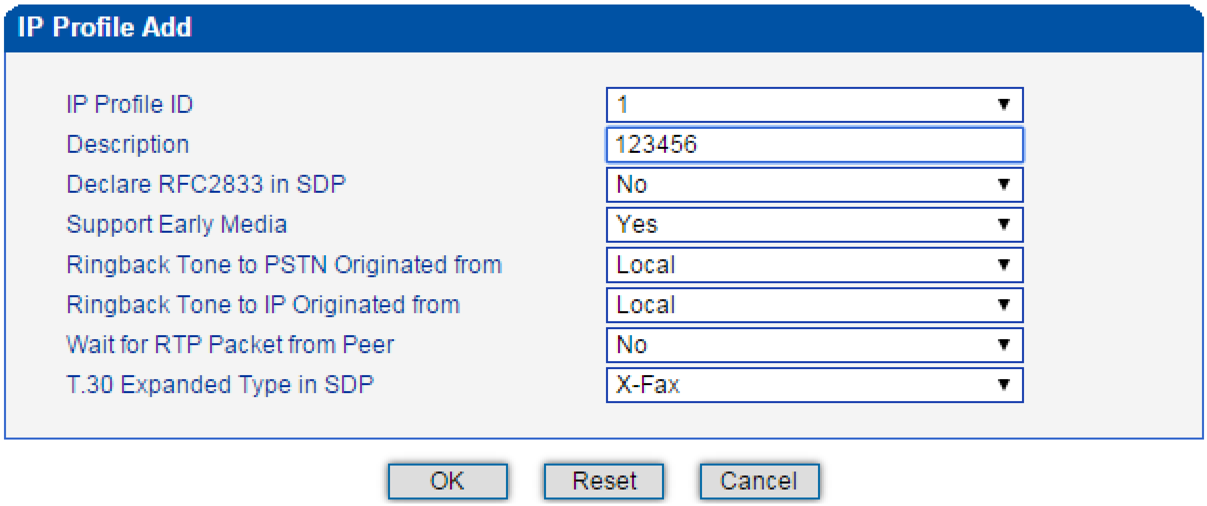
You can group several SIP trunks together, so when one SIP trunk is in an outage, communication can turn to another SIP trunk in the same group.

4.11.1 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.

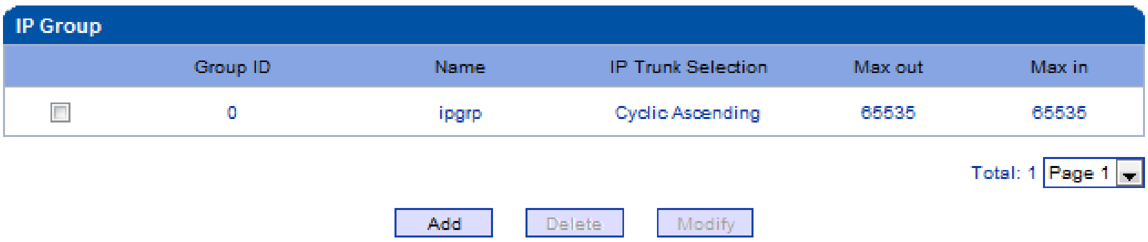


Parameter	**Explanation**
IP Profile ID	The ID of the IP profile, from 1 to 15.
Description	Description of the IP profile
Declare RFC2833 in SDP	Whether to declare RFC2833 in SDP Default value is 'Yes'.
Support Early Media	Whether to support Early Media (183) If 'Yes' is selected, ringback tone will 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 MTG200. 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 MTG200. If 'PSTN' is selected, the ringback tone is played from the PSTN

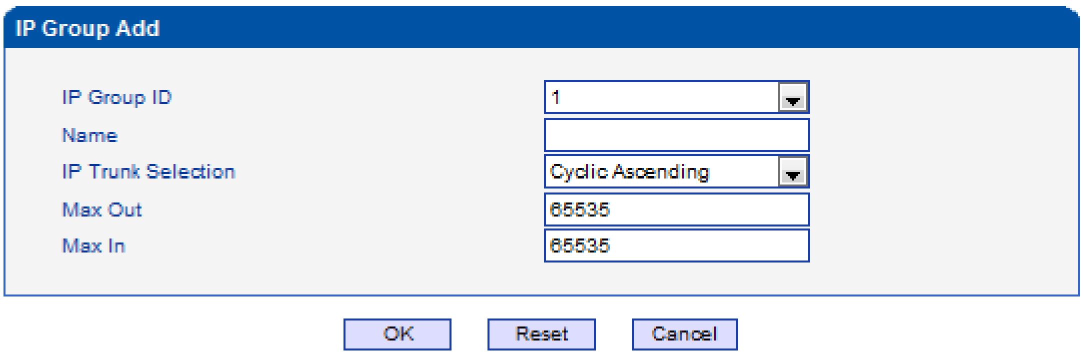
Wait for RTP Packet from Peer	If 'Yes' is selected, RTP packets will be sent from peer device to MTG200 first, and then RTP packets will be sent from MTG to peer device. If 'No' is selected, RTP packets will be sent automatically during calling;
T.30 Expanded Type in SDP	There are two T.30 expanded types: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

4.11.2 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.

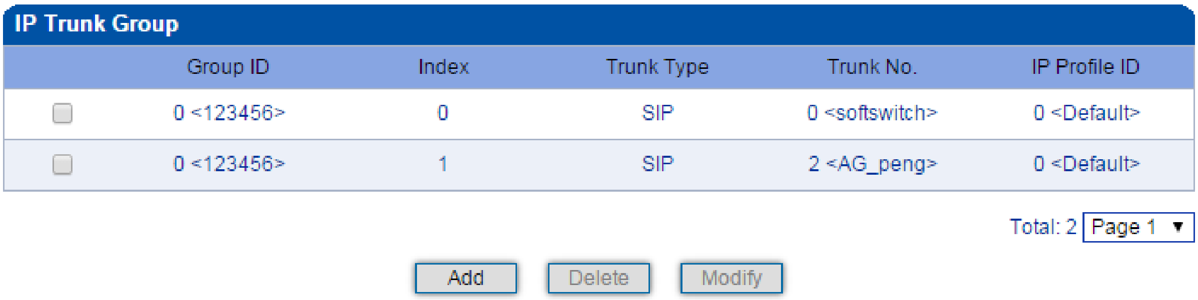


IP Trunk Selection	Ascending	To select IP trunks in an ascending order under a same group.
	Cyclic Ascending:	To select IP trunks in an ascending order, starting from the previous IP trunk that has been selected
	Descending	To select IP trunks in a descending order under a same group
	Cyclic Descending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

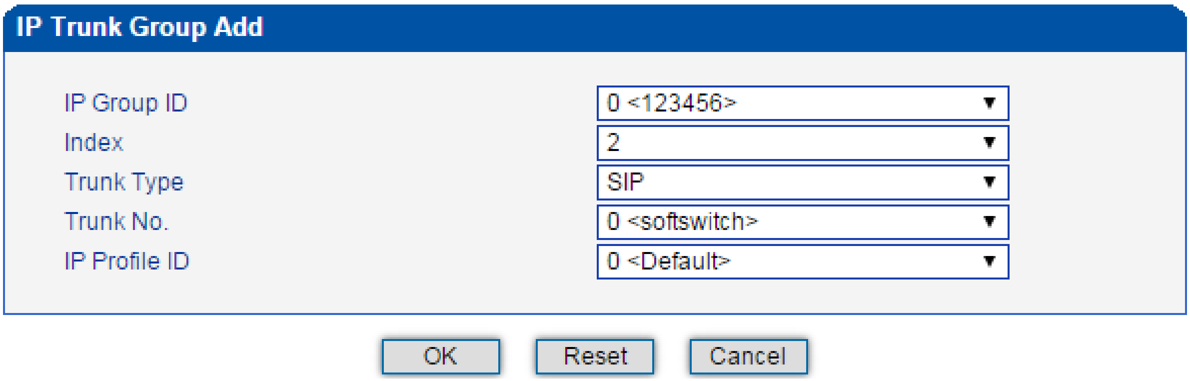
Max Out	Maximum number of callout concurrencies
Max In	Maximum number of callin concurrencies

4.11.3 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.

4.12 Number Filter

This section is mainly to introduce how to configure white & black lists on the MTG200 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.

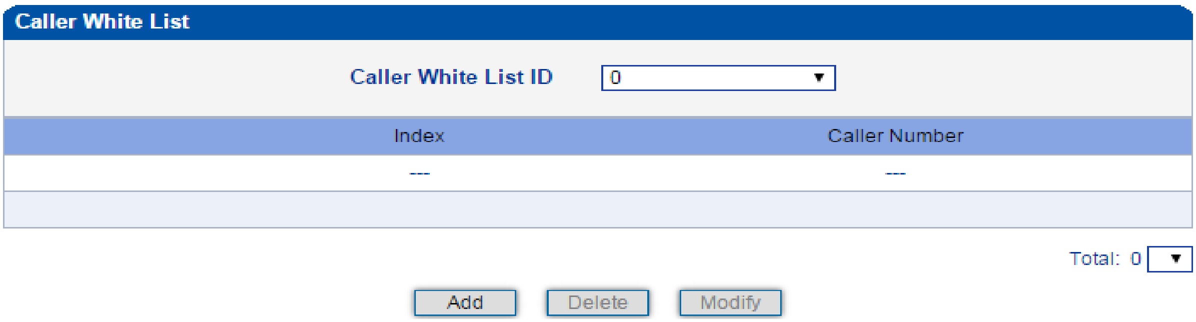
Caller Black List: Calls from the numbers on the Caller Black List will be rejected to pass. If a caller number match with one of the numbers on the Caller Black List, calls will be rejected.

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.

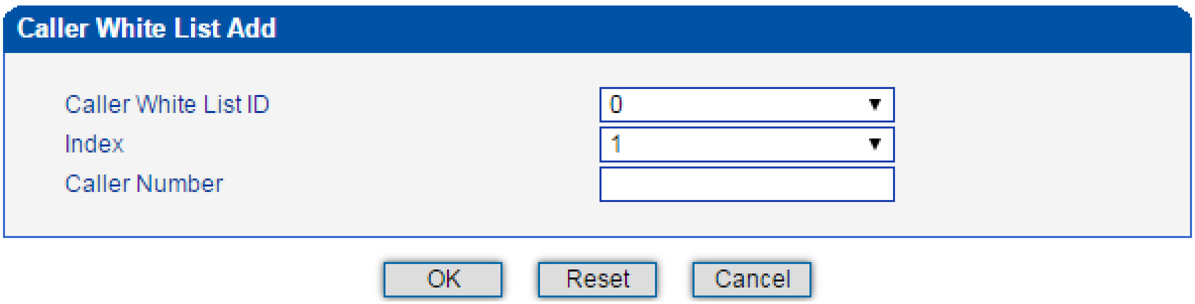
Callee Black List: Calls to the numbers on the Callee Black List will be rejected to pass. If a callee number match with one of the numbers on the Callee Black List, calls to the callee number will be rejected.

4.12.1 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

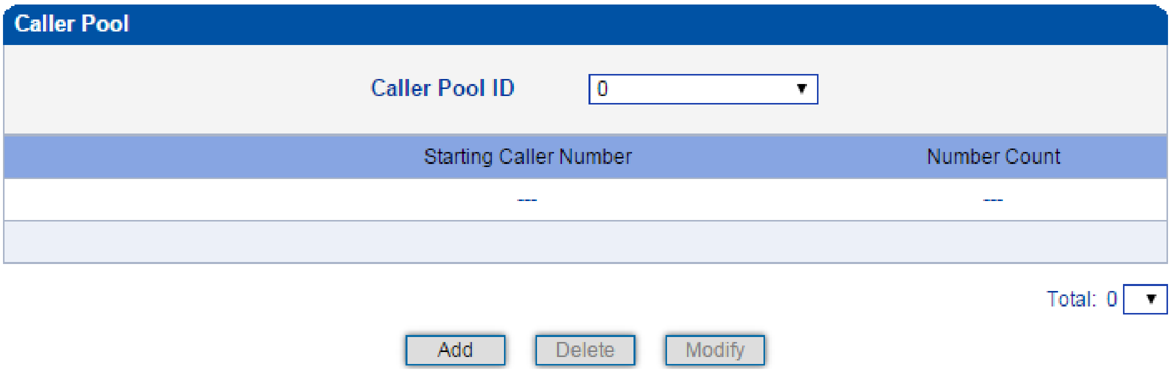
4. Click **OK**.

Note:

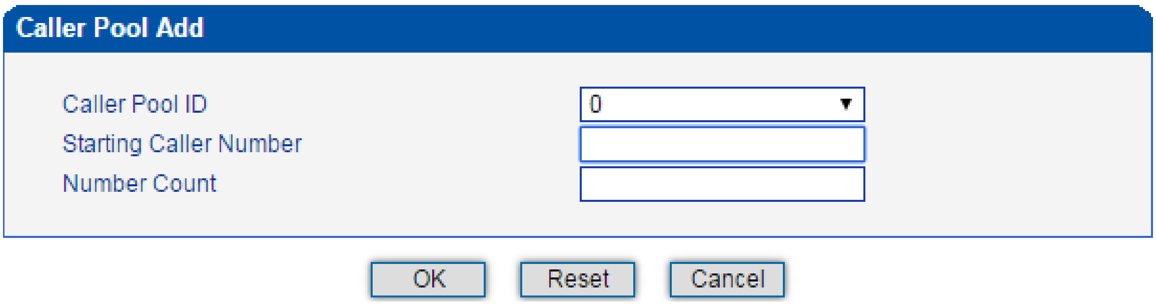
You can add 8 white or black lists at most, with ID from 0 to 7. And each white or black list can contain 1024 numbers at most.

4.12.2 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:

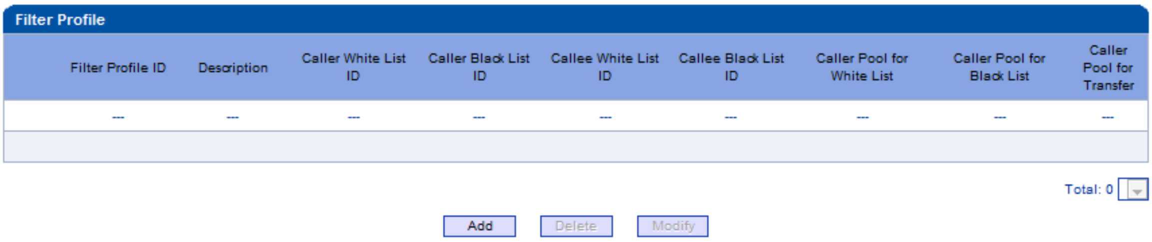
If ‘Starting Caller Number’ is 80080000 and ‘Number Count’ is 100, it means numbers from 80080000 to 80080099 are all in the caller pool.

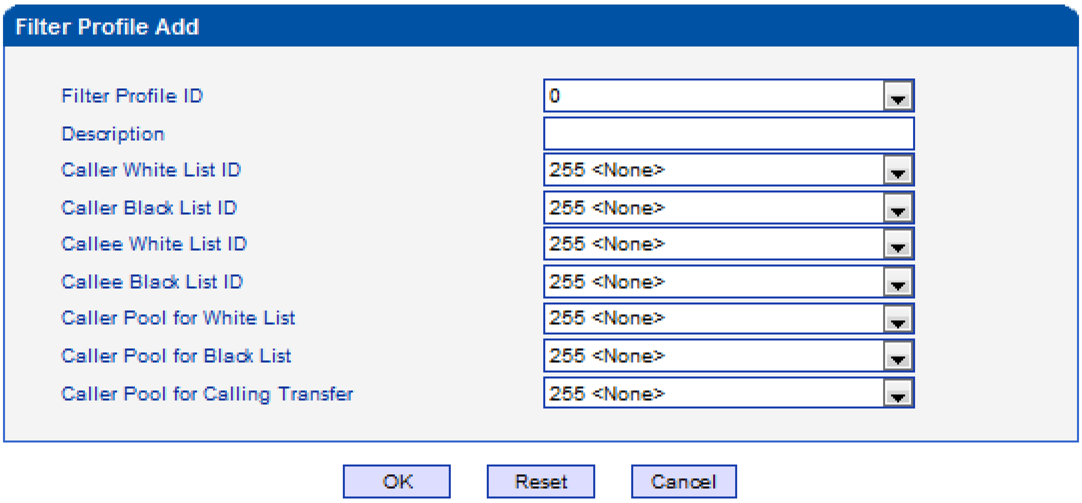
Each caller poor can contain 512 numbers at most, and if there are multiple caller pools, the caller pools can contain up to 1024 numbers in total.

‘Number Count’ cannot be greater than 256.

4.12.3 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.



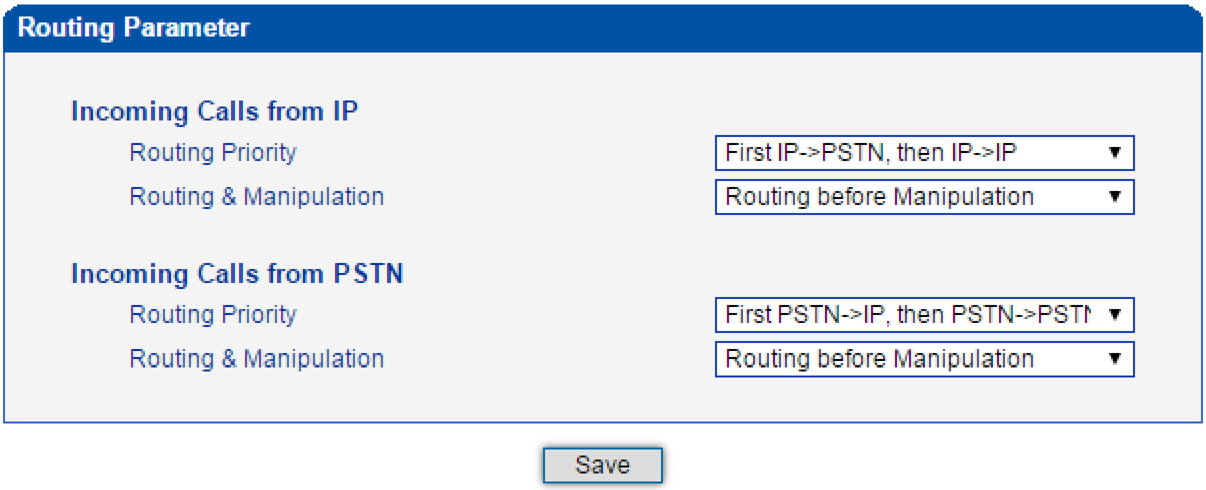


Select a white list ID, and the calls of the numbers on this white list will be passed. Select a black list ID, and the calls of the numbers on this black list will be prohibited.

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

4.13 Call Routing

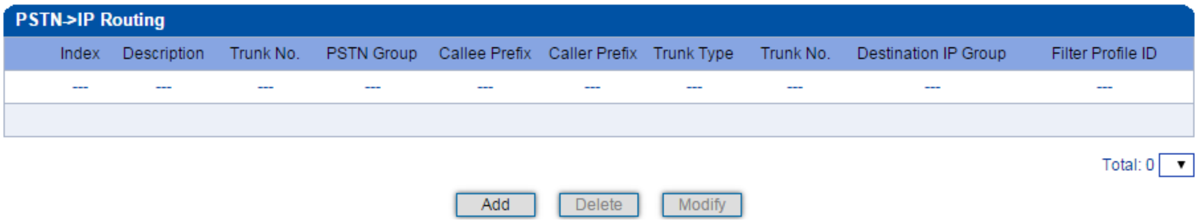
4.13.1 Routing Parameter



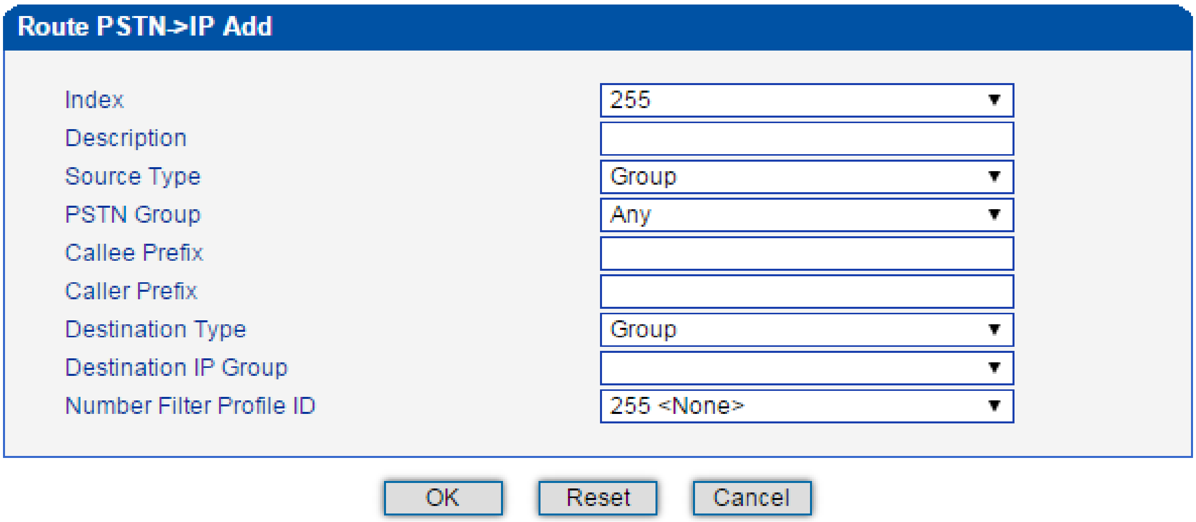
Belong To	**Parameter**	**Explanation**
Incoming Calls from IP	Routing Priority	There are two options: First IP→PSTN, then IP →IP First IP→IP, then IP →PSTN
	Routing & Manipulation	There are two options: Routing before Manipulation Routing after Manipulation
Incoming Calls from PSTN	Routing Priority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

4.13.2 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.

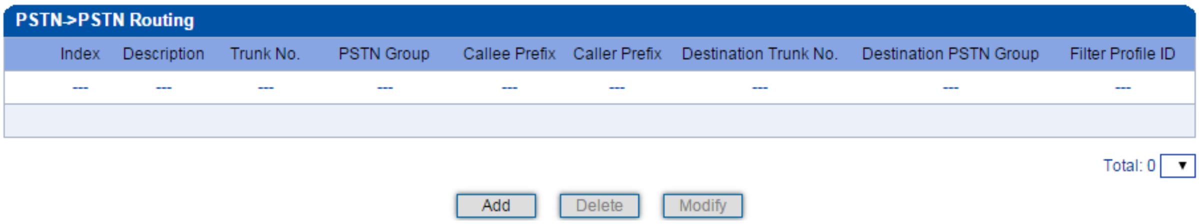


Parameter	**Explanation**
Index	The Index of the PSTN →IP route, from 0 to 255. Greater index value, higher priority for the route.
Description	The description of the PSTN →IP route,
Source Type	Sources include PSTN group and PRI/SS7 trunk.
PSTN Group	If source is PSTN group, please select a specific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

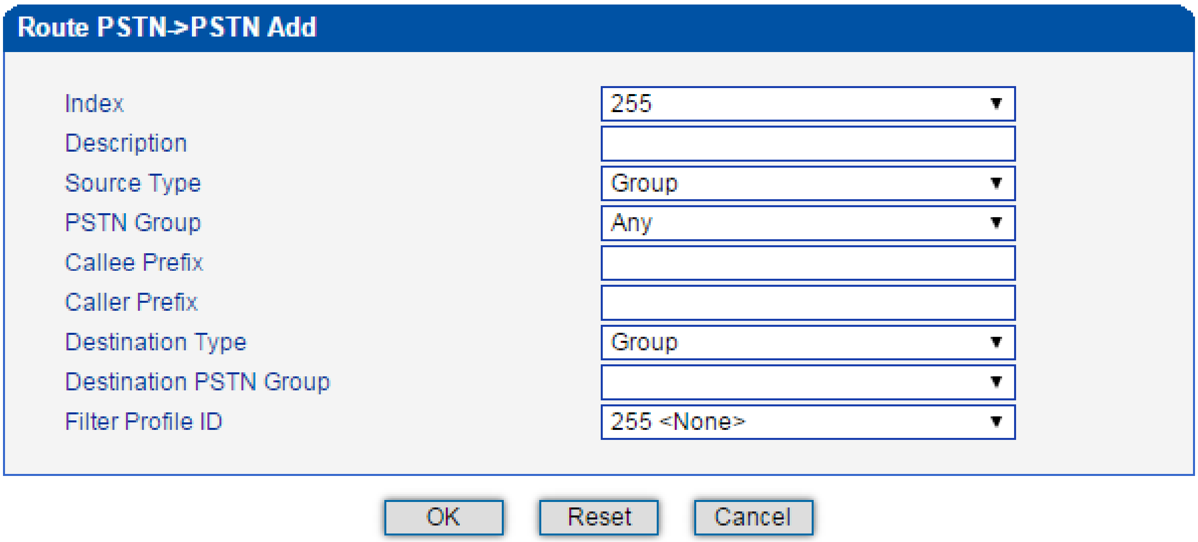
Caller Prefix	The prefix configured for caller number. When a caller number matches the prefix, this PSTN →IP route will be used. '.' is a wildcard, which means this PSTN →IP route will be used, no matter what the caller number is.
Destination Type	Destination is IP group or SIP trunk.
Destination IP 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.

4.13.3 PSTN → PSTN Routing

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



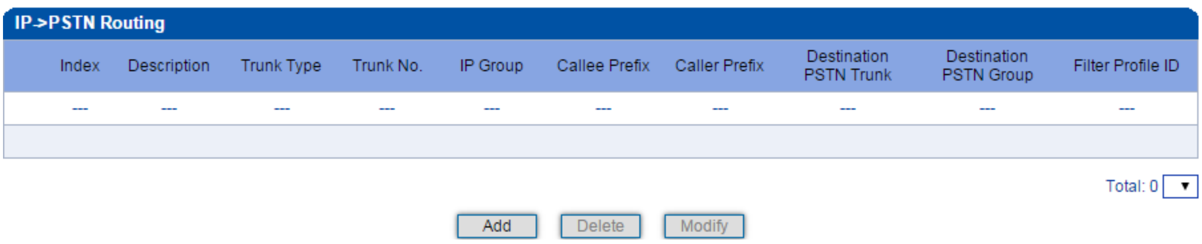
Click **Add**, and the following interface will be displayed.



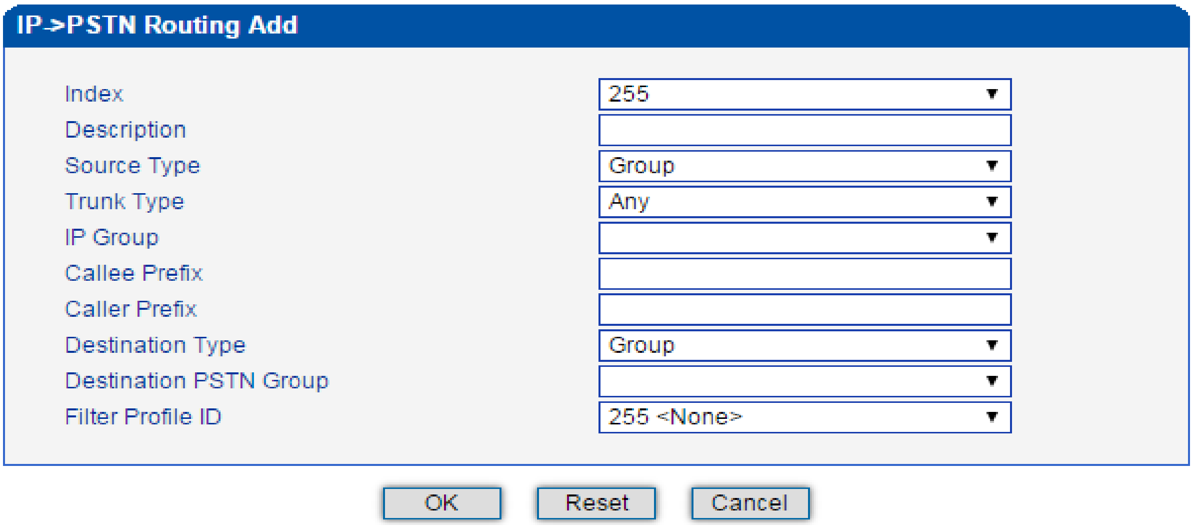
Parameter	**Explanation**
Index	The Index of the PSTN →PSTN route, from 0 to 255. Greater index value, higher priority for the route.
Description	The description of the PSTN →PSTN route,
Source Type	Sources include PSTN group and PRI/SS7 trunk.
PSTN Group	If source is PSTN group, please select a specific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

4.13.4 IP → PSTN Routing

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



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

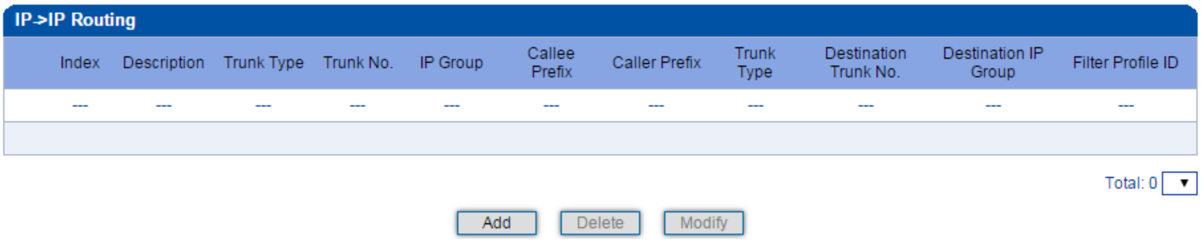


Parameter	Explanation
Index	The Index of the IP→PSTN route, from 0 to 255. Greater index value, higher priority for the route.
Description	The description of the IP →PSTN route,
Source Type	Sources include IP group and IP trunk.
PSTN Group	If source is IP group, please select a specific IP group. If '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.

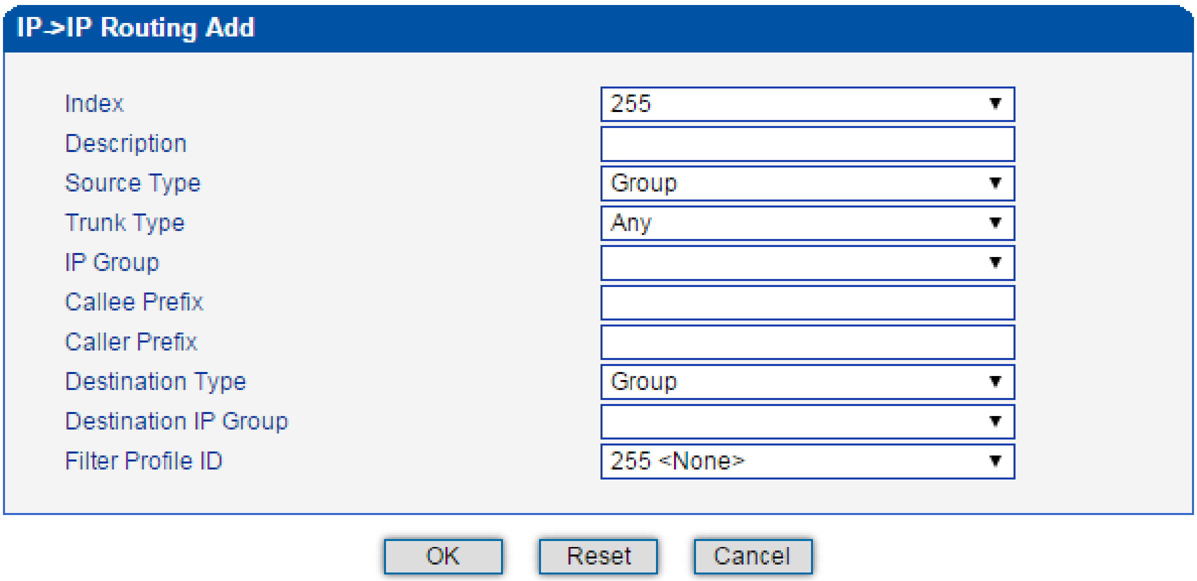
IP Trunk No.	If source is PRI/SS7 trunk, please select a specific PRI/SS7 trunk.
Number Filter Profile ID	The ID of filter profile. The white lists and black lists set in the filter profile will apply to this PSTN →PSTN route.

4.13.5 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.

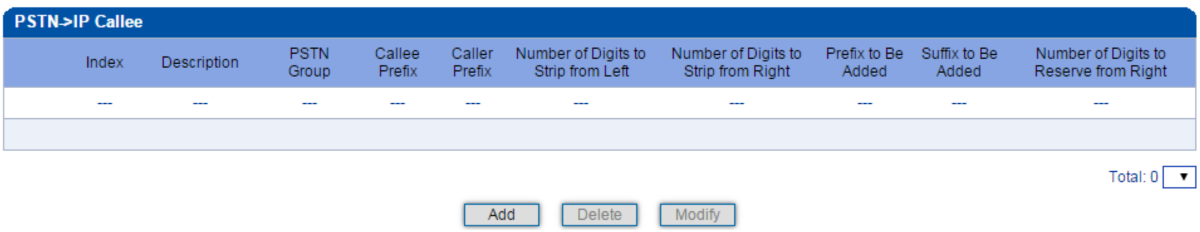
PSTN Group	If source is IP group, please select a specific IP group. If 'Any' is selected, it means the source is any IP group.
PSTN Trunk	If source is IP trunk, please select a specific SIP trunk. If 'Any' is selected, it means the source is any IP trunk. .
Callee Prefix	The prefix configured for callee 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.

4.14 Number Manipulation

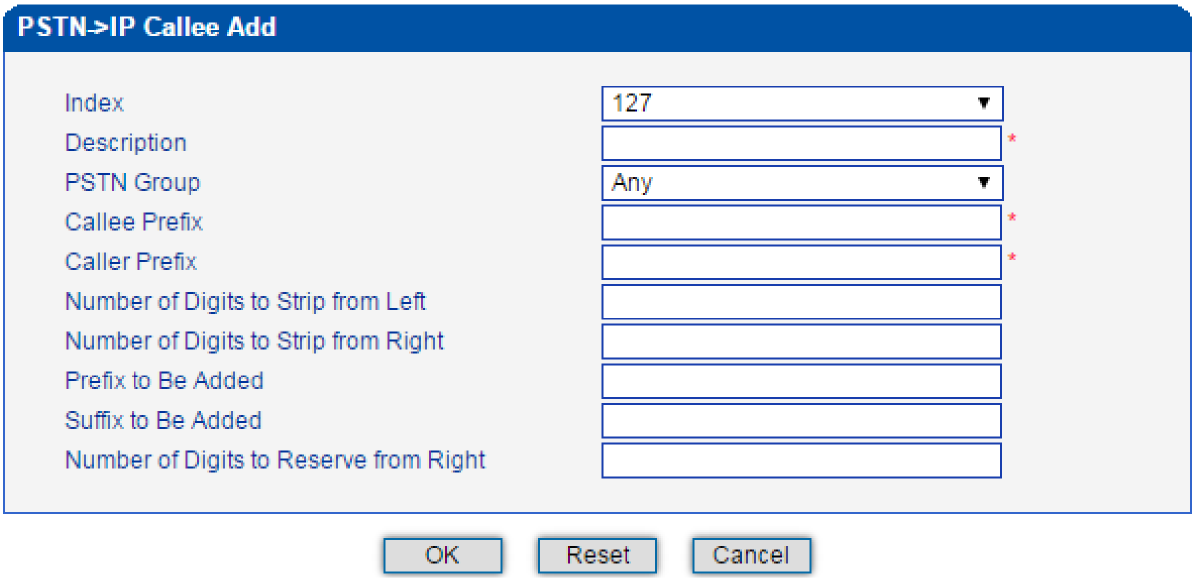
Number manipulation refers to the change of the caller number or callee number during calling process.

4.14.1 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 ?

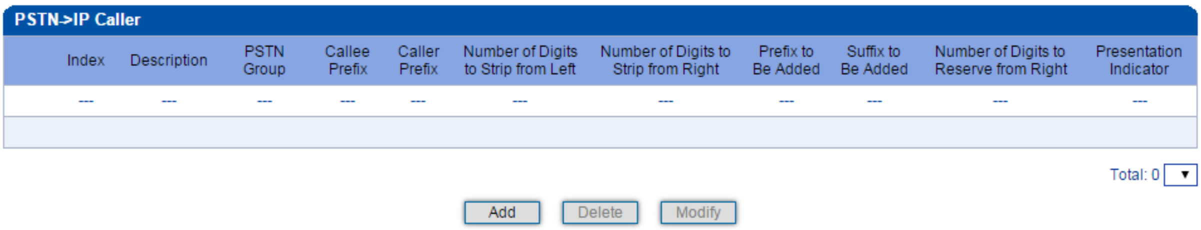
You can enter ‘3’ in the value box for the ‘Number of Digits to Strip from Left’ parameter.

If the called number is 2653101413, how do you change it into 00912653101413?

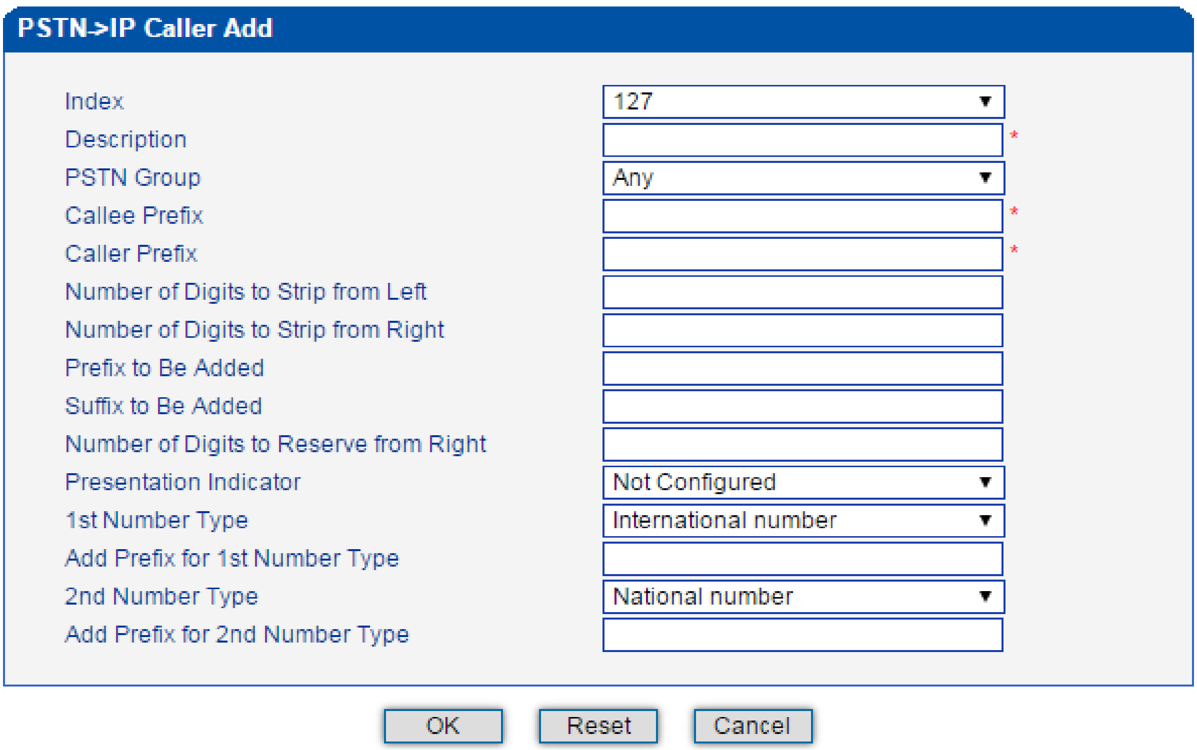
You can enter ‘ 0091’ in the value box for the ‘Callee Prefix’ parameter.

4.14.2 PSTN→IP Caller

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



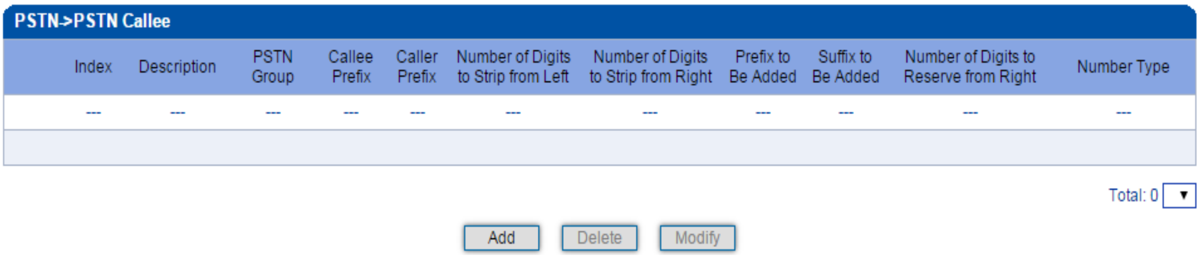
Click **Add**, and the following interface will be displayed.



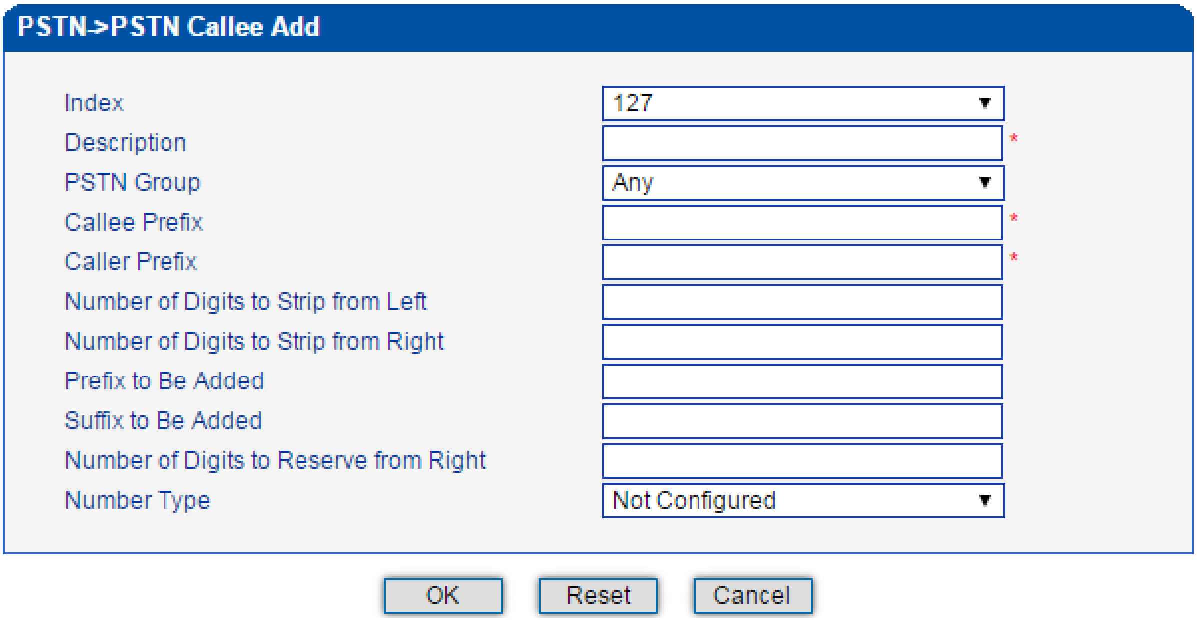
Parameter	Explanation
Index	The index of this PSTN →IP caller number manipulation, from 0 to 127. Each index cannot be used repeatedly.
Description	The description of this PSTN →IP caller number manipulation
PSTN Group	Select a PSTN group. The caller number will be manipulated when a call uses a trunk 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.
1^{st} Number Type	If the caller number belongs to 1^{st} number type, the set prefix will be added to the caller number.
Add Prefix for 1^{st} Number Type	The prefix that will be added to those numbers that belong to 1^{st} number type
2^{nd} Number Type	If the caller number belongs to 2^{nd} 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 2^{nd} number type

4.14.3 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.

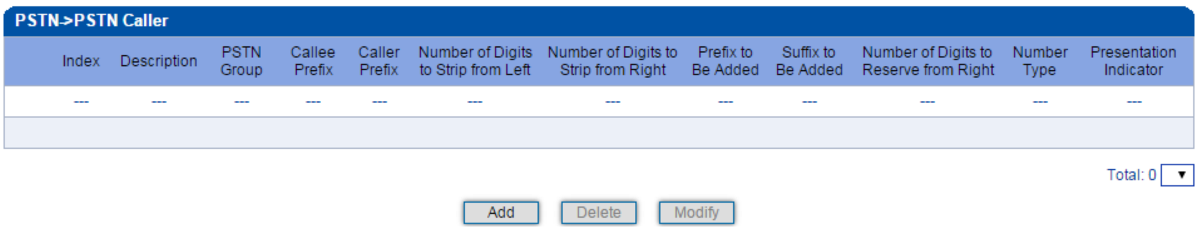


Parameter	Explanation
Index	The index of this PSTN →PSTN callee number manipulation, from 0 to 127. Each index cannot be used repeatedly.
Description	The description of this PSTN →PSTN callee number manipulation
PSTN Group	Select a PSTN group. The callee number will be manipulated when a call uses a 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.

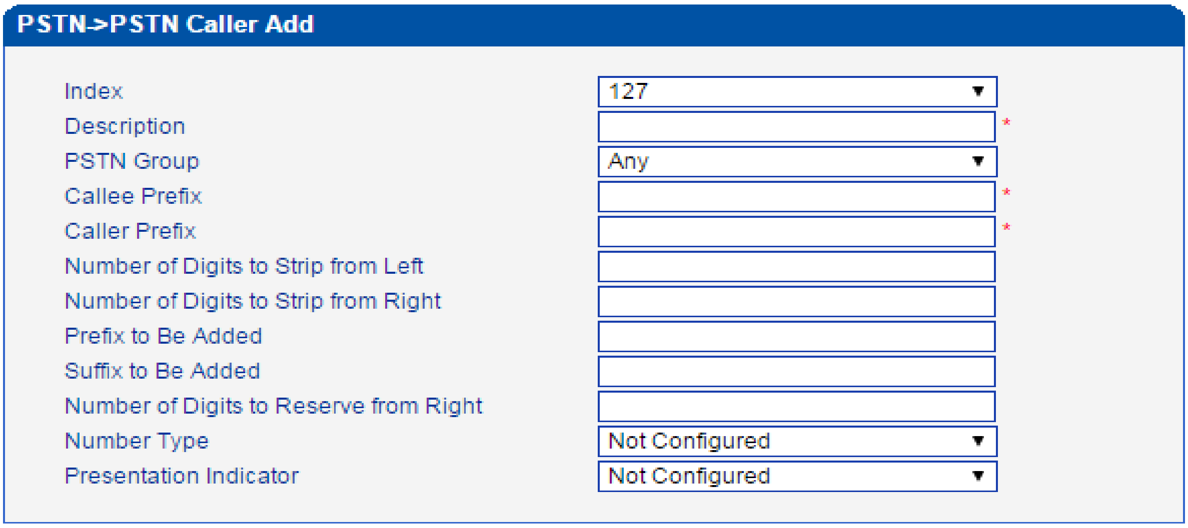
Caller Prefix	Set a prefix for the caller number
Number of Digits to Strip from Left	The number of digits which are lessened from the left of the callee number
Number of Digits to Strip from Right	The number of digits which are lessened from the right of the callee number
Prefix to be added	The prefix 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'

4.14.4 PSTN →PSTN Caller

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



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

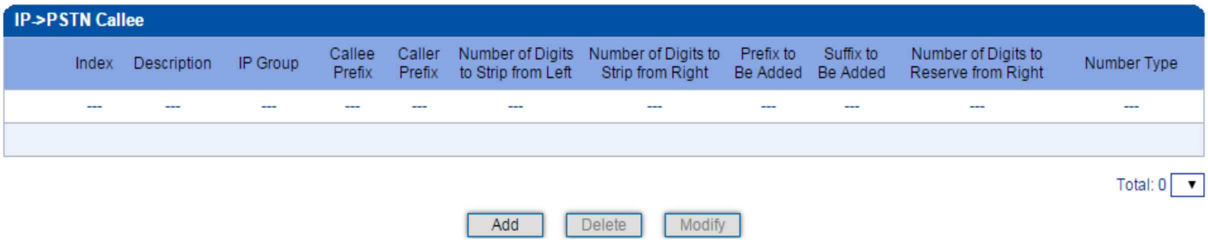


Parameter	**Explanation**
Index	The index of this PSTN →PSTN caller number manipulation, from 0 to 127. Each index cannot be used repeatedly.
Description	The description of this PSTN →PSTN caller number manipulation
PSTN Group	Select a PSTN group. The caller number will be manipulated when a call uses 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

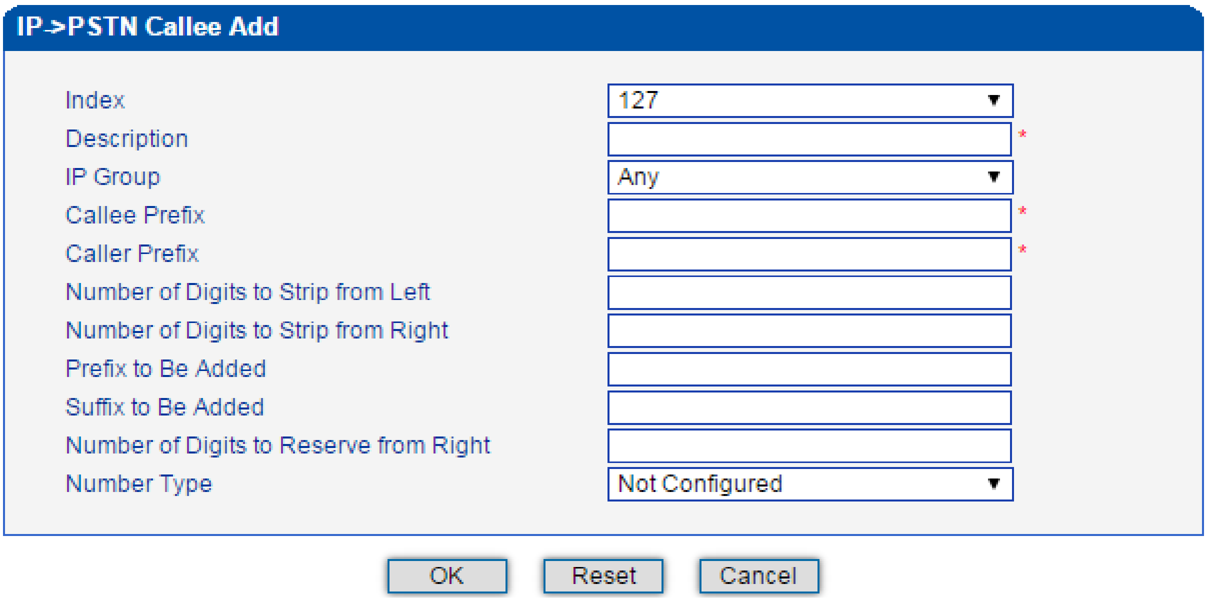
Presentation Indicator	If "Allowed" is selected, the calling number will be presented. If "Restricted" is selected, the calling number will not be presented. If "Not Config" is selected, the parameter does not work.
Number Type	The type of the caller number. Options include 'Not Config', 'International', 'National', 'Unknown', 'Network Specific', 'Subscriber' and 'Abbreviated'

4.14.5 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.

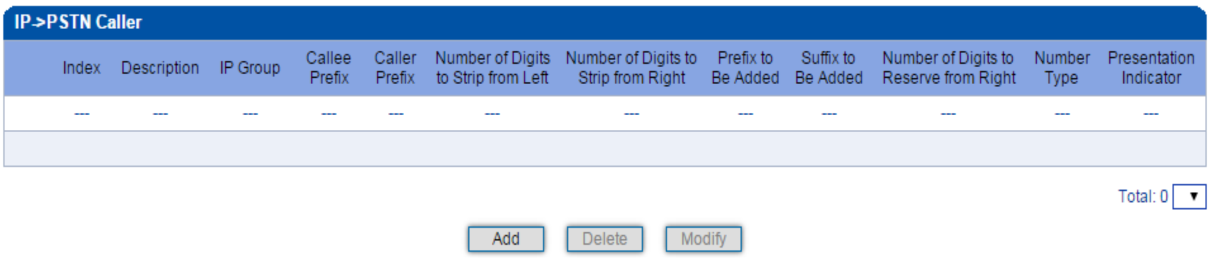


Parameter	Explanation

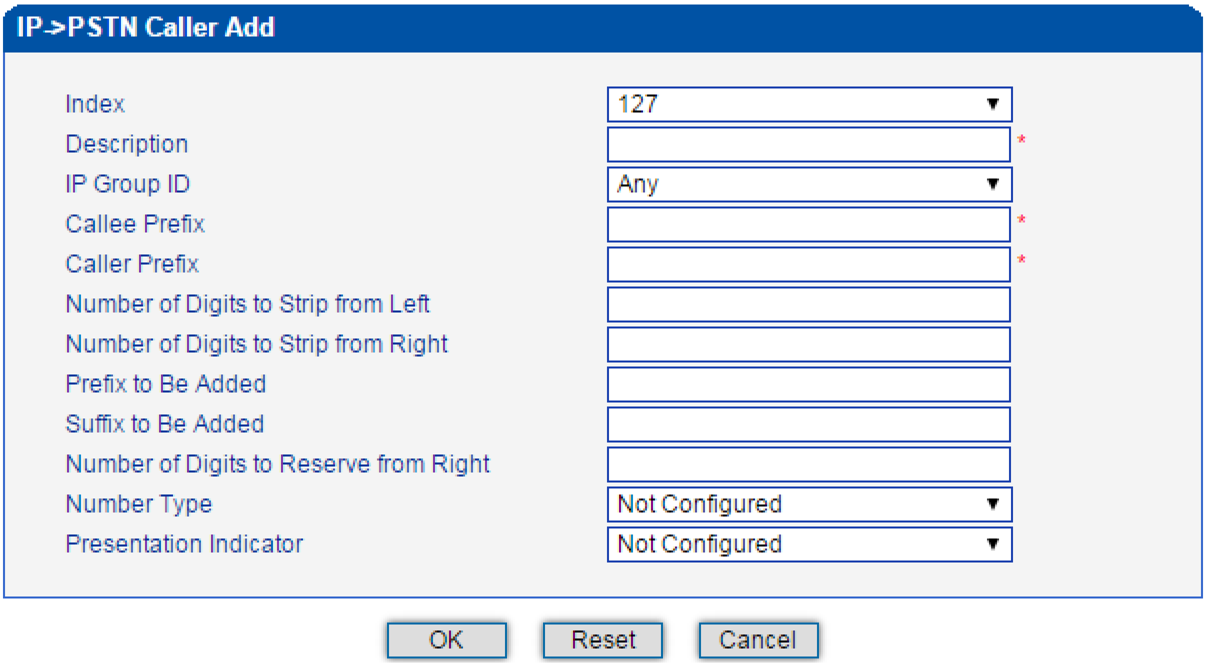
Index	The index of this IP →PSTN callee number manipulation, from 0 to 127. Each index cannot be used repeatedly.
Description	The description of this IP →PSTN callee number manipulation
IP Group	Select an IP group. The callee number will be manipulated when a call uses a trunk of this IP group, actual 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'

4.14.6 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.

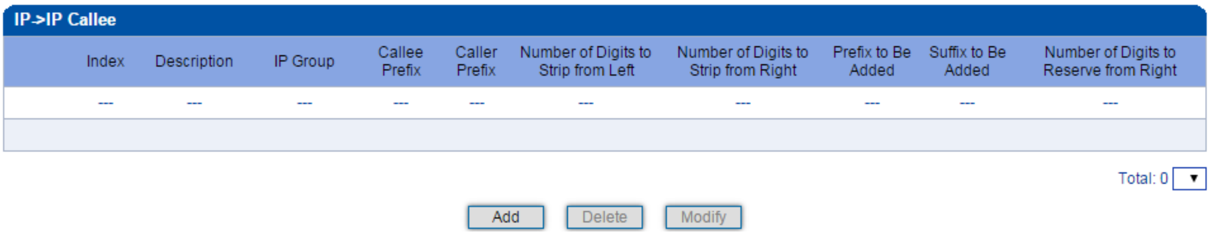


Parameter	Explanation
Index	The index of this IP →PSTN caller number manipulation, from 0 to 127. Each index cannot be used repeatedly.
Description	The description of this IP →PSTN caller number manipulation
IP Group	Select an IP group. The caller number will be manipulated when a call uses a trunk of 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.

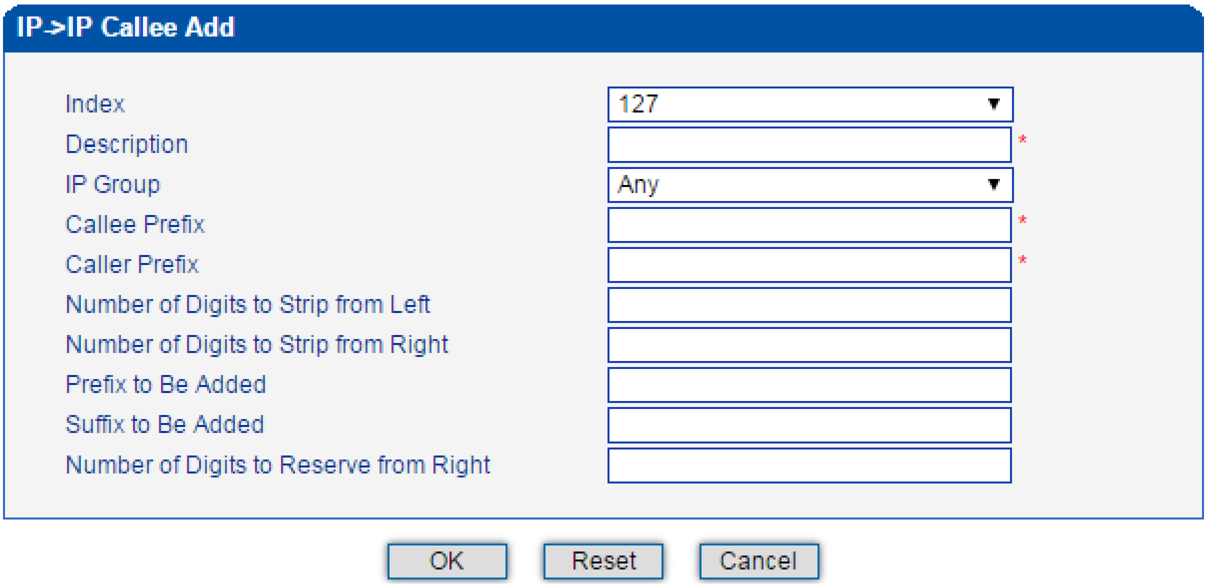
Number of Digits to Reserve from Right	The number of the retained digits which. are counted from the right of the caller number
Presentation Indicator	If “Allowed” is selected, the calling number will be presented. If “Restricted” is selected, the calling number will not be presented. If “Not Config” 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’

4.14.7 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.

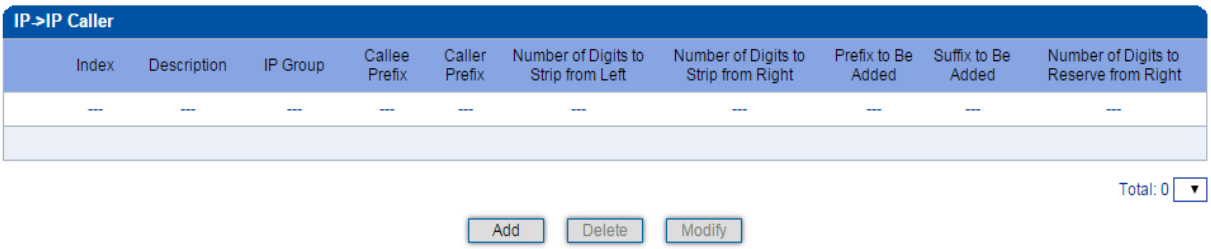


Parameter	**Explanation**

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

4.14.8 IP → IP Caller

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

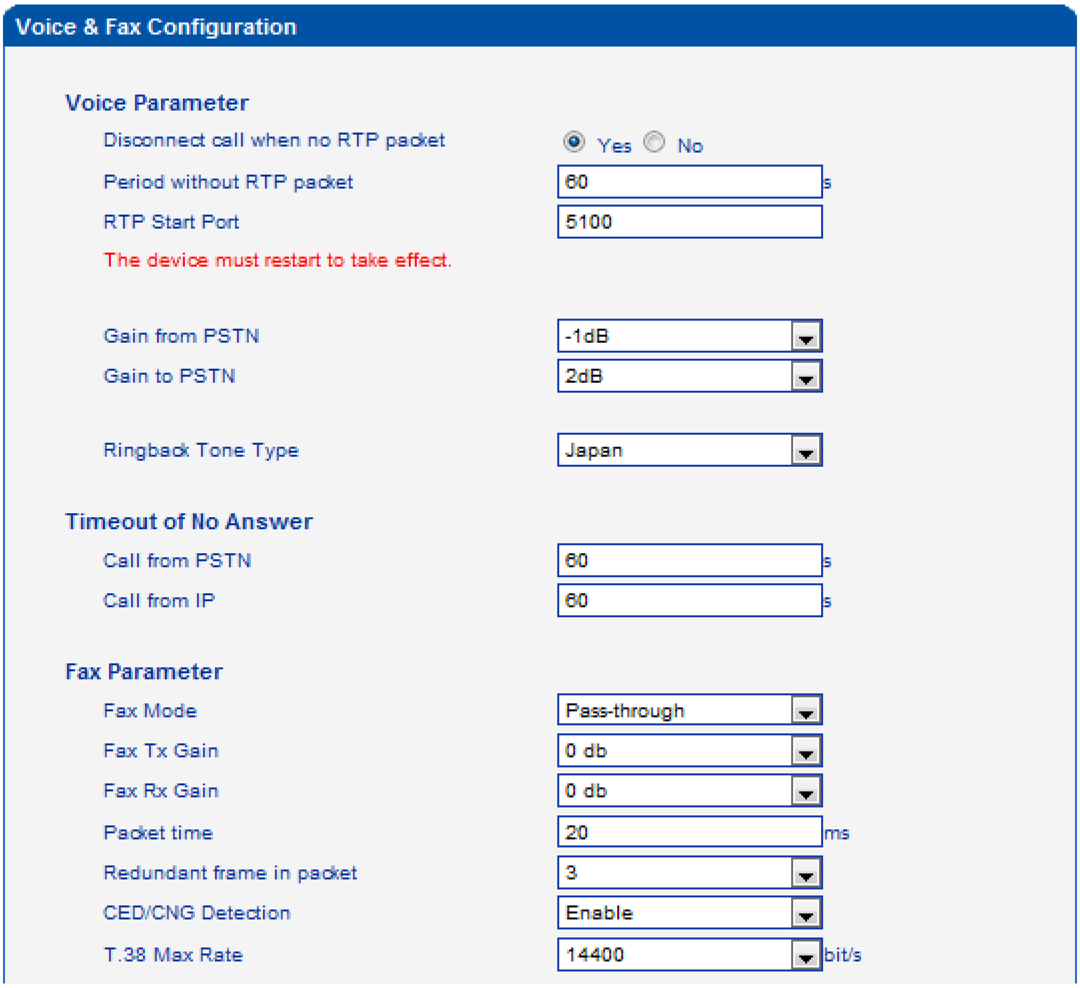


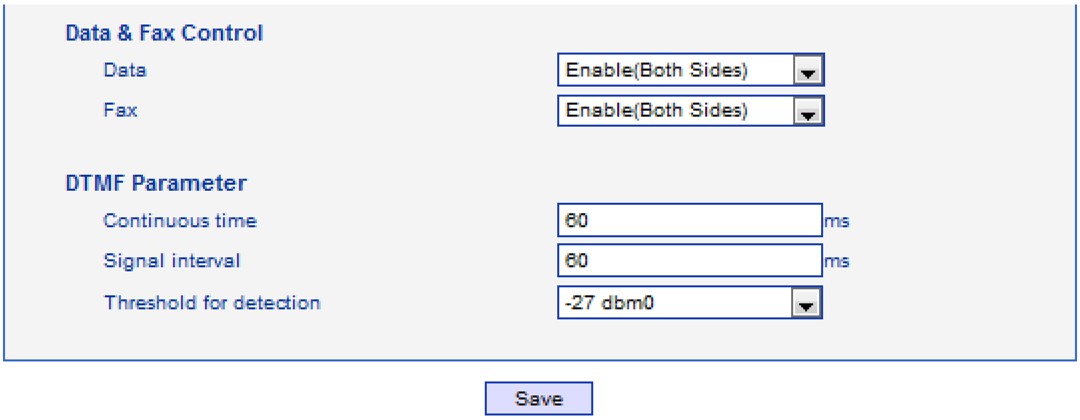
Click **Add**, and the following interface will be displayed.

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

4.15 Voice & Fax



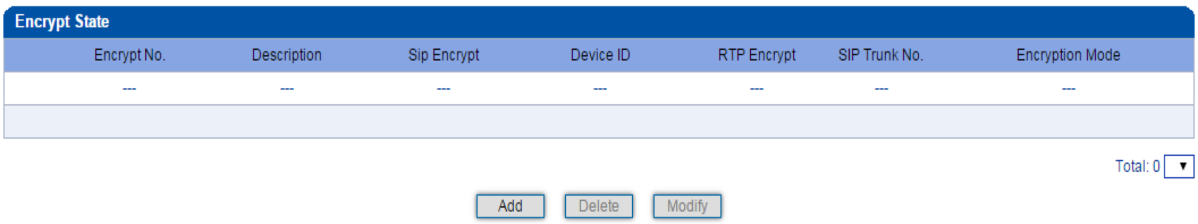


Belong to	Parameter	Explanation
Voice Parameter	Disconnect call when no RTP packet	Options include ‘Yes’ and ‘No’. If ‘Yes’ is selected, the call will be disconnected when it is detected that the call’s silence time is longer than the set maximum time without receiving RTP packets.
	Period without RTP 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 Mode	Options include T.38, Pass-through and Adaptive. Default value is T.38.

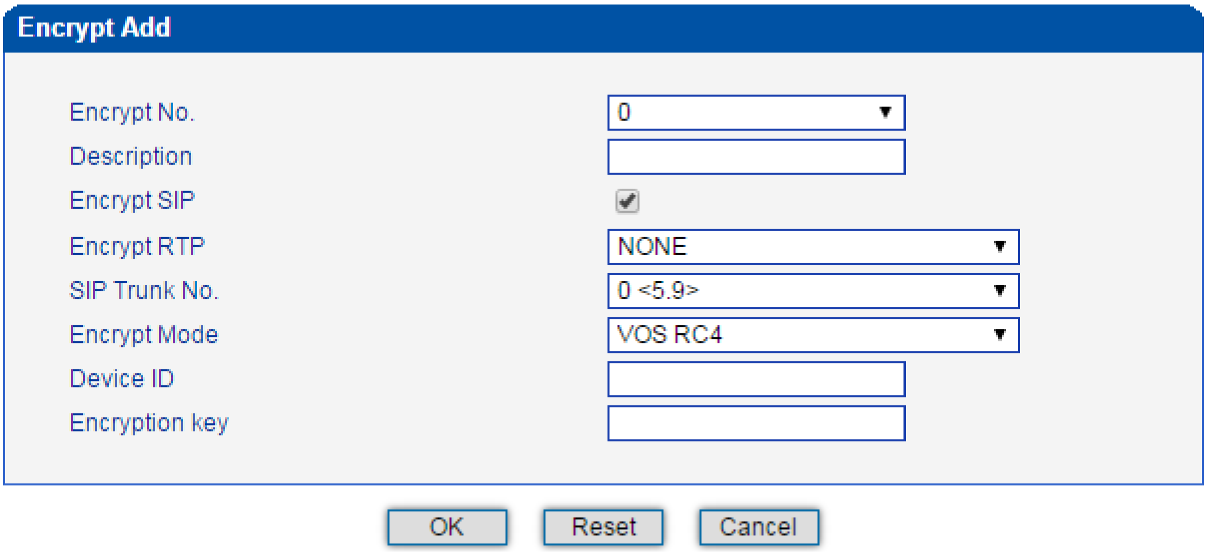
Fax Parameter		Adaptive means auto negotiate with peer side.
	Fax Tx Gain	Gain of sending a fax
	Fax Rx Gain	Gain of receiving a fax
	Packet time	The time for data packing
	Redundant frame in Packet	The length of frame in RTP packet
	CED/CNG Detection	Whether to detect CED/CNG
Data & Fax Control	Data	Whether to enable voice data service on the MTG200
	Fax	Whether to enable fax service on the MTG200
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

4.16 Encrypt Config

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



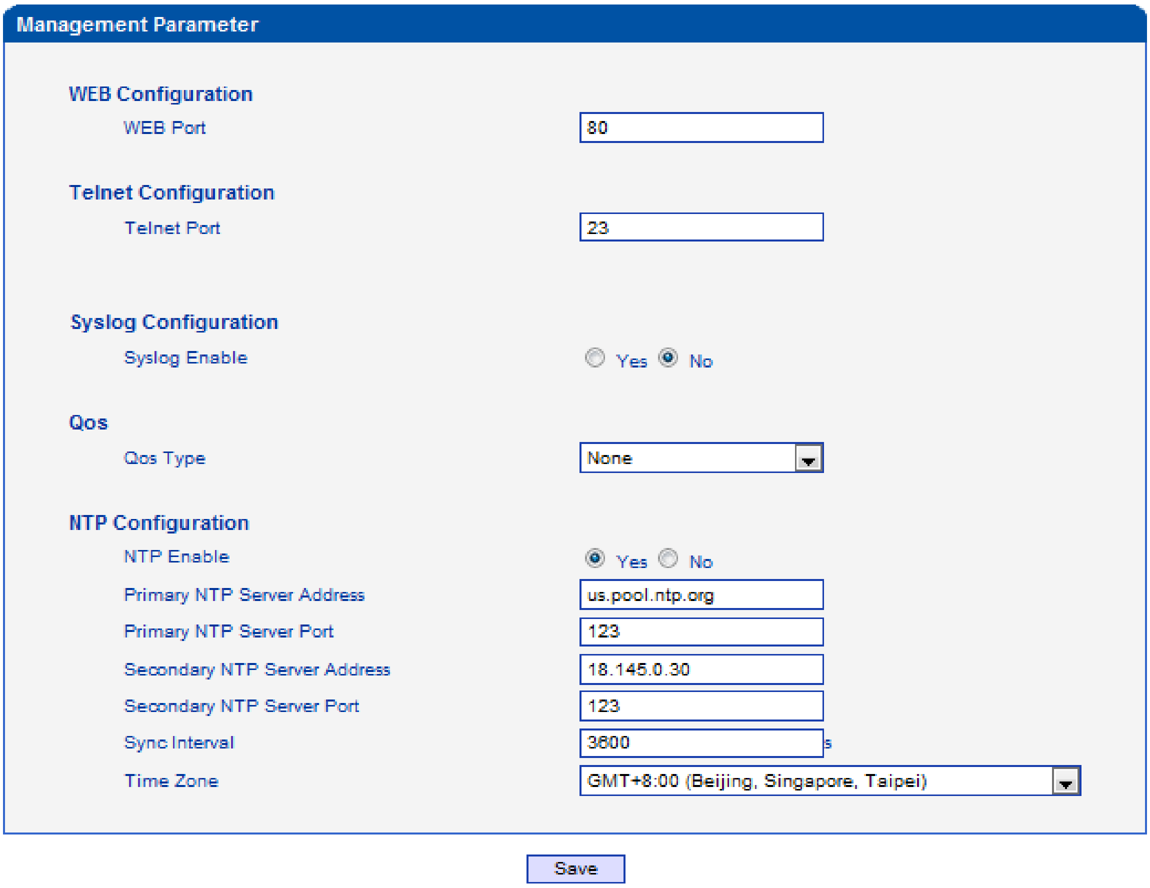
Click **Add**, and the following interface will be displayed.



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

4.17 Maintenance

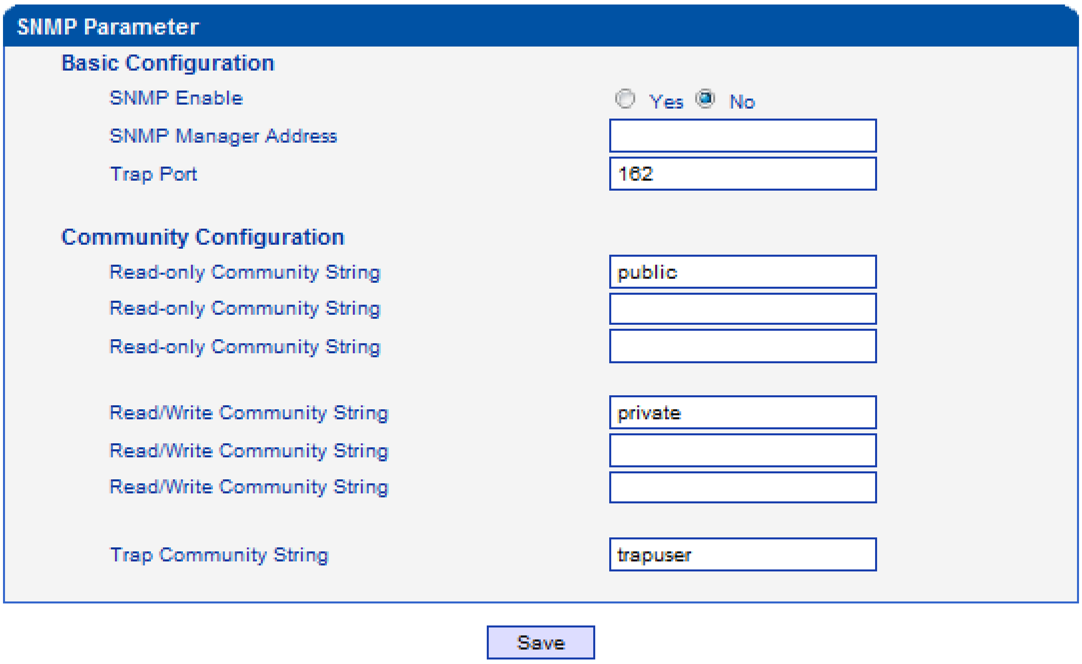
4.17.1 Management Parameter



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.
NTP Configuration	NTP Enable	Whether to enable NTP (network time protocol)
	Primary NTP Server Address	The IP address of primary NTP server
	Primary NTP Server Port	The port of Primary NTP Server
	Secondary NTP Server Address	The IP address of secondary NTP server
	Secondary NTP Server Port	The port of secondary NTP Server
	Sync Interval	The tine interval to synchronize NTP
	Time Zone	Local time zone

4.17.2 SNMP Parameter



SNMP Enable	Whether to enable SNMP (Simple Network Management Protocol)
SNMP Manager Address	IP address of network management server
Trap Port	Default trap port is 162
Read-only Community String	Define a read-only community
Read/Write Community String	Define a read/write community

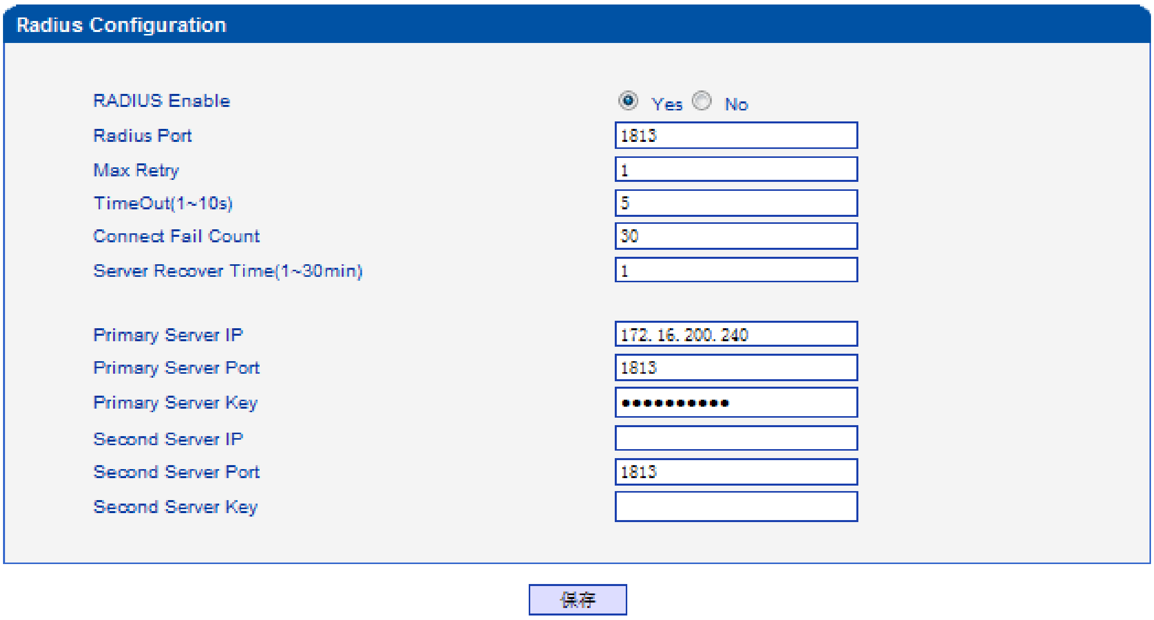
Trap Community String	Define trap community

Note:

After completing the configurations, please restart the device for the configurations to take effect.

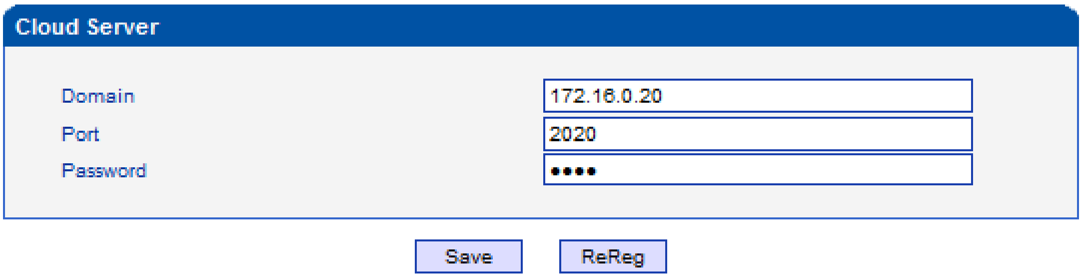
User can manage or configure the gateway on remote NM server through SNMP. But for security consideration, It is recommended that this option is opened only when there is a need.

4.17.3 Radius Parameter



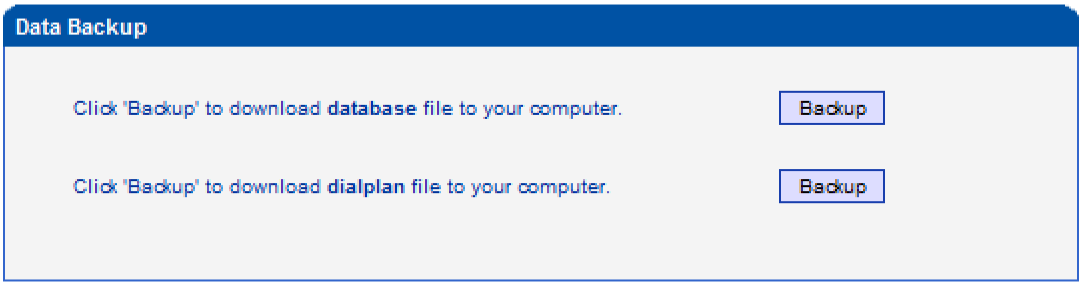
RADIUS Enable	Whether to enable RADIUS (Remote Authentication Dial In User Service)
RADIUS Port	Listening port of RADIUS
Max Retry	Number of retries
Timeout	Timeout for retry
Connect Fail Count	The number of connection failures
Server Recover Time	The time for the server to recover

4.17.4 Cloud Server



4.17.5 Data Backup

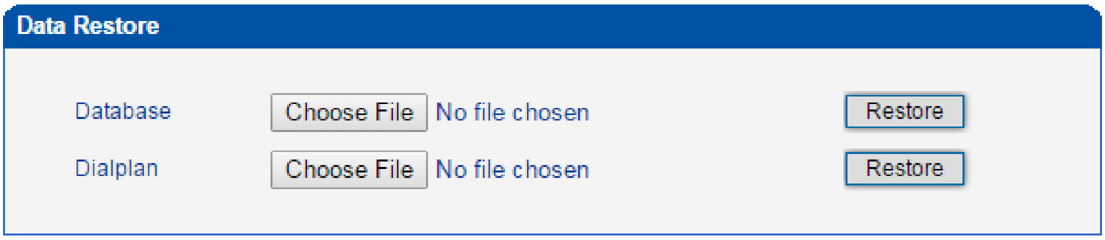
On the **Data Backup** interface, you can click **Backup** to download database file and dialplan file.



4.17.6 Data Restore

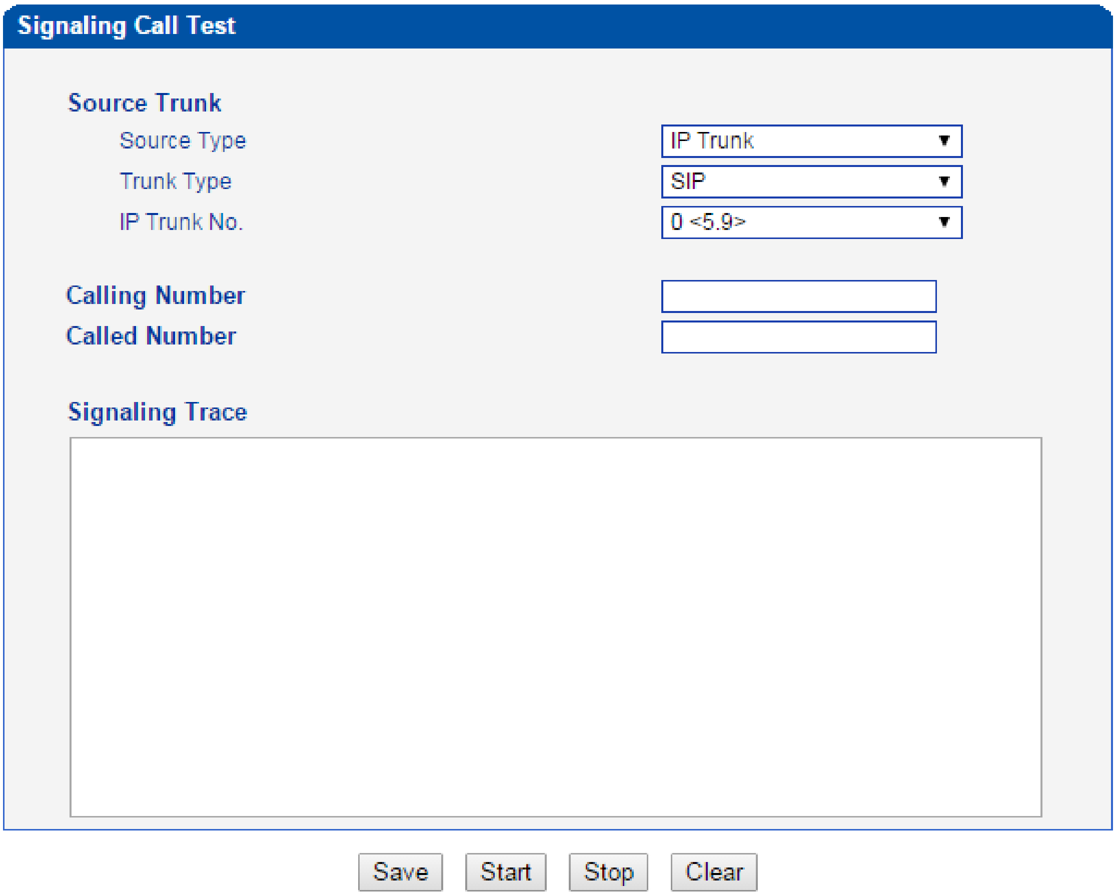
On the **Data Restore** interface, you can restore database and dialplan. If you upload a file that contains default configurations, the MTG200 will be restored to default configurations. You can also upload a dialplan file to restore dialing rules.

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



4.17.7 Signaling Call Test

On the **Signaling Call Test** interface, you can test whether the signaling of a call is successfully connected or not. 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.



4.17.8 Version Information

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

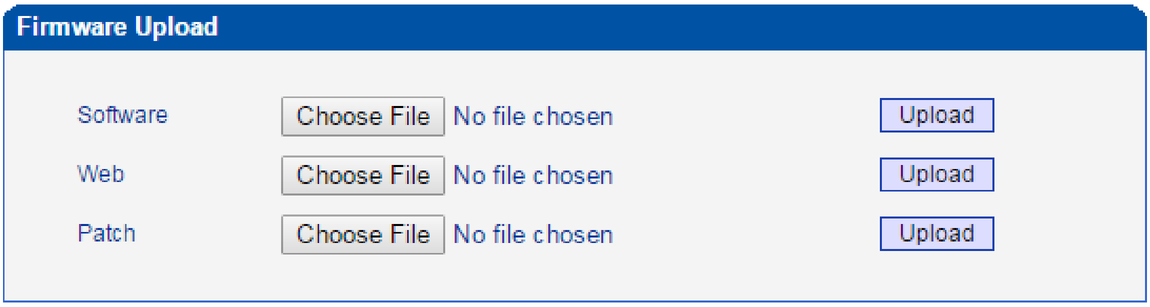
Version Information

File Type	Version	Date Built	Time Built
Software	2.03.05.03	2016-04-13	11:05:54
Database	1.09.47	2016-04-11	17:04:17
Web	2.03.05.03	2016-04-13	11:06:45

Refresh

4.17.9 Firmware Upgrade

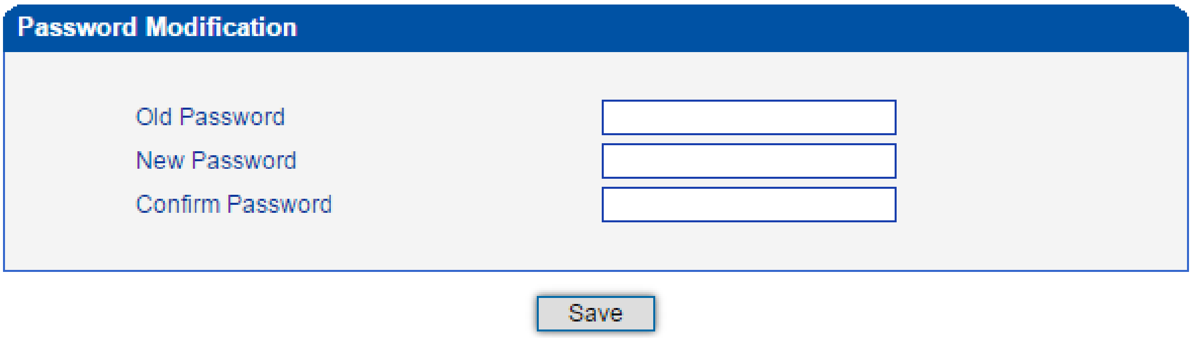
On the **Applications Upload** interface, you can upload files to upgrade the software, the Web, and the Modfile of MTG200. If you select 'Package', it means the upgrading files of the software and Web are packaged and then uploaded.



4.17.10 Password Modification

On the **Password Modification** interface, you can modify password for logging in the MTG200 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.



4.17.11 Device Restart

Click the Restart button, and you can restart the MTG200 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
RADIUS	Remote Authentication Dial In User Service
NTP	Network Time Protocol

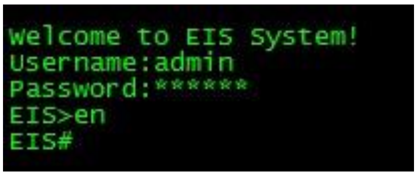
6 Commands

6.1 Troubleshooting and Command Lines

This is a section for some customers who need more details of E1/T1 gateway with command lines. To make sure the system runs successfully, we advise customers to set E1/T1 gateway by GUI. In this manual, some topics such as how to check the IP, signaling and call conversation are covered.

6.1.1 Basic Command

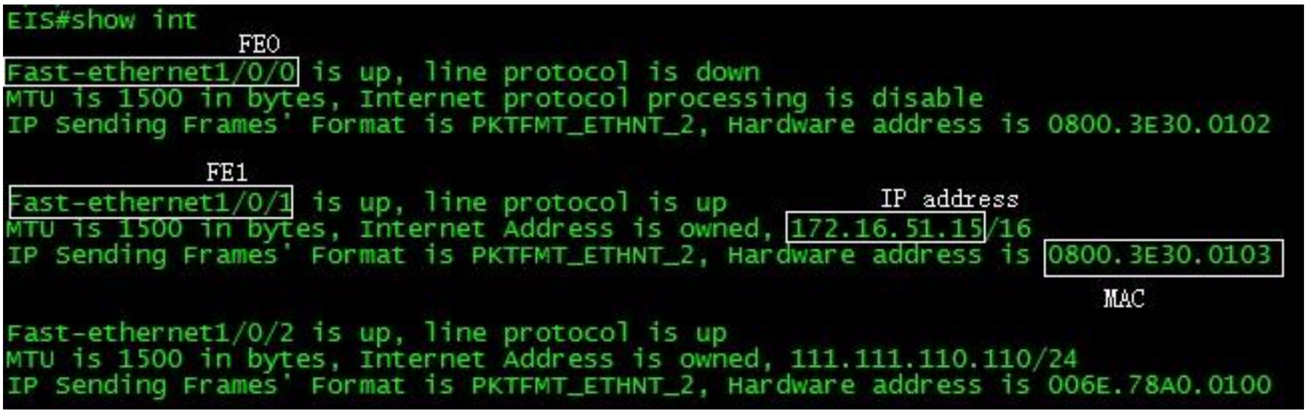
Run system tool --- Telnet to log into the gateway. After entering **username** and **password**, run command **en** to activate the privileged commands.



```
Welcome to EIS System!
Username:admin
Password:******
EIS>en
EIS#
```

6.1.2 Show IP address

Run the command **show int**, the output shows the port names of FE0 and FE1, as well as IP address and MAC address.




```
EIS#show int
Fast-ethernet1/0/0 is up, line protocol is down
MTU is 1500 in bytes, Internet protocol processing is disable
IP Sending Frames' Format is PKTFMT_ETHNT_2, Hardware address is 0800.3E30.0102

Fast-ethernet1/0/1 is up, line protocol is up
MTU is 1500 in bytes, Internet Address is owned, 172.16.51.15/16
IP Sending Frames' Format is PKTFMT_ETHNT_2, Hardware address is 0800.3E30.0103

Fast-ethernet1/0/2 is up, line protocol is up
MTU is 1500 in bytes, Internet Address is owned, 111.111.110.110/24
IP Sending Frames' Format is PKTFMT_ETHNT_2, Hardware address is 006E.78A0.0100
```

6.1.3 Show CPU performance

```
EIS#show perf
    performance now :11
    performance 5s  :10
    performance 60s :11
    performance 600s:10
```

Performance now: cpu load at current time

Performance 5s: cpu load at average 5 seconds

Performance 60s: cpu load at average 60 seconds

Performance 600s: cpu load at average 600 seconds

6.1.4 3.4 Show ss7 status

Run the command **show ss7 sta**, the output is shown as follows:

```
EIS#show ss7 sta
  grpId  linkState  mainLink  backupLink  currentCalls  maxCalls  failCalls  totalCalls  failRatio
----------------------------------------------------------------------------------------------------
  0      OK         ISUP                  7             109       27450      112203      2446%%


                 errors:4400
      current memory usage:70710(bytes)
          max memory usage:100524(bytes)
```

If the system connects with PRI, please run command **show q931 sta.**

6.1.5 3.5 Show ss7 ts

Run the command **ss7 ts**, the status of each channel of each E1/T1 port is displayed.

```
EIS#show ss7 ts
  E1Port: 0
  1     2     3     4     5     6     7     8     9     10    11    12    13    14    15
  used  used  used  used  used  used  used  used  used  used  used  free  free  free  free
  16    17    18    19    20    21    22    23    24    25    26    27    28    29    30    31
  free  free  free  free  free  free  free  free  free  free  free  free  free  free  free  free
--------------------------------------------------------------------------------------------
  E1Port: 1
  1     2     3     4     5     6     7     8     9     10    11    12    13    14    15
  free  free  free  free  free  free  free  free  free  free  free  used  free  free  free
  16    17    18    19    20    21    22    23    24    25    26    27    28    29    30    31
  used  free  free  free  free  free  free  free  free  free  free  free  free  free  free  free
--------------------------------------------------------------------------------------------
  E1Port: 2
  1     2     3     4     5     6     7     8     9     10    11    12    13    14    15
  free  free  free  free  free  free  free  free  free  free  free  free  free  free  free
  16    17    18    19    20    21    22    23    24    25    26    27    28    29    30    31
  free  free  free  free  free  free  free  free  free  free  free  free  free  free  free  free
--------------------------------------------------------------------------------------------
  E1Port: 3
  1     2     3     4     5     6     7     8     9     10    11    12    13    14    15
  free  free  free  free  free  free  free  free  free  free  free  free  free  free  free
  16    17    18    19    20    21    22    23    24    25    26    27    28    29    30    31
  blck  free  free  free  free  free  free  free  free  free  free  free  free  free  free  free
--------------------------------------------------------------------------------------------

  grpNo[0] free ts total: 110
  grpNo[0] used ts total: 13
  grpNo[0] blck ts total: 1
```

Note: This is not available for PRI

6.1.6 3.6 Block ss7 ts

Enter config mode by running command **^config**

```
EIS#
EIS#^config
EIS(config)#
```

Block entire e1

Example:

If you want to block port 2, run the command **busy –cic 2**, and then the system will disable port 2 into a locked status.

```
EIS(config)#busy-cic 2
```

Unblock entire e1

Example:

If you want to unblock port 2 or to actiate the port 2, please run the command **free –cic 2**.

```
EIS(config)#free-cic 2
```

Block specified ts

Example:

If you want to block ts 3 in port 2 or to disable the ts 3 in port 2, run the command **busy–cic 2 3**

```
EIS(config)#busy-cic 2 3
```

Unblock specified ts

Example:

If you want to unblock ts 3 in port 2 or to enable the ts 3 in port 2, run the command **free –cic 2 3**

```
EIS(config)#free-cic 2 3
```

You can check the block status by running **show ss7 ts**

6.1.7 3.7 Show ss7/PRI/cc call information

```
EIS#show ss7 call
grpId: interface ID   userId: CC call ID   callId: SS7 call ID

----------------------------------------------------------------------------------------------------
STATISTICS INFORMATION:
                                      ss7 grpId = 0
                                      ss7 state = OK
                               current call num = 9
                          call num at same time = 109
                                 total call num = 112213
                          total reject call num = 27450
                                   reject ratio = 2446%%

CALL PROCESS INFORMATION:
  grpId userId callId currState time   e1 ts   in/out     calling        called         transNum
----------------------------------------------------------------------------------------------------
  0     2473   101    talking   03:03  0  1    outgoing   48303001       32232050
  0     2443   102    talking   08:32  0  2    outgoing   48303025       42271497
  0     2487   103    talking   00:40  0  3    outgoing   48302541       48200315
  0     2353   104    talking   27:19  0  4    outgoing   48303001       36122170
  0     2479   105    talking   02:17  0  5    outgoing   48303024       42224706
  0     2489   106    release   00:00  0  6    outgoing                  42369583
  0     2431   108    talking   12:16  0  8    outgoing   48303001       22247653
  0     2491   109    release   00:00  0  9    outgoing                  42369583
  0     2451   10c    talking   07:27  0  12   outgoing   48303043       42249070

online total calls: 9
```

If the system connects with PRI, please run **show q931 call**

Customer also can run **show cc call** to list all the activated calls with SS7/PRI (cc = call control)

6.1.8 3.8 Debug call (call control log analyze):

debug call control(recommend)

Set the track condition

If you want to debug all calls, run the command **debug cc detail all**.

```
EIS(config)#debug cc detail all
 Set successfully! current:0
```

Or debug a call by the called or calling number

```
EIS(config)#debug cc detail called 1234567
 Set successfully! current:1
```

```
EIS(config)#debug cc detail calling 987654321
 Set successfully! current:2
```

(replace the called/calling number by yours)

Customer can check the tracking condition by running **debug cc show.**

```
EIS(config)#debug cc show
 Type    TermType DevNo PortNo Target
 ------------------------------------------
 Detail  Called   65535 65535  1234567
 Detail  Calling  65535 65535  987654321

 Trace num:2  All trace:0
```

And then exit config mode, enter into ada mode to turn on a port.

```
EIS(config)#ex
EIS#^ada
EIS(ada)#[107-03:25:55:570]ADA CONNECTED ...,WELCOME!

EIS(ada)#turnon 27
EIS(ada)#
```

(ex = exit)

Run **Cancel debug cc** to turn off the debug mode for cc all.

```
EIS(ada)#turnoff 27
EIS(ada)#ex
EIS#^config
EIS(config)#no debug cc all

 Set successfully! current:0
```

6.1.9 Example 1 : One succ call from IP to PSTN:

EIS(ada)#[069-14:18:49:710]ST: <-1,Sip-t,2,65535,987654321,idle> **<<== SIP_CALL_INVITE, Local:1234567@172.16.51.10, Peer:987654321@172.16.51.10, Std Sdp:v=0**

(note: receive a call from siptrunk)

o=- 12949395404797000 1 IN IP4 172.16.100.172

s=CounterPath X-Lite 4.0

c=IN IP4 172.16.100.172

t=0 0

a=ice-ufrag:2c37f5

a=ice-pwd:a0b4dc8cc787732f66e9e625e69dbbfd

m=audio 50832 RTP/AVP 107 0 8 101

a=rtpmap:107 BV32/16000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

a=sendrecv

a=candidate:1 1 UDP 659136 172.16.100.172 50832 typ host

[070-14:18:49:710]ST: <5,Sip-t,2,65535,987654321,idle> ==>> CC_ST_SETUP, ccb:5, user type:0(Norm), calling:987654321, longnum:987654321, trunkGrpId:255, profileId:255, std sdp:v=0

o=- 12949395404797000 1 IN IP4 172.16.100.172

s=CounterPath X-Lite 4.0

c=IN IP4 172.16.100.172

t=0 0

a=ice-ufrag:2c37f5

a=ice-pwd:a0b4dc8cc787732f66e9e625e69dbbfd

m=audio 50832 RTP/AVP 107 0 8 101

a=rtpmap:107 BV32/16000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

a=sendrecv

a=candidate:1 1 UDP 65913

[071-14:18:49:710]ST: <Sip-t,2,65535,987654321> ====Processed: SIP_CALL_INVITE

[072-14:18:49:710]ST: cr, no:9, ccb:5, State:1(init), cause:0(CCS_NONE(无原因值)), redirect:0

[073-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> <<== CC_ST_SETUP, cr:9, calling:987654321, longNum:987654321, dial:1234567, num_ok:1,calltype:2(msg), rtsType:0, callType:2(ccb), fax dest<65535,65535>, trunkGrpId:255, profileId:255, sigToneTyp:0, std sdp:v=0

o=- 12949395404797000 1 IN IP4 172.16.100.172

s=CounterPath X-Lite 4.0

c=IN IP4 172.16.100.172

t=0 0

a=ice-ufrag:2c37f5

a=ice-pwd:a0b4dc8cc787732f66e9e625e69dbbfd

m=audio 50832 RTP/AVP 107 0 8 101

a=rtpmap:107 BV32/16000

a=rtpmap:101

[074-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> **predispose start calling :987654321 called:1234567**!

[075-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> predispose end calling :987654321 called:1234567!

[076-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> source user auth:0x6, is fxo call in auth pass:0

[077-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> route type:3(Out route) -- before cc number analysis.

[078-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> **[IP2tel]match route succ! srcIpGrpId:0, dstTrkGrpId:0, ChnSelMode:0, callingProfId:0, srcIpGrpId:0.**

(**note :** mach routing successful , if call failed, you can check this if source IP Group id and destination Trunk Group Id is if the same with you expect, if not , please check the routing configure)

[079-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> **[before manipulate number]calling:987654321, called:1234567, longNum:987654321, anl called:1234567, isRouteAfNumManip:0, callerNumTyp:255, calledNumTyp:255, presentId:0.**

[080-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> **[after manipulate number]calling:987654321, called:1234567, longNum:987654321, anl called:1234567, callerNumTyp:0, calledNumTyp:0, presentId:0.**

(**note:** if configure number manipulation, can check the manipulate result here)

[081-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> analysis successfully, service:0(normal), bill:4(normal), route:3(rts_out), dest_term:8(Ss7), dest_dev:65535, dest_port:65535, dest_grp:65535, called:1234567 !

[082-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> number convert, old calling:987654321, old called:1234567

[083-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> before trans num process! caller:987654321, disNum:, called:1234567, g_ulIsTransOrgCalleeNum:0, g_ulNumTransType:1, g_ulAllowMobileTransfer:0!

[084-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> number convert, new calling:987654321, dis num:, new called:1234567

[085-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> ==>> CC_ST_PROCEEDING, called:1234567

[086-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> calling :0.0.0.0   called:255.255.255.255

[087-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> is need reflect:0, callingProfId:0.

[088-14:18:49:710]CC: <5,Ss7,65535,65535,,idle> ==>> CC_ST_SETUP, cr:10, calling:987654321, longNum:987654321, dial:1234567, OrgCallee:, num_ok:1, trunkGrpId:0, profileId:255, isForceReflect(ccb):0, ringback2IP:1,std sdp:v=0

o=- 12949395404797000 1 IN IP4 172.16.100.172

s=CounterPath X-Lite 4.0

c=IN IP4 172.16.100.172

t=0 0

a=ice-ufrag:2c37f5

a=ice-pwd:a0b4dc8cc787732f66e9e625e69dbbfd

m=audio 50832 RTP/AVP 107 0 8 101

a=rtpmap:107 BV32/16000

a=rtpmap:101 telephone-event/8000

a=fmtp:10

[089-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> ccb state change from 'idle' to 'proceeding', ccb no:5

[090-14:18:49:710]CC: <Sip-t,2,65535>, <Ss7,65535,65535>, ====Processed: CC_ST_SETUP

[091-14:18:49:710]CCB: no:5, cr1:9, cr2:10, State:4(proceeding), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[092-14:18:49:710]ST: <5,Sip-t,2,65535,987654321,init> <<== CC_ST_PROCEEDING, Std Sdp:, Priv Sdp:

[093-14:18:49:710]ST: <Sip-t,2,65535,987654321> ====Processed: CC_ST_PROCEEDING

[094-14:18:49:710]ST: cr, no:9, ccb:5, State:4(out_proc), cause:0(CCS_NONE(无原因值)), redirect:0

[095-14:18:49:710]ST: <5,,65535,65535,,idle> <<== CC_ST_SETUP, calling:987654321, long:987654321, dial:1234567, send_ok:1, Std Sdp:v=0

o=- 12949395404797000 1 IN IP4 172.16.100.172

s=CounterPath X-Lite 4.0

c=IN IP4 172.16.100.172

t=0 0

a=ice-ufrag:2c37f5

a=ice-pwd:a0b4dc8cc787732f66e9e625e69dbbfd

m=audio 50832 RTP/AVP 107 0 8 101

a=rtpmap:107 BV32/16000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

a=sendrecv

a=candidate:1 1 UDP 659136 172.16.100.172 50832 typ host

a=candidate:

[096-14:18:49:710]ST: <5,Ss7,65535,65535,,idle> **==>> CC_SETUP_REQ, index:10, if:65535, trunkGrp:0, calling:987654321, called:1234567, callingTyp:0, calledTye:0, presentId:0, trans:**

(**note:** setup a call to pstn)

[097-14:18:49:710]ST: <Ss7,65535,65535,> ====Processed: CC_ST_SETUP

[098-14:18:49:710]ST: cr, no:10, ccb:5, State:6(present), cause:0(CCS_NONE(无原因值)), redirect:0

[099-14:18:49:710]ST: <Ss7,65535,65535,> ====Processed: CC_ST_SETUP

[100-14:18:49:710]ST: cr, no:10, ccb:5, State:6(present), cause:0(CCS_NONE(无原因值)), redirect:0

[101-14:18:49:760]ST: <5,Ss7,65535,65535,,present> **<<== CC_ALERTING_IND, q931id:773, if:2, calling:, called: org_called:, e1:10, ts:5, callingTyp:0, calledTyp:0, presentationInd:0, send_ok:0, cause:0(OK)**

(**note:** the other side in pstn receive the setup msg)

[102-14:18:49:760]ST: <5,Ss7,65535,65535,,present> ==>> CC_ST_SETUP_ACK, cause:0(CCS_NONE(无原因值))

[103-14:18:49:760]ST: <5,Ss7,65535,65535,,in_proc> Tm alloc, e1:10, ts:5

[104-14:18:49:760]ST: <5,Ss7,65535,65535,,in_proc> Tm crcx, connid:196758, ip:172.16.100.172, port:50832, algo:0, pkt:20, zip:0, ZipEia:65535, crypt:0, tcp:0, p2pV2:0, telEventPayload:101, dtmfMode:0.

[105-14:18:49:760]ST: <5,Ss7,65535,65535,,in_proc> play ringBack to IP.

[106-14:18:49:760]ST: <5,Ss7,65535,65535,,in_proc> ==>> CC_ST_ALERTING, ccb:5, user type:0(Norm), calling:987654321, longnum:987654321, std sdp:v=0

o=call 10000 20000 IN IP4 172.16.51.10

s=-

c=IN IP4 172.16.51.10

t=0 0

m=audio 5102 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

a=ptime:20

, priv sdp:

[107-14:18:49:760]ST: <Ss7,65535,65535,> ====Processed: CC_ALERTING_IND

[108-14:18:49:760]ST: cr, no:10, ccb:5, State:8(recving), cause:0(CCS_NONE(无原因值)), redirect:0

[109-14:18:49:770]CC: <5,Ss7,65535,65535,,proceeding> <<== CC_ST_SETUP_ACK, cause:0(CCS_NONE(无原因值)), longnum:

[110-14:18:49:770]CC: <5,Sip-t,2,65535,,proceeding> ccb state change from 'proceeding' to 'wait ack', ccb no:5

[111-14:18:49:770]CC: <Sip-t,2,65535>, <Ss7,65535,65535>, ====Processed: CC_ST_SETUP_ACK

[112-14:18:49:770]CCB: no:5, cr1:9, cr2:10, State:5(wait ack), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[113-14:18:49:770]CC: <5,Ss7,65535,65535,,wait ack> <<== CC_ST_ALERTING, std sdp:v=0

o=call 10000 20000 IN IP4 172.16.51.10

s=-

c=IN IP4 172.16.51.10

t=0 0

m=audio 5102 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

a=ptime:20

, priv sdp:

[114-14:18:49:770]CC: <5,Sip-t,2,65535,,wait ack> ccb state change from 'wait ack' to 'alerting', ccb no:5

[115-14:18:49:770]CC: <5,Sip-t,2,65535,,alerting> route type:3(rts_out), called term type:8(Ss7)

[116-14:18:49:770]CC: <5,Ss7,65535,65535,,alerting> ==>> CC_ST_ALERTING, std sdp:v=0

o=call 10000 20000 IN IP4 172.16.51.10

s=-

c=IN IP4 172.16.51.10

t=0 0

m=audio 5102 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

a=ptime:20

, priv sdp:

[117-14:18:49:770]CC: <Sip-t,2,65535>, <Ss7,65535,65535>, ====Processed: CC_ST_ALERTING

[118-14:18:49:770]CCB: no:5, cr1:9, cr2:10, State:6(alerting), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[119-14:18:49:770]ST: <5,Sip-t,2,65535,987654321,out_proc> <<== CC_ST_ALERTING, Std Sdp:v=0

o=call 10000 20000 IN IP4 172.16.51.10

s=-

c=IN IP4 172.16.51.10

t=0 0

m=audio 5102 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

a=ptime:20

, Priv Sdp:

[120-14:18:49:770]ST: <5,Sip-t,2,65535,987654321,out_proc> ==>> ST_SIP_CALL_PRE_ACCEPT, index:9, local:1234567@172.16.51.10, peer:987654321@172.16.51.10, std sdp:v=0

o=call 10000 20000 IN IP4 172.16.51.10

s=-

c=IN IP4 172.16.51.10

t=0 0

m=audio 5102 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

a=ptime:20

, priv sdp:

[121-14:18:49:770]ST: <Sip-t,2,65535,987654321> ====Processed: CC_ST_ALERTING

[122-14:18:49:770]ST: cr, no:9, ccb:5, State:5(deliver), cause:0(CCS_NONE(无原因值)), redirect:0

EIS(ada)#[123-14:18:52:470]**ST: <5,Ss7,65535,65535,,recving> <<== CC_SETUP_CFM, q931id:773, if:2, calling:, called: org_called:, e1:10, ts:5, callingTyp:0, calledTyp:0, presentationInd:0, send_ok:0, cause:0(OK)**

(**note:** called answer the call)

[124-14:18:52:470]ST: <5,Ss7,65535,65535,,recving> connId:0x30096, isPlayLocalRingback2IP:1.

[125-14:18:52:470]ST: <5,Ss7,65535,65535,,recving> ==> CC_ST_CONNECT,

[126-14:18:52:470]ST: <Ss7,65535,65535,> ====Processed: CC_SETUP_CFM

[127-14:18:52:470]ST: cr, no:10, ccb:5, State:9(active), cause:0(CCS_NONE(无原因值)), redirect:0

[128-14:18:52:470]CC: <5,Ss7,65535,65535,,alerting> <<== CC_ST_CONNECT, calling:987654321, long:987654321, called:1234567, calling dial num:1234567, Std Sdp:v=0

o=call 10000 20000 IN IP4 172.16.51.10

s=-

c=IN IP4 172.16.51.10

t=0 0

m=audio 5102 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

a=ptime:20

, Priv Sdp:

[129-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> stop queue seat timer!

[130-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> stop queue timer!

[131-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> clear bill end time(cc connect).

[132-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> stop hint at port:65535 ,connid:4294967295

[133-14:18:52:470]CC: <5,Ss7,65535,65535,,alerting> stop hint at port:65535, connid:4294967295

[134-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> vpbx process flag:0, ippbx process flag:0

[135-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> route type:3(rts_out), called term type:8(Ss7)

[136-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> get bill start time:14-18-52

[137-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> payer info(DevNo-2, PortNo-65535, callDirect-1, termType-Sip-t), Is ccb stpayer.pstPort NULL:yes. Service type(ccb):normal, is need settle:no.

[138-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> ==>> CC_ST_CONNECT, called:1234567

[139-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> ccb state change from 'alerting' to 'active', ccb no:5

[140-14:18:52:470]CC: <Sip-t,2,65535>, <Ss7,65535,65535>, ====Processed: CC_ST_CONNECT

[141-14:18:52:470]CCB: no:5, cr1:9, cr2:10, State:7(active), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[142-14:18:52:470]ST: <5,Sip-t,2,65535,987654321,deliver> <<== CC_ST_CONNECT, Std Sdp:v=0

o=call 10000 20000 IN IP4 172.16.51.10

s=-

c=IN IP4 172.16.51.10

t=0 0

m=audio 5102 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

a=ptime:20

, Priv Sdp:

[143-14:18:52:470]ST: <5,Sip-t,2,65535,987654321,deliver> ==>> SIP_CALL_ACCEPT, index:9, calltype:0 local:1234567@172.16.51.10, peer:987654321@172.16.51.10, std sdp:v=0

o=call 10000 20000 IN IP4 172.16.51.10

s=-

c=IN IP4 172.16.51.10

t=0 0

m=audio 5102 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

a=ptime:20

, priv sdp:, ext:

[144-14:18:52:470]ST: <5,Sip-t,2,65535,987654321,active> start wait peer conn timer, len:15s

[145-14:18:52:470]ST: <Sip-t,2,65535,987654321> ====Processed: CC_ST_CONNECT

[146-14:18:52:470]ST: cr, no:9, ccb:5, State:9(active), cause:0(CCS_NONE(无原因值)), redirect:0

[147-14:18:52:510]ST: <5,Sip-t,2,65535,987654321,active> <<== SIP_ACCEPT_ACK, Index:9, Local:1234567@172.16.51.10, Peer:987654321@172.16.51.10

[148-14:18:52:510]ST: <5,Sip-t,2,65535,987654321,active> ==> CC_ST_CONNECT_ACK

[149-14:18:52:510]ST: <5,Sip-t,2,65535,987654321,active> stop wait peer conn timer

[150-14:18:52:510]CC: <5,Sip-t,2,65535,,active> <<== CC_ST_CONNECT_ACK

[151-14:18:52:510]CC: <Sip-t,2,65535>, <Ss7,65535,65535>, ====Processed: CC_ST_CONNECT_ACK

[152-14:18:52:510]CCB: no:5, cr1:9, cr2:10, State:7(active), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

EIS(ada)#[153-14:19:20:680]ST: <5,Ss7,65535,65535,,active> **<<== CC_DISCONNECT_IND, q931id:773, if:2, calling:, called: org_called:, e1:10, ts:5, callingTyp:0, calledTyp:0, presentationInd:0, send_ok:0, cause:16(正常的呼叫清除)**

(**note:** called disconnect the call)

[154-14:19:20:680]ST: <5,Ss7,65535,65535,,disconn> Tm dlcx, connid:196758

[155-14:19:20:680]ST: <5,Ss7,65535,65535,,disconn> Release the call, cause:CCS_NORM_CLEAR(正常释放)(1) !

[156-14:19:20:680]ST: <5,Ss7,65535,65535,,disconn> ==>> CC_RELEASE_REQ, index:10, if:2, q931_id:773, cause:16

[157-14:19:20:680]ST: <Ss7,65535,65535,> ====Processed: CC_DISCONNECT_IND

[158-14:19:20:680]ST: cr, no:10, ccb:5, State:11(release), cause:1(CCS_NORM_CLEAR(正常释放)), redirect:0

[159-14:19:20:690]ST: <5,Ss7,65535,65535,,release> <<== CC_RELEASE_CFM, q931id:773, if:2, calling:, called: org_called:, e1:10, ts:5, callingTyp:0, calledTyp:0, presentationInd:0, send_ok:0, cause:16(正常的呼叫清除)

[160-14:19:20:690]ST: <5,Ss7,65535,65535,,release> Release the call, cause:CCS_NORM_CLEAR(正常释放)(1) !

[161-14:19:20:690]ST: <5,Ss7,65535,65535,,release> ==> CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放))

[162-14:19:20:690]ST: <5,Ss7,65535,65535,,release> Free CR 10, cause:1(CCS_NORM_CLEAR(正常释放))

[163-14:19:20:690]ST: <,65535,65535,> ====Processed: CC_RELEASE_CFM

[164-14:19:20:690]ST: cr, no:10, ccb:4294967295, State:0(idle), cause:0(CCS_NONE(无原因值)), redirect:0

[165-14:19:20:690]CC: <5,Sip-t,2,65535,,active> [cc release comp]ccb no:5, sub ccb no:4294967295

[166-14:19:20:690]CC: <-1,Ss7,65535,65535,,idle> <<== CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放)))

[167-14:19:20:690]CC: <5,Sip-t,2,65535,,active> State(active) is not match, refuse resel route!

[168-14:19:20:690]CC: <5,Sip-t,2,65535,,active> bill start time:14-18-52, bill end time: 0- 0- 0.

[169-14:19:20:690]CC: <5,Sip-t,2,65535,,active> [bill end time]bill type:normal, service type(ccb):normal, is need settle:no.redirect flag:0, called term type:Ss7, Is ccb stpayer.pstPort NULL:yes.

[170-14:19:20:690]CC: <5,Sip-t,2,65535,,active> ==>> CC_ST_RELEASE, cause:1(CCS_NORM_CLEAR(正常释放))

[171-14:19:20:690]CC: <5,Sip-t,2,65535,,active> ccb state change from 'active' to 'release', ccb no:5

[172-14:19:20:690]CC: <Sip-t,2,65535>, <Ss7,65535,65535>, ====Processed: CC_ST_REL_COMP

[173-14:19:20:690]CCB: no:5, cr1:9, cr2:10, State:9(release), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:1(CCS_NORM_CLEAR(正常释放))

[174-14:19:20:690]ST: <5,Sip-t,2,65535,987654321,active> <<== CC_ST_RELEASE, cause:CCS_NORM_CLEAR(正常释放)

[175-14:19:20:690]ST: <5,Sip-t,2,65535,987654321,active> ==>> SIP_CALL_BYE, index:9, local:1234567@172.16.51.10, peer:987654321@172.16.51.10, cause:CCS_NORM_CLEAR(正常释放)

[176-14:19:20:690]ST: <5,Sip-t,2,65535,987654321,active> ==> CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放))

[177-14:19:20:690]ST: <5,Sip-t,2,65535,987654321,active> Free CR 9, cause:1(CCS_NORM_CLEAR(正常释放))

[178-14:19:20:690]CC: <-1,Sip-t,2,65535,,idle> [cc release comp]ccb no:5, sub ccb no:4294967295

[179-14:19:20:690]CC: <-1,Sip-t,2,65535,,idle> stop queue seat timer!

[180-14:19:20:690]CC: <-1,Sip-t,2,65535,,idle> stop queue timer!

[181-14:19:20:690]**CC: <-1,Sip-t,2,65535,,idle> <<== CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放)))**

(**note:** release complete)

[182-14:19:20:690]CC: <-1,Sip-t,2,65535,,idle> Free CCB 5, cause:1(CCS_NORM_CLEAR(正常释放))

[183-14:19:20:690]CC: <-1,Sip-t,2,65535,,idle> stop queue seat timer!

[184-14:19:20:690]CC: <-1,Sip-t,2,65535,,idle> stop queue timer!

6.1.10 Example 2: One succ call from PSTN to IP:

EIS(ada)#[031-00:14:01:640]**ST: <-1,Ss7,65535,65535,,idle> <<== CC_SETUP_IND, q931id:779, if:2, calling:987654321, called:1234567 org_called:, e1:10, ts:11, callingTyp:0, calledTyp:0, presentationInd:0, send_ok:1, cause:0(OK)**

(**note:** receive a call from pstn)

[032-00:14:01:640]ST: <-1,Ss7,65535,65535,,idle> Can't recognize calling :987654321, with format locolwihtarea:0, longwith0:1

[033-00:14:01:640]ST: <-1,Ss7,2,65535,00000000,idle> Tm alloc succ, e1:10, ts:11, conn id:196782, port:5120

[034-00:14:01:640]ST: <-1,Ss7,2,65535,00000000,idle> @@@ add called:1234567, lines:1

[035-00:14:01:640]ST: <11,Ss7,2,65535,00000000,idle> ==>> CC_ST_SETUP, ccb:11, user type:0(Norm), calling:987654321, longnum:987654321, trunkGrpId:2, profileId:0, std sdp:v=0

o=call 10000 20000 IN IP4 172.16.51.15

s=-

c=IN IP4 172.16.51.15

t=0 0

m=audio 5120 RTP/AVP 4 18 8 0 101

a=rtpmap:4 G723/8000

a=rtpmap:18 G729/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

, priv sdp:a=X-ACrypt

a=X-Tcp

a=X-P2PV2

a=X-P2PDst:67241984.104333337

[036-00:14:01:640]ST: <Ss7,2,65535,00000000> ====Processed: CC_SETUP_IND

[037-00:14:01:640]ST: cr, no:21, ccb:11, State:1(init), cause:0(CCS_NONE(无原因值)), redirect:0

[038-00:14:01:640]CC: <11,Ss7,2,65535,,idle> <<== CC_ST_SETUP, cr:21, calling:987654321, longNum:987654321, dial:1234567, num_ok:1,calltype:7(msg), rtsType:0, callType:7(ccb), fax dest<65535,65535>, trunkGrpId:2, profileId:0, sigToneTyp:0, std sdp:v=0 o=call 10000 20000 IN IP4 172.16.51.15

s=-

c=IN IP4 172.16.51.15

t=0 0

m=audio 5120 RTP/AVP 4 18 8 0 101

a=rtpmap:4 G723/8000

a=rtpmap:18 G729/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0

[039-00:14:01:640]CC: <11,Ss7,2,65535,,idle> predispose start calling :987654321 called:1234567!

[040-00:14:01:640]CC: <11,Ss7,2,65535,,idle> predispose end calling :987654321 called:1234567!

[041-00:14:01:640]CC: <11,Ss7,2,65535,,idle> Invoke cc_pstn_in_proc()!

[042-00:14:01:640]CC: <11,Ss7,2,65535,,idle> PSTN in call process start! called:1234567, pstnInUserGrp:65535, numRecvIsComp:1, isSbnFlow:1.

[043-00:14:01:640]CC: <11,Ss7,2,65535,,idle> search destination port by long number fail!called:1234567, firstCalled:1234567.

[044-00:14:01:640]**CC: <11,Ss7,2,65535,,idle> [before manipulate number]calling:987654321, called:1234567, longNum:987654321, anl called:1234567, isRouteAfNumManip:0, callerNumTyp:0, calledNumTyp:0, presentId:0.**

[045-00:14:01:640]CC: <11,Ss7,2,65535,,idle> [after manipulate number]calling:987654321, called:1234567, longNum:987654321, anl called:1234567, callerNumTyp:0, calledNumTyp:0, presentId:0.

[046-00:14:01:640]CC: <11,Ss7,2,65535,,idle> [tel2IP]match route succ! ipGrpId:3, trkSelMode:0.

[047-00:14:01:640]CC: <11,Ss7,2,65535,,idle> select ip trunk succ! trunkGrpId:3, trunkType:4(Sip trunk), trunkNo:4, trunkPriority:0, calledProfId:0.

(**note :** mach routing successful , if call failed, you can check this if source IP Group id and destination Trunk Group Id is if the same with you expect, if not , please check the routing configure)

[048-00:14:01:640]CC: <11,Ss7,2,65535,,idle> analysis successfully, service:0(normal), bill:4(normal), route:3(rts_out), dest_term:4(Sip-t), dest_dev:4, dest_port:65535, dest_grp:65535, called:1234567 !

[049-00:14:01:640]CC: <11,Ss7,2,65535,,idle> ==>> CC_ST_PROCEEDING, called:1234567

[050-00:14:01:640]CC: <11,Ss7,2,65535,,idle> calling :0.0.0.0   called:255.255.255.255

[051-00:14:01:640]CC: <11,Ss7,2,65535,,idle> is need reflect:0, callingProfId:0.

[052-00:14:01:640]CC: <11,Sip-t,4,65535,,idle> ==>> CC_ST_SETUP, cr:22, calling:987654321, longNum:987654321, dial:1234567, OrgCallee:, num_ok:1, trunkGrpId:3, profileId:0, isForceReflect(ccb):0, ringback2IP:0,std sdp:v=0

o=call 10000 20000 IN IP4 172.16.51.15

s=-

c=IN IP4 172.16.51.15

t=0 0

m=audio 5120 RTP/AVP 4 18 8 0 101

a=rtpmap:4 G723/8000

a=rtpmap:18 G729/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

, priv sdp:

[053-00:14:01:640]CC: <11,Ss7,2,65535,,idle> ccb state change from 'idle' to 'proceeding', ccb no:11

[054-00:14:01:640]ST: <11,Ss7,2,65535,00000000,init> <<== CC_ST_PROCEEDING, calling:, long:, dial:1234567, send_ok:1, Std Sdp:, Priv Sdp:, cause:0(CCS_NONE(无原因值))

[055-00:14:01:640]ST: <11,Ss7,2,65535,00000000,init> ==>> CC_PROCEEDING_REQ, index:21, if:2, q931_id:779

[056-00:14:01:640]ST: <11,Ss7,2,65535,00000000,init> [custom ringback] call type:2, called:1234567, call forward flag:0, vpbx flag:0

[057-00:14:01:640]ST: <Ss7,2,65535,00000000> ====Processed: CC_ST_PROCEEDING

[058-00:14:01:640]ST: cr, no:21, ccb:11, State:4(out_proc), cause:0(CCS_NONE(无原因值)), redirect:0

[059-00:14:01:640]ST: <11,Sip-t,4,65535,,idle> <<== CC_ST_SETUP, presentId:0, Std Sdp:v=0

o=call 10000 20000 IN IP4 172.16.51.15

s=-

c=IN IP4 172.16.51.15

t=0 0

m=audio 5120 RTP/AVP 4 18 8 0 101

a=rtpmap:4 G723/8000

a=rtpmap:18 G729/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

, Priv Sdp:

[060-00:14:01:640]ST: <11,Sip-t,4,65535,00000000,present> ==>> CC_ST_SETUP_ACK, cause:0(CCS_NONE(无原因值))

[061-00:14:01:640]**ST: <11,Sip-t,4,65535,00000000,present> ==>> SIP_CALL_INVITE, index:22, local:sip:987654321@172.16.51.15, peer:sip:1234567@172.16.50.170 (ip:172.16.50.170, port:5060), std sdp:v=0**

(**note:** send a sip invite msg to destination sip trunk)

o=call 10000 20000 IN IP4 172.16.51.15

s=-

c=IN IP4 172.16.51.15

t=0 0

m=audio 5120 RTP/AVP 4 18 8 0 101

a=rtpmap:4 G723/8000

a=rtpmap:18 G729/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

, priv sdp:, ext:

[062-00:14:01:640]ST: <Sip-t,4,65535,00000000> ====Processed: CC_ST_SETUP

[063-00:14:01:640]ST: cr, no:22, ccb:11, State:7(in_proc), cause:0(CCS_NONE(无原因值)), redirect:0

[064-00:14:01:640]CC: <11,Sip-t,4,65535,,proceeding> <<== CC_ST_SETUP_ACK, cause:0(CCS_NONE(无原因值)), longnum:

[065-00:14:01:640]CC: <11,Ss7,2,65535,,proceeding> ccb state change from 'proceeding' to 'wait ack', ccb no:11

[066-00:14:01:640]CC: <Ss7,2,65535>, <Sip-t,4,65535>, ====Processed: CC_ST_SETUP_ACK

[067-00:14:01:640]CCB: no:11, cr1:21, cr2:22, State:5(wait ack), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[068-00:14:01:680]ST: <11,Sip-t,4,65535,00000000,in_proc> <<== SIP_CALL_RING, Local:1234567@172.16.50.170, Peer:987654321@172.16.51.15, Std Sdp:, Priv Sdp:, Ext:

[069-00:14:01:680]ST: <11,Sip-t,4,65535,00000000,in_proc> ==>> CC_ST_ALERTING, ccb:11, user type:0(Norm), calling:987654321, longnum:987654321, std sdp:, priv sdp:

[070-00:14:01:680]ST: <Sip-t,4,65535,00000000> ====Processed: SIP_CALL_RING

[071-00:14:01:680]ST: cr, no:22, ccb:11, State:8(recving), cause:0(CCS_NONE(无原因值)), redirect:0

[072-00:14:01:680]CC: <11,Sip-t,4,65535,,wait ack> <<== CC_ST_ALERTING, std sdp:, priv sdp:

[073-00:14:01:680]CC: <11,Ss7,2,65535,,wait ack> ccb state change from 'wait ack' to 'alerting', ccb no:11

[074-00:14:01:680]CC: <11,Ss7,2,65535,,alerting> route type:3(rts_out), called term type:4(Sip-t)

[075-00:14:01:680]CC: <11,Sip-t,4,65535,,alerting> ==>> CC_ST_ALERTING, std sdp:, priv sdp:

[076-00:14:01:680]CC: <Ss7,2,65535>, <Sip-t,4,65535>, ====Processed: CC_ST_ALERTING

[077-00:14:01:680]CCB: no:11, cr1:21, cr2:22, State:6(alerting), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[078-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> <<== CC_ST_ALERTING, calling:, long:, dial:, send_ok:1, Std Sdp:, Priv Sdp:, cause:0(CCS_NONE(无原因值))

[079-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> Tm crcx, connid:196782, ip:172.16.51.15, port:5121, algo:4, pkt:30, zip:0, ZipEia:65535, crypt:0, tcp:0, p2pV2:0

[080-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> called dev no:4, called term type:4, called profile id:0, call type:2.

[081-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> [calling] std sdp:v=0

o=call 10000 20000 IN IP4 172.16.51.15

s=-

c=IN IP4 172.16.51.15

t=0 0

m=audio 5120 RTP/AVP 4 18 8 0 101

a=rtpmap:4 G723/8000

a=rtpmap:18 G729/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

, priv sdp:a=X-ACrypt

a=X-Tcp

a=X-P2PV2

a=X-P2PDst:67241984.1043333379.2886742799.4000.20072.65535.65535

[082-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> [called] std sdp:, priv sdp:.

[083-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> get ip profile succ!

[084-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> Is need send local ringback tone to tel:yes, call type:2

[085-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> ==>> CC_ALERTING_REQ, index:21, if:2, q931_id:779

[086-00:14:01:680]ST: <Ss7,2,65535,00000000> ====Processed: CC_ST_ALERTING

[087-00:14:01:680]ST: cr, no:21, ccb:11, State:5(deliver), cause:0(CCS_NONE(无原因值)), redirect:0

EIS(ada)#[088-00:14:02:010]**ST: <11,Sip-t,4,65535,00000000,recving> <<== SIP_CALL_ACCEPT, Local:1234567@172.16.50.170, Peer:987654321@172.16.51.15, Std Sdp:v=0**

(**note:** called answer the call)

o=Qtech 8723835 8723836 IN IP4 172.16.50.170

s=-

c=IN IP4 172.16.50.170

t=0 0

m=audio 8000 RTP/AVP 4 101

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

, Priv Sdp:, Ext:

[089-00:14:02:010]ST: <11,Sip-t,4,65535,00000000,recving> ==> CC_ST_CONNECT,

[090-00:14:02:010]ST: <Sip-t,4,65535,00000000> ====Processed: SIP_CALL_ACCEPT

[091-00:14:02:010]ST: cr, no:22, ccb:11, State:9(active), cause:0(CCS_NONE(无原因值)), redirect:0

[092-00:14:02:010]CC: <11,Sip-t,4,65535,,alerting> <<== CC_ST_CONNECT, calling:987654321, long:987654321, called:1234567, calling dial num:1234567, Std Sdp:v=0

o=Qtech 8723835 8723836 IN IP4 172.16.50.170

s=-

c=IN IP4 172.16.50.170

t=0 0

m=audio 8000 RTP/AVP 4 101

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

, Priv Sdp:

[093-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> stop queue seat timer!

[094-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> stop queue timer!

[095-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> clear bill end time(cc connect).

[096-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> stop hint at port:65535 ,connid:4294967295

[097-00:14:02:010]CC: <11,Sip-t,4,65535,,alerting> stop hint at port:65535, connid:4294967295

[098-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> vpbx process flag:0, ippbx process flag:0

[099-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> route type:3(rts_out), called term type:4(Sip-t)

[100-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> get bill start time:00-14-02

[101-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> payer info(DevNo-2, PortNo-65535, callDirect-1, termType-Ss7), Is ccb stpayer.pstPort NULL:yes. Service type(ccb):normal, is need settle:no.

[102-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> ==>> CC_ST_CONNECT, called:1234567

[103-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> ccb state change from 'alerting' to 'active', ccb no:11

[104-00:14:02:010]CC: <Ss7,2,65535>, <Sip-t,4,65535>, ====Processed: CC_ST_CONNECT

[105-00:14:02:010]CCB: no:11, cr1:21, cr2:22, State:7(active), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[106-00:14:02:010]ST: <11,Ss7,2,65535,00000000,deliver> <<== CC_ST_CONNECT, calling:, long:, dial:1234567, send_ok:1, Std Sdp:v=0

o=Qtech 8723835 8723836 IN IP4 172.16.50.170

s=-

c=IN IP4 172.16.50.170

t=0 0

m=audio 8000 RTP/AVP 4 101

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11

, Priv Sdp:, cause:0(CCS_NONE(无原因值))

[107-00:14:02:010]ST: <11,Ss7,2,65535,00000000,deliver> Tm mdcx, connid:196782, ip:172.16.50.170, port:8000, algo:4, pkt:30, zip:0, ZipEia:65535, crypt:0, tcp:0, p2pV2:0, faxMode:0

[108-00:14:02:010]ST: <11,Ss7,2,65535,00000000,deliver> [custom ringback] call type:2, called:1234567, call forward flag:0, vpbx flag:0

[109-00:14:02:010]ST: <11,Ss7,2,65535,00000000,deliver> ==>> CC_SETUP_RSP, index:21, if:2, q931_id:779

[110-00:14:02:010]ST: <Ss7,2,65535,00000000> ====Processed: CC_ST_CONNECT

[111-00:14:02:010]ST: cr, no:21, ccb:11, State:9(active), cause:0(CCS_NONE(无原因值)), redirect:0

[112-00:14:02:010]ST: <11,Ss7,2,65535,00000000,active> <<== CC_SETUP_COMPL_IND, q931id:779, if:2, calling:, called: org_called:, e1:10, ts:11, callingTyp:0, calledTyp:0, presentationInd:0, send_ok:0, cause:0(OK)

[113-00:14:02:010]ST: <11,Ss7,2,65535,00000000,active> <<== CC_SETUP_COMPL_IND

[114-00:14:02:010]ST: <11,Ss7,2,65535,00000000,active> ==> CC_ST_CONNECT_ACK

[115-00:14:02:010]ST: <Ss7,2,65535,00000000> ====Processed: CC_SETUP_COMPL_IND

[116-00:14:02:010]ST: cr, no:21, ccb:11, State:9(active), cause:0(CCS_NONE(无原因值)), redirect:0

[117-00:14:02:010]ST: <Ss7,2,65535,00000000> ====Processed: CC_SETUP_COMPL_IND

[118-00:14:02:010]ST: cr, no:21, ccb:11, State:9(active), cause:0(CCS_NONE(无原因值)), redirect:0

[119-00:14:02:010]CC: <11,Ss7,2,65535,,active> <<== CC_ST_CONNECT_ACK

[120-00:14:02:010]CC: <Ss7,2,65535>, <Sip-t,4,65535>, ====Processed: CC_ST_CONNECT_ACK

[121-00:14:02:010]CCB: no:11, cr1:21, cr2:22, State:7(active), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

EIS(ada)#[122-00:14:04:060]**ST: <11,Sip-t,4,65535,00000000,active> <<== SIP_CALL_BYE, Local:987654321@172.16.51.15, Peer:1234567@172.16.50.170, Std Sdp:, Priv Sdp:, Ext:**

(**note:** called release the call)

[123-00:14:04:060]ST: <11,Sip-t,4,65535,00000000,active> ==> CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放))

[124-00:14:04:060]ST: <11,Sip-t,4,65535,00000000,active> Free CR 22, cause:1(CCS_NORM_CLEAR(正常释放))

[125-00:14:04:060]CC: <11,Ss7,2,65535,,active> [cc release comp]ccb no:11, sub ccb no:4294967295

[126-00:14:04:060]CC: <-1,Sip-t,4,65535,,idle> <<== CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放)))

[127-00:14:04:060]CC: <11,Ss7,2,65535,,active> State(active) is not match, refuse resel route!

[128-00:14:04:060]CC: <11,Ss7,2,65535,,active> bill start time: 0-14- 2, bill end time: 0- 0- 0.

[129-00:14:04:060]CC: <11,Ss7,2,65535,,active> [bill end time]bill type:normal, service type(ccb):normal, is need settle:no.redirect flag:0, called term type:Sip-t, Is ccb stpayer.pstPort NULL:yes.

[130-00:14:04:060]CC: <11,Ss7,2,65535,,active> ==>> CC_ST_RELEASE, cause:1(CCS_NORM_CLEAR(正常释放))

[131-00:14:04:060]CC: <11,Ss7,2,65535,,active> ccb state change from 'active' to 'release', ccb no:11

[132-00:14:04:060]CC: <Ss7,2,65535>, <Sip-t,4,65535>, ====Processed: CC_ST_REL_COMP

[133-00:14:04:060]CCB: no:11, cr1:21, cr2:22, State:9(release), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:1(CCS_NORM_CLEAR(正常释放))

[134-00:14:04:060]ST: <11,Ss7,2,65535,00000000,active> <<== CC_ST_RELEASE, calling:, long:, dial:, send_ok:1, Std Sdp:, Priv Sdp:, cause:1(CCS_NORM_CLEAR(正常释放))

[135-00:14:04:060]ST: <11,Ss7,2,65535,00000000,active> needPlaySigTone2Tel:0, isReflectRoute:0, cause:1.

[136-00:14:04:060]ST: <11,Ss7,2,65535,00000000,active> Tm dlcx, connid:196782

[137-00:14:04:060]ST: <11,Ss7,2,65535,00000000,active> @@@ free called:1234567, lines:0

[138-00:14:04:060]ST: <11,Ss7,2,65535,00000000,active> Release the call, cause:CCS_NORM_CLEAR(正常释放)(1) !

[139-00:14:04:060]ST: <11,Ss7,2,65535,00000000,active> ==>> CC_DISCONNECT_REQ, index:21, if:2, q931_id:779

[140-00:14:04:060]ST: <Ss7,2,65535,00000000> ====Processed: CC_ST_RELEASE

[141-00:14:04:060]ST: cr, no:21, ccb:11, State:11(release), cause:1(CCS_NORM_CLEAR(正常释放)), redirect:0

[142-00:14:04:070]ST: <11,Ss7,2,65535,00000000,release> <<== CC_RELEASE_IND, q931id:779, if:2, calling:, called: org_called:, e1:10, ts:11, callingTyp:0, calledTyp:0, presentationInd:0, send_ok:0, cause:16(正常的呼叫清除)

[143-00:14:04:070]ST: <11,Ss7,2,65535,00000000,release> Release the call, cause:CCS_NORM_CLEAR(正常释放)(1) !

[144-00:14:04:070]ST: <11,Ss7,2,65535,00000000,release> ==> CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放))

[145-00:14:04:070]ST: <11,Ss7,2,65535,00000000,release> Free CR 21, cause:1(CCS_NORM_CLEAR(正常释放))

[146-00:14:04:070]ST: <,65535,65535,> ====Processed: CC_RELEASE_IND

[147-00:14:04:070]ST: cr, no:21, ccb:4294967295, State:0(idle), cause:0(CCS_NONE(无原因值)), redirect:0

[148-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> [cc release comp]ccb no:11, sub ccb no:4294967295

[149-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> stop queue seat timer!

[150-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> stop queue timer!

[151-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> <<== CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放)))

[152-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> Free CCB 11, cause:1(CCS_NORM_CLEAR(正常释放))

[153-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> stop queue seat timer!

[154-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> stop queue timer!