# MTG1000 Trunk Gateway User Manual V2.0

**Dinstar Technologies Co., Ltd.**

# Revision Records

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# Content

# 1. Product Introduction

## 1.1 Overview

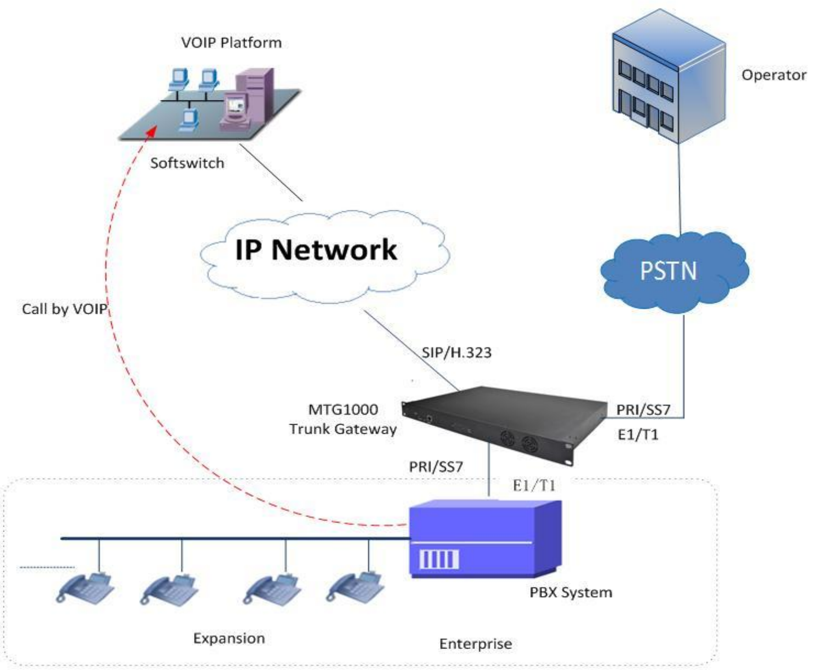
MTG1000 is a trunk gateway aimed at operators and call center, and used to help enterprise to realize the evolution from the traditional PBX to voice IP. On the one hand, it supports PRI/SS7 protocol and adopts standard T1/E1 trunk interface to realize docking with traditional PBX. On the other hand, adopt standard SIP protocol docking with various soft switch to ensure PSTN seamless access to IP voice/NGN network, and achieving VoIP/FoIP and more value-added service. MTG1000 supports intelligent multiple trunk routing technology, makes the operator easy to manage trunk routing by price optimum rule, and the automatic switch-over between multiple trunk routing makes the network have high reliability.

MTG1000 has good call processing ability, and provides 4/8 T1/E1 interface. It is able to handle a variety of signaling protocol and voice decoding. It supports the rich GUI configuration, the user easily set and maintenance system. Mainly includes the following kinds of models:

- MTG1000-4E1
- MTG1000-8E1

A typical network diagram shows the function of MTG1000 as below.

Figure 1-1-1 Application topology




## 1.2 Equipment Structure

### 1.2.1 Rear View


Figure 1-2-1 MTG1000 Rear View

Table 1-2-1 MTG1000 Rear View Description

| Interface | Description |
|---|---|
| PWR | Connecting the power adapter, 110~240VAC, 50~60HZ, 1.2A, double power |
| Port0-Port7 | E1/T1 port, There are 8E1/T1 ports |
| FE0 | The Service Ethernet Interface, standard 10/100BASE-TX Ethernet interfaces. Default IP address is 192.168.1.111, default subnet mask is 255.255.255.0 |
| FE1 | Management Ethernet Interface. Default IP address is 192.168.11.1, default subnet mask is 255.255.255.0 |

### 1.2.2 Front View


Figure 1-2-2 MTG1000 Front View

Table 1-2-2 MTG1000 Front View Description

| LED | Function | Color | Work Status |
|---|---|---|---|
| POWER | Power status indicator | Green | Off：Power is off |
| | | | On: Power is on |
| RUN | Register indicator | Green | Slow blinking: Unregister |
| | | | Fast blinking: Register |
| ALM | The failure of device indicator | Yellow | Off: Normal |
| | | | On: Failed |
| RST | Reset button, it is used to restart the device | | |
| CONSOLE | RS232 console port: it can be used to debug and configure the device. The baud rate is 115200 bps. | | |
| E1/T1 | Indicating the connection state of device E1/T1. | Green | Off：E1/T1 port connection normal |
| | | | On: E1/T1 port connection and sending/ receiving message normal |
| | | | Flash:E1/T1 port connection failed |
| LINK | Indicating the connection state of the network | Green | Off: Network connection failed |
| | | | On: Network connection normal, and 0 indicates FE0 and 1 indicates FE1 |
| SPEED | Indicating the network bandwidth | Yellow | Off:10Mbps bandwidth |
| | | | On:100Mbps bandwidth |

### 1.2.3 RJ-48c Line sequence

| RJ-48 Pin (on T1/E1 PIC) (Data numbering form) | RJ-48 Pin (Data numbering form) | Signal |
|---|---|---|
| 1 | 1 | RX, Ring, - |
| 2 | 2 | RX, Tip, + |
| 4 | 4 | TX, Ring, - |
| 5 | 5 | TX, Tip, + |
| 3 | 3 | Shield/Return/Ground |
| 6 | 6 | Shield/Return/Ground |
| 7 | No connect | No connect |
| 8 | No connect | No connect |

MTG1000 trunk gateway adopts standard RJ-48C interface and impedance value is 120Ω. Connected end device by cross lines sequence.

## 1.3 Functions and Features

### 1.3.1 Protocol standard supported

- Standard SIP /SIP-T/H.323/PRI/SS7 protocol
- NAT Traversing (STUN)
- Hypertext Transfer Protocol (HTTP)
- Domain Name System (DNS)
- Dynamic host configuration protocol (DHCP)
- ITU-T G.711A-Law/U-Law、G.723.1、G.729AB、iLBC (optional)

### 1.3.2 System Function

- Comfort Noise Generation (CNG)
- Voice Activity Detection (VAD)
- Adaptive (Dynamic) Jitter Buffer (DJB)
- DTMF mode: RFC 2833, SIP INFO and INBAND
- T.38/ Pass-Through FAX over IP
- HTTP/Telnet configuration
- Firmware upgrade by TFTP/Web

### 1.3.3 Industrial standards supported

- Stationary use environment: EN 300 019: Class 3.1
- Storage environment: EN 300 019: Class 1.2
- Transportation environment: EN 300 019: Class 2.3
- Acoustic noise: EN 300 753
- CE EMC directive 2004/108/EC
- EN55022: 2006+A1:2007
- EN61000-3-2: 2006,
- EN61000-3-3: 1995+A1: 2001+A2: 2005
- EN55024: 1998+A1: 2001+A2: 2003
- Certifications: FCC, CE

### 1.3.4 General hardware specification

- Power supply: 220VAC, 1.2A
- Temperature: 0~40℃ (operational),-20~70℃(storage)
- Humidity: 10%~90%, no condensation
- Max power consumption: 25W
- Dimension (mm): 436*300*44
- Net Weight: 1.9 kg

# 2. Parameter Setting

## 2.1 Login

First, device FE0 port connect PC with string, and then fill FE0 IP address in browser, FE0 default IP address is 192.168.1.111. It will request customer to input user name and password. Default user name and password are “admin”.

If customer modified the default IP or forgot the IP, that can’t enter the configuration page. Please connect PC and device serial with the serial line. Enter the CLI to view or modify the equipment IP. Here IP is set to 172.16.99.120. In addition, hold down the RST button to restart the device, customer can regain the port’s default IP. Then enter the IP address of device in the browser address bar. Customer will see the following page.

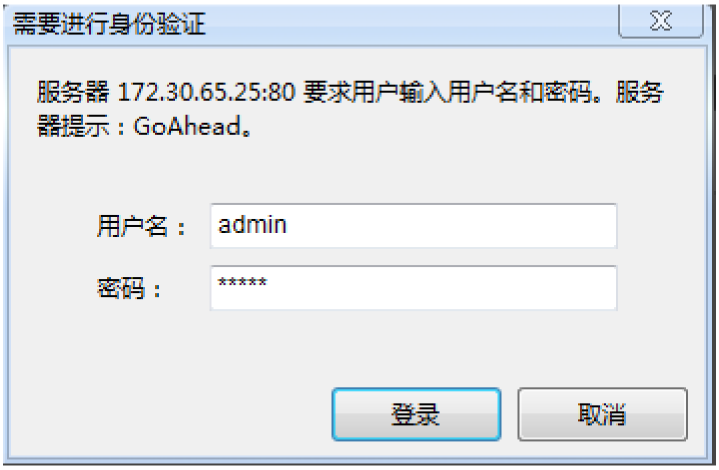



Figure 2-1-1 Login Interfaces

The default user name and password is "admin". To guarantee the system safety, when login for the first time. The system will prompt the user to modify the password. The interface is shown as below.

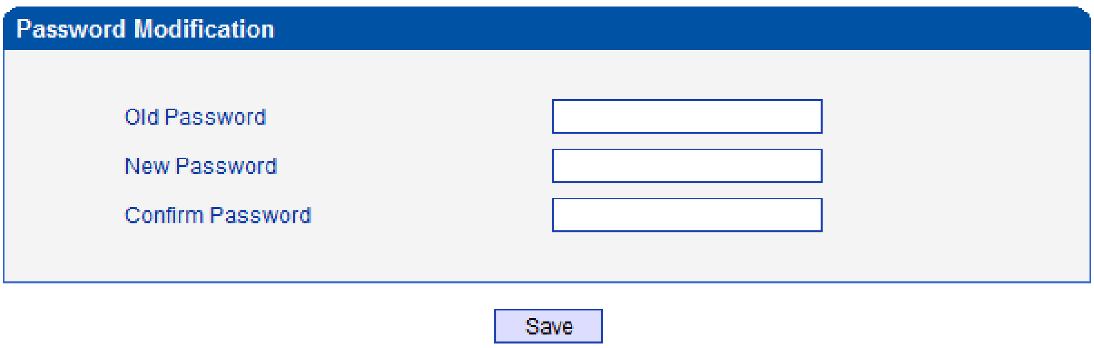



Figure 2-1-2 Modify Password

Users through to traverse the left navigation tree, and can complete view, edit and configuration device in the right configuration interface.

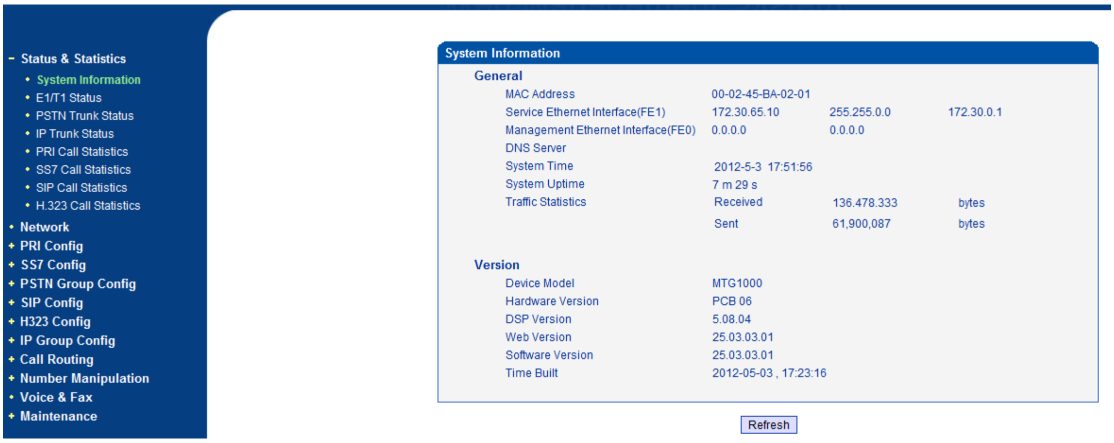



Figure 2-1-3 Description of System Information

## 2.2 Web interface structure and navigation tree

After entering configuration page, according to demand choose Chinese interface or English interface, the default is English interface.

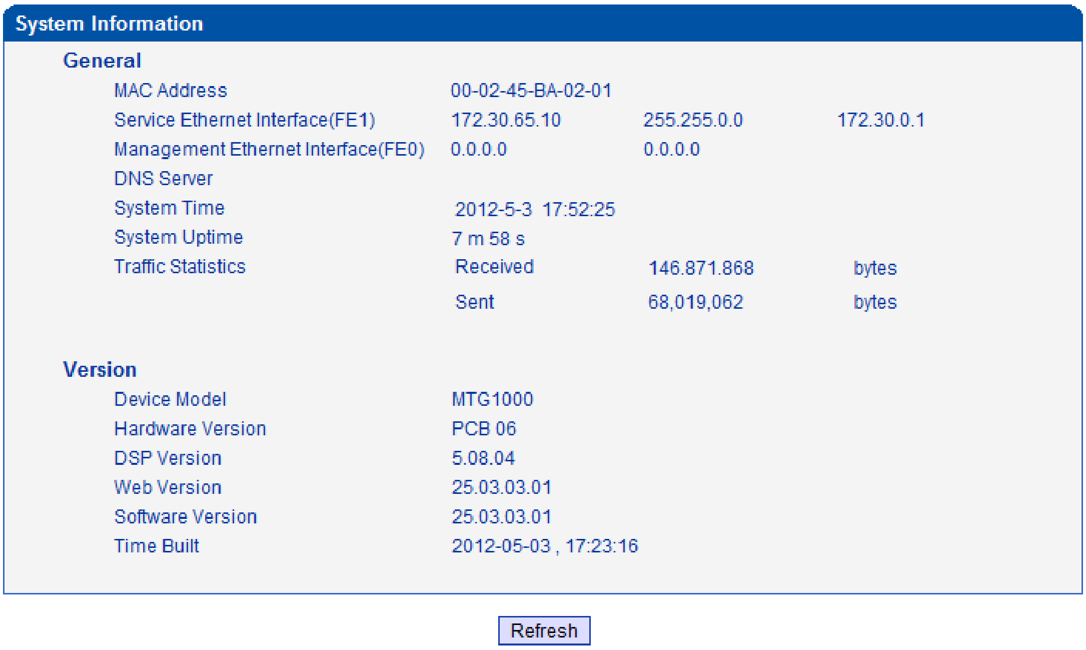
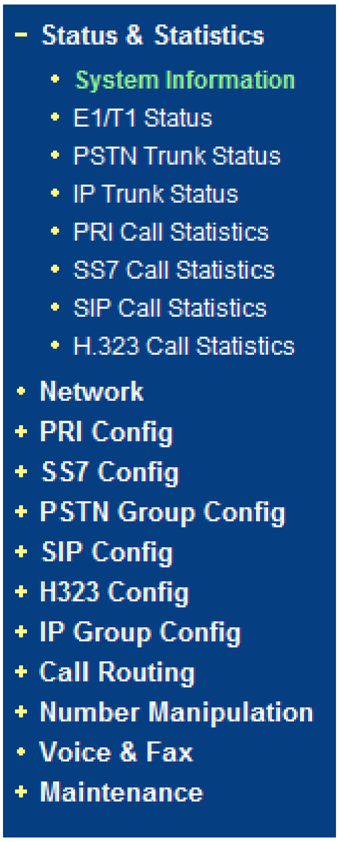


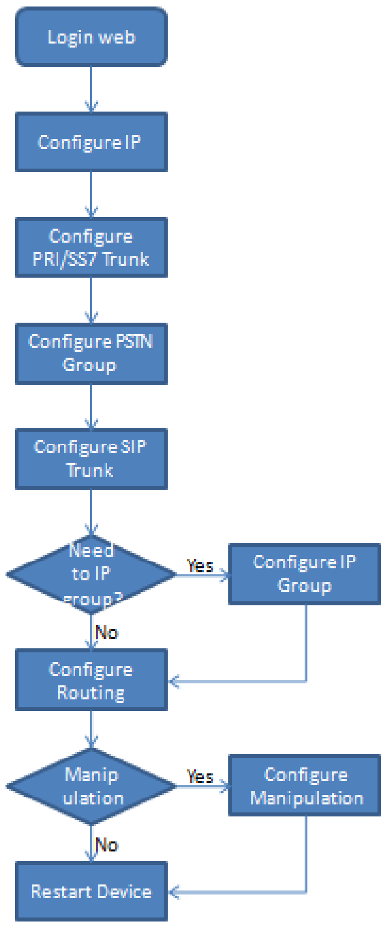

Figure 2-2-1 System Information

Users through to traverse the left navigation tree, and can complete view, edit and configuration device in the right configuration interface.

Figure 2-2-2 Navigation tree



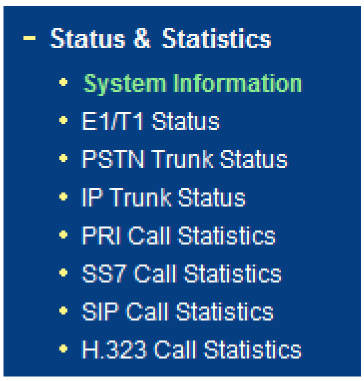

MTG configuration flow chart below:




## 2.3 Status & Statistics

Open the operation of the navigation tree information node, and can view the device information and state system.

Figure 2-3-1 Status & Statistics



### 2.3.1 System Information

System information interface shows the general information and version information.

Figure 2-3-1 System Information

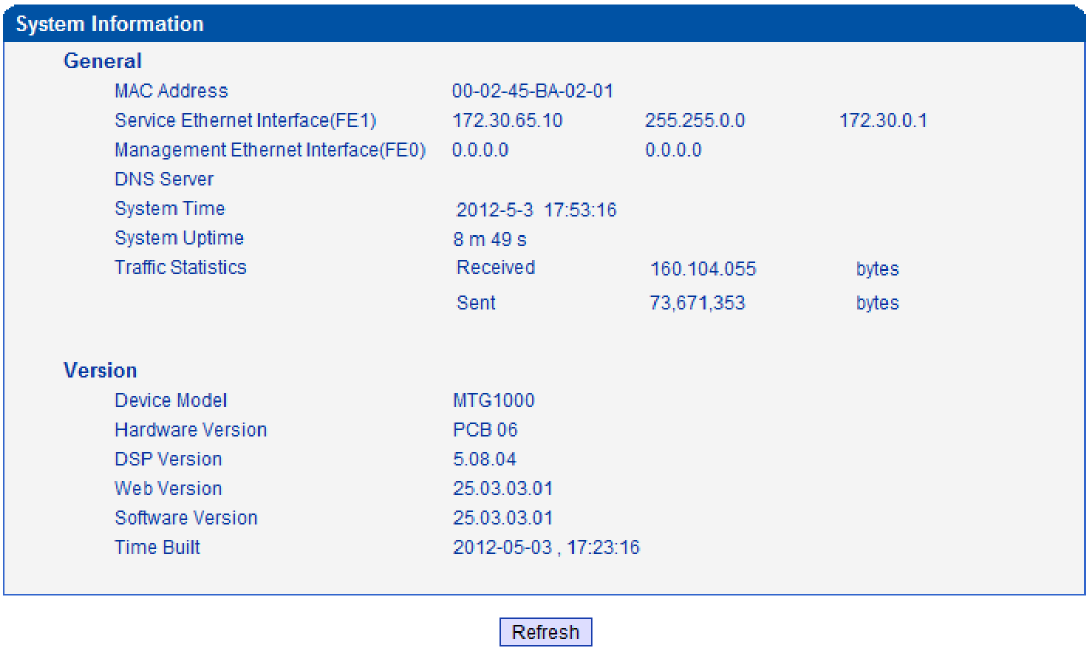



Table 2-3-1 System Information

| | |
|---|---|
| MAC address | Hardware address of FE0 port |
| Service Ethernet Mode | Network mode of FE0, include: static and DHCP. |
| Service Ethernet Interface | Include: IP address, subnet mask, FE0 port default gateway |
| Management Ethernet Interface | Include IP address、subnet mask of FE1 |
| DNS | DNS server IP address |
| System Up Time | Time elapsed from device power on to now |
| Traffic Statics | Total bytes of message received and sent by FE0 port |
| Equipment Type | Equipment type; this equipment is: MTG1000 |
| Hardware Version | Hardware version of device |
| DSP Version | Digital signal processing chip driver version |
| Web Version | Version of current WEB interface of device |
| Software Version | Software version of device running currently |
| Built Time | The build time of current software version |

### 2.3.2 E1/T1 Status

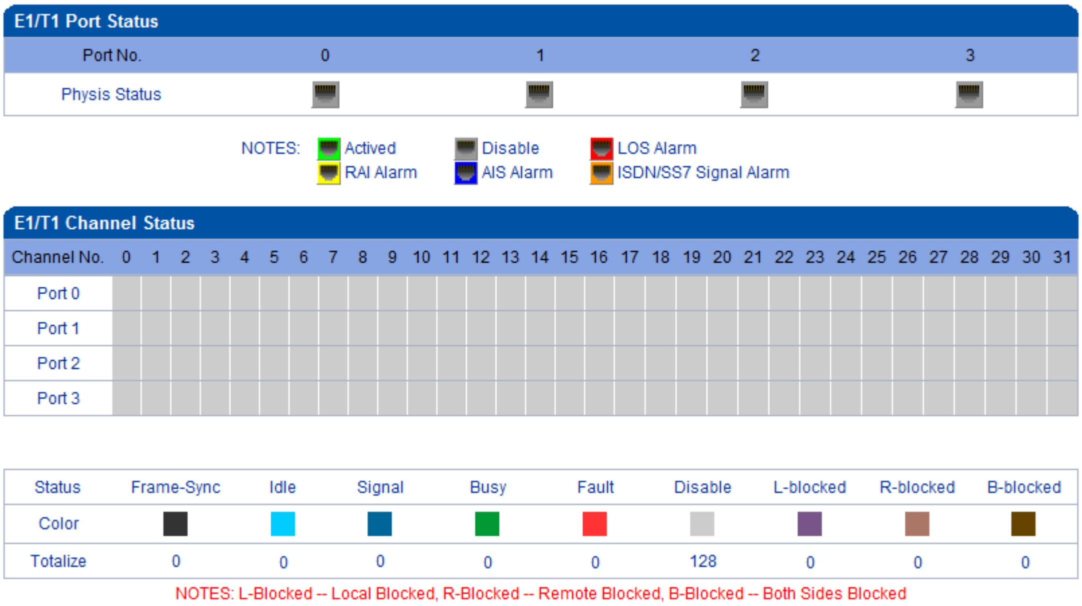



Figure 2-3-2 E1/T1 Status

Table 2-3-2 Description of E1/T1 status

| | |
|---|---|
| E1/T1 Port Status | **1**. LOS Alarm: Signal loss alarm, this alarm is created when receiving is lost; please check the physical connection whether disconnected. |
| | **2**. RAI Alarm: Receive remote alarm indication, it is a signal transmitted in the outgoing direction when a terminal determines that it has lost the incoming signal. Receiving remote alarm indication (RAI) means the far-end equipment over the T1 line has a problem with the signal it is receiving from the upstream equipment. |
| | **3**. AIS Alarm: The Alarm Indication Signal (AIS) failure is declared when an AIS defect is detected at the input and the AIS defect still exists after the Loss of frame failure which is caused by the unframed nature of the 'all-ones' signal is declared. The AIS failure is cleared when the Loss Of Frame failure is cleared. |
| | **4**. Disable: Means that this E1/T1 is not used. |
| | **5**. ISDN/SS7 Signal Alarm: Means physical connection is normal, signaling link has problem. |
| | **6**. Active-OK: Means that physical connection and signaling link are normal. |
| E1/T1Channel Status | Frame-Sync: Non voice channel, which used as a synchronization channel |
| | Idle: Means this channel is idle, when the channel is enabled and the cable is connected OK. |
| | **3**.Signal: Signal channel |
| | **4**.Busy: Means this channel is occupied by voice |
| | **5**. Fault: The channel is enabled but the cable is not connected. |
| | **6**.Disable: Have not use this E1/T1 trunk |
| | **7**.L-blocked:<br>Local blocked, means that communication can only be initiated from local |
| | **8**.R-blocked: |

| | |
|---|---|
| | Remote blocked, means that communication can only be initiated from remote |
| | **9.** B-blocked:<br>Both Sides blocked, means that the two sides cannot communication |

### 2.3.3 PSTN Trunk Status

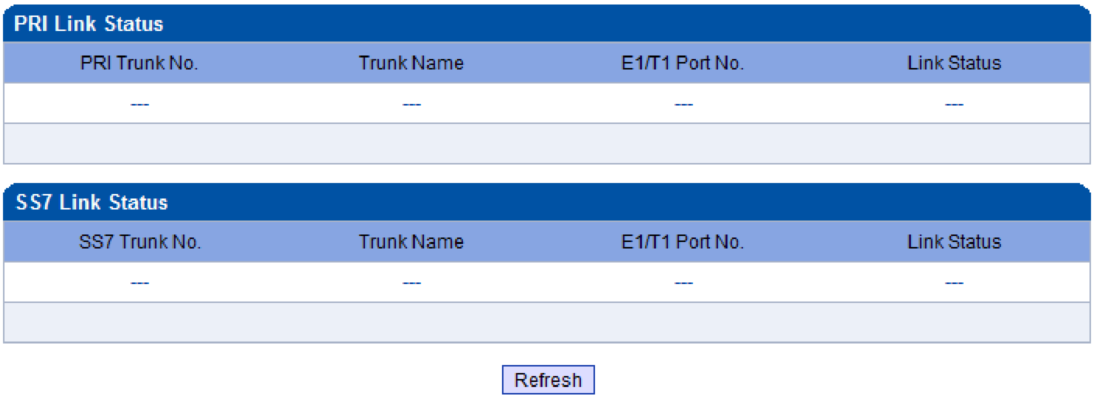



Figure 2-3-3 PSTN Trunk Status

**PSTN trunk status description:**

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 name of the trunk |
| E1/T1Port No | Indicate the E1/T1 line occupied by the PRI trunk. |
| Link Status | Indicate whether the PRI link is established. |

2） SS7 Link Status

| | |
|---|---|
| SS7 Trunk No. | SS7 trunk number, each relay takes up 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. |

### 2.3.4 IP Trunk Status

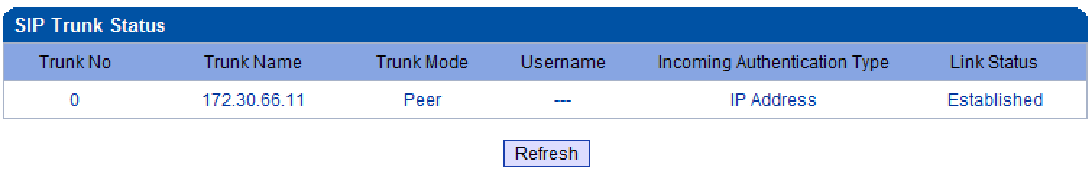



Figure 2-3-4 SIP Trunk Status

**IP trunk status**

| | |
|---|---|
| SIP Trunk No | The number of SIP trunk |
| Username | When SIP trunk is under registered mode, change the value in the configuration shown in the account registration, If SIP trunk is under non-registered mode, the value is meaningless, as '---' |

| | |
|---|---|
| Trunk Mode | Peer and Access two modes |
| Register Status | Indicate the status of SIP trunk（access mode）, register or unregister, when is under peer to peer mode, the values is meaningless, as '---' |
| Link Status | Established and Fault status. |
| SIP Trunk No | The number of SIP trunk |

### 2.3.5 PRI Call Statistics

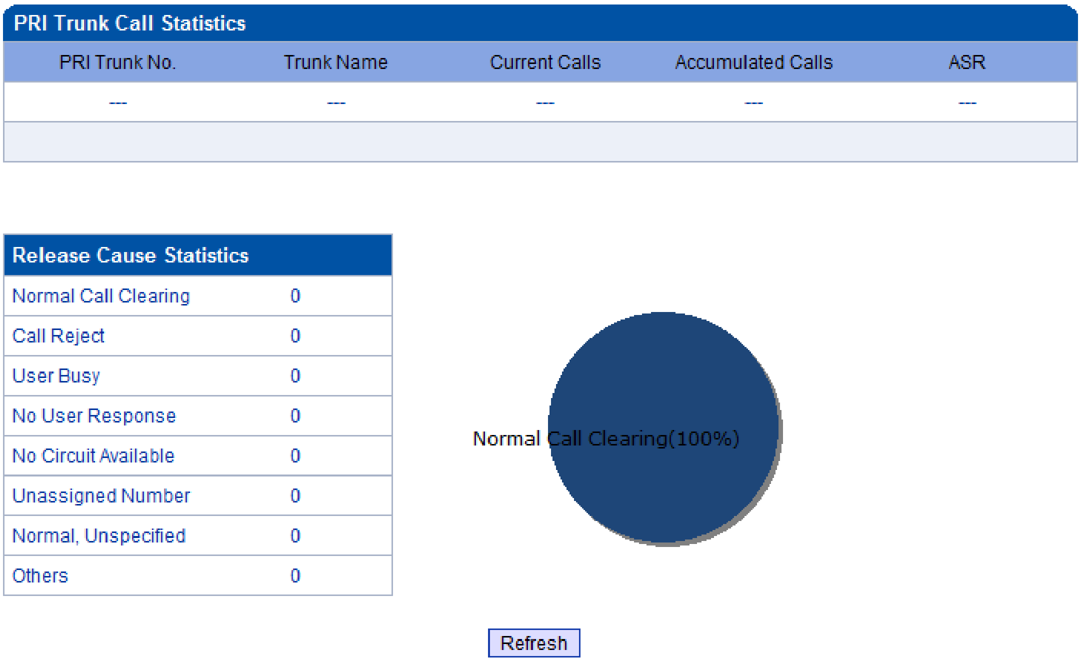



Figure 2-3-5 PRI Call Statistics description

**PRI call statistics description**

| | |
|---|---|
| PRI Trunk No | The number of PRI trunk |
| Trunk Name | The name used to describe the PRI trunk |
| Current Calls | Number of lines that are being called currently |
| Accumulated Calls | Total number of calls from running start of system to current time. |
| ASR | The percent of calls completed in total calls. |

This statistics page show the reasons for release of the call, including: Normal Call Clearing, Call Rejected, User Busy, No User Response, No Circuit Available, Unassigned Number, Normal Unspecified and others. Statistical information in an intuitive would be reflected on the pie char.

### 2.3.6 SS7 Trunk Call Statistics

| SS7 Trunk Call Statistics | | | | |
|---|---|---|---|---|
| SS7 Trunk No. | Trunk Name | Current Calls | Accumulated Calls | ASR |
| --- | --- | --- | --- | --- |

| 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 |


Refresh

Figure 2-3-6 SS7 Trunk Call Statistics

The parameters of SS7 trunk call statistics are the same with PRI parameters. Please refer to PRI trunk call statistics.

### 2.3.7 SIP Call Statistics

| SIP Trunk Call Statistics | | |
|---|---|---|
| SIP Trunk No. | Trunk Name | Current Calls |
| 0 | 172.30.66.16 | 0 |

Refresh

Figure 2-3-7 SIP Trunk Call Statistics

**SIP call statistics description**

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

### 2.3.8 H.323 Call Statistics

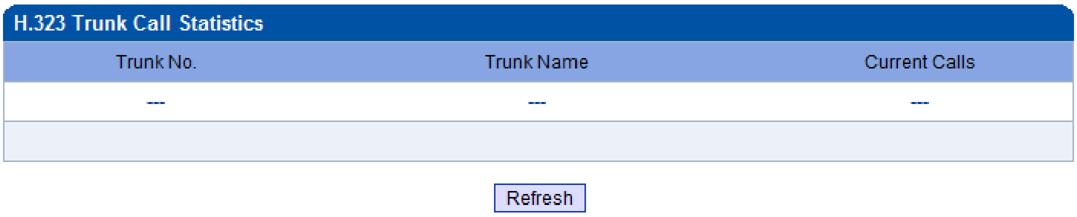



Figure 2-3-8 H.323 Trunk Call Statistics

H. 323 call statistical parameters and SIP call statistical parameters is same, can be reference SIP parameters statistics show.

## 2.4 Network

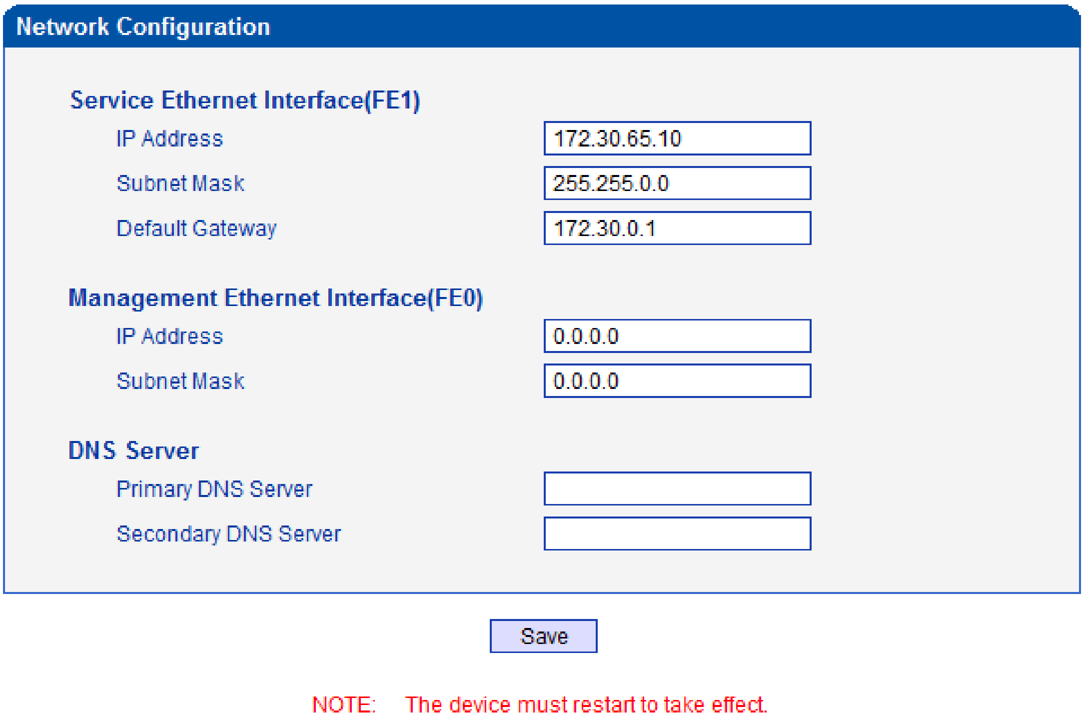



Figure 2-4-1 Network Configuration

**Network Configuration**

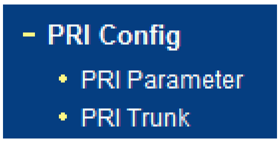
| | | |
|---|---|---|
| Service Ethernet Interface（FE0） | IP address | Set FE0 port static IP address. |
| | Subnet Mask | Fill in subnet mask |
| | Default Gateway | Fill in default gateway |
| Management Ethernet Interface（FE1） | IP address | Set FE1 port static IP address |
| | Subnet Mask | Fill in subnet mask |
| DNS Server | Primary DNS | Fill in DNS Server IP address. |
| | Secondary DNS | The secondary DNS server is option. |

**Ntoe：** FE0 port IP and FE1 port IP should be set in different segments. After configure the network address, and restart the gateway configuration to take effect.

## 2.5 PRI Config

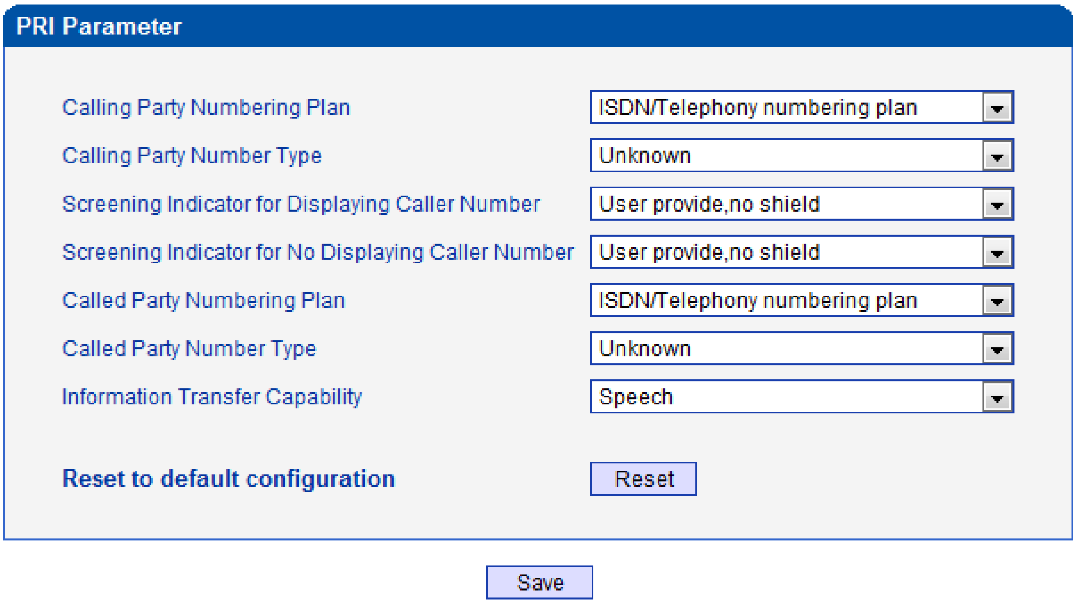
PRI configuration includes PRI parameter and PRI trunk configuration

Figure 2-5-1 PRI Config




### 2.5.1 PRI Parameter

Figure 2-5-2 PRI Parameter



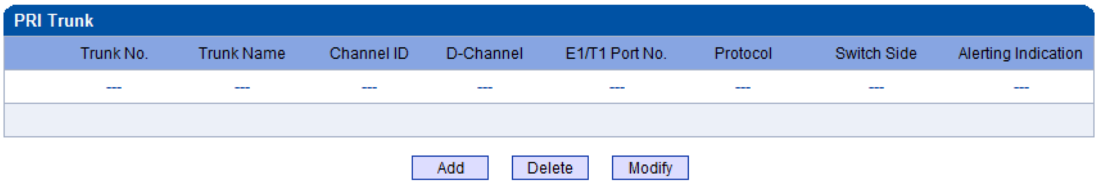

**PRI parameter description**

| | |
|---|---|
| Calling Party Numbering Plan | Provide six plans: Unknown, ISDN/Telephony numbering plan, data numbering plan, telegraph numbering plan, national standard numbering plan, private numbering plan. The default is ISDN/Telephony numbering plan. |
| Calling Party Number Type | Six optional types are provided for calling party: Unknown, International number, National number, Network special number, User number, Short code dialing. The default option is Unknown. |
| Screening Indicator for Displaying Caller Number | Four options available: User provider, no shield; User provide, check and send; User provide, check and having failure; Network provide. The default option is: User provider, no shield. |
| Screening Indicator for No Displaying Caller Number | Four options available: User provider, no shield; User provide, check and send; User provide, check and having failure; Network provide. The default option is: User provider, no shield. |
| Called Party Numbering Plan | Provide six plans: Unknown, ISDN/Telephony numbering plan, data numbering plan, telegraph numbering plan, national standard numbering plan, private numbering plan. The default is ISDN/Telephony numbering plan. |

| | |
|---|---|
| Called Party Number Type | Six optional types are provided for called party: Unknown, International number, National number, Network special number, User number, Short code dialing. The default option is Unknown. |
| Information Transfer Capability | Support speech and 3.1khz audio. The default option is speech. |

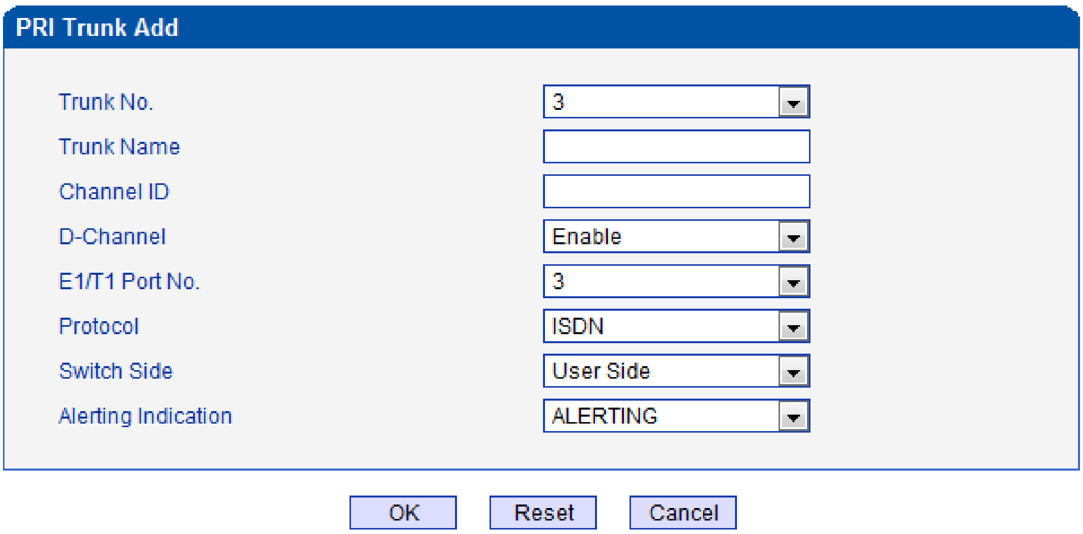
### 2.5.2 PRI Trunk

Figure 2-5-3 PRI Trunk




Users can add/delete/modify PRI trunk in this configuration option.

Figure 2-5-4 Add PRI Trunk




**PRI trunk description**

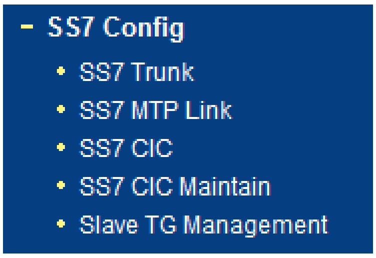
| | |
|---|---|
| Trunk No | The number of PRI trunk; when user add PRI trunk, 0~7 number will appear in the pull-down menu to be selected (the number here depends on E1/T1 physical port number actually existed in equipment). After trunk number is established, filling in corresponding port number in "E1/T1 Port No.", so as to assign E1/T1 to designated trunk; Each PRI trunk corresponds to a E1/T1 port. |
| Trunk Name | Description of PRI trunk |
| Channel ID | Channel ID of E1/T1 ports, this number definition generally starts from 0. |
| D-channel | Indicate whether E1/T1 supports D channel, the default is Yes. |
| E1/T1 Port No | E1/T1 port number is numbered according to the physical position of E1/T1, it generally starts from 0. |
| Protocol | Interface type of PRI. There are two types are available: ISDN and QSIG; the default is ISDN. |

| | |
|---|---|
| Switch Side | Indicate PRI network property of E1/T1, it is divided into: “User side” and“Network side”. When PRI loopback is carried out, the network properties of E1/T1 port at both receiving and sending sides must be different. |
| Alerting Indication | The ring signal include Alerting and Progress |

## 2.6 SS7 Config

SS7 configuration includes: SS7trunk, SS7 MTP Link, SS7 CIC, SS7 CIC Maintain and Slave TG Management.

Figure 2-6-1 Add PRI Trunk




### 2.6.1 SS7 Trunk

Figure 2-6-2 SS7 Trunk

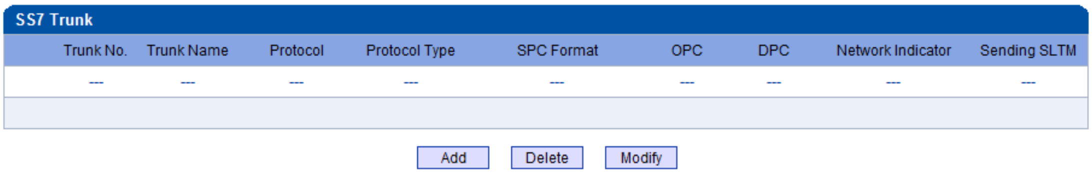
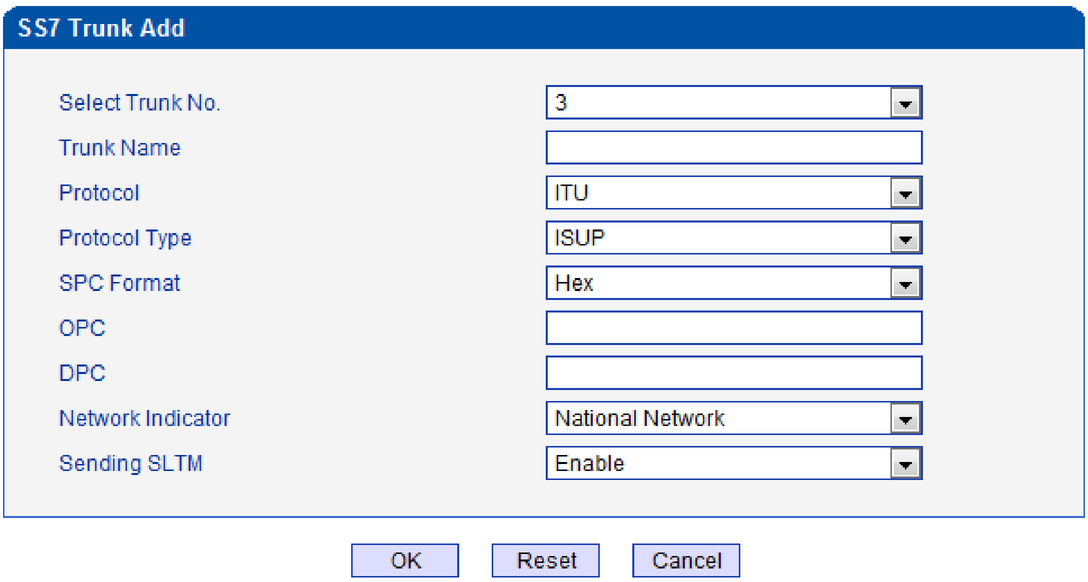



Figure 2-6-3 SS7 Trunk Add



SS7 is a standard protocol to initiate a calling connection with SPC exchange.

**Notes:**

1. "Trunk No." is a shared data, therefore, SS7 „Trunk No.‟ can't be the same as PRI "Trunk No."
2. SPC length is 24bits when option "ANSI" or "ITU-CHINA" is selected in item "Standard Type".
3. SPC length is 14bits when option "ITU" is selected in item "Standard Type".
4. SPC Length represents the structure of OPC/DPC. SPC View Mode indicates which input format is selected for OPC/DPC structure.
5. When SPC length is 24bits and 'Hex' are selected, the structure is like xyz, and x,y,z must be hex number between 00-FF. eg., 33AA55.
6. When SPC length is 14bits and 'ITU Pointcode Structure' are selected, the structure is like x-y-z, and x,z must be decimal number between 0-7, and y must be decimal number between 0-255. eg., 6-222-3.
7. When SPC length is 14bits and 'Hex' are selected, the structure is like xyz, and x/z is a 3 bit hex number, y is a 8 bit hex number. eg., 202E(100 00000101 110).

**SS7 trunk add**

| | |
|---|---|
| Select Trunk No | The number of SS7 trunk. Generally, a DPC will establish a SS7 trunk number respectively, SS7 trunk number cannot be conflict with PRI trunk number. After SS7 trunk is established, assign E1/T1 to SS7 trunk in "SS7 Circuit" option. |
| Trunk Name | Name of trunk, it can be edited to any name user want. |
| Protocol | SPC types: ITU-T (14 bit), ANSI (24 bit), ITU-CHINA (24 bit) |
| Protocol Type | Supported two protocol types: ISUP and TUP |
| SPC Format | Signaling Point Code format includes hexadecimal system and ITU pointcode structure (decimal system) |
| OPC | Original Point Code |
| DPC | Destination Point Code |
| Service Type | SS7 service types: ISUP (ISDN User Part) and TUP (Telephone User Part). |
| Network Indicator | Indicate the network property of SS7, including International Network, International Spare, National Network, National Spare; the default is "National Network" (this type is used in China, USA, and Japan), "International Network" is generally used in inter-office switch room; others will be selected according to physical circumstances. |

**Note：**

1. If protocol standard chose 'ANSI' or 'ITU-CHINA', and then the SPC length is 24 bits.
2. If protocol standard chose'ITU', and then the SPC length is 14 bits.

3. SPC length performance on the OPC/DPC structure; SPC pattern instructions of the different structure OPC/DPC input formats.

4. When the SPC length is 24 bits, and chosen ITU, OPC/DPC structure format is :x-y-z; x、 y、 z is a number of 0-255, such as: 22-222-77

5. When the SPC length is 24 bits, and chosen Hex, OPC/DPC structure format is :xyz; x、y、z must be Hex number of 00-FF, such as: 33AA55

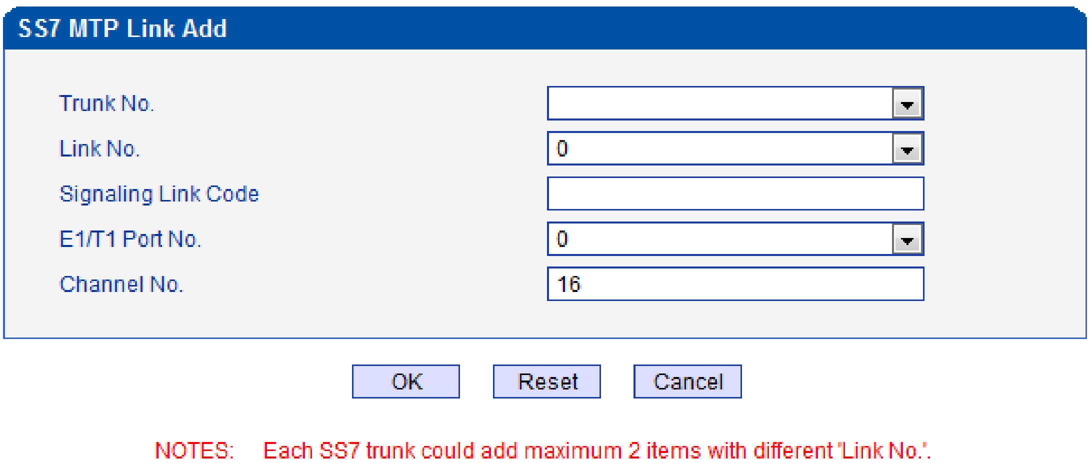
6. When the SPC length is 24 bits, and chosen ITU, OPC/DPC structure format is : x-y-z；x、z must be decimal value; y is decimal number 0-255, such as: 6-222-3

7. When the SPC length is 24 bits, and chosen Hex, OPC/DPC structure format is :xyz; x、 z must be three bitts hex value; y is 8 bitts hex value, such as:（202E）100 00000101 110

### 2.6.2 SS7 MTP Link

Figure 2-6-4 SS7 MTP Link


Figure 2-6-5 SS7 MTP Link Add



**SS7 MTP link description**

| | |
|---|---|
| Trunk No | It is consistent with foregoing "Trunk No" of SS7 trunk. |
| Link No | Equipment maximum support 2 signaling links, these two links share workload, when one link fails, the other link will take over the load until restore from failure, and then they will share the load again. |
| Signaling Link Code | If a signaling point has established several signaling links, then the code of each signaling link will begin from 0. |
| E1/T1 Port No | Indicate which E1/T1 this link is established on, it is stipulated that such numbering is carried out according to the physical position of E1/T1. |
| Channel No | Indicate time slot that link is established on. It is assigned to 1 or 16 for time slot, the default is 16 time slot. |

### 2.6.3 SS7 Circuit

Figure 2-6-5 SS7 Circuit

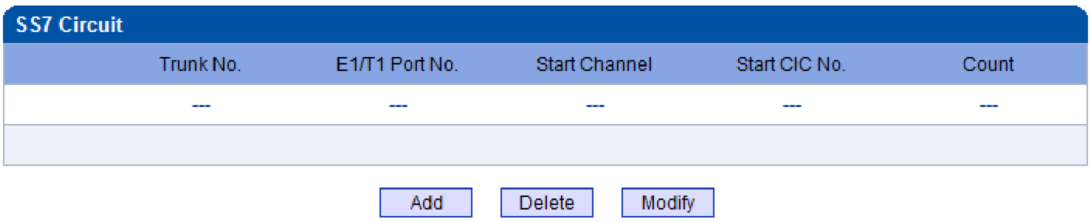
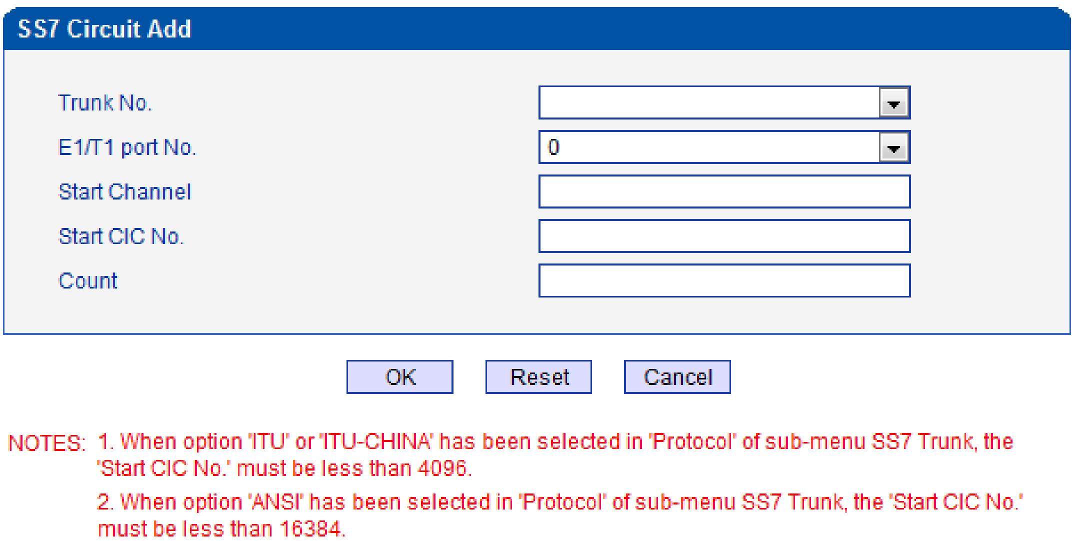



Figure 2-6-6 SS7 Circuit description




CIC (circuit identification code) is an important parameter of SS7 circuit. It should be confirmed with service provider. If the CIC is mismatched, it will result in one-way voice communication.

**SS7 Circuit Add**

| | |
|---|---|
| Trunk No | The "Trunk No." here corresponds to the "Trunk No." of SS7 trunk. |
| E1/T1 port No | Fill in the port number of E1/T1. Assign E1/T1 to selected SS7 trunk. |
| Start Channel | The start of SS7 channel trunk |

| | |
|---|---|
| Start CIC No | An initial circuit number to this E1/T1 matches by both parties |
| Count | A total of 32 channels |

### 2.6.4 SS7 Circuit Maintain

According to the different operating modes, 7 circuit maintenance objects into two categories: ports and channel.

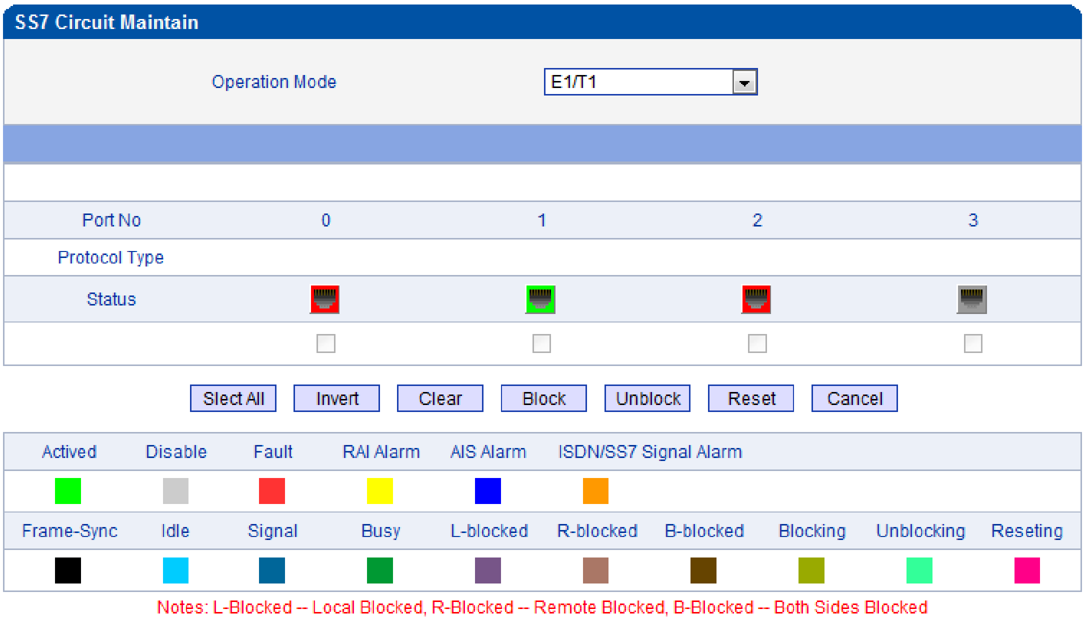



Figure 2-6-7 SS7 Circuit Maintain-E1/T1

**SS7 Circuit Maintain-E1/T1 description**

| | |
|---|---|
| Operation Mode | There are port operation and channel optional |
| Port No | Display the port number |
| Protocol Type | TUP or ISUP |
| Status | There are 16 status with ports, each state corresponds to a color: activated, disable, fault, RAI Alarm, ISDN/SS7 Signal Alarm, Frame-Sync, Idle, Signal, Busy, L-blocked, R-blocked, B-blocked, Blocking, Unblocking and Resetting. |

These ports can work in many ways: Select All, Invert, Clear, Block, Unblock, Reset and Cancel.

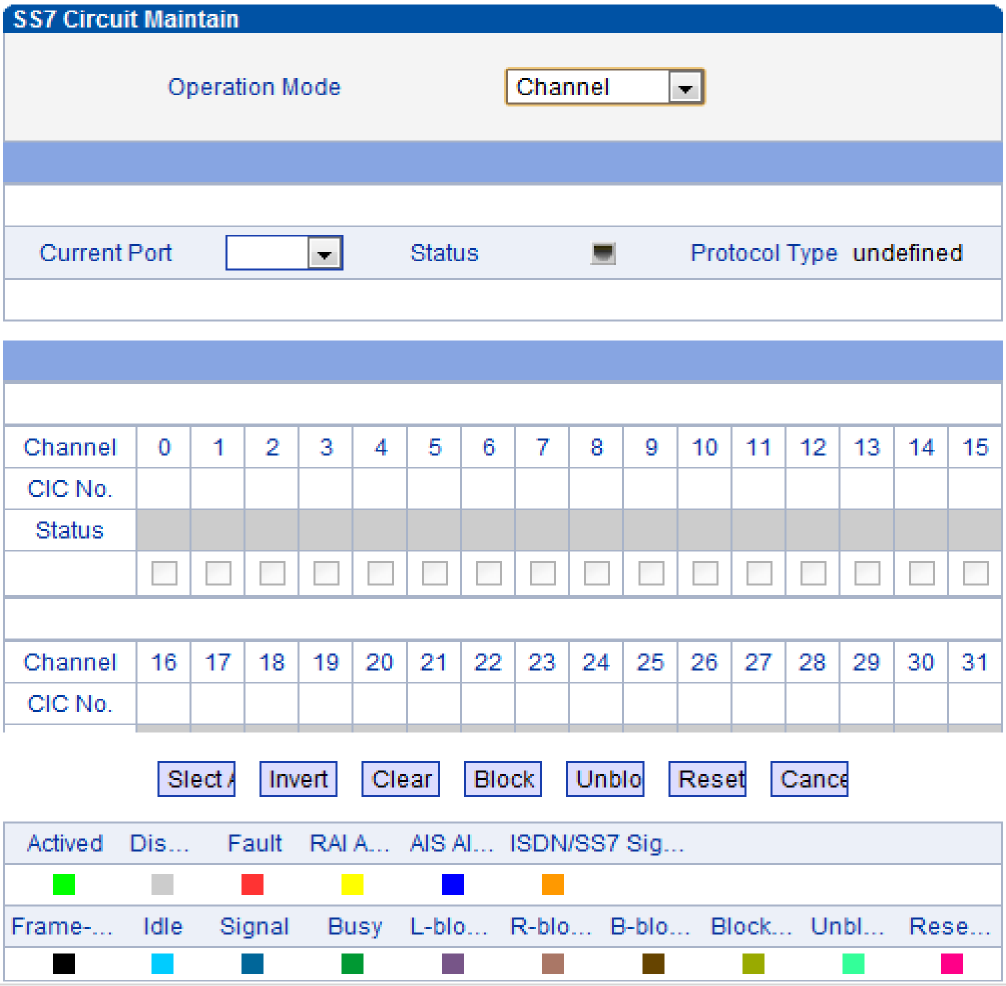



Figure 2-6-8 SS7 Circuit Maintain-Channel

If user wants to manage the channel, please select operation mode to channel.

Select current port, use will see port status and protocol type. The following will show the slot and channel status. There are 16 kinds of channel states and each state corresponds to a color

### 2.6.5 Slave TG Management


Figure 2-6-9 Slave TG Management

When need to share 7 signaling point, add slave TG, so as to realize the multiple TG sharing a link.

## 2.7 PSTN Group Config

### 2.7.1 E1/T1 Parameter

Clock source of E1/T1can be selected “Remote” or “Local”. If selecting E1/T1 port to port0, when user modified port0, port0-3 will be changed together with port0. Port4-7 changed following the port4.

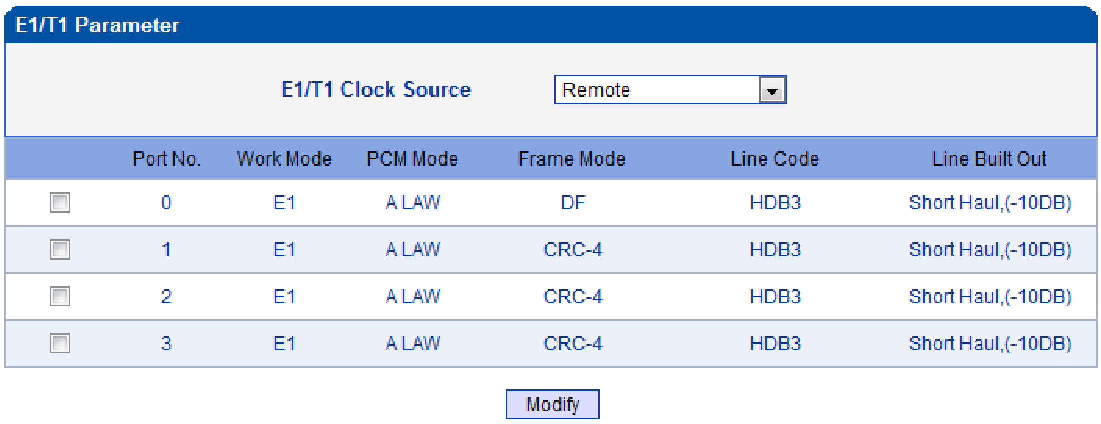



Figure 2-7-1 E1/T1 Parameter

**E1/T1 parameter description**

| | |
|---|---|
| Work Mode | E1/T1, the default is E1. |
| PCM Mode | PCM mode: A LAW and Mu LAW, the default is A LAW |
| Frame Mode | The frame modes of E1 are: DF, CRC-4, CRC4_ITU, the default is CRC-4; the frame modes of T1 are: F12, F4, ESF, F72, the default is F4. |
| Line Code | Line codes of E1 are: NRZ, CMI, AMI, HDB3, the default is HDB3. The Line codes of T1 are: NRZ, CMI, AMI, B8ZS, the default is B8ZS. |
| Line Built Out | Cable length. E1 lines docking, the environment will affect the E1 line signal strength, signal strength according to (DB value) to select the long-term or short-term. |

### 2.7.2 Coder Group

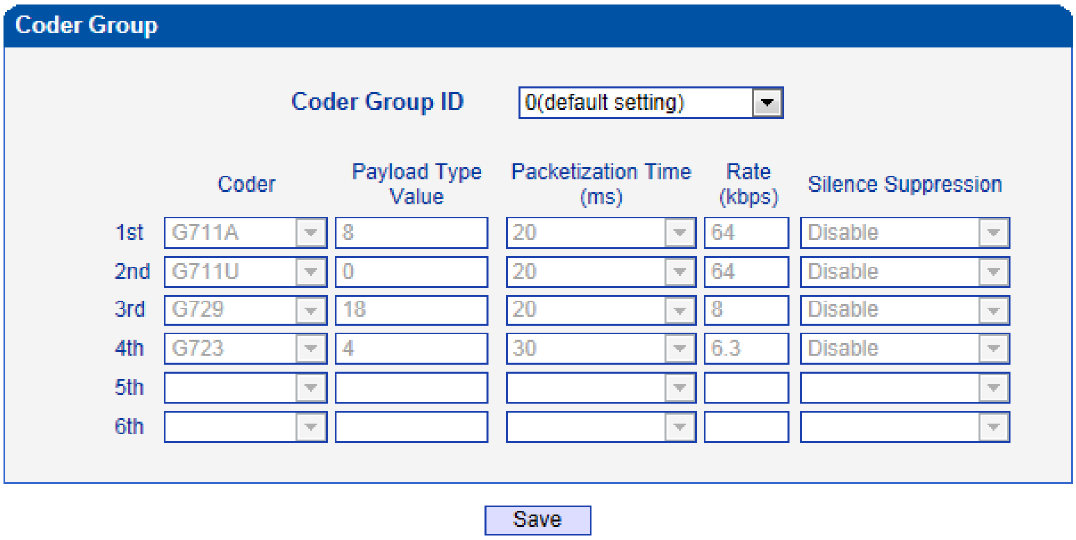



Figure 2-7-2 Coder Group

**Coder group description**

| | |
|---|---|
| Coder Group ID | ID standard for Voice ability, total with 8 groups, where 0 is the default group ID number, the codec that equipment supports in the grouping will be displayed in 0 group. Default value cannot be modified. |
| Coder | Support 3 kinds of voice codec: G.711A/U/G.729/G.723 |
| Payload Type Value | Each codec has a unique value, refer to RFC3551 |
| Packetization Time(ms) | Voice Codec packetization time, user can define different kinds of coding and decoding minimum packetization time |
| Rate(kbps) | Show the rate. |
| Silence Suppression | It is disabled by default. During talking, the bandwidth occupied by voice transmission will be released automatically for silence party or when talk is paused. |
| Coder Group ID | ID standard for Voice ability, total with 8 groups, where 0 is the default group ID number, the codec that equipment supports in the grouping will be displayed in 0 group. Default value cannot be modified. |

### 2.7.3 Dial Plan

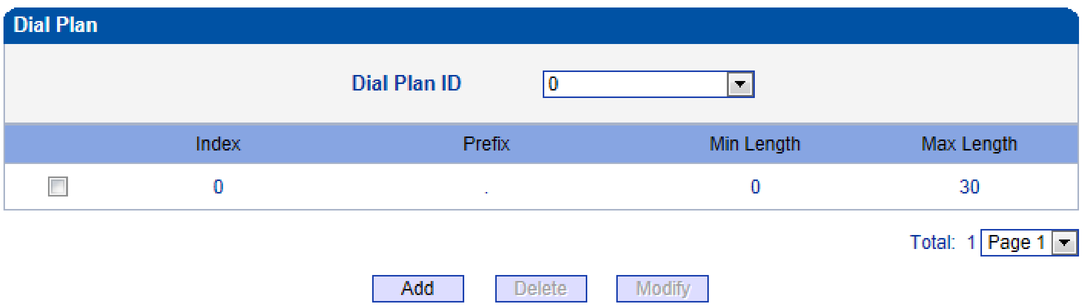



Figure 2-7-3 Dial Plan

Dial plan used for configuring the receiving number, user can configure different prefix number, these rules can be divided into 5 groups with a dial plan ID, where 0 is the default setting.

**Notes**:

1. In order to ensure each rule can take effect, long matching numbers (prefix) rule dial plan index value need smaller.
2. Maximum length is 30, this value is the number of the total length and including the prefix length.

Click “Add” to add dial plan, configuration page as follow:

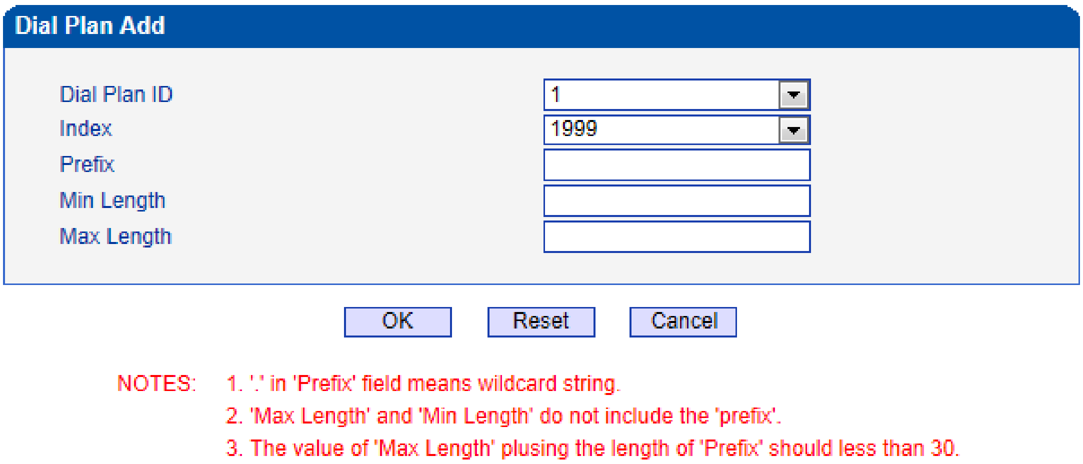



Figure 2-7-4 Dial Plan Add

**Dial Plan description**

| | |
|---|---|
| Dial Plan ID | The number to identify a dial plan |
| Index | Dial plan priority rules take effect in accordance with dial plan index size, and not according to the maximum number received. |
| Prefix | Match number, "." representative of any number |
| Min Length | The minimum receiving Number length (0 to 30). If receiving a number equal to the minimum length greater than, less than equal to the maximum length, the number will be used to continue the call. If the maximum length determine the number to receive a complete, will no longer receive a new number, and immediately began to number analysis. If there are numbers continue to be received, the system will give up these numbers. |
| Max Length | The largest received number length (0 to 30) |

**special version:**

1. Dial plan can be backup and restore in management configuration.
2. “Min Length” and “Max Length” are equal to the total number of possible length minus the prefix length.
3. When overlap dialing, called number length sure, and then the “Min Length” and “Max Length” will

be set to the same value to accelerate connection rate.

4. Prefix configuration, compatible "digit map" mode.

### 2.7.4 Dial Timeout

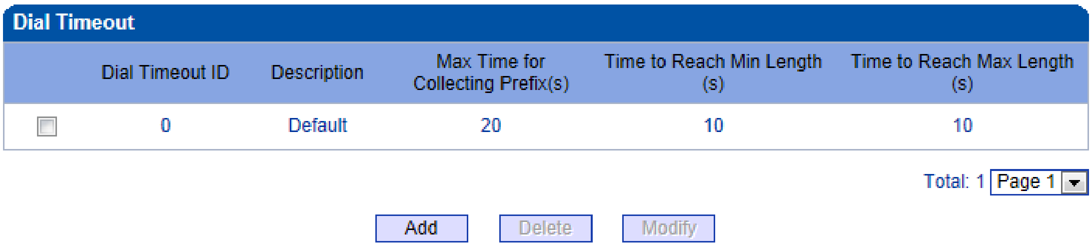



Figure 2-7-5 Dial Timeout

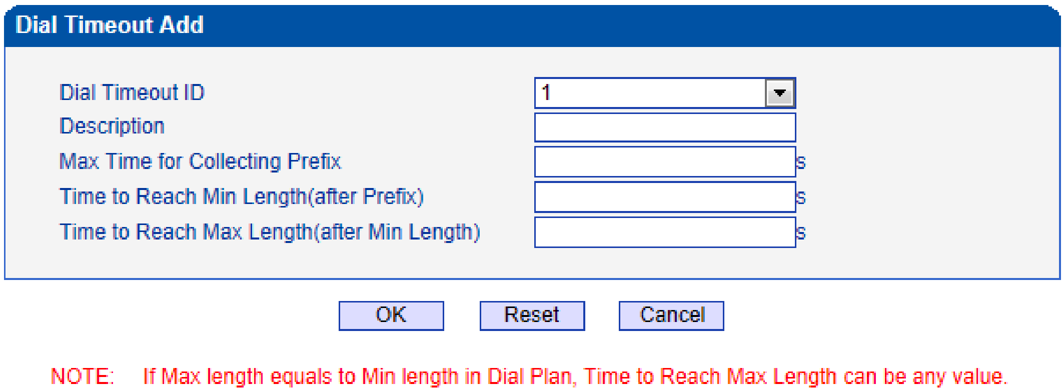



Figure 2-7-6 Dial Timeout Add

**Dial timeout description**

| | |
|---|---|
| Dial Time ID | The number to identify a dial timeout rule |
| Description | Description of dial timeout |
| Max Time for Collecting Prefix | Generally refer to the time from user dial first digit to harvest in prefix number. |
| Time to Reach Min Length(after Prefix) | After receiving prefix number, the number has not yet reached the length of the minimum receiving number, the length of timeout |
| Time to Reach Max Length(after Min Length) | After receiving number, the number has reached the minimum length, but not reached the maximum length of the dial timeout |

### 2.7.5 PSTN Profile

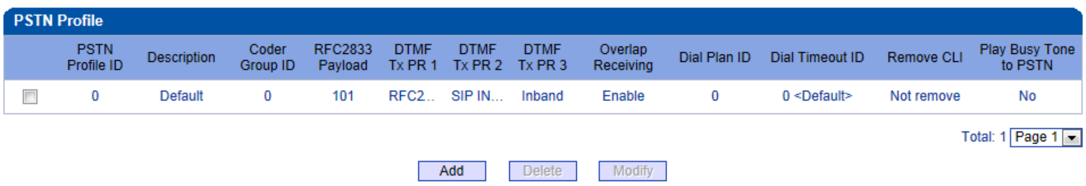


Figure 2-7-7 PSTN Profile

PSTN profile is used to configure PSTN call number rules and parameter.

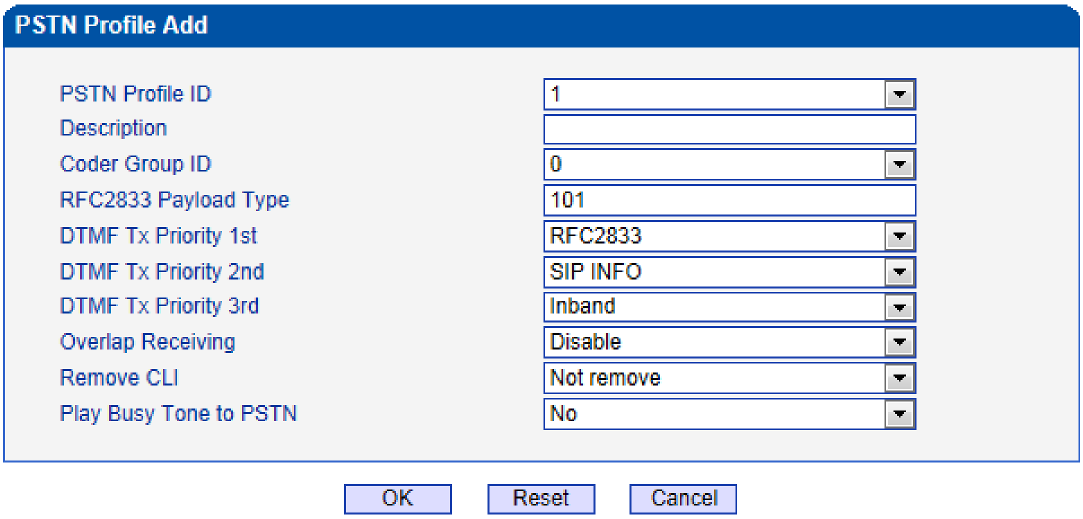



Figure 2-7-8 PSTN Profile Add

**PSTN profile add description**

| | |
|---|---|
| PSTN Profile ID | The number to the PSTN Profile |
| Description | Description of the PSTN Profile |
| Code Group ID | Refer to "Coder Group" |
| RFC2833 Payload Type | The item is 101 by default. |
| 1st/2nd/3rd Tx DTMF Option | There are three ways to send DTMF: RFC2833/SIP INFO/ INBAND, in accordance with the priority choice to send the configuration mode |
| Overlap Receiving | Not enabled by default, only user enables this feature, “Dial plan” and “Dial timeout” would work. |
| Remove CLI | Default does not remove CLI |
| Play busy tone to PSTN | Equipment will play busy tone from IP to PSTN |
| PSTN Profile ID | The number to the PSTN Profile |
| Description | Description of the PSTN Profile |

## 2.7.6 PSTN Group

PSTN group configuration can be different E1/T1ports or the same port in different time slots to form a PSTN trunk group based on different channel selection.

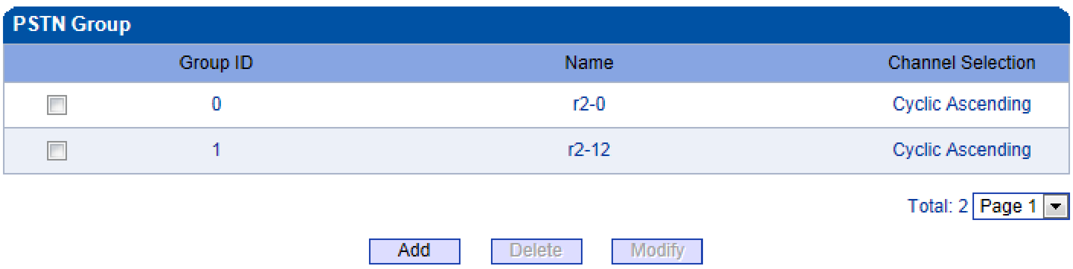



Figure 2-7-9 PSTN Group

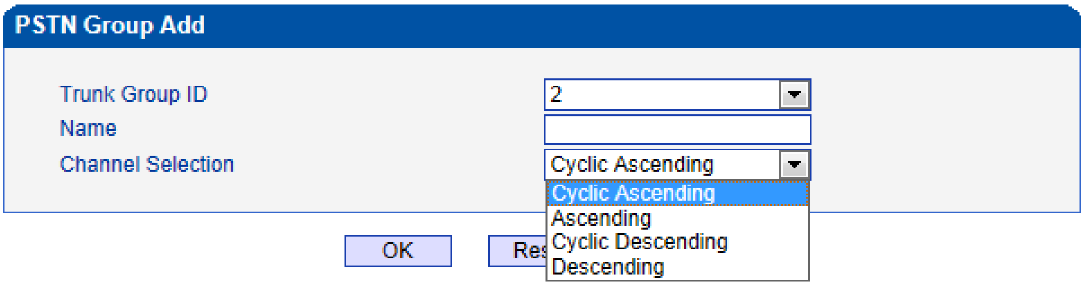



Figure 2-7-10 PSTN Group Add

Adding PSTN group needs to fill three parameters: trunk group Numbers, trunk group Name. Channel selection mode and at most, can add up to 16 set of data. Channel selection mode refers to E1/T1 timeslot allocation strategy in a trunk group. There are four options: Ascending, Descending, Cyclic Ascending and Cyclic Descending for routing.

### 2.7.7 PSTN Group Management

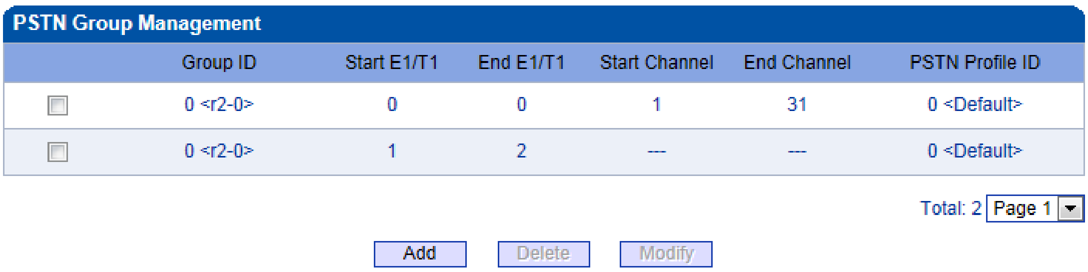



Figure 2-7-11 PSTN Group Management

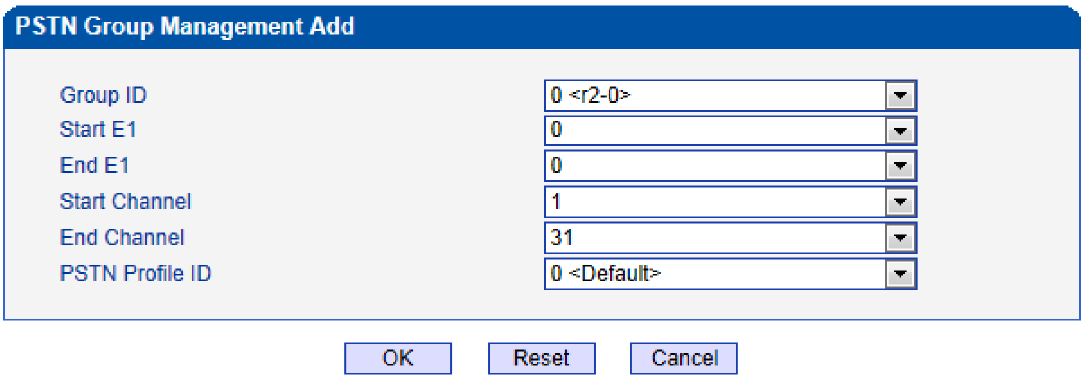



Figure 2-7-12 PSTN Group Management Add

**PSTN group management add**

| | |
|---|---|
| Group ID | PSTN group ID |
| Start E1 | E1/T1 trunk group port number in the initial |
| End E1 | Last a E1/T1 trunk group port number |
| Start Channel | The beginning of time slot, assigned a precise time slot for a group of trunk |
| End Channel | The end of time slot, assigned a precise time slot for a group of trunk |
| PSTN Profile ID | Refer to PSTN Profile |

When cross E1 port operation, don't choose start/termination of the time.

## 2.8 SIP Config

### 2.8.1 SIP Parameter

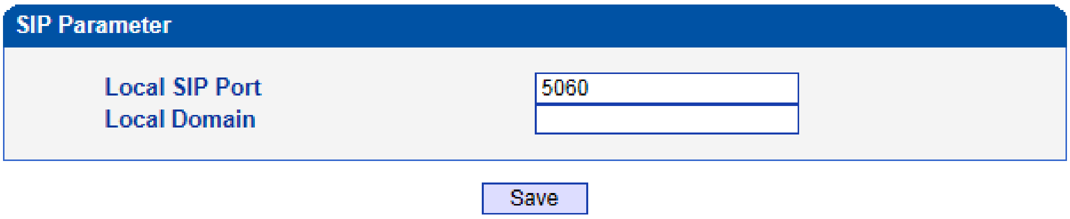



Figure 2-8-1 SIP Parameter

The default Local SIP Port is 5060, and Local Domain set here can replace SIP account.

### 2.8.2 SIP Trunk

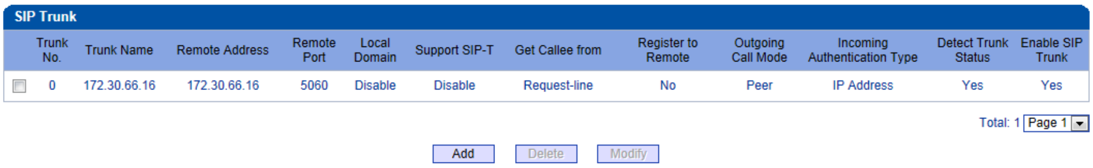



Figure 2-8-2 SIP Trunk

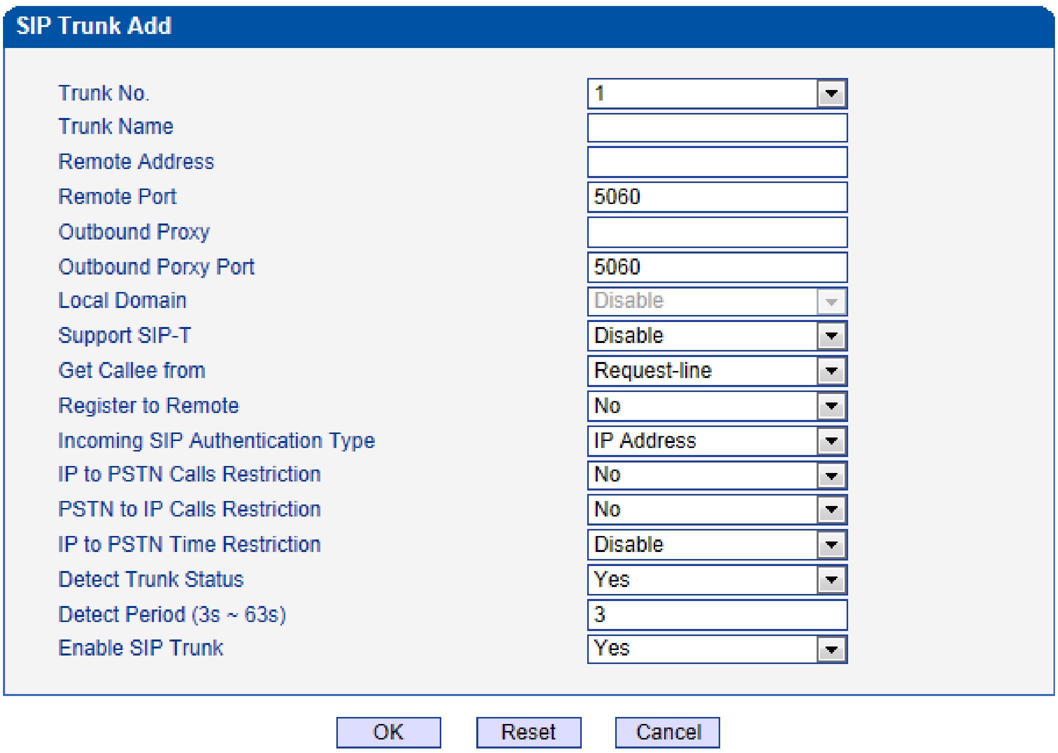



Figure 2-8-3 SIP Trunk Add

**SIP trunk description**

| | |
|---|---|
| Trunk No | The range of number is 1~99 |
| Trunk Name | Description the trunk |
| Remote Address | IP address of remote platform interfacing with this equipment. |
| Remote Port | Q.931 port of SIP of remote platform interfacing with this equipment, the default is 5060 |
| Outbound Proxy | SIP proxy IP address |
| Outbound Proxy Port | The default proxy port is 5060. |
| Local Domain | Refer to SIP parameter |
| Support SIP-T | Not the target configuration, the parameter is always no. it is for SS7. |
| Get Callee from | Received the called number from request domain or “To header” filed |
| Register to Remote | Defined by IETF work group RFC3372, it is a standard used to establish remote communication between SIP and ISUP; the default is “Yes”; if SIP trunk does not support, then set it to “No”. |
| Incoming SIP Authentication Type | There are two modes: IP address and Password. If user selects “password”, then password will be filled. |
| IP to PSTN Calls Restriction | IP to PSTN side of the limitation on the number of calls; the range is 0~65535, the default is no limitation; If Yes is selected, then input limitation number of calls in the edit box appeared. |
| PSTN to IP Calls Restriction | PSTN to IP side of the limitation on the number of calls; the range is 0~65535, the default is no limitation; If Yes is selected, then input limitation number of calls in the edit box appeared. |
| IP to PSTN Time Restriction | The default setting is disabled. If Enabled is selected, then user can edit the start and stop time of prohibition time interval. Within this time interval, all calls from IP to PSTN are prohibited. (Calls from PSTN to IP are not limited) |
| Detect Trunk Status | Detect the status of SIP trunk. If select it, the equipment will send HEARTBEAT message to peer to make sure the link status is OK. |
| Enable SIP Trunk | A switch used to enable this SIP trunk or not; user can select “Yes” or “No”, when “No” is selected, this SIP trunk is invalid. |

### 2.8.3 SIP Account

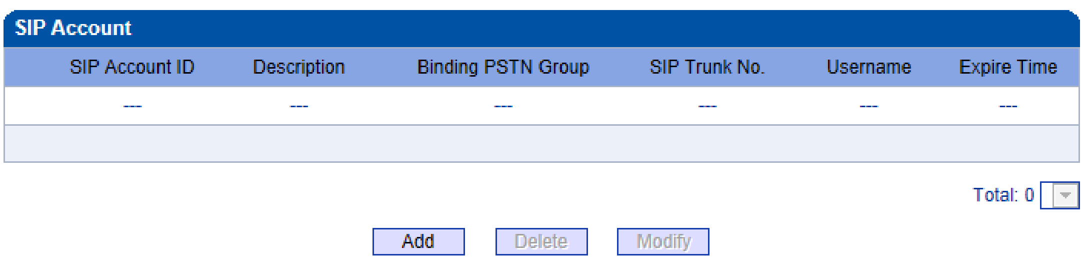



Figure 2-8-4 SIP Account

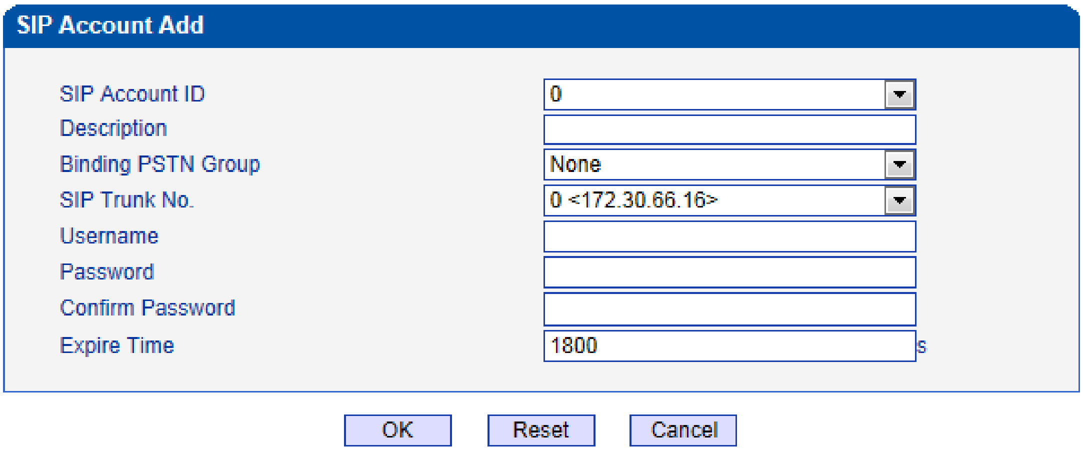



Figure 2-8-5 SIP Account Add

This option is when the equipment is in the registered mode, used to manage SIP trunk account.

**SIP trunk account**

| | |
|---|---|
| SIP Account ID | SIP Account Number, from 0-127 |
| Description | Description of the SIP account |
| Binding PSTN Group | IP trunk group number, “any” indicates any trunk group |
| SIP Trunk No | The corresponding number and name of the SIP trunk |
| Username | SIP registration user name, the same SIP trunk can configure multiple SIP accounts, corresponding to different trunk group ID |
| Password | Registered password |
| Confirm Password | Enter the password again. |
| Expire Time | SIP registration interval, default is 1800s |

# 2.9 H323 Config

## 2.9.1 H.323 Parameter

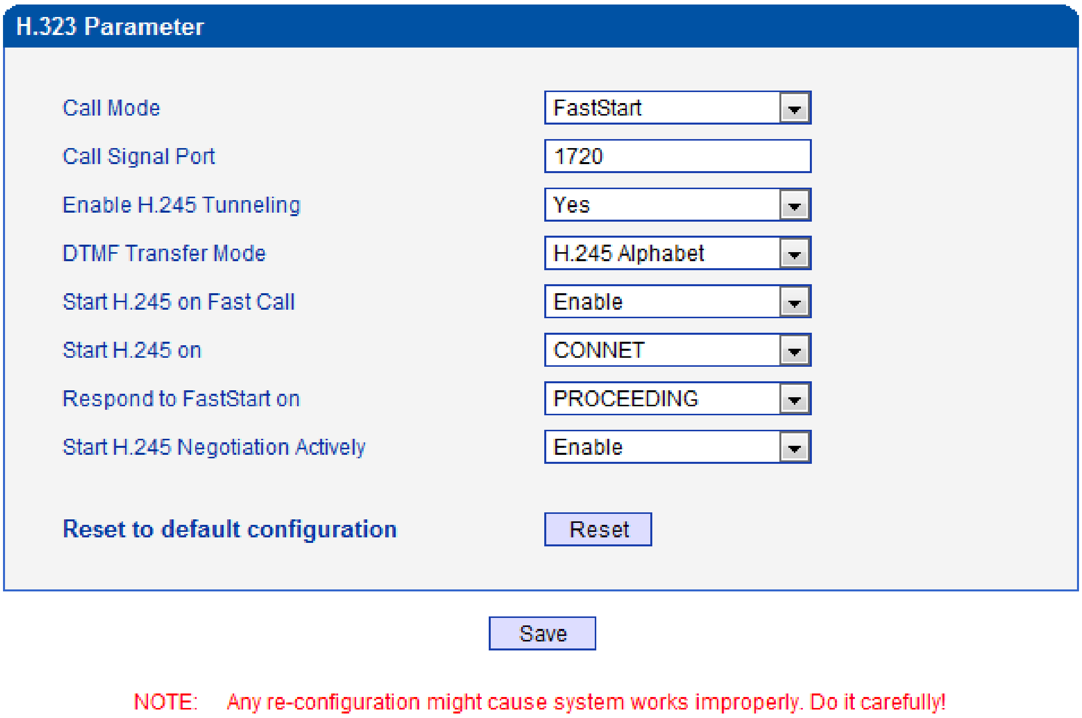



Figure 2-9-1 H.323 Parameter

**H.323 Parameter description**

| | |
|---|---|
| Call Mode | Supports faststart mode and conventional mode, faststart mode through faster. |
| Call Signal Port | Default call signal port is 1720 |
| Enable H245 Tunneling | H. 245 is the multimedia communication control signaling protocol in H.323, and its control of information running in H.245 control channels. Default, the channels open forever. |
| DTMF Transfer Mode | Send mode has two: H.245 Alphabet and H.245 Signal, default is H.245 Alphabet mode. |
| Start H245 on Fast Call | Whether establish H.245 agreement |
| Start H245 on | There are three steps building H.245: Call Connection, Signal Sending and Proceeding, default is Connect. |
| Respond to Faststart on | When call mode is faststart mode, response phase is divided into three stages: Call Connection, Signal Sending and Proceeding, default is Proceeding phase. |
| Start H.245 Negotiation Actively | Whether establish H.245, terminal equipment will be sent H.245 negotiation news consult. |
| Reset to default configuration | Click the button to recover factory configuration. |

## 2.9.2 H.323 Trunk

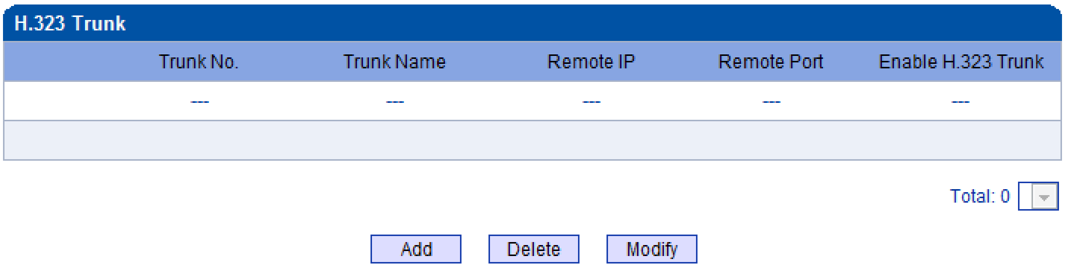



Figure 2-9-2 H.323 Trunk

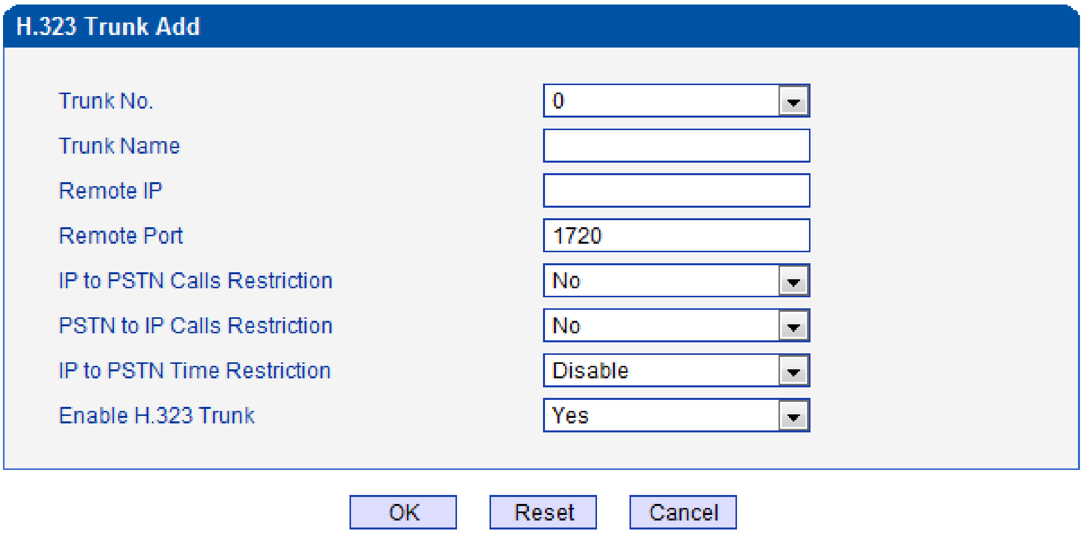



Figure 2-9-3 H.323 Trunk Add

**H.323 trunk description**

| | |
|---|---|
| Trunk No. | Can add up to 63 trunk |
| Trunk Name | Named for the trunk |
| Remote IP | Equipment to end interface platform IP |
| Remote Port | Equipment to end interface platform port, default is 1720. |
| IP to PSTN Calls Restriction | IP to the side of the PSTN concurrent call the default without restriction. If select Yes, and then fill in limited number of concurrent call in edit box. The max is 65535. |
| PSTN to IP Calls Restriction | PSTN to the side of the IP concurrent call the default without restriction. If select Yes, and then fill in limited number of concurrent call in edit box. The max is 65535. |
| IP to PSTN Time Restriction | Default disables the function. If select enable, users will edit banning call of the start time and end time. All call from IP to PSTN will be prohibited in this period time. |
| Enable H.323 Trunk | After configuration, whether restart device. |

## 2.10 IP Group Config

The user can group manage SIP/H.323 trunk through IP packet configuration.

### 2.10.1 IP Profile

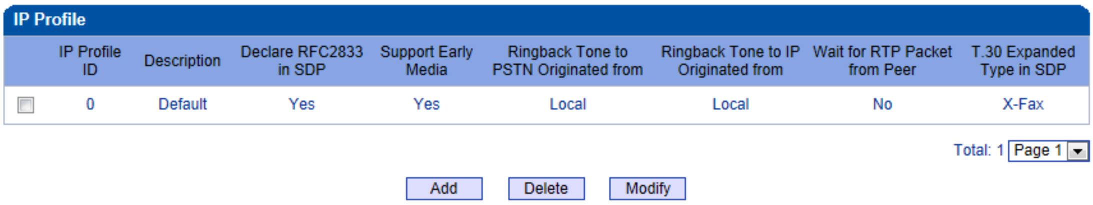



Figure 2-10-1 IP Profile

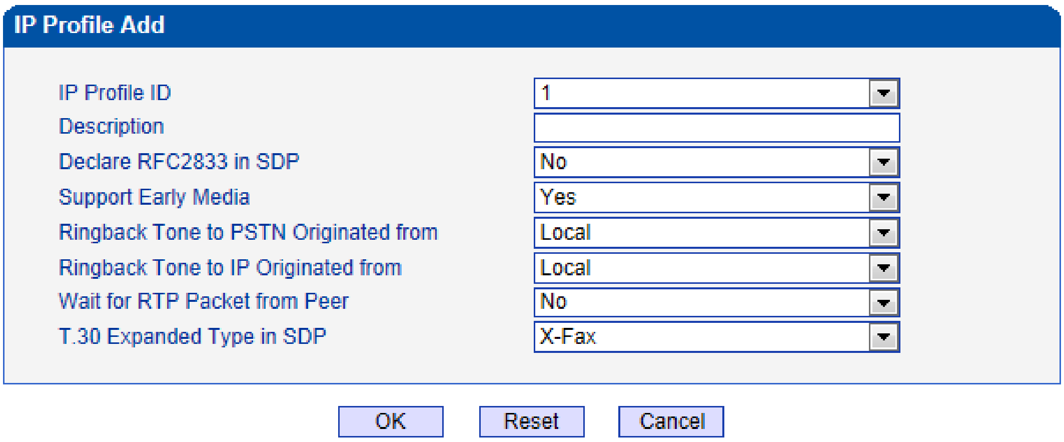



Figure 2-10-2 IP Profile Add

**IP profile add**

| | |
|---|---|
| IP Profile ID | IP property identification number can be configured to 15 properties |
| Description | Description of the IP Profile |
| Declare RFC2833 in SDP | Default support |
| Support Early Media | Whether support Early Media(183). If select “Yes”, the called side to the early media to provide ring back tone to the caller. |
| Ring back Tone to PSTN Originated from | IP-> PSTN call ring back tone player side, if setting to local, it will play from the equipment. If setting to IP , it will play by the called |
| Ring back Tone to IP Originated from | PSTN->IP call ring back tone player side, if setting to local, it will play from the equipment and set to PSTN, it will play by the called |
| Wait for RTP Packet from Peer | If set to No, it will auto send RTP packets during the call and if set to Yes, it will wait the RTP packet was sent by the back side first ,then send out RTP packets |
| T.30 Expanded Type in SDP | T30 extended types in SDP: x-fax or fax |

### 2.10.2 IP Group

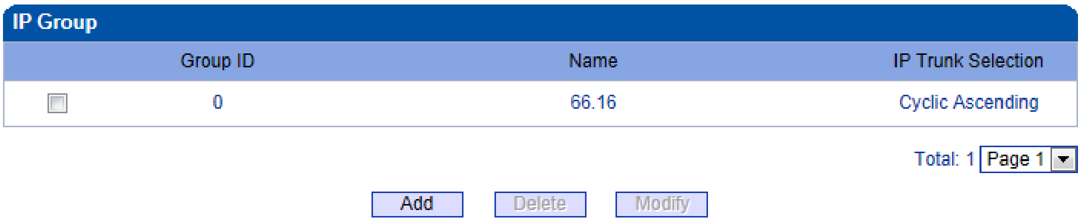



Figure 2-10-3 IP Group

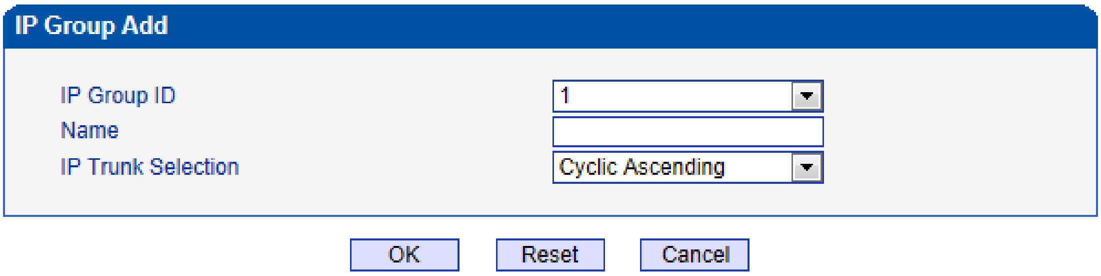



Figure 2-10-4 IP Group Add

Add the IP group including the IP group ID, IP group name, IP trunk selection. User can add a total of 16 IP group. IP routing mod is to show in an IP group SIP time distribution strategy. There are four options: Ascending, Descending, Cyclic ascending, Cyclic descending. (According to SIP trunk number to choice)

### 2.10.3 IP Group Management

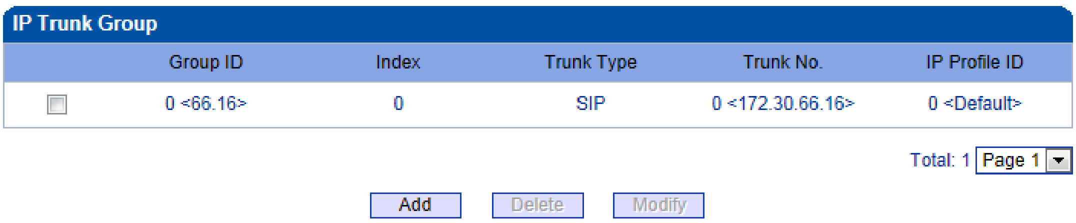



Figure 2-10-5 IP Trunk Group

**IP trunk group description**

| | |
|---|---|
| Group ID | IP group ID |
| Index | The priority value of 0-15 |
| Trunk Type | Currently only supports SIP, H.323 will be also supported in future |
| Trunk No | SIP trunk number |
| IP Profile ID | Refer to IP Profile |

## 2.11 Call Routing

### 2.11.1 Routing Parameter

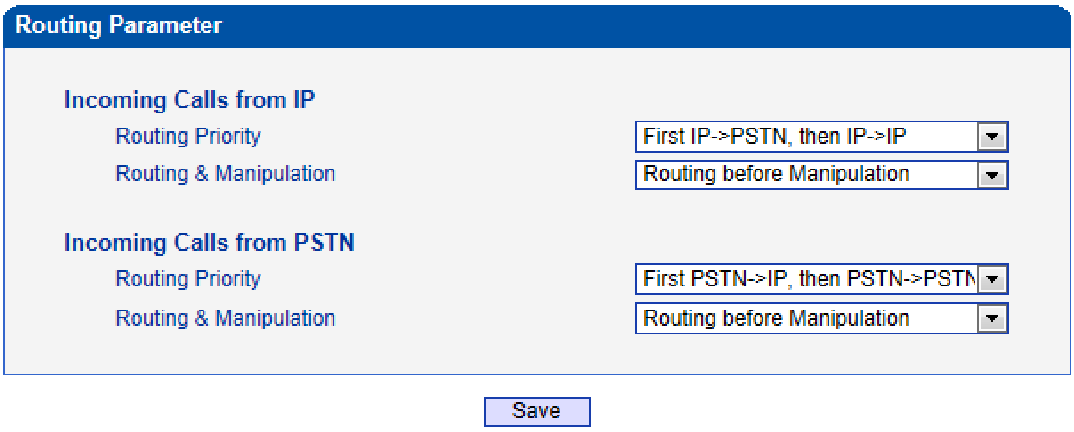



Figure 2-11-1 Routing Parameter

Inbound and outbound call routing configuration

The key steps how to Configure routing：

The more accurate routing configuration, index values should be smaller.

“Any” and "." are useful; suggesting configuration, to avoid cannot match the routing.

### 2.11.2 PSTN->IP Routing

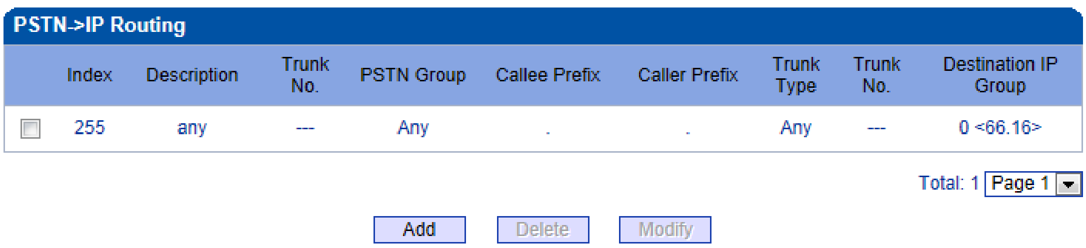



Figure 2-11-2 PSTN->IP Routing

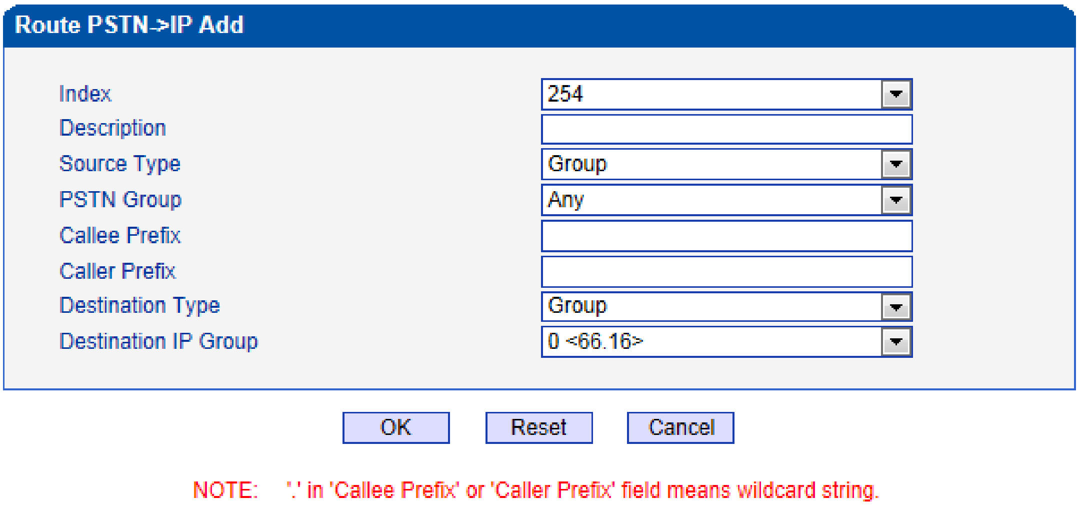



Figure 2-11-3 PSTN->IP Add

"PSTN -> IP Routing": Routing Call from PSTN to IP

**PSTN->IP routing description**

| | |
|---|---|
| Index | Routing index number (0 ~ 255) , "PSTN->IP Routing" priority rule is according to the index to set. Reference dial plan. |
| Description | Describe the routing |
| Source Type | Source type is PSTN group or PRI/SS7 trunk. |
| PSTN Group | Refer to "PSTN Group Config", any means any trunk group. |
| Callee Prefix | Callee number matches prefix number, "." Is a wildcard, representing any callee number |
| Caller Prefix | Caller number matches prefix number, "." Is a wildcard, representing any caller number |
| Destination Type | Destination type is IP group or SIP/H.323 trunk. |
| Destination IP Group | Refer to "IP Group" |
| Trunk Type | Trunk type means IP side trunk type-SIP/H.323. |
| Trunk No. | Trunk number |

### 2.11.3 PSTN->PSTN Routing

Figure 2-11-4 PSTN->PSTN Routing

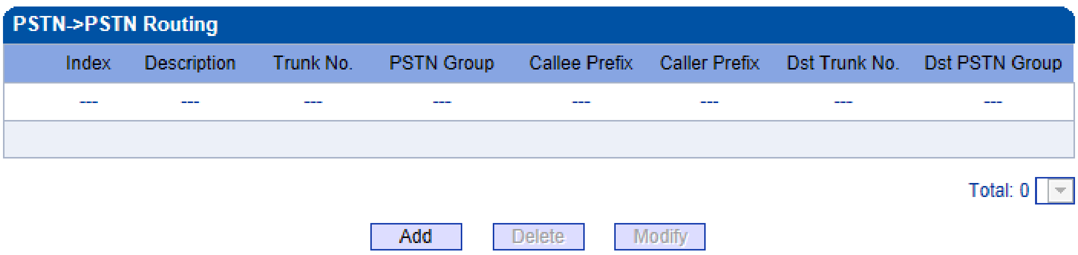
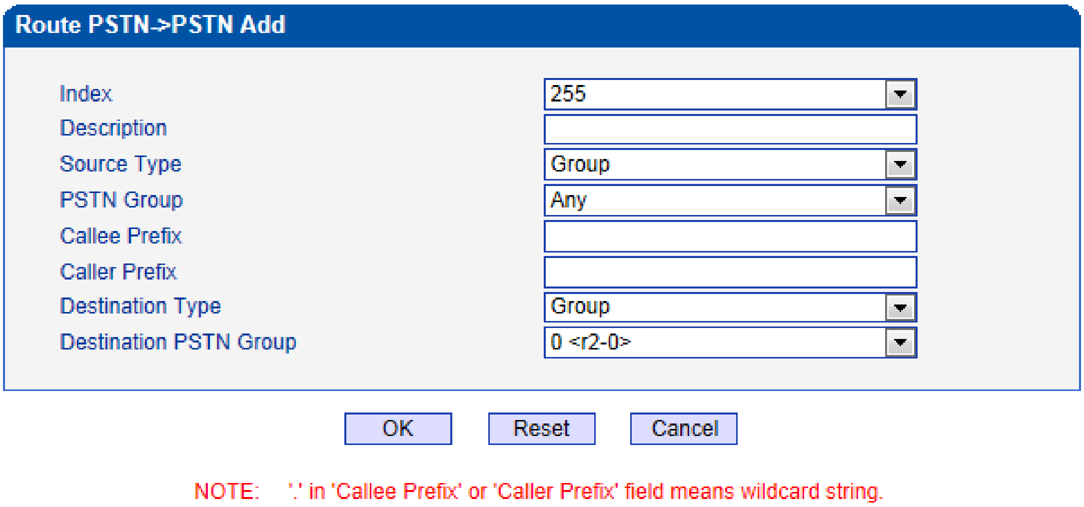


Figure 2-11-5 PSTN->PSTN Add




"PSTN->PSTN Routing": Routing Call from PSTN to PSTN

**PSTN->PSTN Routing**

| | |
|---|---|
| Index | Routing index number (0 ~ 255) , "PSTN->IP Routing" priority rule is according to the index to set. Reference dial plan. |
| Description | Describe the routing |
| Source Type | Source type is PSTN group or PRI/SS7 trunk. |
| PSTN Group | Refer to "PSTN Group Config", any means any trunk group. |
| PSTN Trunk | Reference "PRI Trunk" or "SS7 Trunk" |
| Callee Prefix | Callee number matches prefix number, "." Is a wildcard, representing any callee number |
| Caller Prefix | Caller number matches prefix number, "." Is a wildcard, representing any caller number |
| Destination Type | Destination type is PSTN group or SIP/H.323 trunk. |
| Destination PSTN Group | Refer to "PSTN Group Config" |

### 2.11.4 IP->PSTN Routing

Figure 2-11-6 IP->PSTN Routing

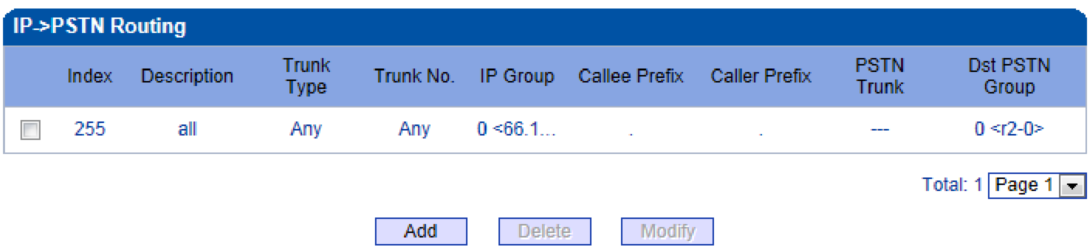
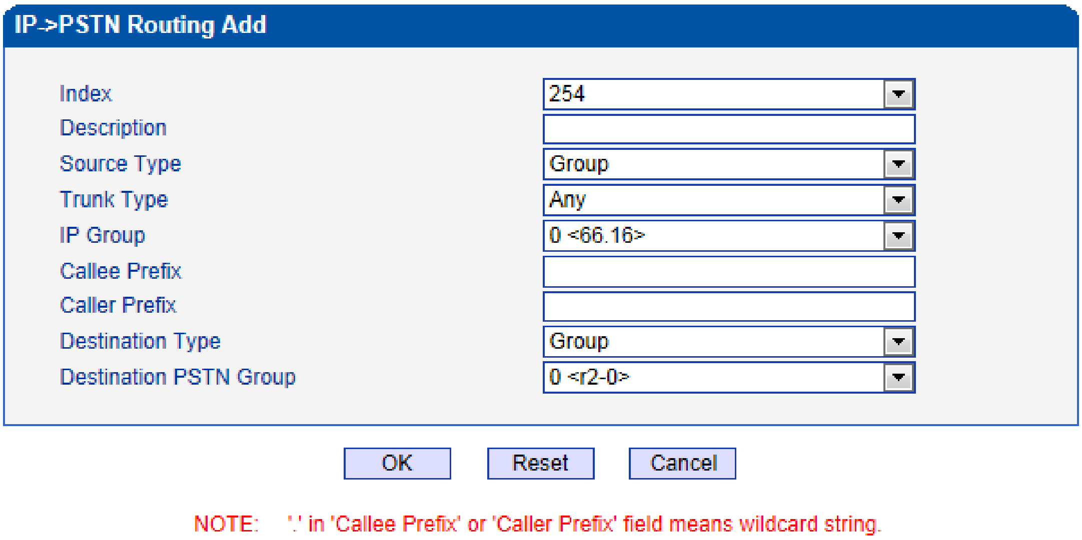


Figure 2-11-7 IP->PSTN Routing




"IP -> PSTN Routing"： Routing Call from IP to PSTN

IP->PSTN routing configuration and PSTN->PSTN routing configuration are similar, the only difference is PSTN destination group.

### 2.11.5 IP->IP Routing

Figure 2-11-8 IP->IP Routing

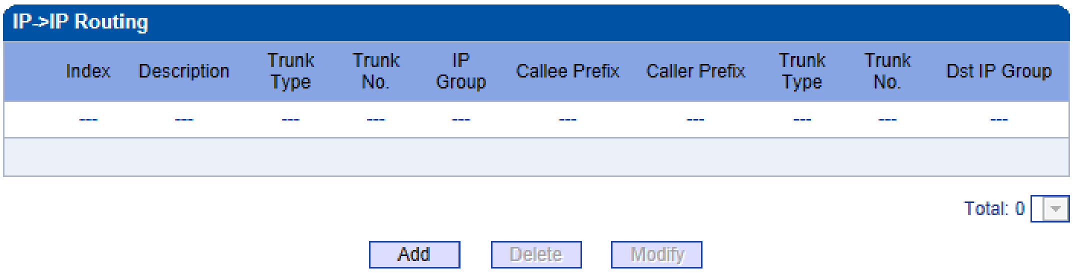
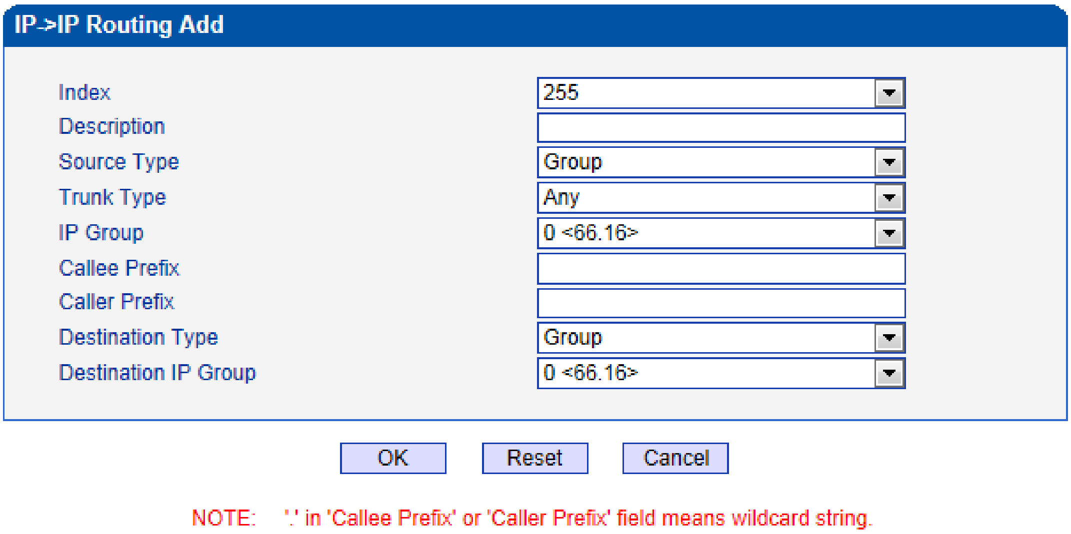



Figure 2-11-9 IP->IP Add

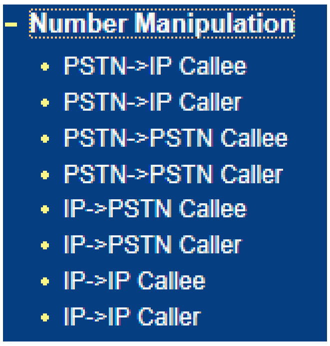



IP->IP routing configuration and PSTN->IP configuration are similar. The only difference is that the destination is the IP group.

## 2.12 Number Manipulation

Select "Number Manipulation" in navigation tree, the display interface is shown as below:

Figure 2-12-1 Number Manipulation




"Number Manipulation" is used to replace numbers. User can replace and remove the inbound and outbound calling / called number.

**Notes:**

1. The more precise configuration, index values should be smaller.

2. Suggesting configure "Any" and ".", avoid missing the call for the replace number。

3. When configuring data, it is suggested that index starts from large index value, to avoid adding an exact match data, not directly use the data.

4. When configuring data, it is suggested that keep using index value.

### 2.12.1 PSTN->IP Callee

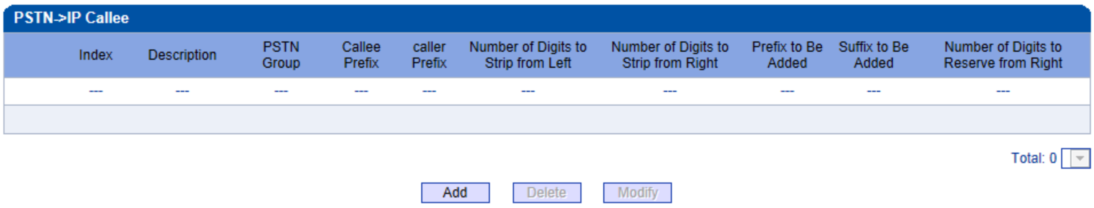



Figure 2-12-2 PSTN->IP Callee

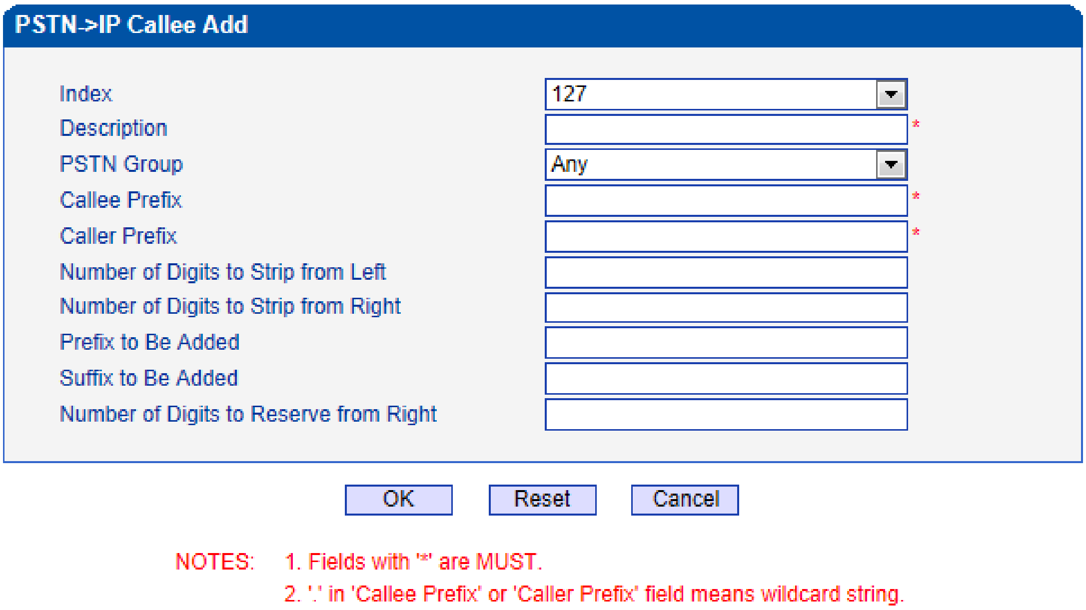


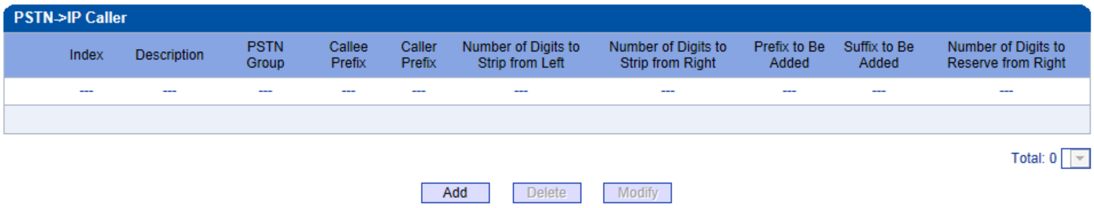

Figure 2-12-3 PSTN->IP Callee Add

"PSTN->IP Callee"： Replace the called number from PSTN

**PSTN->IP destination number**

| | |
|---|---|
| Index | Index number (0 ~ 127) |
| Description | Describe the transformation of the number |
| PSTN Group | Refer to "PSTN Group", "any" means any trunk group |
| Callee Prefix | Called number prefix, "." mean any called number |
| Caller Prefix | Caller number prefix, "." Mean any caller number |
| Number of Digits to Strip from left | Remove the called number digits from the left |
| Number of Digits to Strip from right | Remove the called number digits from the right |
| Prefix to be Add | Add a called number prefix |
| Suffix to be Add | Add a called number suffix |
| Number of Digits to Reserve from Right | Starting from the right to retain the called number digits |

### 2.12.2 PSTN->IP Caller



PSTN->IP Caller

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

Total: 0

Add Delete Modify

Figure 2-12-4 PSTN->IP Caller

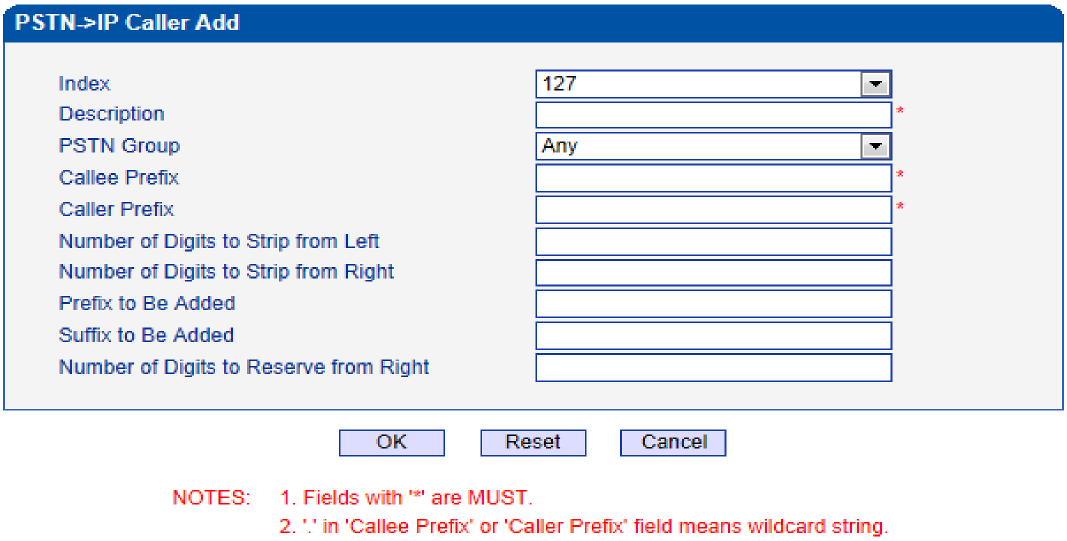



Figure 2-12-5 PSTN->IP Caller Add

PSTN->IP Callee configuration parameters and IP->PSTN Caller configuration parameters are the same.

PSTN->PSTN Callee

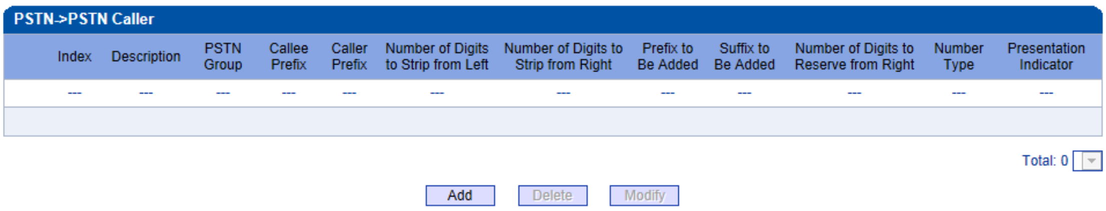
| Index | Description | PSTN Group | Callee Prefix | Caller Prefix | Number of Digits to Strip from Left | Number of Digits to Strip from Right | Prefix to Be Added | Suffix to Be Added | Number of Digits to Reserve from Right | Number Type |
|---|---|---|---|---|---|---|---|---|---|---|
| --- | ---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

Total: 0

Add Delete Modify

Figure 2-12-6 PSTN->PSTN Callee

PSTN->PSTN Callee configuration parameters with the above is basically same, only more of a "number type" parameter. Common number types are: Not Configured, Unknown, International, National, Network Specific, Subscriber, Abbreviated.



PSTN->PSTN Caller

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

Total: 0

Add Delete Modify

Figure 2-12-7 PSTN->PSTN Caller

"Presentation indicator" parameter used to indicate the status of the operation.

The operation of the option the right are: Not configured, Allowed, Restricted.

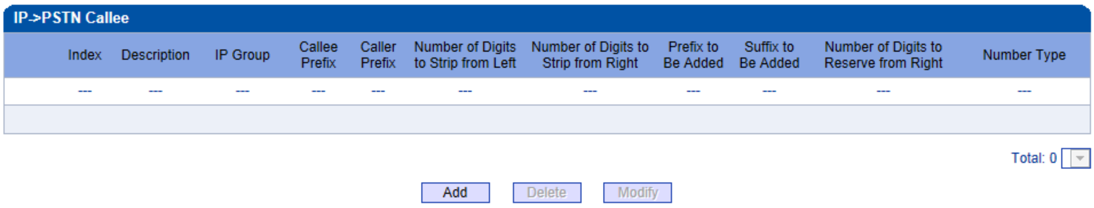



Figure 2-12-8 IP->PSTN Callee

**IP->PSTN callee description**

| | |
|---|---|
| Index | Index number (0 ~ 127) |
| Description | Describe the transformation of the number |
| IP Group | Refer to “IP Group”, “any” means any trunk group |
| Callee Prefix | Called number prefix, “.” means any called number |
| Caller Prefix | Caller number prefix, “.” Means any caller number |
| Number of Digits to Strip from left | Remove the called number digits from the left |
| Number of Digits to Strip from right | Remove the called number digits from the right |
| Prefix to be Add | Add a called number prefix |
| Suffix to be Add | Add a called number suffix |
| Number of Digits to Reserve from Right | Starting from the right to retain the called number digits |
| Number Type | Common number types are: Not Configured, Unknown, International, National, Network Specific, Subscriber and Abbreviated. |

“IP->PSTN Caller”, “IP->IP Callee”, “IP->IP Caller” configuration parameters in the previous number manipulation rules have been mentioned, please refer that section.

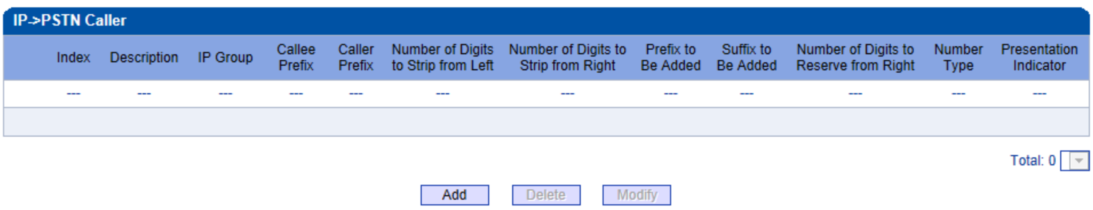



Figure 2-12-9 IP->PSTN Caller

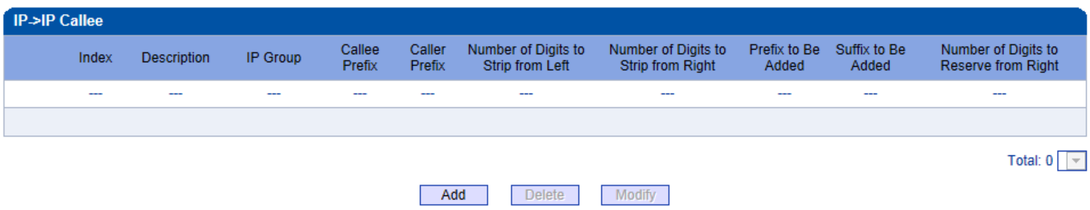


Figure 2-12-10 IP->IP Callee

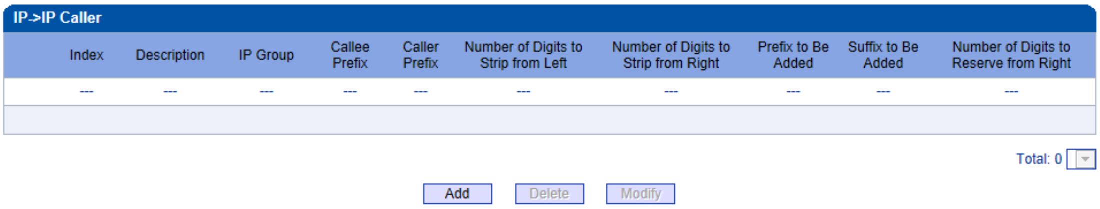



Figure 2-12-11 IP->IP Caller

## 2.13 Voice & Fax

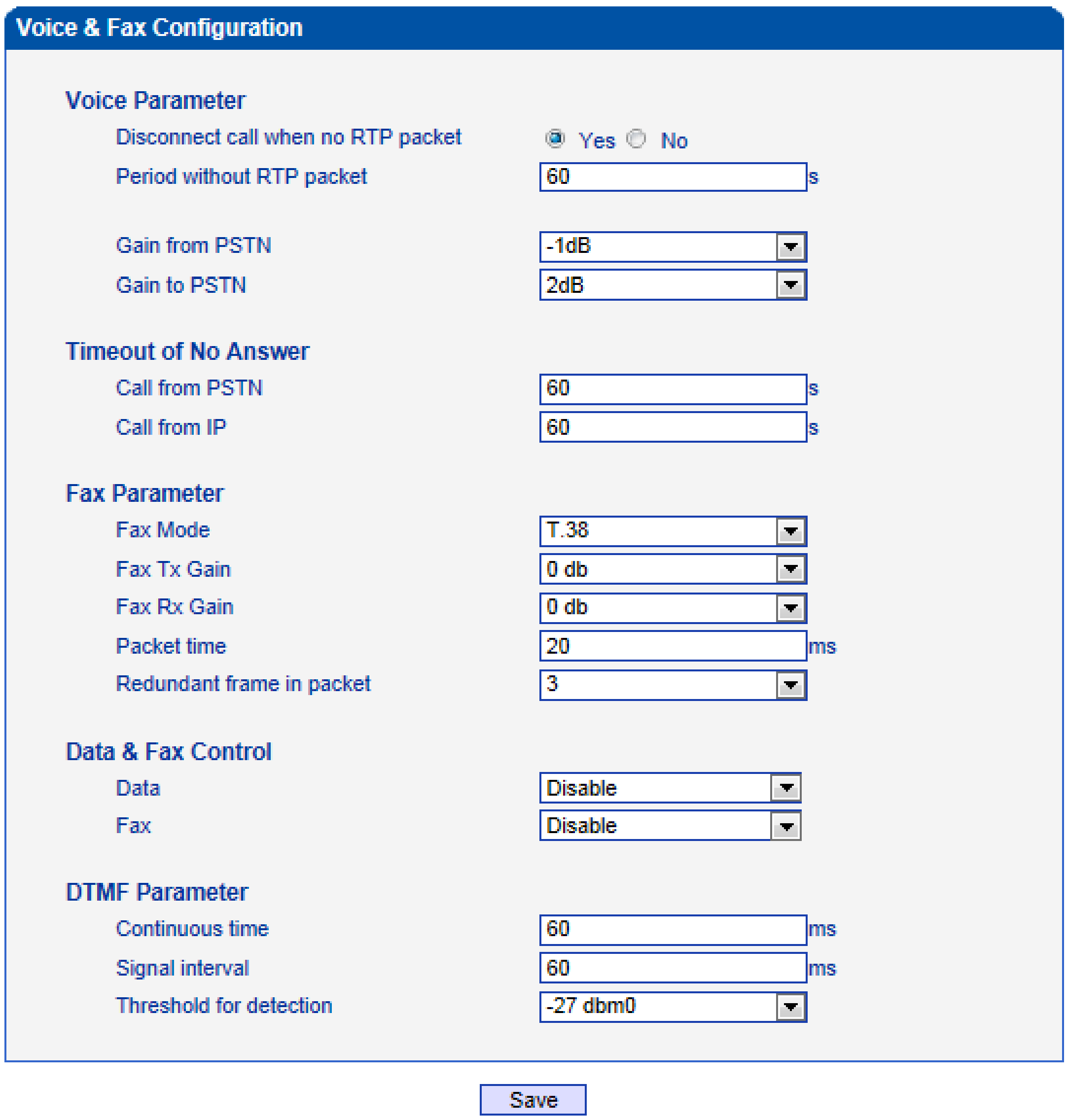



Figure 2-13-1 Voice & Fax

**Voice & Fax description**

| | | |
|---|---|---|
| Voice Parameter | Disconnect Call when no RTP packet | When selected “Yes”, detected call’s silence time longer than silence timeout that for a long time not received RTP packets, then hangup the call. |
| | Period without RTP packet | The maximum time length of silence |
| | PSTN in Gain | Incoming PSNT gain |
| | IP in Gain | Incoming IP gain |
| Timeout of no answer | Call from PSTN | Call timeout of no answer from PSTN |
| | Call from IP | Call timeout of no answer from IP |
| Fax Parameter | Fax Mode | Two modes are provided: T.38/Pass-through; default option is T.38. |
| | Fax Tx Gain | Gain of sending a fax |
| | Fax Rx Gain | Gain of receiving a fax |
| | Packet time | Data packing duration |
| | Redundant frame in packet | The length of frame in RTP packet |
| Data & Fax Control | Data | Whether to allow the control of voice data |
| | Fax | Whether to allow the control of fax |
| DTMF Parameter | Continuous time | The level of a frequency duration |
| | Signal interval | The time interval between two different frequency signals |
| | Threshold for detection | Frequency detection threshold |

## 2.14 Management Parameter

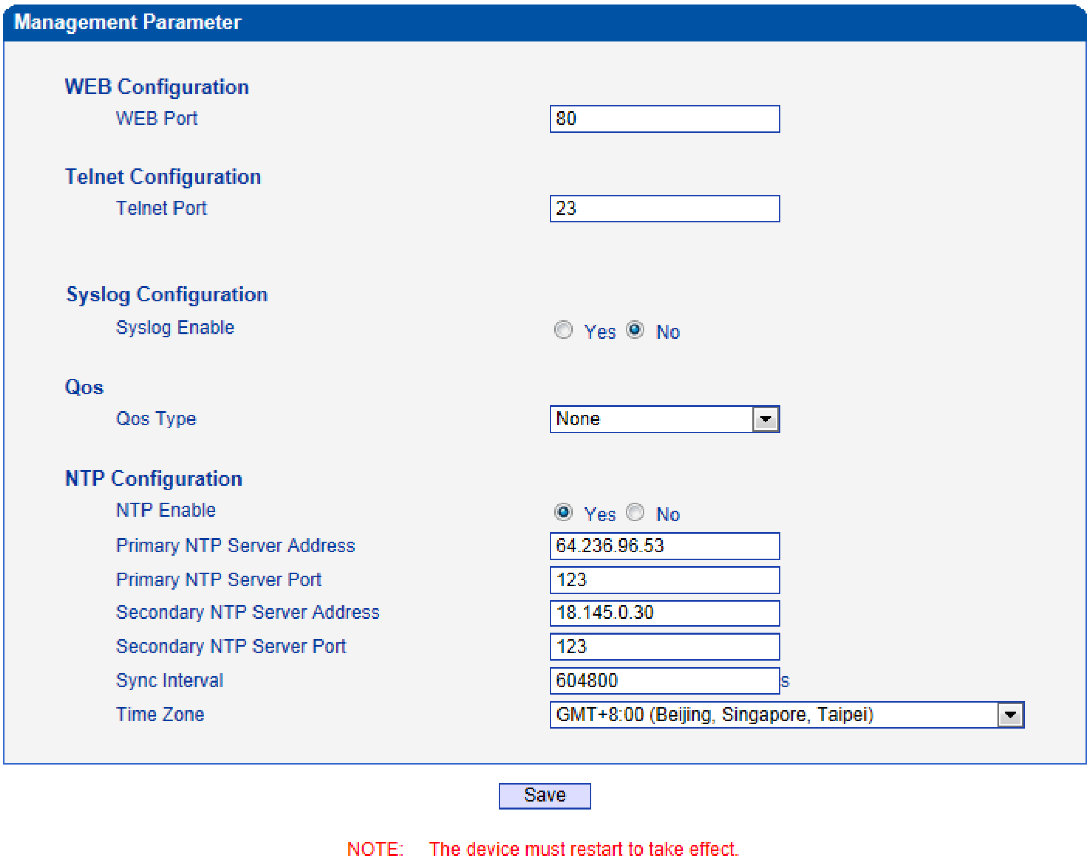



Figure 2-14-1 Management Parameter

Management parameter description

| | |
|---|---|
| WEB Port | Listening port of local WEB service, the default is 80. |
| Telnet Port | Listening port of local Telnet service, the default is 23. |
| Syslog Enable | The default is "No". |
| Server Address | Address for saving system log |
| Syslog Level | None, Debug, Notice, Warning, Error |
| Send CDR | Whether send Call Detail Record |
| Qos Type | There are three options: none, TOS and DS. TOS only supports IPv4. |
| NTP Enable | Simple Network Management Protocol is enabled or not; the default is Yes. |
| Primary NTP server Address | The Primary IP address of SNMP management host computer. The host computer of the IP address will carry out monitoring and management to equipment. |
| Primary NTP server Port | The port that managed device provides trap message (it is generally alarm message) to SNMP management host computer, the default is 123. |
| Secondary NTP server Address | The Secondary IP address of SNMP |
| Sync Interval | Time interval of check |
| Time Zone | The time zone of local |

### 2.14.2 SNMP Parameter

Simple Network Management Protocol (SNMP) is application layer protocol, and used to manage communication line.

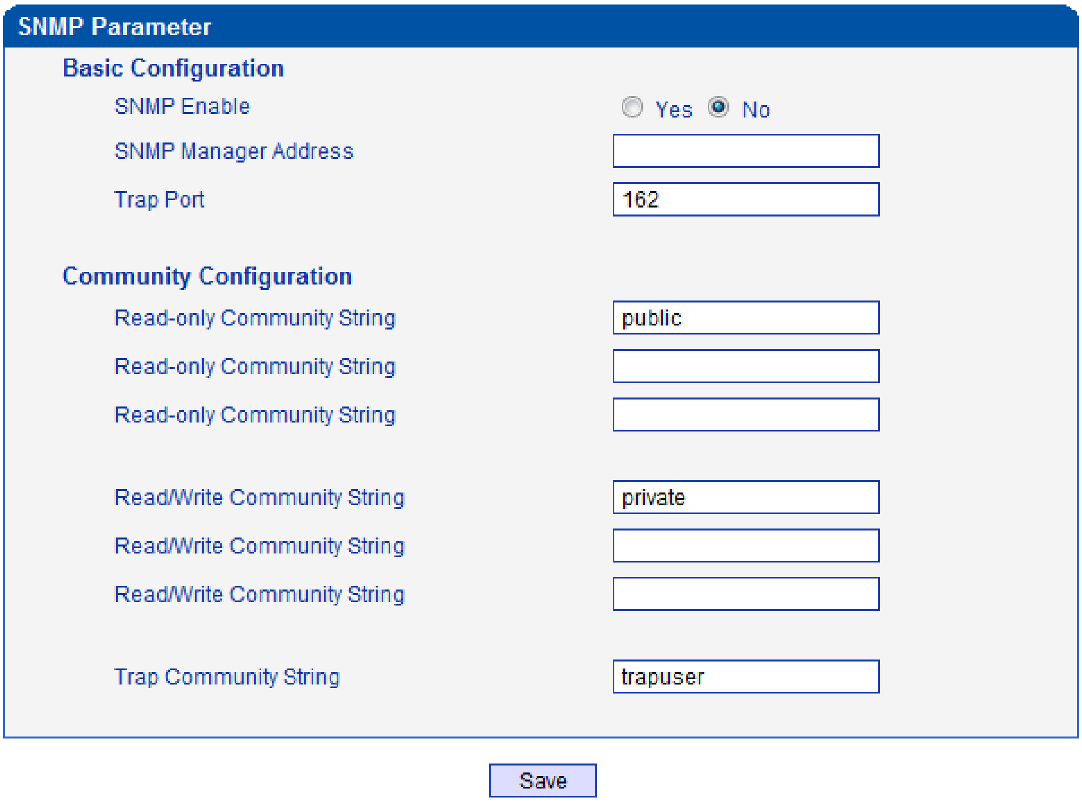



Figure 2-14-3 SNMP Parameter

**SNMP Parameter description**

| | |
|---|---|
| SNMP Enable | Whether enable SNMP function |
| SNMP Manager Address | Network management server IP address |
| 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 configuration, please restart equipment to take effect.

Users can manage and configure gateway on remote NM server through SNMP configuration. But in order to security, recommend this option to open when needed.

### 2.14.3 Data Backup

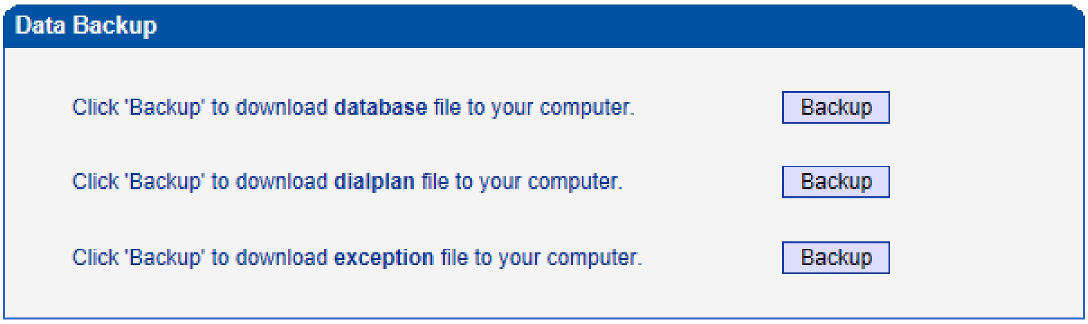



Figure 2-14-4 Data Backup

Database and dial rules will be saved to the local computer system logs through data backup.

### 2.14.4 Data Restore

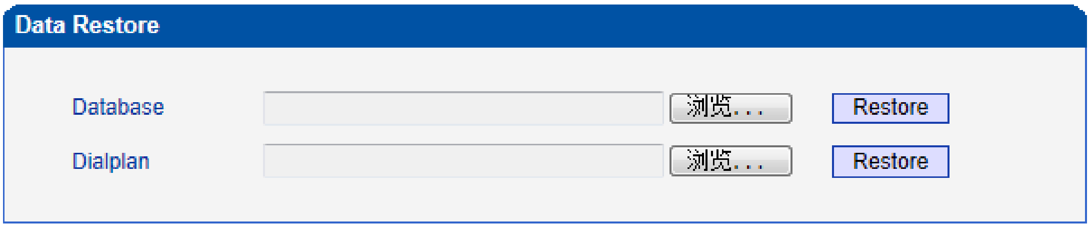



Figure 2-14-5 Data Restore

**Data restore description**

| | |
|---|---|
| Database | Click "Browse" to select the Database file, and then click "Restore". |
| Dial plan | Click "Browse" to select the Dial plan file, and then click "Restore". |

### 2.14.5 Version Information

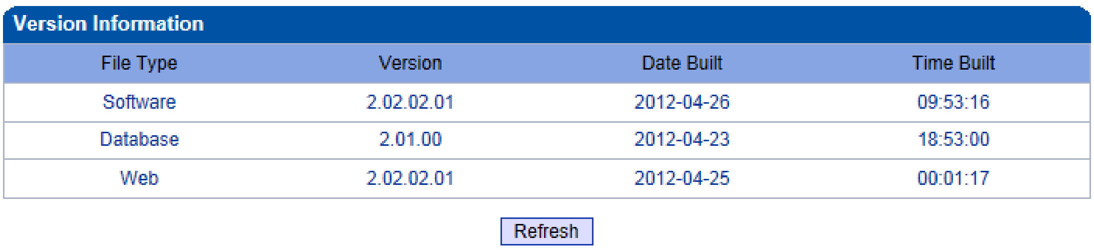



Figure 2-14-6 Version Information

Version information description version and built time of program, database and web file.

### 2.14.6 Firmware Upload

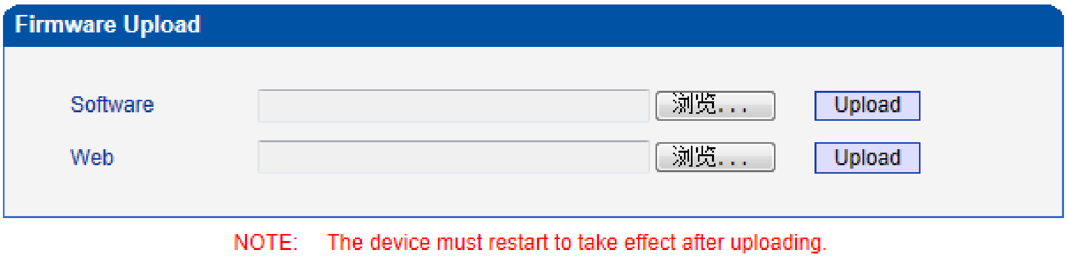



Figure 2-14-7 Firmware Upload

**Firmware upload description**

| | |
|---|---|
| Software | Click "Browse" to select the firmware, and then click "Upload". |
| Web | Click "Browse" to select the Web software, and then click "Upload". |

### 2.14.7 Modify Password

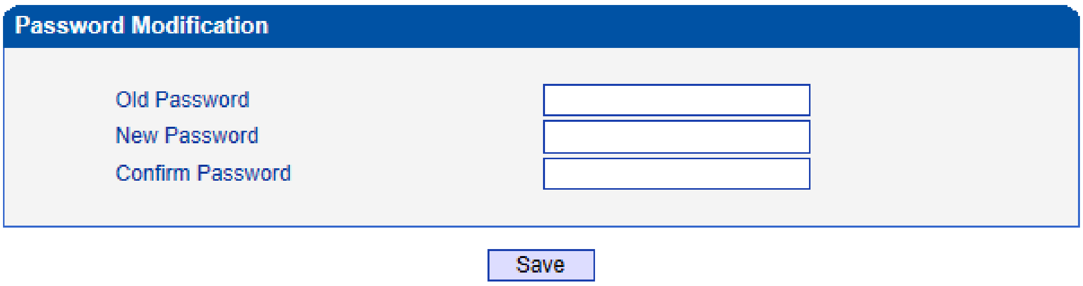



Figure 2-14-8 Modify Password

After entering configuration page, please modify password to ensure the system security.

### 2.14.8 Restart Device


Figure 2-14-9 Restart Device

If user click Restart, a message ("Are you sure?") will be popped up, and then click OK.

# 3. FAQ

## **3.1** How to get the IP address if user modified or forgot the default IP?

There are one way to get the IP address:

1) Connect the CONSOLE with your PC Serial Port. The baud rate is 9600 bps. The user name and password is "admin". When users logged in system, and then run command "show int" for getting the IP.

   Please refer to http://www.dinstar.com/service/faq_145.aspx

## **3.2** If meet other questions, please from Dinstar website and download trouble shootingV4.0.URL is: http://www.dinstar.com/service/Training.aspx

# 4. Glossary

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

IMSI: International Mobile Subscriber Identification Number

IMEI: International Mobile Equipment Identity

DMZ: Demilitarized Zone