



## MTG1000 Trunk Gateway User Manual V2.0



### **Dinstar Technologies Co., Ltd.**

**Address:** Floor 6 Guoxing Building Changxing Road Nanshan District Shenzhen China 518052

**Telephone:** 86-755-26456664

**Fax:** 86-755-26456659

**Email:** sales@dinstar.com, support@dinstar.com

**Website:** www.dinstar.com

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## 1. Product Introduction

### 1.1 Overview

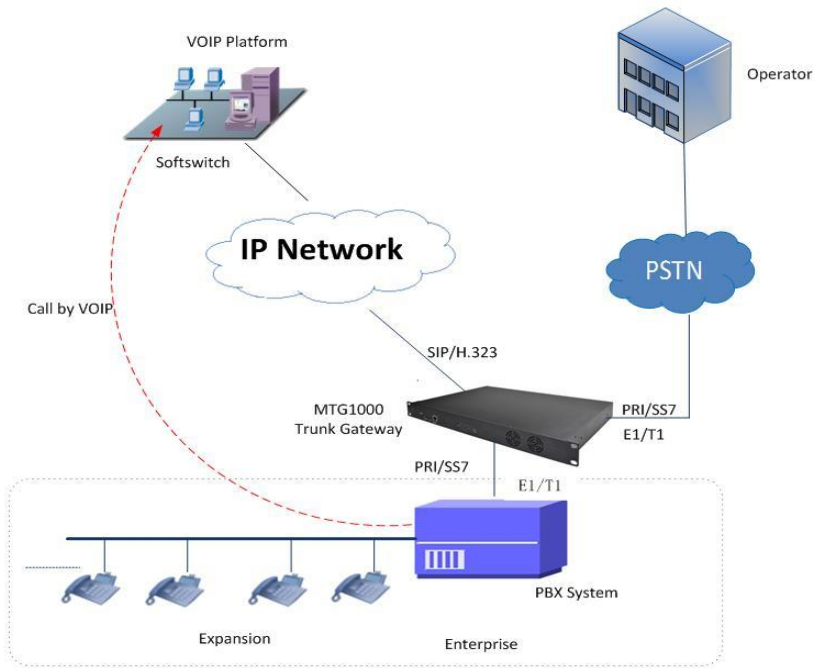
MTG1000 is a trunk gateway aimed at operators and call center, and used to help enterprise to realize the evolution from the traditional PBX to voice IP. On the one hand, it supports PRI/SS7 protocol and adopts standard T1/E1 trunk interface to realize docking with traditional PBX. On the other hand, 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°C (operational), -20~70°C(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.

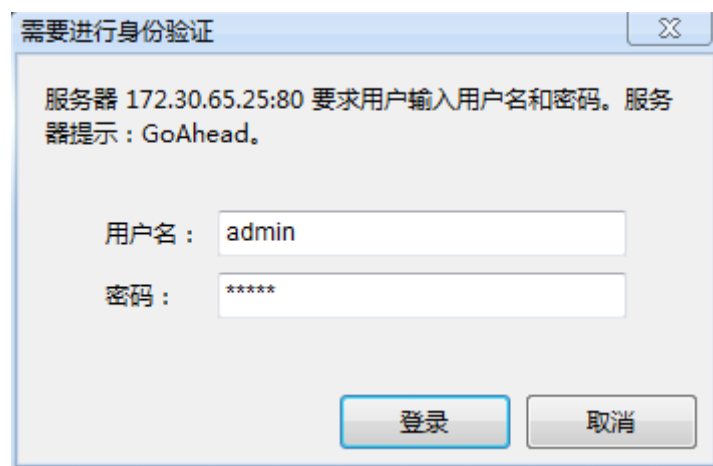


Figure 2-1-1 Login Interfaces

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.

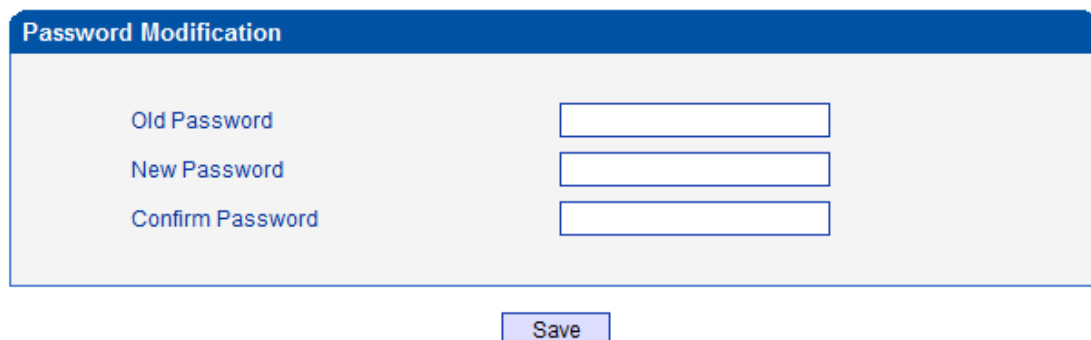


Figure 2-1-2 Modify Password

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

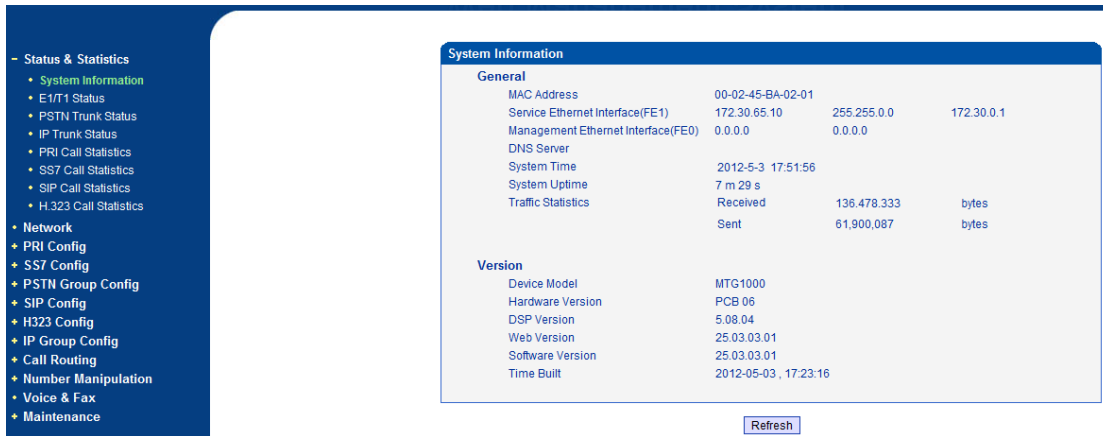


Figure 2-1-3 Description of System Information

## 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.

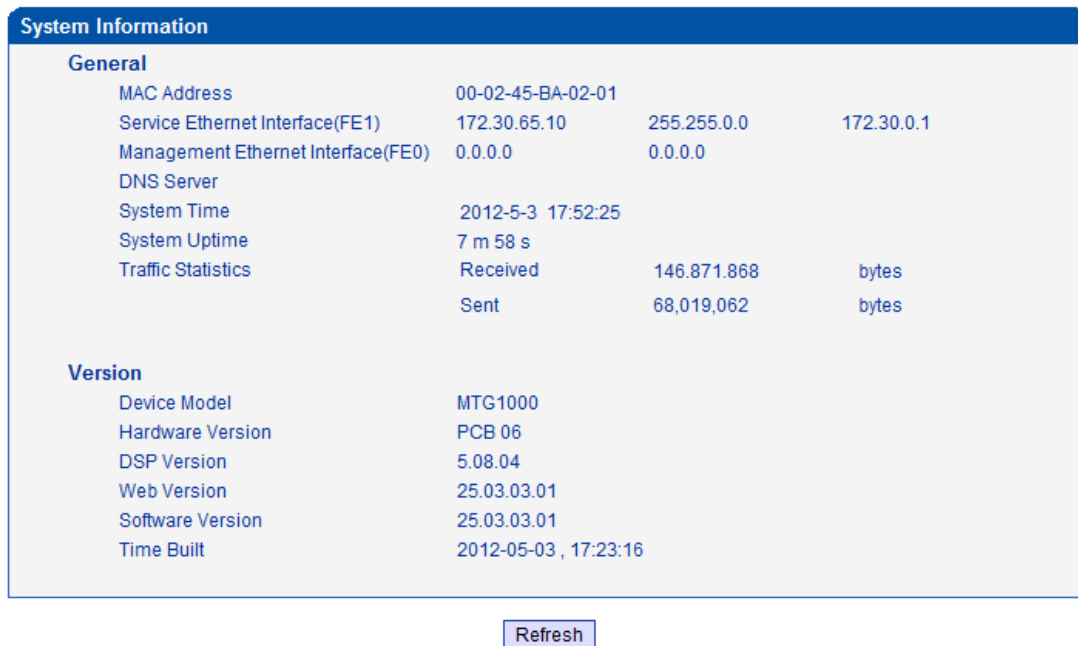


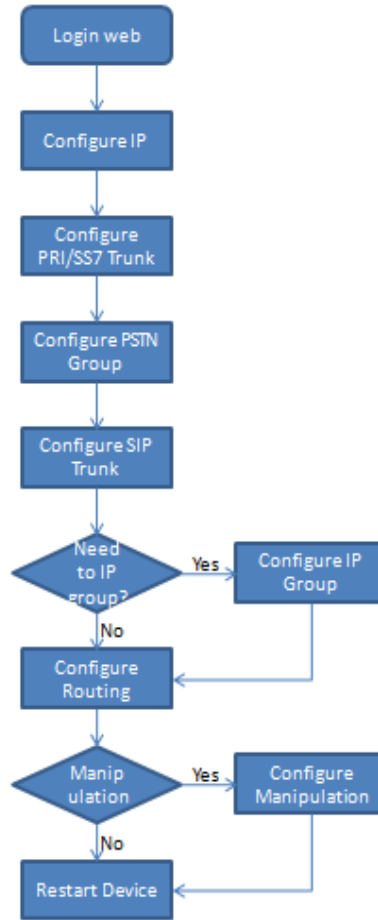
Figure 2-2-1 System Information

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

- **Status & Statistics**
  - **System Information**
  - E1/T1 Status
  - PSTN Trunk Status
  - IP Trunk Status
  - PRI Call Statistics
  - SS7 Call Statistics
  - SIP Call Statistics
  - H.323 Call Statistics
- **Network**
- + PRI Config
- + SS7 Config
- + PSTN Group Config
- + SIP Config
- + H323 Config
- + IP Group Config
- + Call Routing
- + Number Manipulation
- Voice & Fax
- + Maintenance

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

- **Status & Statistics**
  - **System Information**
  - E1/T1 Status
  - PSTN Trunk Status
  - IP Trunk Status
  - PRI Call Statistics
  - SS7 Call Statistics
  - SIP Call Statistics
  - H.323 Call Statistics

### 2.3.1 System Information

System information interface shows the general information and version information.

Figure 2-3-1 System Information

**System Information**

**General**

MAC Address	00-02-45-BA-02-01		
Service Ethernet Interface(FE1)	172.30.65.10	255.255.0.0	172.30.0.1
Management Ethernet Interface(FE0)	0.0.0.0	0.0.0.0	
DNS Server			
System Time	2012-5-3 17:53:16		
System Uptime	8 m 49 s		
Traffic Statistics	Received	160,104,055	bytes
	Sent	73,671,353	bytes

**Version**

Device Model	MTG1000
Hardware Version	PCB 06
DSP Version	5.08.04
Web Version	25.03.03.01
Software Version	25.03.03.01
Time Built	2012-05-03 , 17:23:16

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

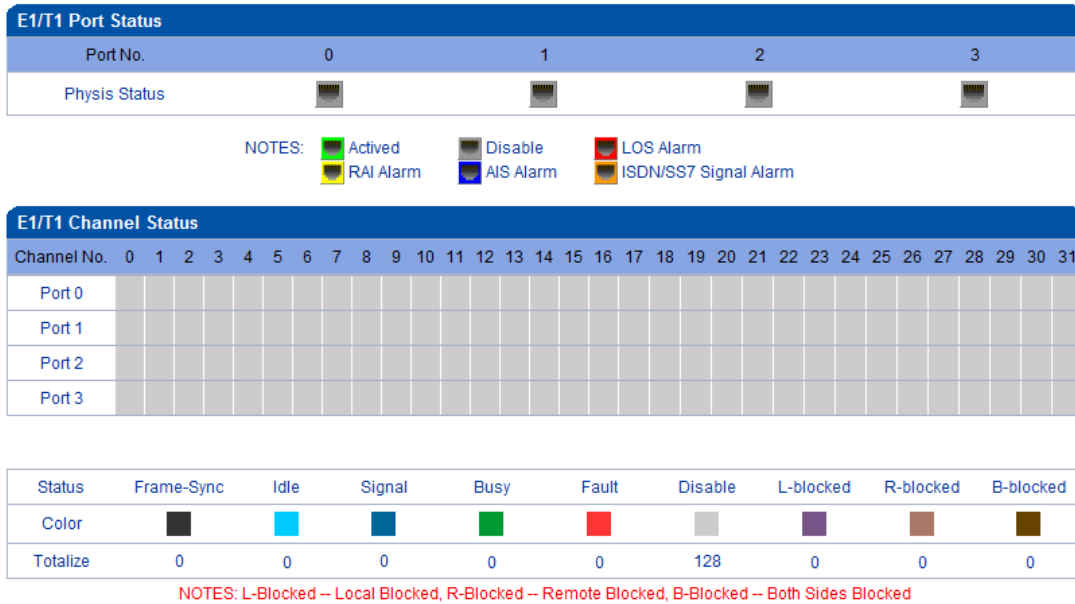


Figure 2-3-2 E1/T1 Status

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: 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: Both Sides blocked, means that the two sides cannot communication

### 2.3.3 PSTN Trunk Status

PRI Link Status			
PRI Trunk No.	Trunk Name	E1/T1 Port No.	Link Status
---	---	---	---

SS7 Link Status			
SS7 Trunk No.	Trunk Name	E1/T1 Port No.	Link 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

SIP Trunk Status					
Trunk No	Trunk Name	Trunk Mode	Username	Incoming Authentication Type	Link Status
0	172.30.66.11	Peer	---	IP Address	Established

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

PRI Trunk Call Statistics				
PRI 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 2-3-5 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

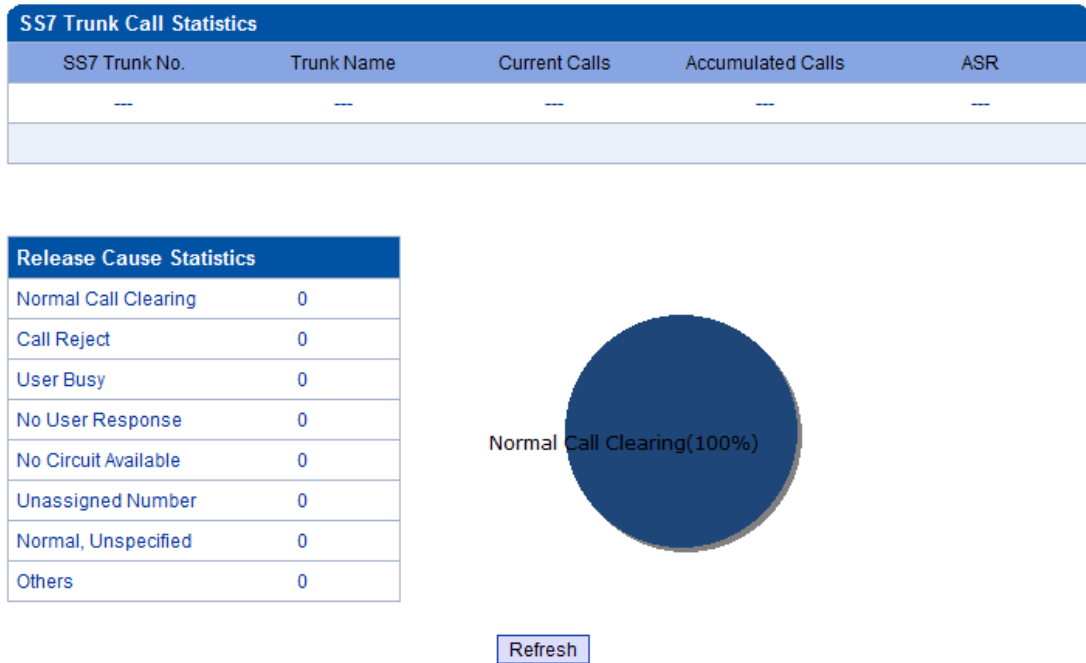


Figure 2-3-6 SS7 Trunk Call Statistics

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



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

H.323 Trunk Call Statistics		
Trunk No.	Trunk Name	Current Calls
---	---	---

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

Network Configuration

**Service Ethernet Interface(FE1)**

IP Address

Subnet Mask

Default Gateway

**Management Ethernet Interface(FE0)**

IP Address

Subnet Mask

**DNS Server**

Primary DNS Server

Secondary DNS Server

**NOTE:** The device must restart to take effect.

Figure 2-4-1 Network Configuration

### Network Configuration

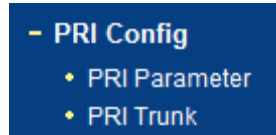
Service Ethernet Interface (FE0)	IP address	Set FE0 port static IP address.
	Subnet Mask	Fill in subnet mask
	Default Gateway	Fill in default gateway
Management Ethernet Interface (FE1)	IP address	Set FE1 port static IP address
	Subnet Mask	Fill in subnet mask
DNS Server	Primary DNS	Fill in DNS Server 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**

Calling Party Numbering Plan	ISDN/Telephony numbering plan ▼
Calling Party Number Type	Unknown ▼
Screening Indicator for Displaying Caller Number	User provide,no shield ▼
Screening Indicator for No Displaying Caller Number	User provide,no shield ▼
Called Party Numbering Plan	ISDN/Telephony numbering plan ▼
Called Party Number Type	Unknown ▼
Information Transfer Capability	Speech ▼

Reset to default configuration

#### 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

PRI Trunk							
Trunk No.	Trunk Name	Channel ID	D-Channel	E1/T1 Port No.	Protocol	Switch Side	Alerting Indication
---	---	---	---	---	---	---	---

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

Figure 2-5-4 Add PRI Trunk

PRI Trunk Add	
Trunk No.	<input type="text" value="3"/>
Trunk Name	<input type="text"/>
Channel ID	<input type="text"/>
D-Channel	<input type="text" value="Enable"/>
E1/T1 Port No.	<input type="text" value="3"/>
Protocol	<input type="text" value="ISDN"/>
Switch Side	<input type="text" value="User Side"/>
Alerting Indication	<input type="text" value="ALERTING"/>

#### PRI trunk description

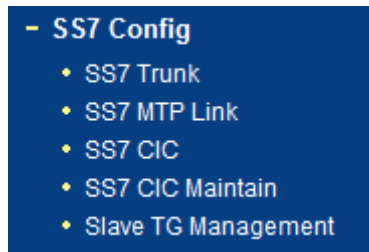
Trunk No	The number of PRI trunk; when user add PRI trunk, 0~7 number will appear in the pull-down menu to be selected (the number here depends on E1/T1 physical port number actually existed in equipment). After trunk number is established, filling in corresponding port number in "E1/T1 Port No.", so 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

SS7 Trunk								
Trunk No.	Trunk Name	Protocol	Protocol Type	SPC Format	OPC	DPC	Network Indicator	Sending SLTM
---	---	---	---	---	---	---	---	---

Figure 2-6-3 SS7 Trunk Add

**SS7 Trunk Add**

Select Trunk No.

Trunk Name

Protocol

Protocol Type

SPC Format

OPC

DPC

Network Indicator

Sending SLTM

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

SS7 MTP Link					
Trunk No.	Link No.	Signaling Link Code	E1/T1 Port No.	Channel No.	
--	--	--	--	--	

Figure 2-6-5 SS7 MTP Link Add

**SS7 MTP Link Add**

Trunk No.	<input style="width: 90%;" type="text"/>
Link No.	<input style="width: 90%;" type="text" value="0"/>
Signaling Link Code	<input style="width: 90%;" type="text"/>
E1/T1 Port No.	<input style="width: 90%;" type="text" value="0"/>
Channel No.	<input style="width: 90%;" type="text" value="16"/>

**NOTES:** Each SS7 trunk could add maximum 2 items with different 'Link No.'.



**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

Trunk No.	E1/T1 Port No.	Start Channel	Start CIC No.	Count
--	--	--	--	--

Figure 2-6-6 SS7 Circuit description

**SS7 Circuit Add**

Trunk No.

E1/T1 port No.

Start Channel

Start CIC No.

Count

- NOTES:**
1. When option 'ITU' or 'ITU-CHINA' has been selected in 'Protocol' of sub-menu SS7 Trunk, the 'Start CIC No.' must be less than 4096.
  2. When option 'ANSI' has been selected in 'Protocol' of sub-menu SS7 Trunk, the 'Start CIC No.' must be less than 16384.

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.

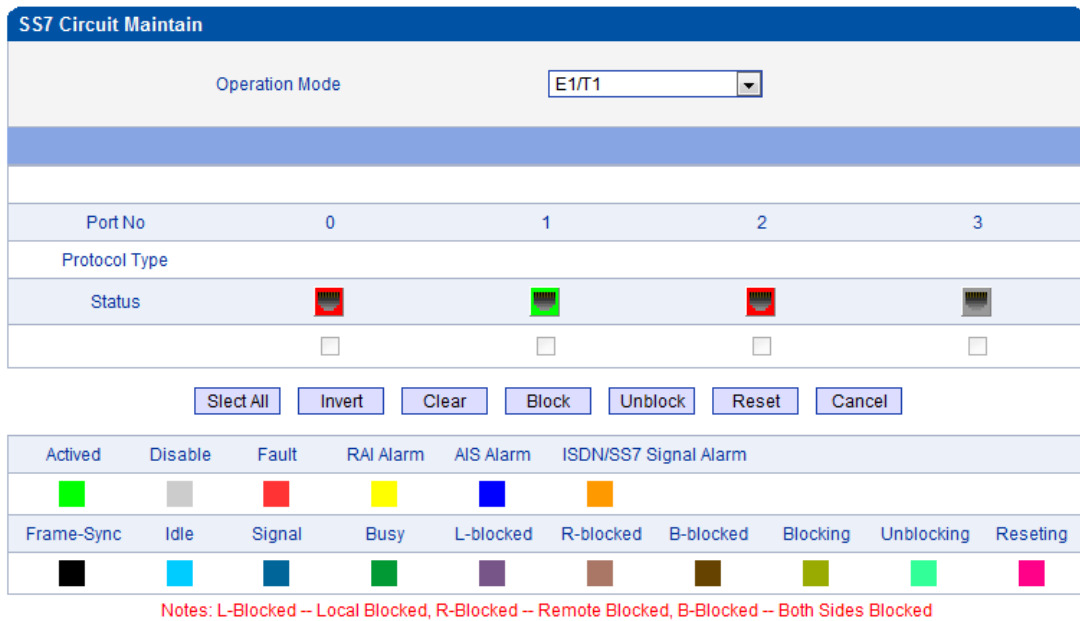


Figure 2-6-7 SS7 Circuit Maintain-E1/T1

#### 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.

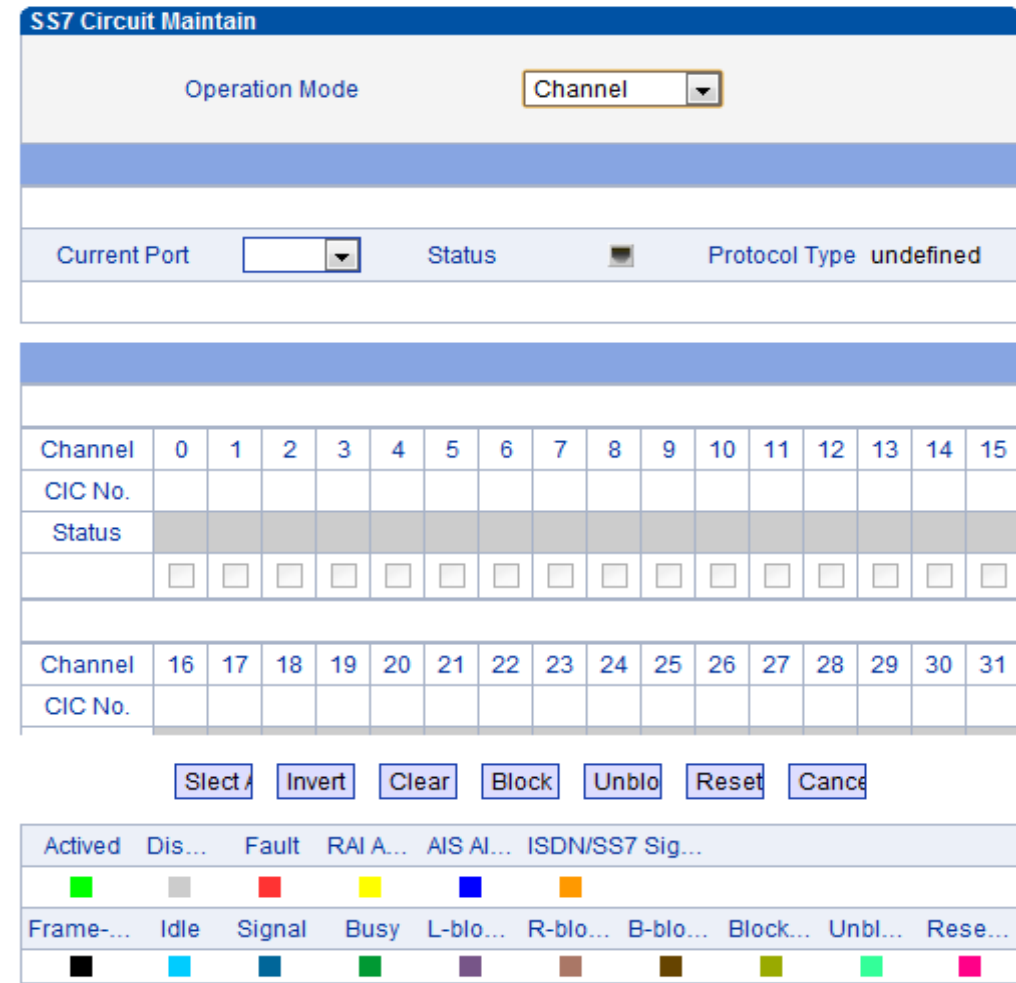


Figure 2-6-8 SS7 Circuit Maintain-Channel

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

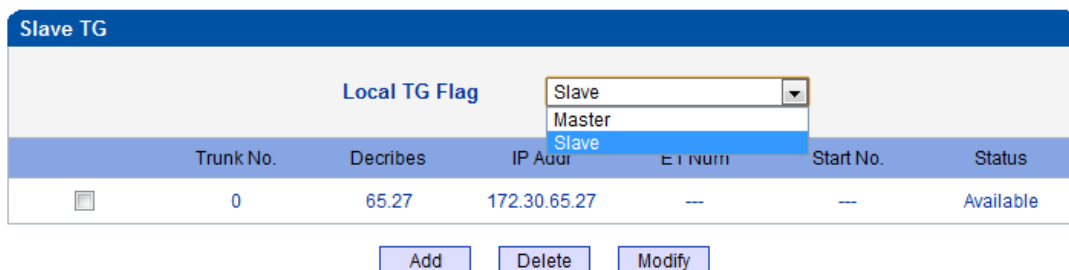


Figure 2-6-9 Slave TG Management

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/T1 can 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.

**E1/T1 Parameter**

**E1/T1 Clock Source** Remote

	Port No.	Work Mode	PCM Mode	Frame Mode	Line Code	Line Built Out
<input type="checkbox"/>	0	E1	A LAW	DF	HDB3	Short Haul,(-10DB)
<input type="checkbox"/>	1	E1	A LAW	CRC-4	HDB3	Short Haul,(-10DB)
<input type="checkbox"/>	2	E1	A LAW	CRC-4	HDB3	Short Haul,(-10DB)
<input type="checkbox"/>	3	E1	A LAW	CRC-4	HDB3	Short Haul,(-10DB)

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

Coder Group

Coder Group ID 0(default setting) ▼

	Coder	Payload Type Value	Packetization Time (ms)	Rate (kbps)	Silence Suppression
1st	G711A ▼	8	20 ▼	64	Disable ▼
2nd	G711U ▼	0	20 ▼	64	Disable ▼
3rd	G729 ▼	18	20 ▼	8	Disable ▼
4th	G723 ▼	4	30 ▼	6.3	Disable ▼
5th	▼		▼		▼
6th	▼		▼		▼

Save

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

Dial Plan

Dial Plan ID 0 ▼

Index	Prefix	Min Length	Max Length
0	.	0	30

Total: 1 Page 1 ▼

Add
Delete
Modify

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:

- NOTES:**
1. '.' in 'Prefix' field means wildcard string.
  2. 'Max Length' and 'Min Length' do not include the 'prefix'.
  3. The value of 'Max Length' plusing the length of 'Prefix' should less than 30.

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

Dial Timeout					
	Dial Timeout ID	Description	Max Time for Collecting Prefix(s)	Time to Reach Min Length (s)	Time to Reach Max Length (s)
<input type="checkbox"/>	0	Default	20	10	10

Total: 1 Page 1

Figure 2-7-5 Dial Timeout

**Dial Timeout Add**

Dial Timeout ID:

Description:

Max Time for Collecting Prefix:  s

Time to Reach Min Length(after Prefix):  s

Time to Reach Max Length(after Min Length):  s

**NOTE:** If Max length equals to Min length in Dial Plan, Time to Reach Max Length can be any value.

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

PSTN Profile												
	PSTN Profile ID	Description	Coder Group ID	RFC2833 Payload	DTMF Tx PR 1	DTMF Tx PR 2	DTMF Tx PR 3	Overlap Receiving	Dial Plan ID	Dial Timeout ID	Remove CLI	Play Busy Tone to PSTN
<input type="checkbox"/>	0	Default	0	101	RFC2..	SIP IN...	Inband	Enable	0	0 <Default>	Not remove	No

Total: 1 Page 1

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.
1 <sup>st</sup> /2 <sup>nd</sup> /3 <sup>rd</sup> 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.

Group ID	Name	Channel Selection
0	r2-0	Cyclic Ascending
1	r2-12	Cyclic Ascending

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

	Group ID	Start E1/T1	End E1/T1	Start Channel	End Channel	PSTN Profile ID
<input type="checkbox"/>	0 <r2-0>	0	0	1	31	0 <Default>
<input type="checkbox"/>	0 <r2-0>	1	2	--	--	0 <Default>

Total: 2 Page 1

Add Delete Modify

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

Trunk No.	Trunk Name	Remote Address	Remote Port	Local Domain	Support SIP-T	Get Callee from	Register to Remote	Outgoing Call Mode	Incoming Authentication Type	Detect Trunk Status	Enable SIP Trunk
0	172.30.66.16	172.30.66.16	5060	Disable	Disable	Request-line	No	Peer	IP Address	Yes	Yes

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**

SIP Account					
SIP Account ID	Description	Binding PSTN Group	SIP Trunk No.	Username	Expire Time
---	---	---	---	---	---

Total: 0

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

H.323 Parameter

Call Mode	<input type="text" value="FastStart"/>
Call Signal Port	<input type="text" value="1720"/>
Enable H.245 Tunneling	<input type="text" value="Yes"/>
DTMF Transfer Mode	<input type="text" value="H.245 Alphabet"/>
Start H.245 on Fast Call	<input type="text" value="Enable"/>
Start H.245 on	<input type="text" value="CONNET"/>
Respond to FastStart on	<input type="text" value="PROCEEDING"/>
Start H.245 Negotiation Actively	<input type="text" value="Enable"/>
<b>Reset to default configuration</b>	<input type="button" value="Reset"/>

**NOTE:** Any re-configuration might cause system works improperly. Do it carefully!

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

H.323 Trunk				
Trunk No.	Trunk Name	Remote IP	Remote Port	Enable H.323 Trunk
--	--	--	--	--

Total: 0

Figure 2-9-2 H.323 Trunk

H.323 Trunk Add	
Trunk No.	<input type="text" value="0"/>
Trunk Name	<input type="text"/>
Remote IP	<input type="text"/>
Remote Port	<input type="text" value="1720"/>
IP to PSTN Calls Restriction	<input type="text" value="No"/>
PSTN to IP Calls Restriction	<input type="text" value="No"/>
IP to PSTN Time Restriction	<input type="text" value="Disable"/>
Enable H.323 Trunk	<input type="text" value="Yes"/>

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

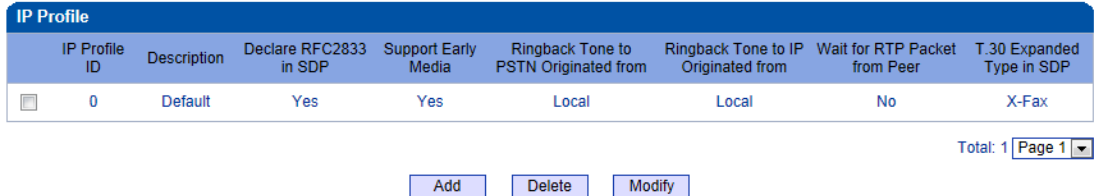


Figure 2-10-1 IP Profile

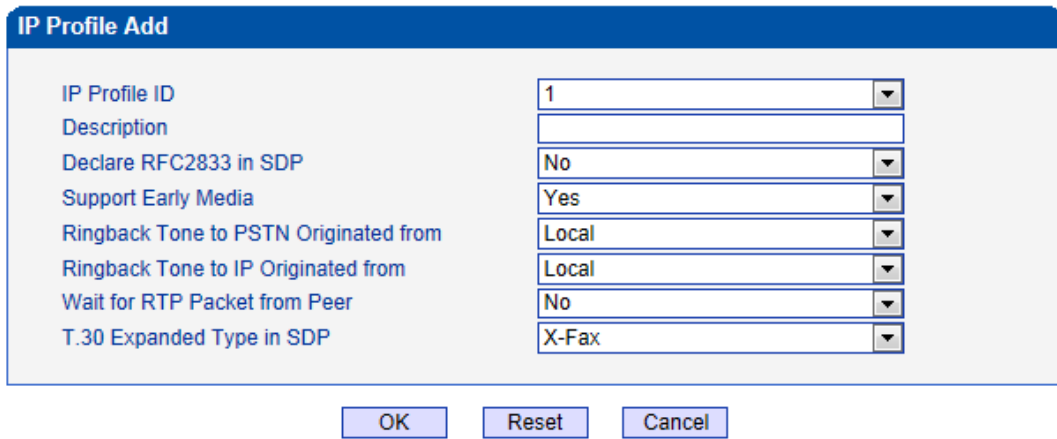


Figure 2-10-2 IP Profile Add

#### 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

IP Group			
	Group ID	Name	IP Trunk Selection
<input type="checkbox"/>	0	66.16	Cyclic Ascending

Total: 1 Page 1

Figure 2-10-3 IP Group

**IP Group Add**

IP Group ID:

Name:

IP Trunk Selection: Cyclic Ascending

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

IP Trunk Group					
	Group ID	Index	Trunk Type	Trunk No.	IP Profile ID
<input type="checkbox"/>	0 <66.16>	0	SIP	0 <172.30.66.16>	0 <Default>

Total: 1 Page 1

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

**Routing Parameter**

**Incoming Calls from IP**

Routing Priority:

Routing & Manipulation:

**Incoming Calls from PSTN**

Routing Priority:

Routing & Manipulation:

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

PSTN->IP Routing									
	Index	Description	Trunk No.	PSTN Group	Callee Prefix	Caller Prefix	Trunk Type	Trunk No.	Destination IP Group
<input type="checkbox"/>	255	any	---	Any	.	.	Any	---	0 <66.16>

Total: 1

Figure 2-11-2 PSTN->IP Routing

**Route PSTN->IP Add**

Index	254
Description	
Source Type	Group
PSTN Group	Any
Callee Prefix	
Caller Prefix	
Destination Type	Group
Destination IP Group	0 <66.16>

NOTE: '.' in 'Callee Prefix' or 'Caller Prefix' field means wildcard string.

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

**PSTN->PSTN Routing**

Index	Description	Trunk No.	PSTN Group	Callee Prefix	Caller Prefix	Dst Trunk No.	Dst PSTN Group
---	---	---	---	---	---	---	---

Total: 0

Figure 2-11-5 PSTN->PSTN Add

Route PSTN->PSTN Add

Index	<input style="width: 90%;" type="text" value="255"/>
Description	<input style="width: 90%;" type="text"/>
Source Type	<input style="width: 90%;" type="text" value="Group"/>
PSTN Group	<input style="width: 90%;" type="text" value="Any"/>
Callee Prefix	<input style="width: 90%;" type="text"/>
Caller Prefix	<input style="width: 90%;" type="text"/>
Destination Type	<input style="width: 90%;" type="text" value="Group"/>
Destination PSTN Group	<input style="width: 90%;" type="text" value="0 &lt;r2-0&gt;"/>

NOTE: '.' in 'Callee Prefix' or 'Caller Prefix' field means wildcard string.

“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

IP->PSTN Routing

	Index	Description	Trunk Type	Trunk No.	IP Group	Callee Prefix	Caller Prefix	PSTN Trunk	Dst PSTN Group
<input type="checkbox"/>	255	all	Any	Any	0 <66.1...	.	.	---	0 <r2-0>

Total: 1

Figure 2-11-7 IP->PSTN Routing

IP->PSTN Routing Add

Index	<input type="text" value="254"/>
Description	<input type="text"/>
Source Type	<input type="text" value="Group"/>
Trunk Type	<input type="text" value="Any"/>
IP Group	<input type="text" value="0 &lt;66.16&gt;"/>
Callee Prefix	<input type="text"/>
Caller Prefix	<input type="text"/>
Destination Type	<input type="text" value="Group"/>
Destination PSTN Group	<input type="text" value="0 &lt;r2-0&gt;"/>

NOTE: '!' in 'Callee Prefix' or 'Caller Prefix' field means wildcard string.

“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

IP->IP Routing

Index	Description	Trunk Type	Trunk No.	IP Group	Callee Prefix	Caller Prefix	Trunk Type	Trunk No.	Dst IP Group
---	---	---	---	---	---	---	---	---	---

Total: 0

Figure 2-11-9 IP->IP Add

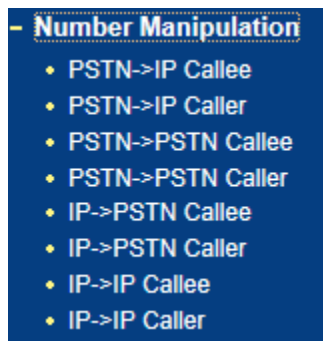
**NOTE:** '!' in 'Callee Prefix' or 'Caller Prefix' field means wildcard string.

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

PSTN->IP 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
---	---	---	---	---	---	---	---	---	---

Total: 0

Figure 2-12-2 PSTN->IP Callee

**PSTN->IP Callee Add**

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:

- NOTES:**
1. Fields with '\*' are MUST.
  2. '.' in 'Callee Prefix' or 'Caller Prefix' field means wildcard string.

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

Figure 2-12-4 PSTN->IP Caller

**PSTN->IP Caller Add**

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:

- NOTES: 1. Fields with '\*' are MUST.  
 2. '.' in 'Callee Prefix' or 'Caller Prefix' field means wildcard string.

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

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

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.

IP->PSTN Callee											
Index	Description	IP 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

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.

IP->PSTN Caller												
Index	Description	IP 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

Figure 2-12-9 IP->PSTN Caller

IP->IP Callee										
Index	Description	IP 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



Figure 2-12-10 IP->IP Callee

IP->IP Caller									
Index	Description	IP 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

Figure 2-12-11 IP->IP Caller

### 2.13 Voice & Fax

#### Voice & Fax Configuration

##### Voice Parameter

Disconnect call when no RTP packet  Yes  No

Period without RTP packet  s

Gain from PSTN  ▼

Gain to PSTN  ▼

##### Timeout of No Answer

Call from PSTN  s

Call from IP  s

##### Fax Parameter

Fax Mode  ▼

Fax Tx Gain  ▼

Fax Rx Gain  ▼

Packet time  ms

Redundant frame in packet  ▼

##### Data & Fax Control

Data  ▼

Fax  ▼

##### DTMF Parameter

Continuous time  ms

Signal interval  ms

Threshold for detection  ▼

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

Management Parameter

**WEB Configuration**  
 WEB Port

**Telnet Configuration**  
 Telnet Port

**Syslog Configuration**  
 Syslog Enable  Yes  No

**Qos**  
 Qos Type

**NTP Configuration**  
 NTP Enable  Yes  No  
 Primary NTP Server Address   
 Primary NTP Server Port   
 Secondary NTP Server Address   
 Secondary NTP Server Port   
 Sync Interval  s  
 Time Zone

**NOTE:** The device must restart to take effect.

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.

SNMP Parameter

**Basic Configuration**

SNMP Enable  Yes  No

SNMP Manager Address

Trap Port

**Community Configuration**

Read-only Community String

Read-only Community String

Read-only Community String

Read/Write Community String

Read/Write Community String

Read/Write Community String

Trap Community String

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

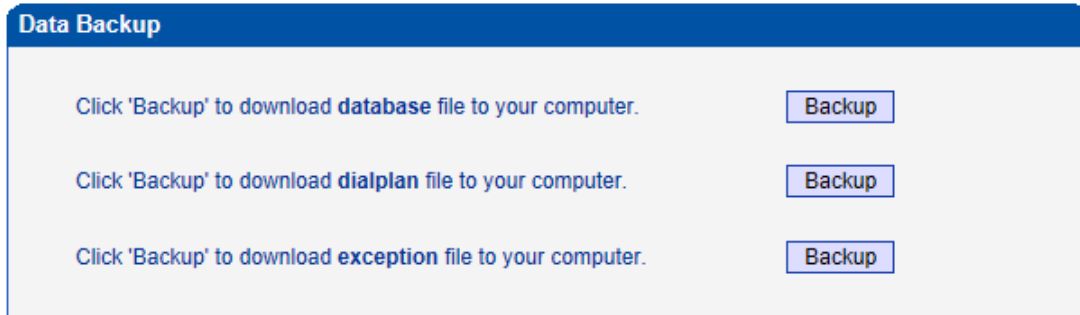


Figure 2-14-4 Data Backup

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

### 2.14.4 Data Restore

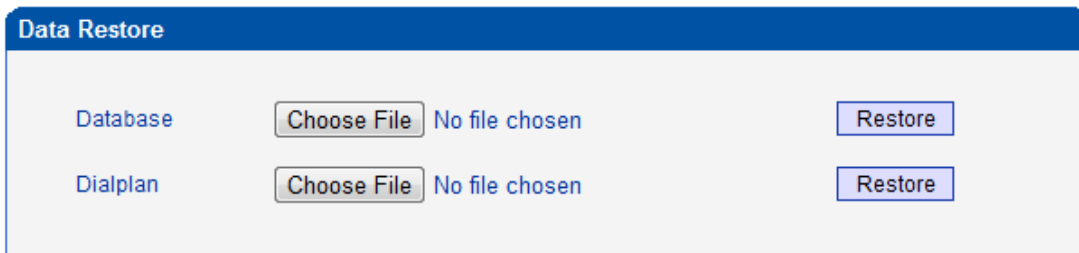


Figure 2-14-5 Data Restore

#### 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

Version Information			
File Type	Version	Date Built	Time Built
Software	2.02.02.01	2012-04-26	09:53:16
Database	2.01.00	2012-04-23	18:53:00
Web	2.02.02.01	2012-04-25	00:01:17

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](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