MTG1000(B) Trunk Gateway User Manual V2.0

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Revision Records

File Name	MTG1000(B) Trunk Gateway User Manual
Document Version	2.0
Firmware Version	2.03.04.01/2.04.04.01
Date	03/10/2013
Revised by	Technical Support Department

Content

1. Product Introduction

1.1 Overview

E1/T1 Trunk Gateway (Hereafter named 'TG' or '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. The gateway 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.

E1/T1 Gateway has good call processing ability, and provides 1/2/4/8 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.

1.2 Equipment Structure

This section mainly introduce hardware structure.

1.2.1 Rear View

Figure 1-2-1 Rear View (8 Ports gateway)

Table 1-2-1 Rear View Description

Interface	Description
PWR	Connecting the power adapter, 110~240VAC, 50~60HZ, 1.2A,
Port0-Port7	E1/T1 ports. The port number are different on different models.
FE0	**-1/2E1/T1 Gateway** **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 **-4/8 Port Gateway** **Management interface**, standard 10/100BASE-TX Ethernet interfaces, Default IP address is 192.168.11.1, default subnet mask is 255.255.255.0
FE1	**-1/2E1/T1 Gateway** **Management interface**, standard 10/100BASE-TX Ethernet interfaces. Default IP address is 192.168.1.111, default subnet mask is 255.255.255.0 **-4/8 Port Gateway** **Service Ethernet Interface**, standard 10/100BASE-TX Ethernet interfaces, 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 Front View

Table 1-2-2 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	1/2E1/T1 Gateway: RS232, 115200bps 4/8 E1/T1 Gateway: RS232, 9600bps		

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

trunk gateway adopts standard RJ-48C interface and iimpedance 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. Web Configuration Guide

2.1 Login

Firstly, connect network from pc to management port of gateway directly and input default IP address in the browser. 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 was set to 172.16.65.27. Enter the IP address of device in the browser address bar. Customer will see the following page.

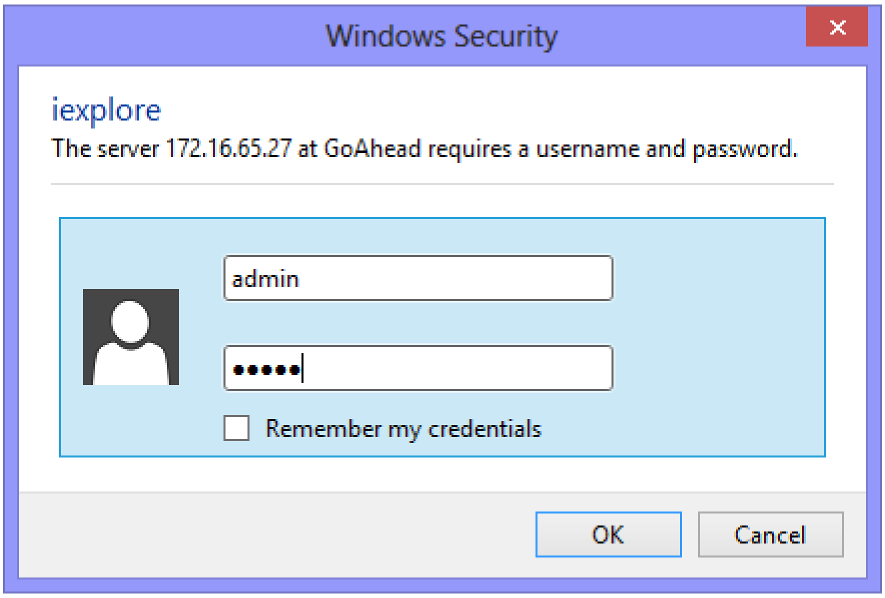


Figure 2-1-1 Login Interfaces

The default user name and password is "admin". To guarantee the system safety, when login for the first time, please modify the default username and password **Maintenance->Password Modification.**

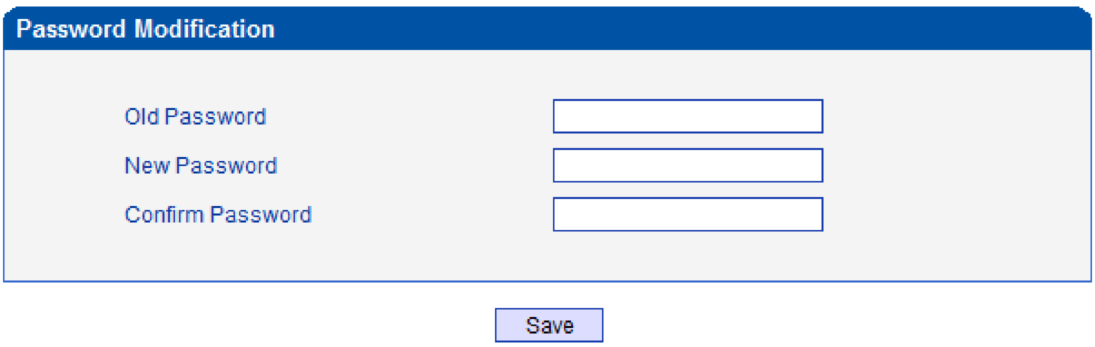


Figure 2-1-2 Modify Password

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.

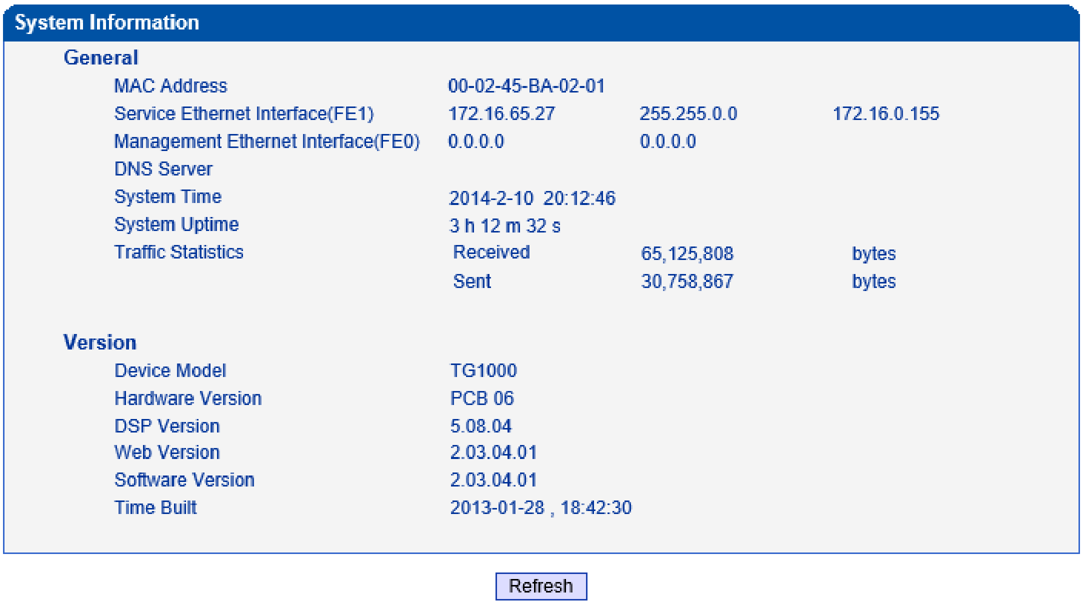
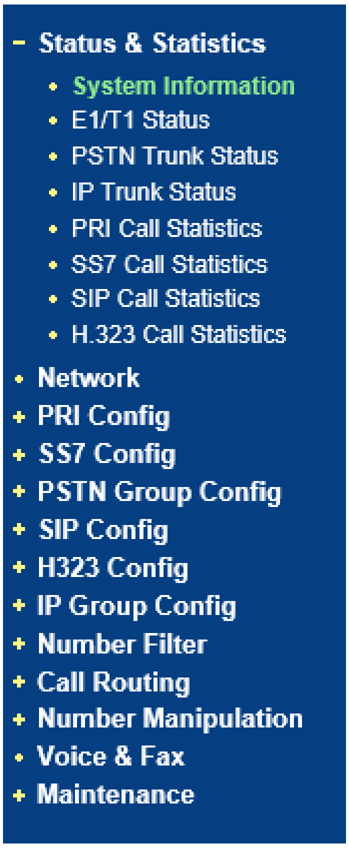


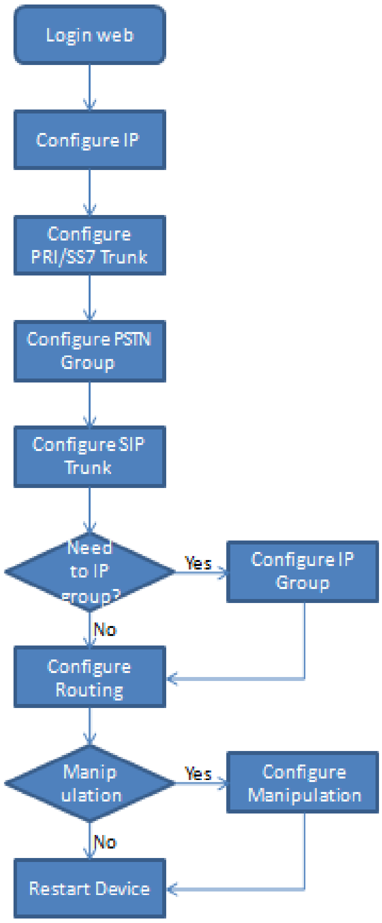
Figure 2-2-1 System Information

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



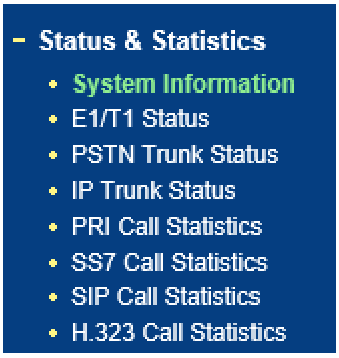
Configuration flow chart below:



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

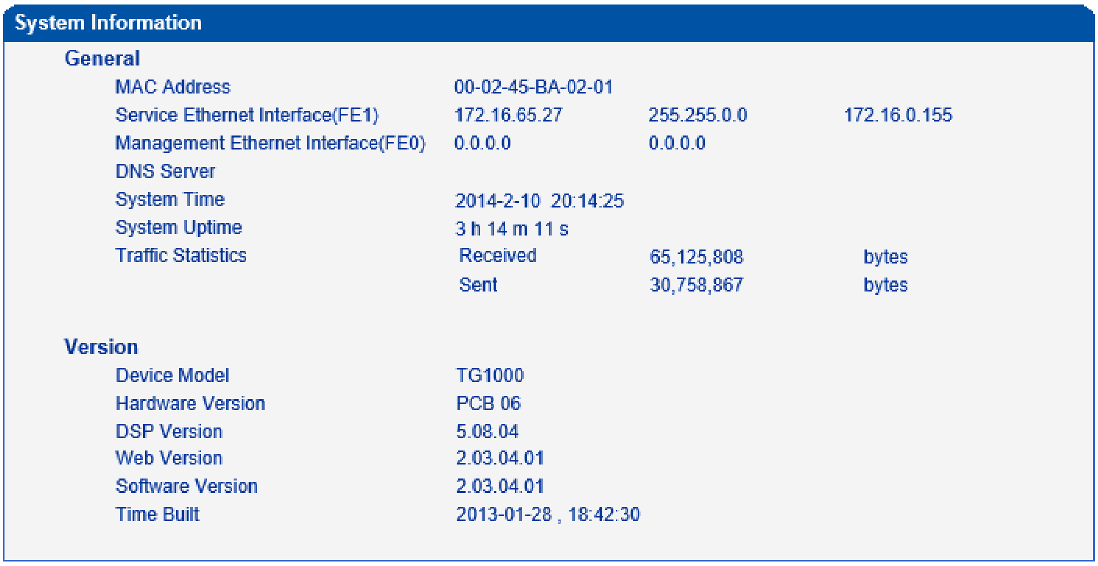


Table 2-3-1 System Information

MAC address	Hardware address of Service interface
Service Ethernet Interface	Transmit all IP based services like signaling, web access, voice stream etc.
Management Ethernet Interface	Mainly use to local access only. A backup access once service interface down then the user could be able to get web access from management interface
DNS	DNS server IP address
System Time	Current time of the system
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	Model name
Hardware Version	Hardware version of device
DSP Version	DSP firmware version.
Web Version	Version of current WEB interface, the web version must be matched with software version
Software Version	Software version running on the gateway
Built Time	The build time of current software version

2.3.2 E1/T1 Status

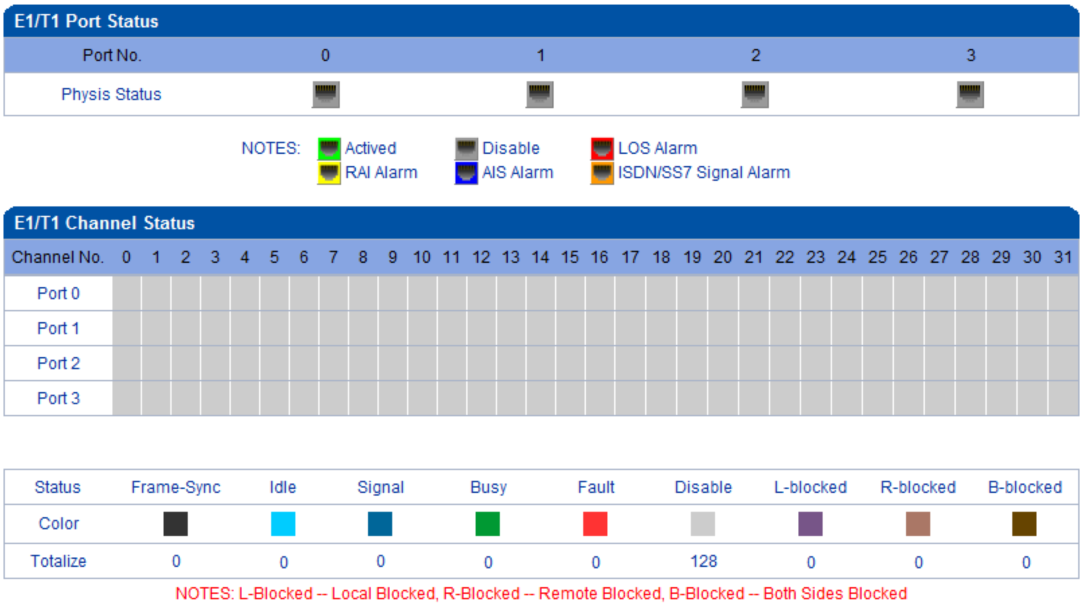


Figure 2-3-2 E1/T1 Status

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
---	---	---	---

Refresh

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

Refresh

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

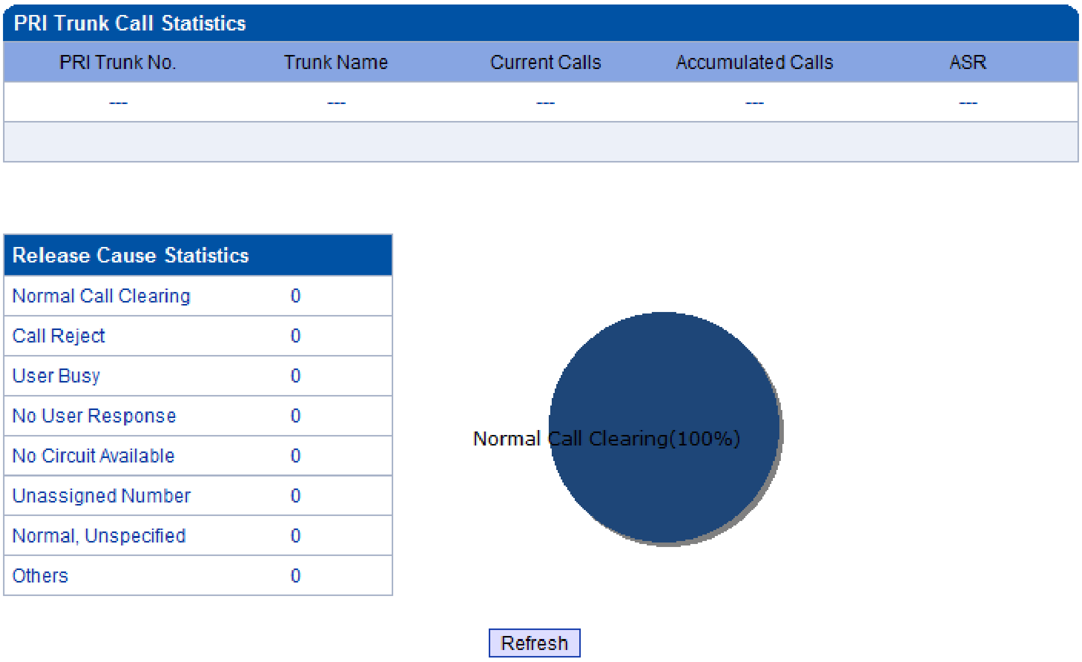


Figure 2-3-5 PRI Call Statistics description

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

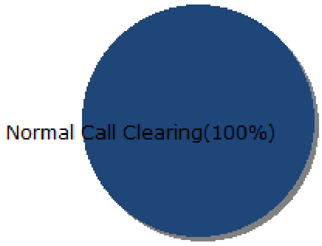


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Figure 2-3-6 SS7 Trunk Call Statistics

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

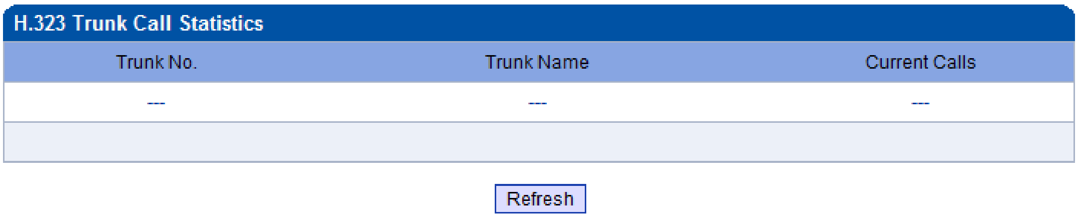


Figure 2-3-8 H.323 Trunk Call Statistics

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

2.4 Network

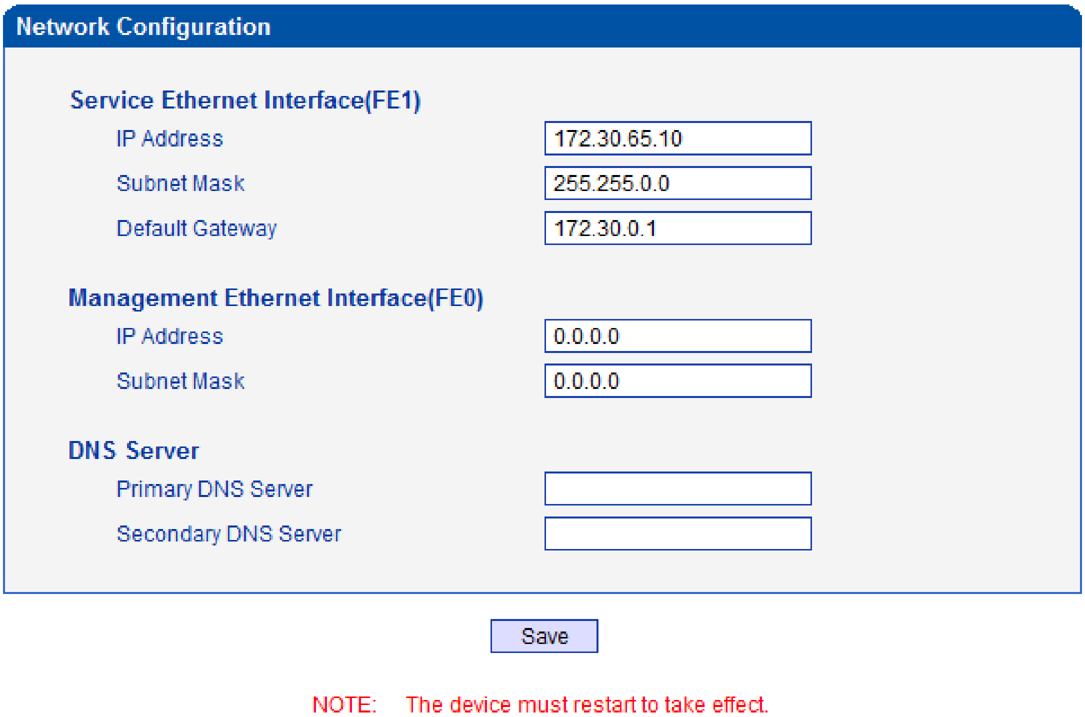


Figure 2-4-1 Network Configuration

Network Configuration

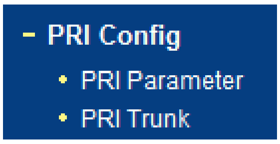
Service Ethernet Interface（FE1	IP address	Set FE1port static IP address.
	Subnet Mask	Fill in subnet mask
	Default Gateway	Fill in default gateway
Management Ethernet Interface（FE0	IP address	Set FE0port static IP address
	Subnet Mask	Fill in subnet mask
DNS Server	Primary DNS	Fill in DNS Server IP address.
	Secondary DNS	The secondary DNS server is option.

Note：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

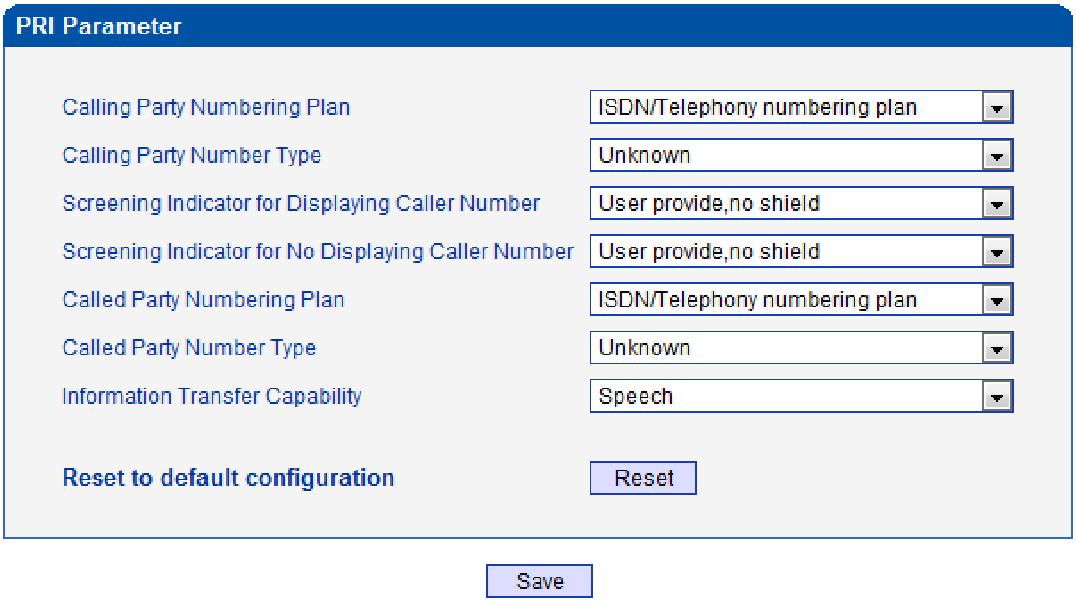
This section is mainly introduce how to configure PRI trunk.

Figure 2-5-1 PRI Config



2.5.1 PRI Parameter

Figure 2-5-2 PRI Parameter



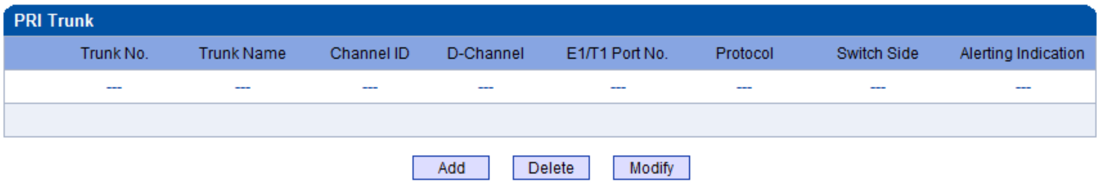
PRI parameter description

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.

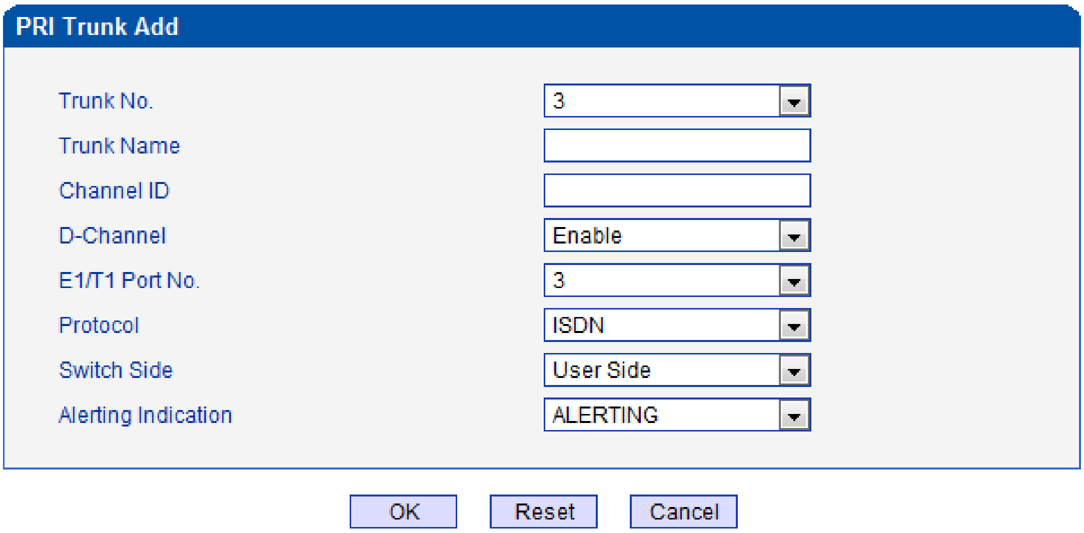
2.5.2 PRI Trunk

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.

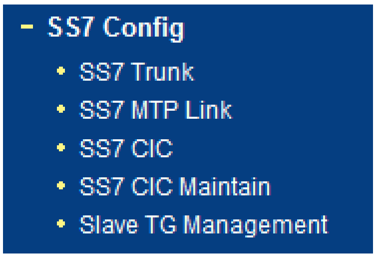
Switch Side	Indicate PRI network property of E1/T1, it is divided into: “User side” and “Network side”. When PRI loopback is carried out, the network properties of E1/T1 port at both receiving and sending sides must be different.
Alerting Indication	The ring signal include Alerting and Progress

2.6 SS7 Config

This section is mainly introduce how to configure SS7 trunk.

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

Figure 2-6-1 Add SS7 Trunk



2.6.1 SS7 Trunk

Figure 2-6-2 SS7 Trunk

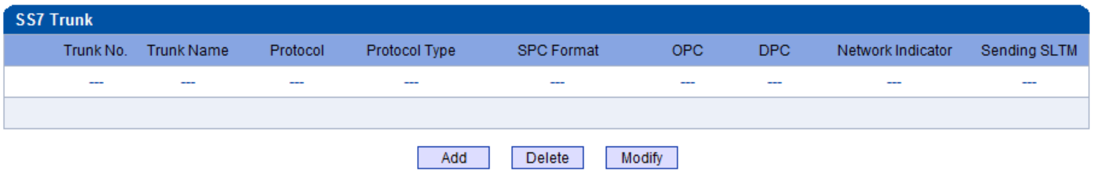
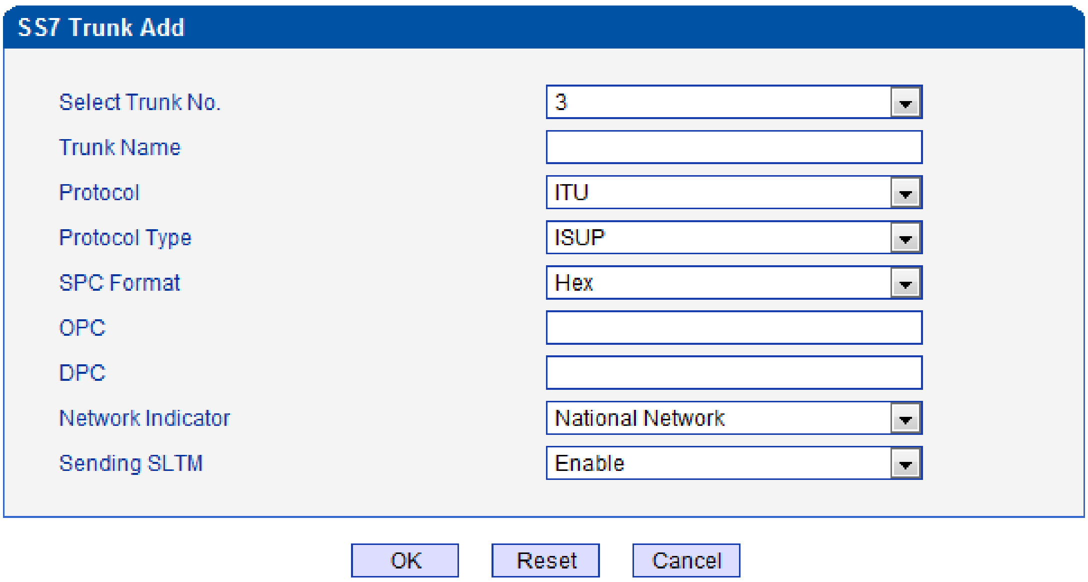


Figure 2-6-3 SS7 Trunk Add



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 point code 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

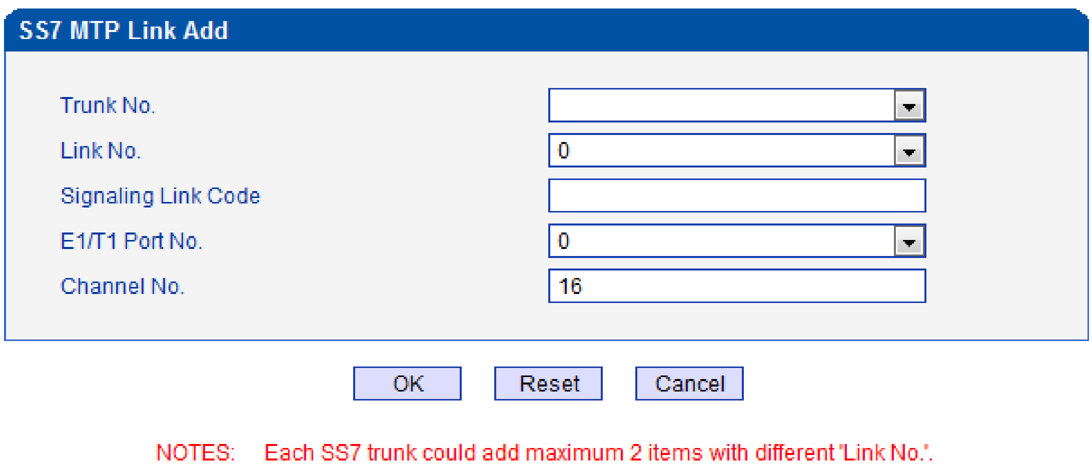
6. When the SPC length is 24 bits, and chosen ITU, OPC/DPC structure format is : x-y-z；x、z must be decimal value; y is decimal number 0-255, such as: 6-222-3

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

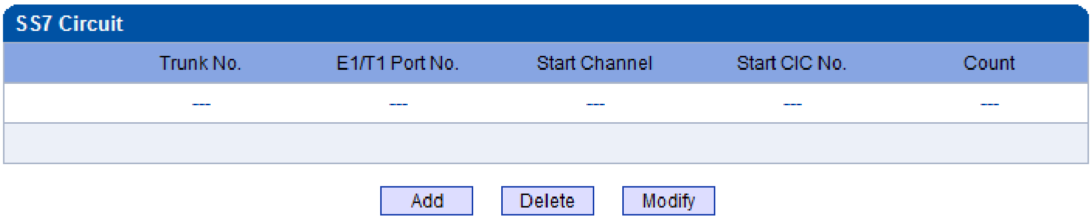
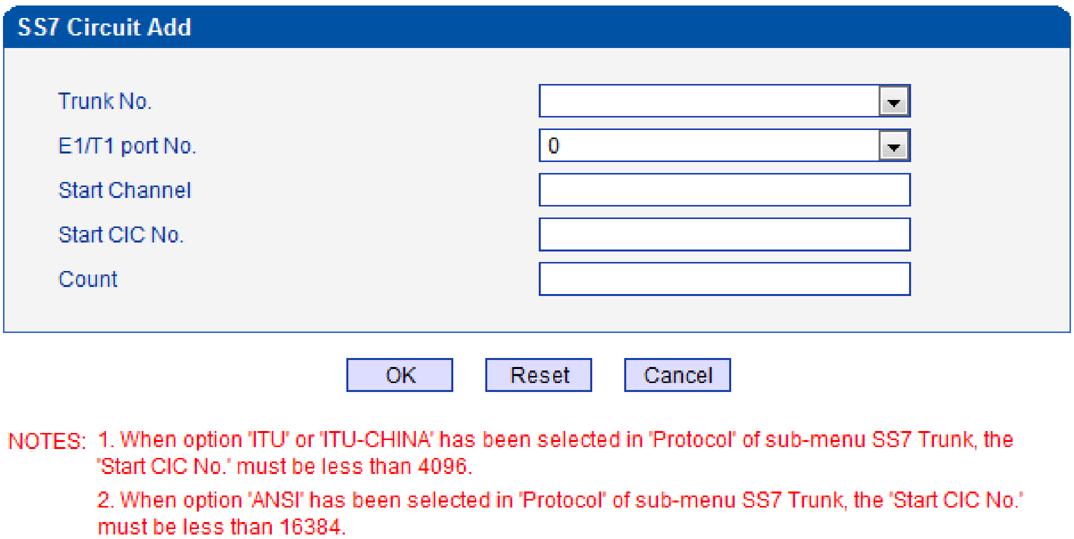


Figure 2-6-6 SS7 Circuit description



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.

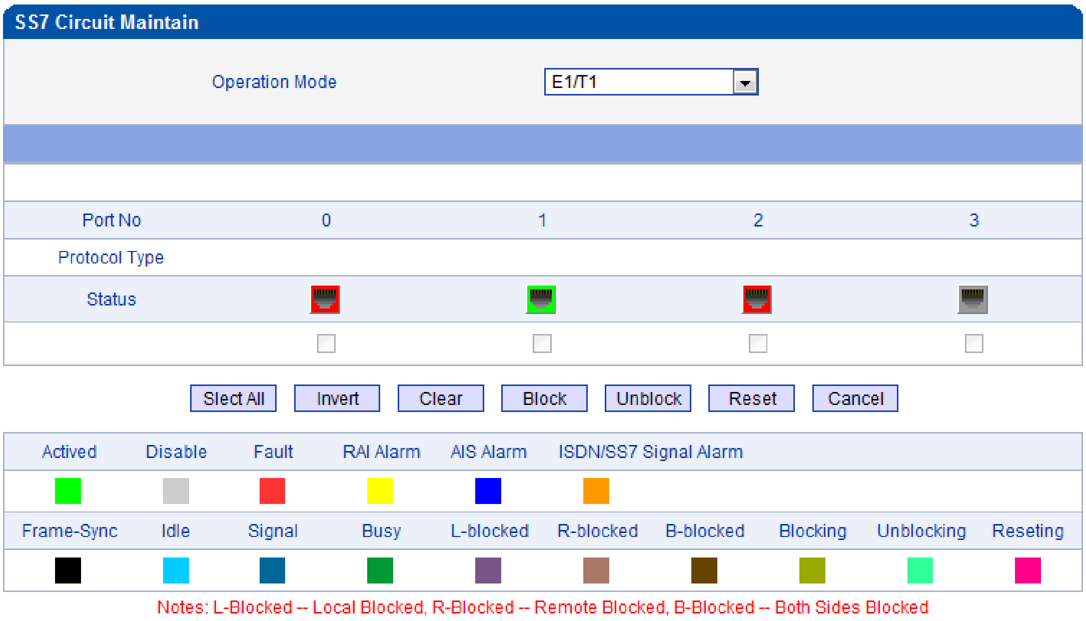


Figure 2-6-7 SS7 Circuit Maintain-E1/T1

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.

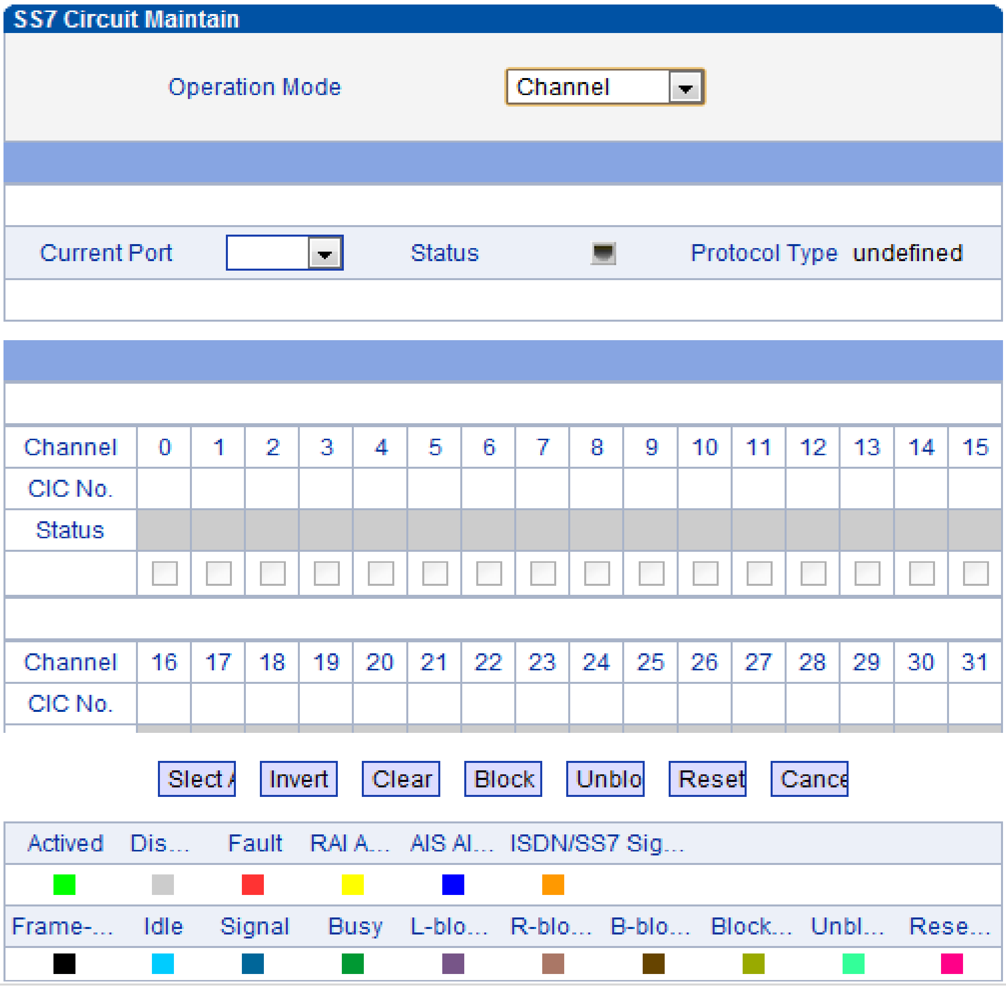


Figure 2-6-8 SS7 Circuit Maintain-Channel

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.

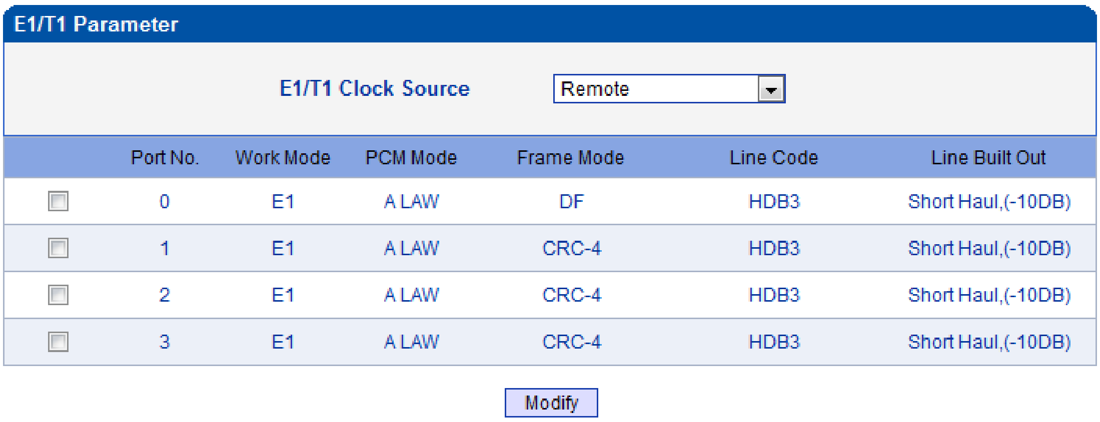


Figure 2-7-1 E1/T1 Parameter

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

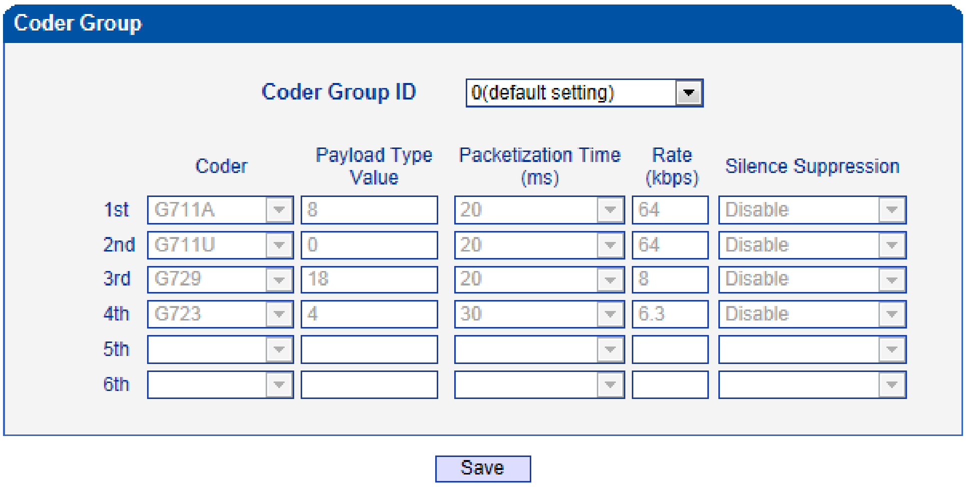


Figure 2-7-2 Coder Group

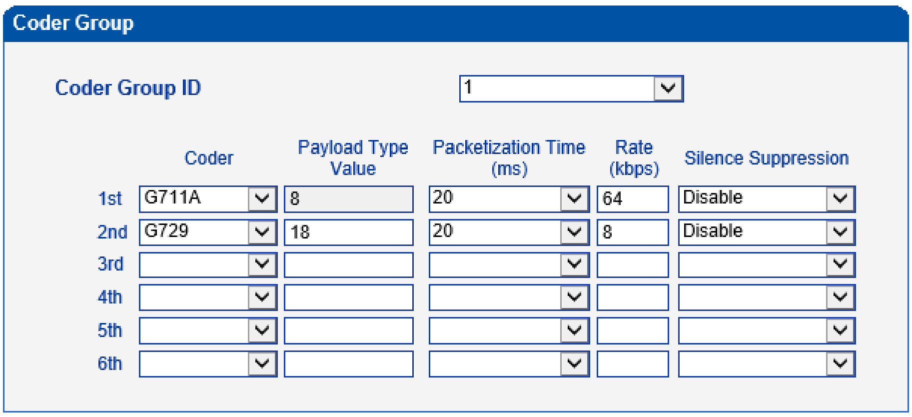
Coder group description

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.

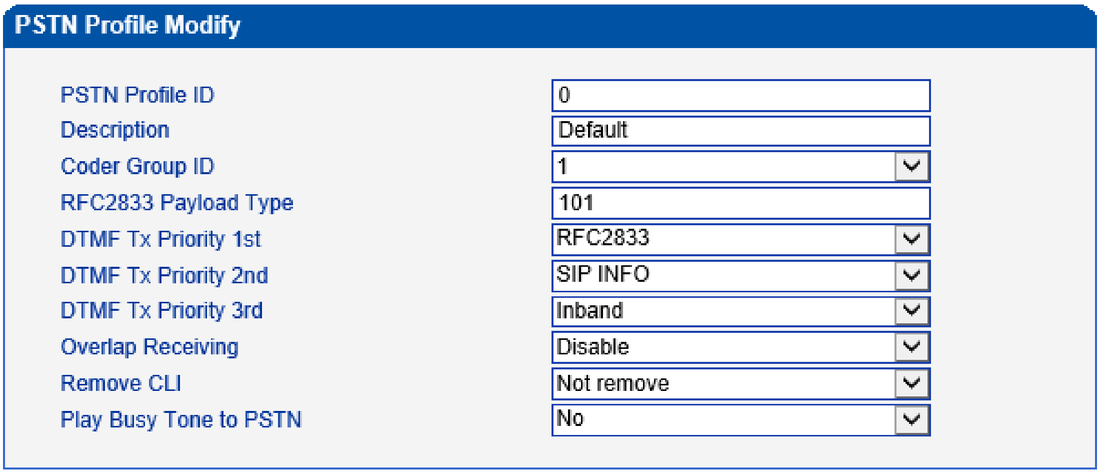
- Example: adjust preferred codec

  Step1: go to codec group page and select codec group ID 1 to create new codec group

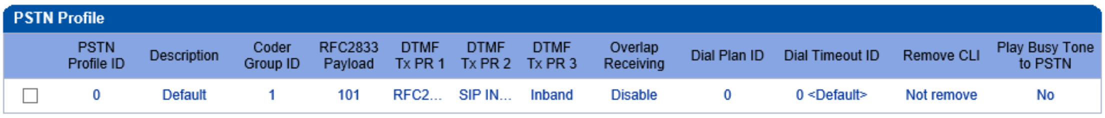
  Step2: select preferred voice codec, allow G711 ALAW and G729 in this example, as below:



Step3: modify PSTN profile and change the codec group ID



Click OK to save the configuration.


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
☐	0	Default	1	101	RFC2...	SIP IN...	Inband	Disable	0	0 <Default>	Not remove	No

2.7.3 Dial Plan

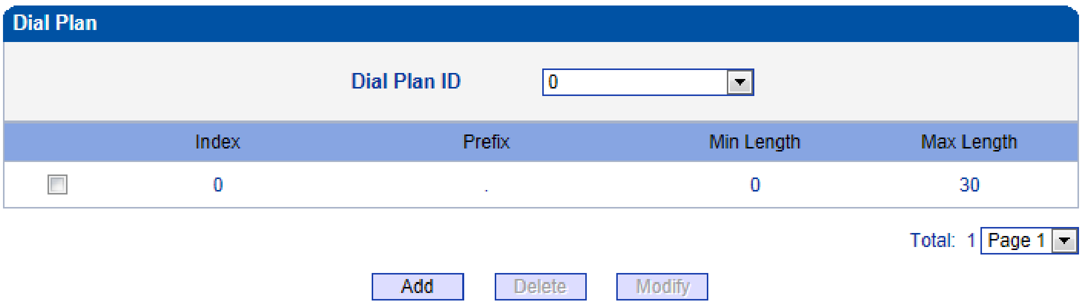


Figure 2-7-3 Dial Plan

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:

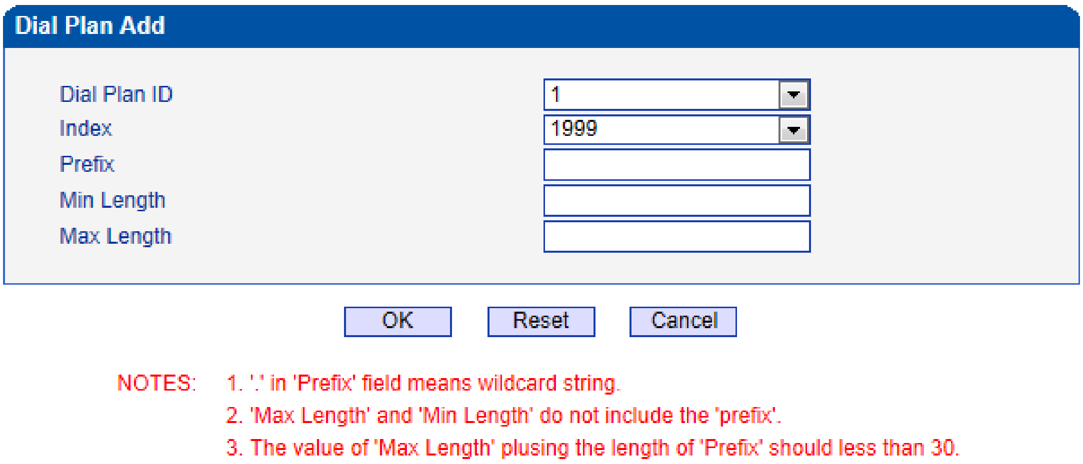


Figure 2-7-4 Dial Plan Add

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

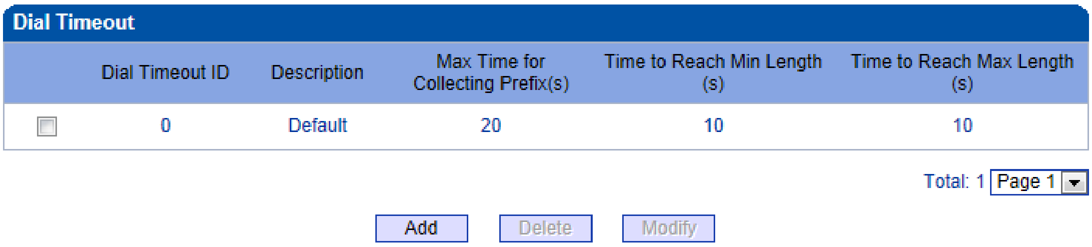


Figure 2-7-5 Dial Timeout

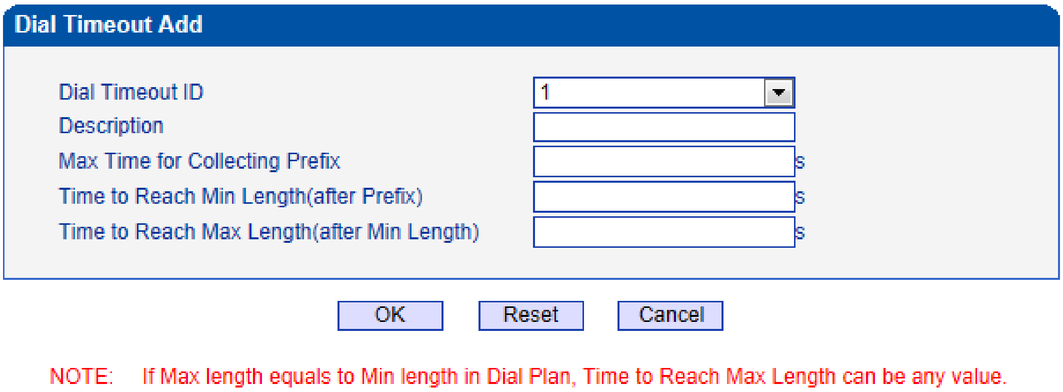


Figure 2-7-6 Dial Timeout Add

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

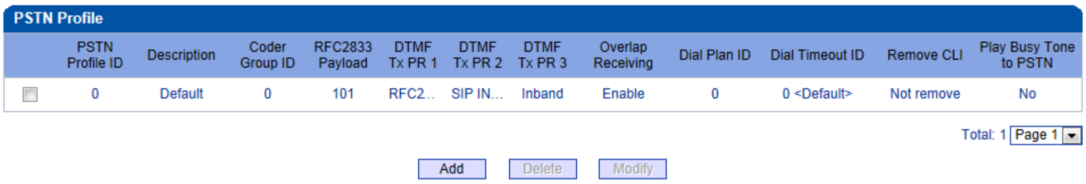


Figure 2-7-7 PSTN Profile

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

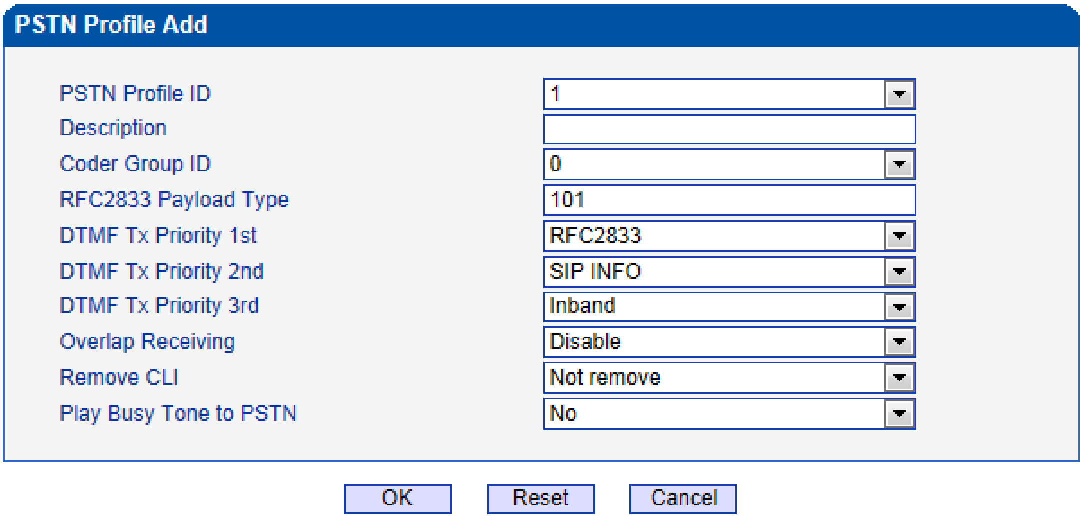


Figure 2-7-8 PSTN Profile Add

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.

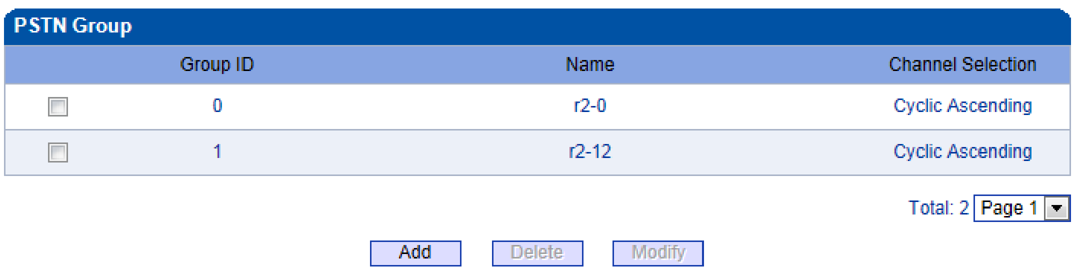


Figure 2-7-9 PSTN Group

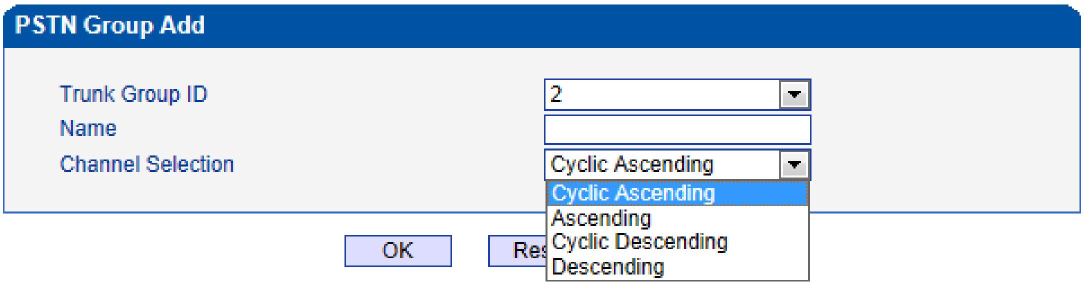


Figure 2-7-10 PSTN Group Add

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

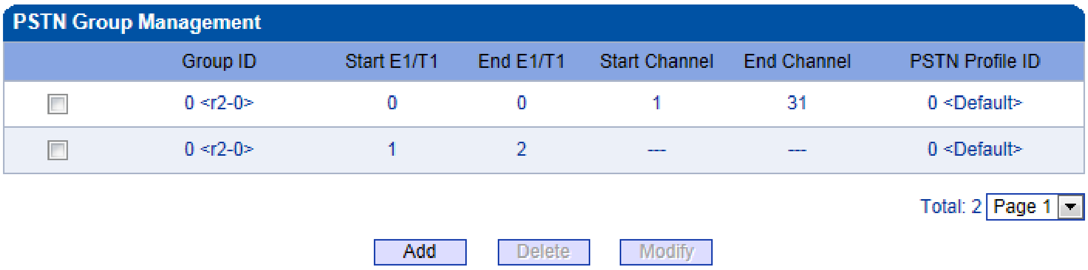


Figure 2-7-11 PSTN Group Management

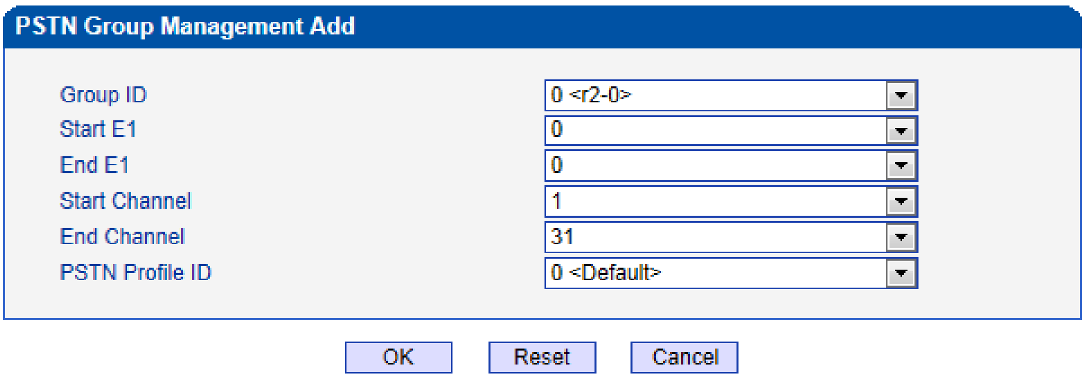


Figure 2-7-12 PSTN Group Management Add

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

2.8 SIP Config

2.8.1 SIP Parameter

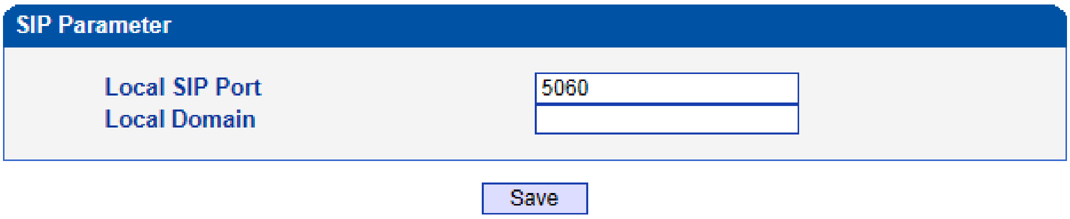


Figure 2-8-1 SIP Parameter

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

2.8.2 SIP Trunk

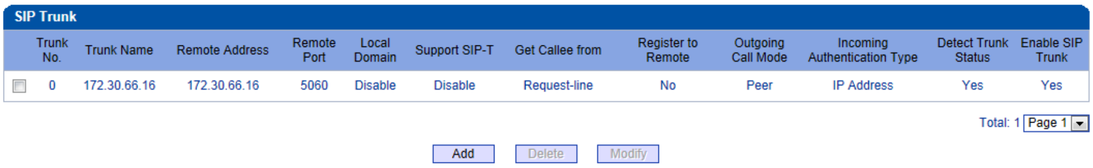


Figure 2-8-2 SIP Trunk

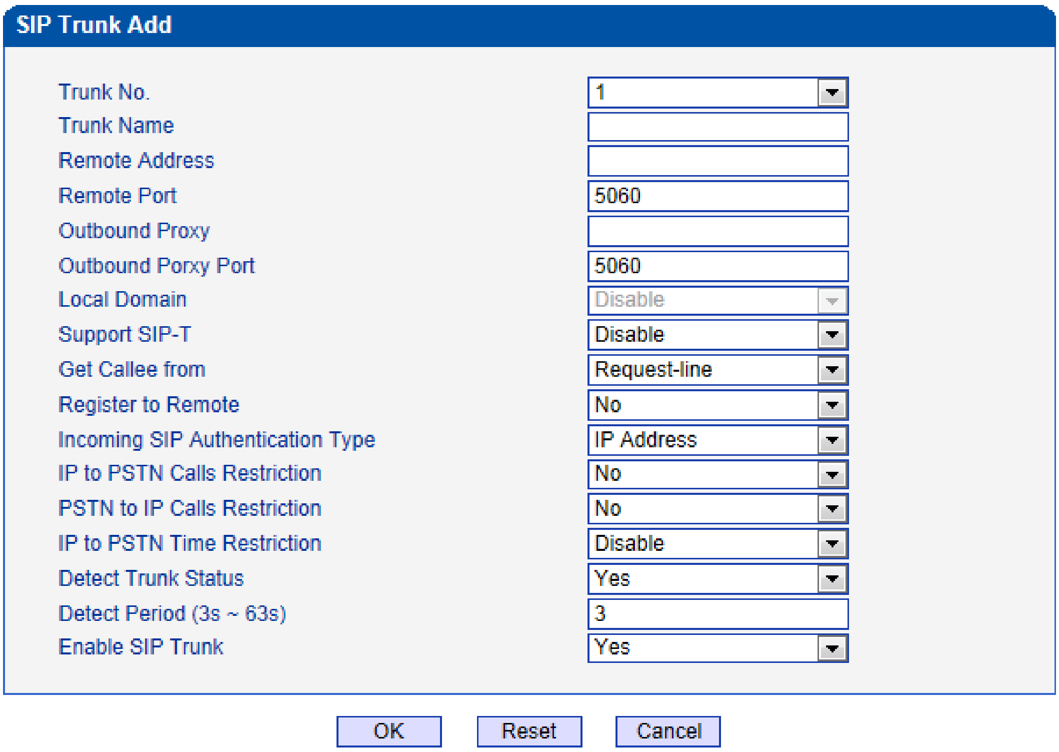


Figure 2-8-3 SIP Trunk Add

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	SIP port 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

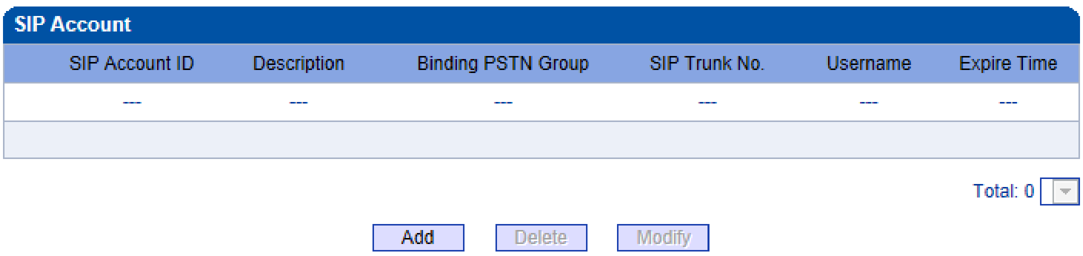


Figure 2-8-4 SIP Account

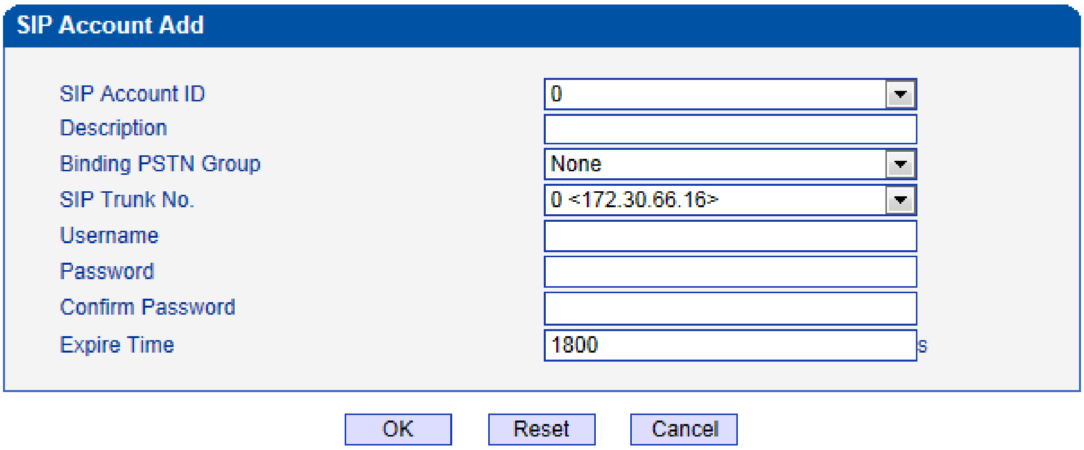


Figure 2-8-5 SIP Account Add

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

This section is available on 4/8 Ports E1/T1 gateway only.

2.9.1 H.323 Parameter

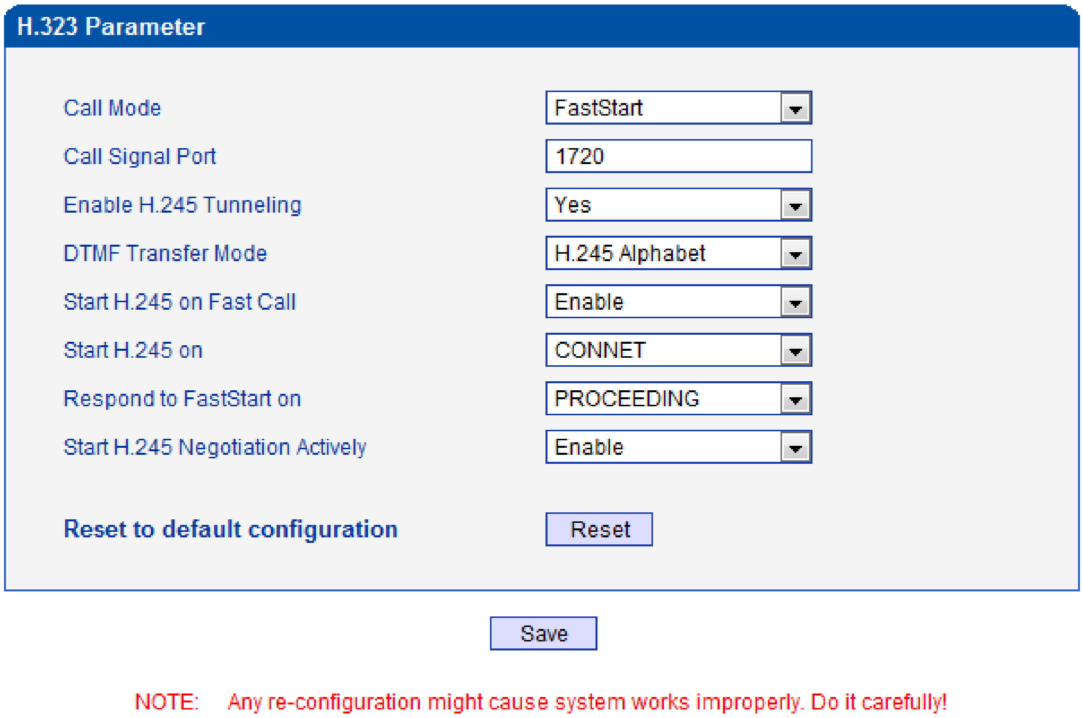


Figure 2-9-1 H.323 Parameter

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

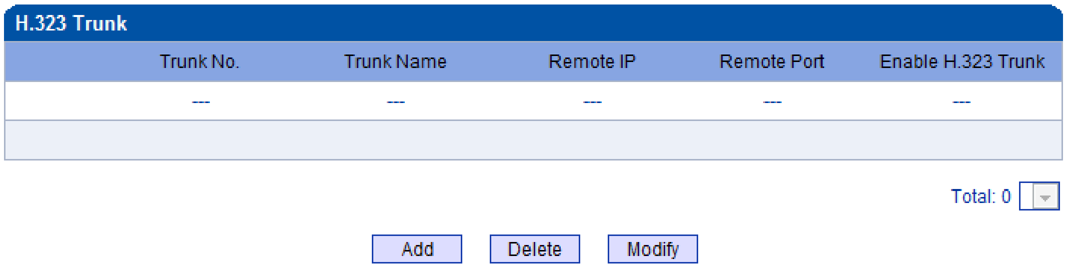


Figure 2-9-2 H.323 Trunk

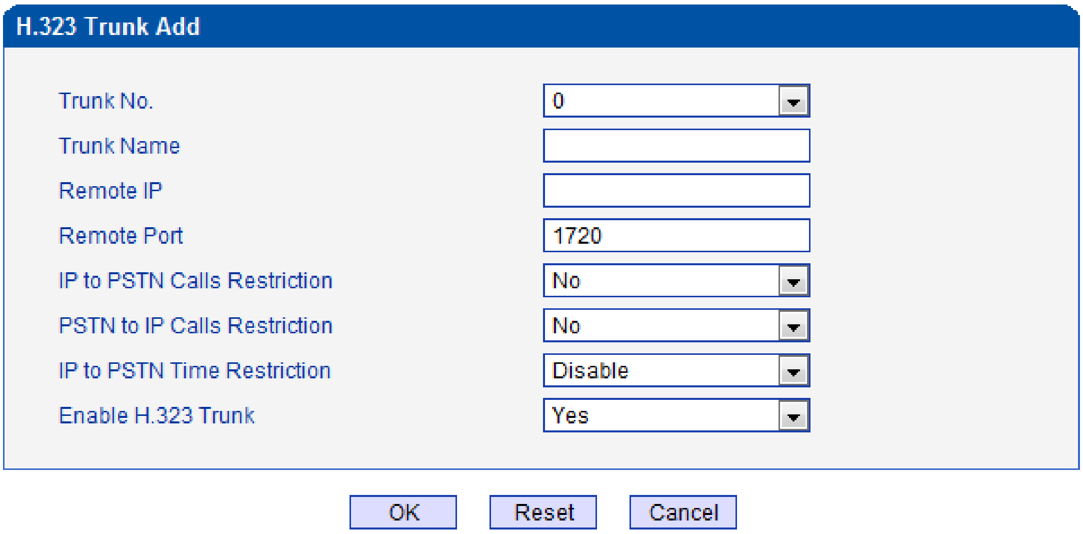


Figure 2-9-3 H.323 Trunk Add

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

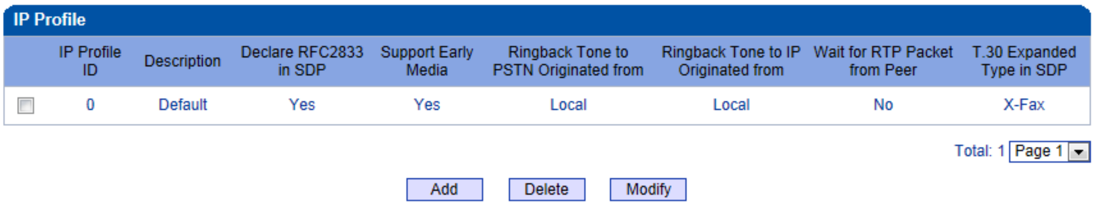


Figure 2-10-1 IP Profile

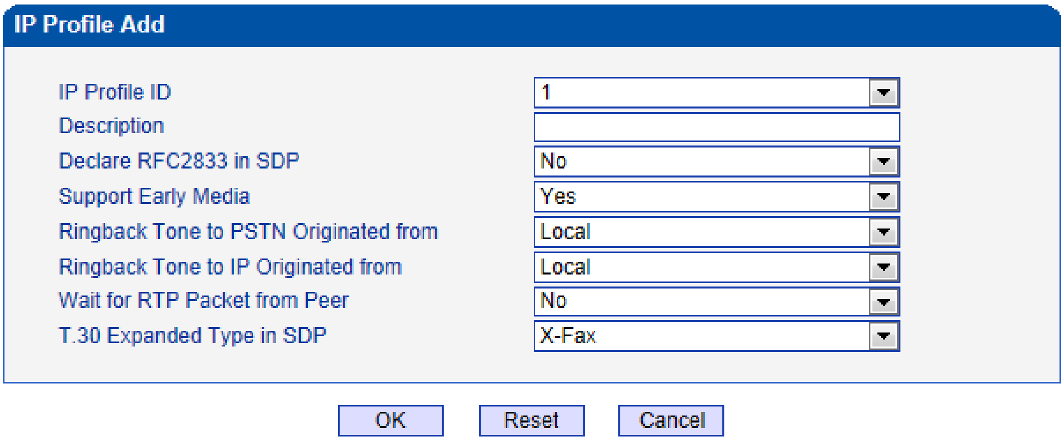


Figure 2-10-2 IP Profile Add

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

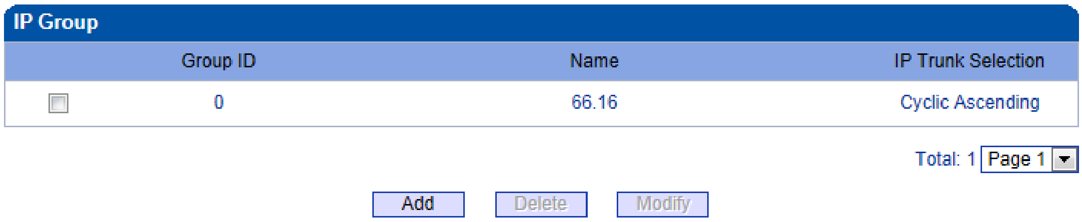


Figure 2-10-3 IP Group

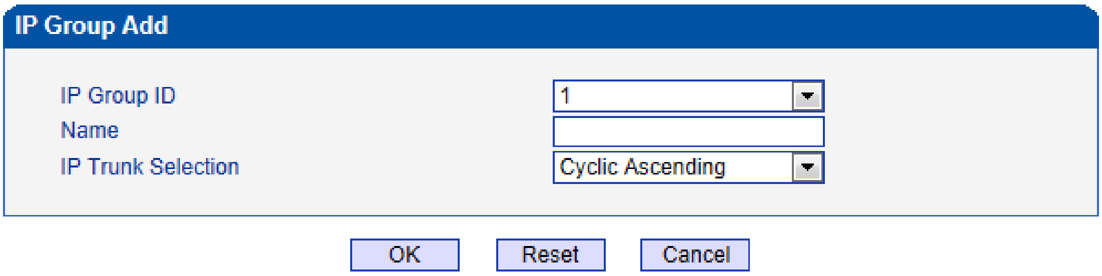


Figure 2-10-4 IP Group Add

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

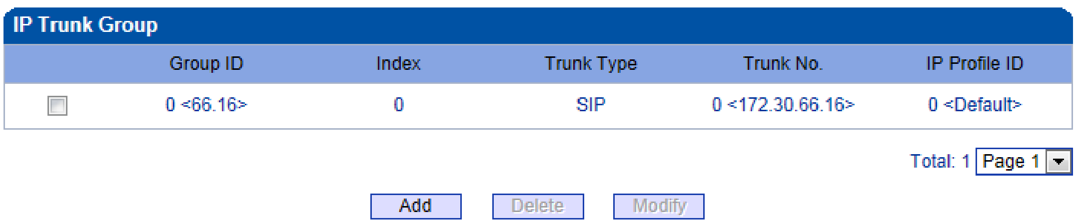


Figure 2-10-5 IP Trunk Group

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 Number filter

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

- Caller White list: to create a batch of callers or prefixes which allow to pass the call to PSTN. All calls coming from IP side will match this caller white list before deliver to PSTN
- Caller Black list: the number or prefix which added in this list would not allow to pass the call to PSTN. All calls coming from IP side will match this caller black list before deliver to PSTN.
- Callee white list: the called number white list which allow to pass the call to PSTN. All calls coming from IP side will match this called white list before deliver to PSTN
- Callee black list: the called number black list which not allow to pass the call to PSTN. All calls coming from IP side will match this called black list before deliver to PSTN
- Caller Pool: sometimes, the caller ID from IP side are illegal which not allow to deliver to PSTN. Caller Pool is to add a batch of number to replace the caller ID from IP side.
  - Example: add caller and callee white list

Add Caller white list, index 0:

Caller White List

Caller White List ID: 0

	Index	Caller Number
☐	1	xxxx
☐	2	2645666
☐	3	xxxxxxxx

Index 1 means allow caller with 4 digit length to call out

Index 2 means allow the caller number or caller with prefix 2645666 to call out

Index 3 means allow the caller with 8 digits length to call out

Add callee white list, index 0:

Callee White List

Callee White List ID: 0

	Index	Callee Number
☐	0	13788900
☐	1	xxxxxxxx

Index 0 means allow the called number or called prefix 13788900 to call to PSTN

Index 1 means allow the called number with has 8 digits length to call to PSTN

Add filter profile to add both caller and callee white list works

Filter Profile Add

Filter Profile ID	0
Description	caller&callee white list
Caller White List ID	0
Caller Black List ID	255 <None>
Callee White List ID	0
Callee Black List ID	255 <None>
Caller Pool for White List	255 <None>
Caller Pool for Black List	255 <None>

Click OK to save the configuration.

Filter Profile

	Filter Profile ID	Description	Caller White List ID	Caller Black List ID	Callee White List ID	Callee Black List ID	Caller Pool for White List	Caller Pool for Black List
☐	0	caller&callee w...	0	None	0	None	None	None

▶ Example: add 100 caller nubmer to replace all caller ID from IP side

Caller Pool

Caller Pool ID: 0

	Starting Caller Number	Number Count
☐	26452500	100

Start number 26452500, total count 100. All the caller number would be replace the number range from 26452500 to 26452599

2.12 Call Routing

2.12.1 Routing Parameter

Routing Parameter

Incoming Calls from IP	
Routing Priority	First IP->PSTN, then IP->IP
Routing & Manipulation	Routing before Manipulation
Incoming Calls from PSTN	
Routing Priority	First PSTN->IP, then PSTN->PSTN
Routing & Manipulation	Routing before Manipulation

Save

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.

2.12.2 PSTN->IP Routing

PSTN->IP Routing

	Index	Description	Trunk No.	PSTN Group	Callee Prefix	Caller Prefix	Trunk Type	Trunk No.	Destination IP Group	Filter Profile ID
☐	255	PSTN_IP	---	Any	.	.	Any	---	0 <IPgrp>	0 <caller&callee whi...

Figure 2-11-2 PSTN->IP Routing

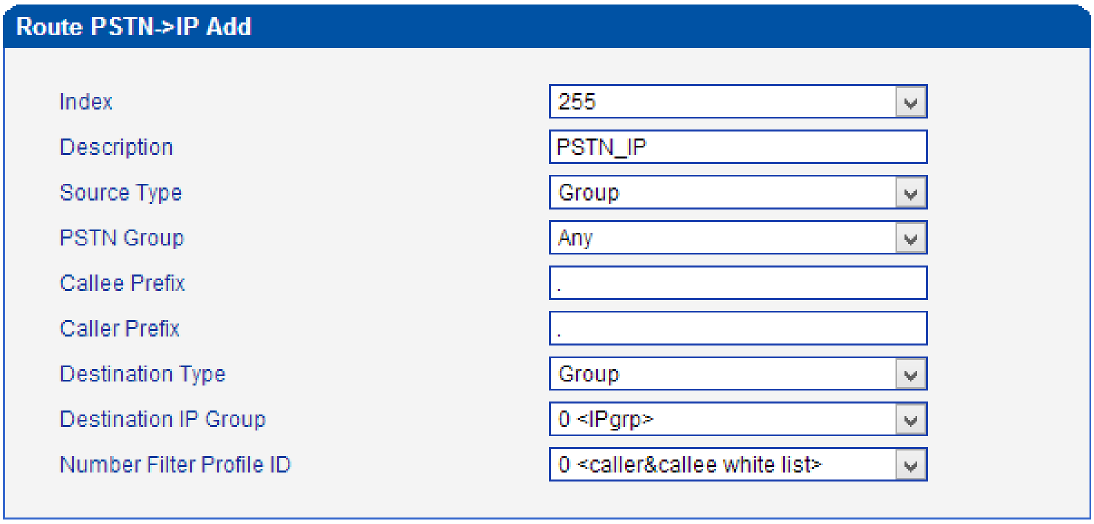


Figure 2-11-3 PSTN->IP Add

"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
Number Filter Profile ID	The profile ID which added on number filter, refer to section 2.12

2.12.3 PSTN->PSTN Routing

To add E1 to E1 call routing

Figure 2-11-4 PSTN->PSTN Routing

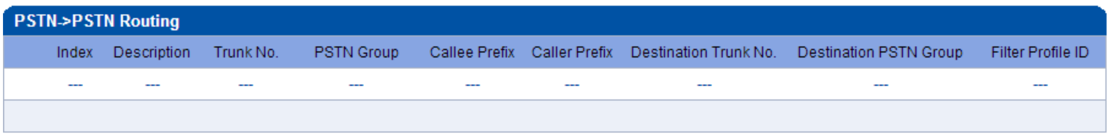
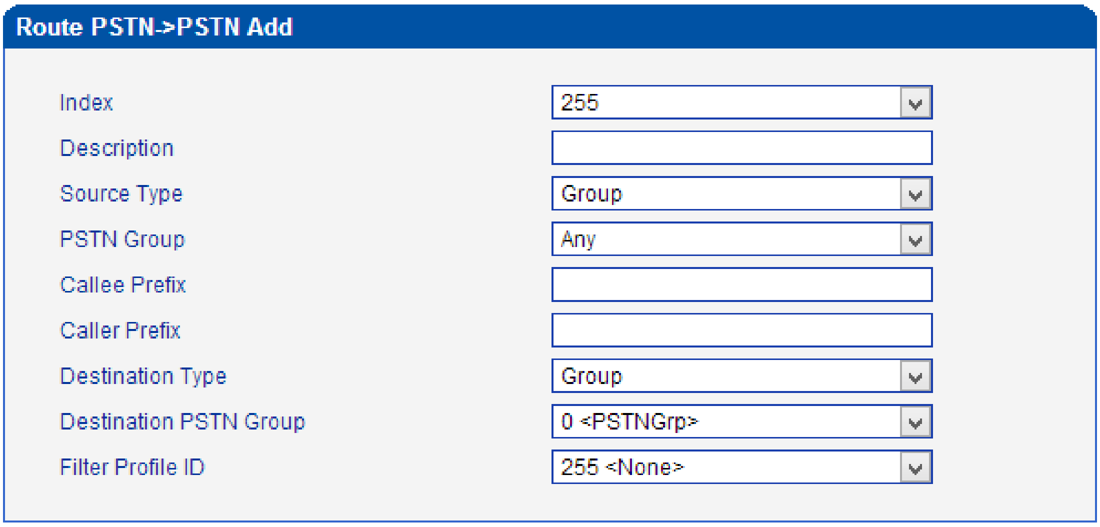


Figure 2-11-5 PSTN->PSTN Add



"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"
Filter Profile ID	The profile ID which added on number filter, refer to section 2.12

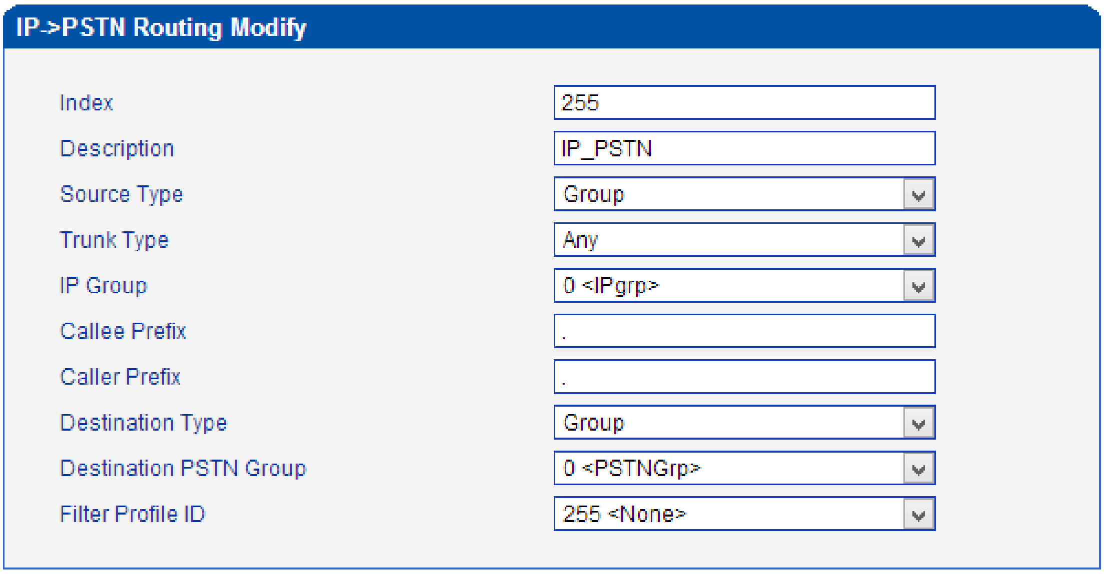
2.12.4 IP->PSTN Routing

Figure 2-11-6 IP->PSTN Routing


IP->PSTN Routing

	Index	Description	Trunk Type	Trunk No.	IP Group	Callee Prefix	Caller Prefix	Destination PSTN Trunk	Destination PSTN Group	Filter Profile ID
☐	255	IP_PSTN	Any	Any	0 <IPgrp>	.	.	---	0 <PSTNGrp>	None

Figure 2-11-7 IP->PSTN Routing

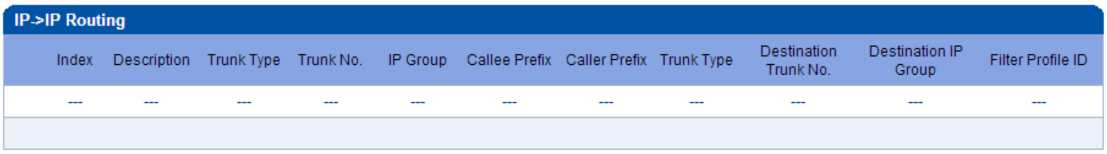


“IP -> PSTN Routing”： Routing Call from IP to PSTN

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

2.12.5 IP->IP Routing

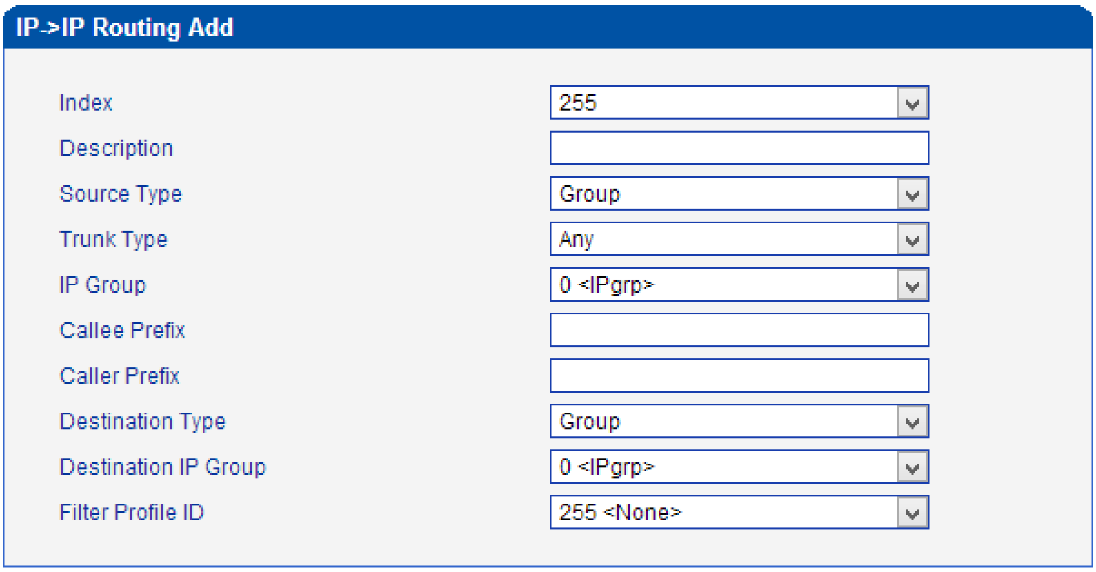
Figure 2-11-8 IP->IP Routing



IP->IP Routing

Index	Description	Trunk Type	Trunk No.	IP Group	Callee Prefix	Caller Prefix	Trunk Type	Destination Trunk No.	Destination IP Group	Filter Profile ID
---	---	---	---	---	---	---	---	---	---	---

Figure 2-11-9 IP->IP Add

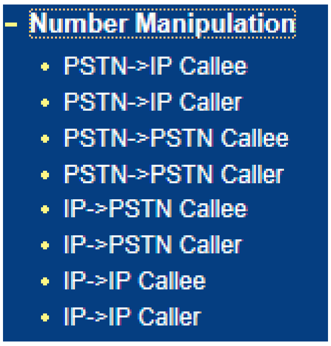


IP->IP routing configuration and PSTN->IP configuration are similar. The only difference is that the destination is the IP group.

2.13 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.

2.13.1 PSTN->IP Callee

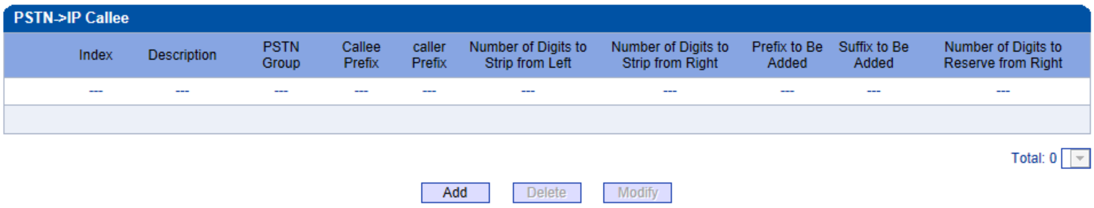


Figure 2-12-2 PSTN->IP Callee

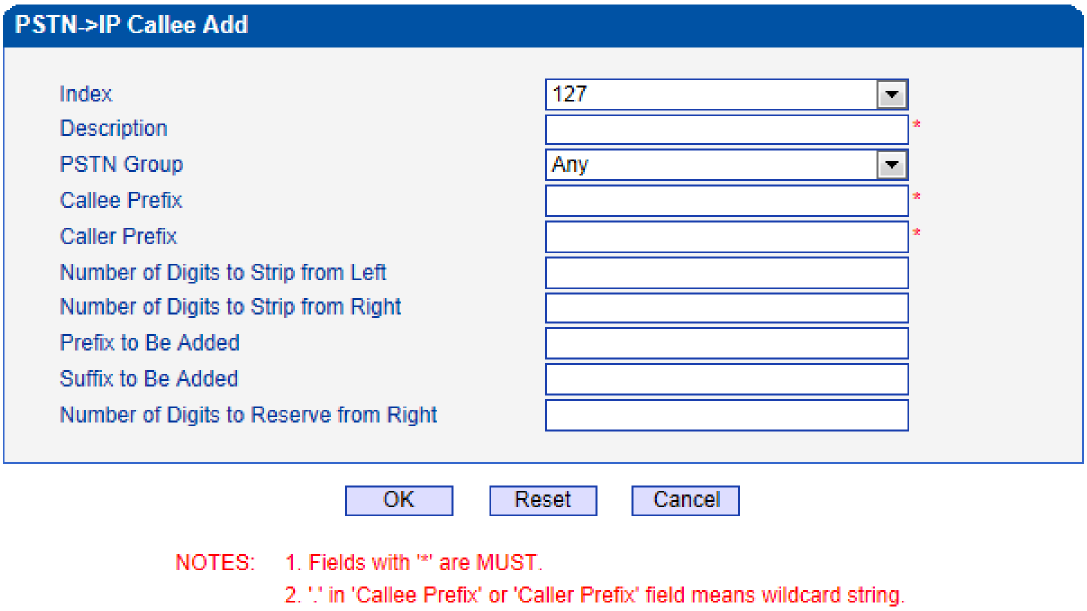


Figure 2-12-3 PSTN->IP Callee Add

"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

2.13.2 PSTN->IP Caller

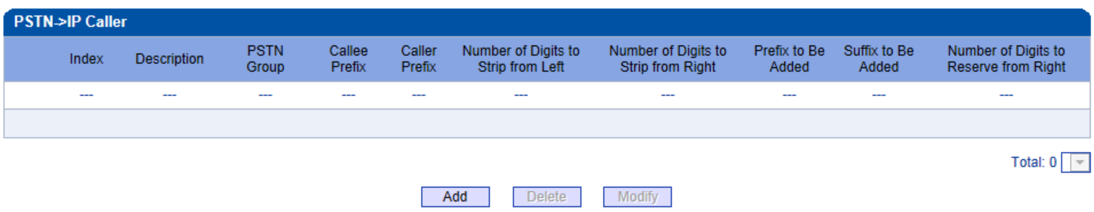


Figure 2-12-4 PSTN->IP Caller

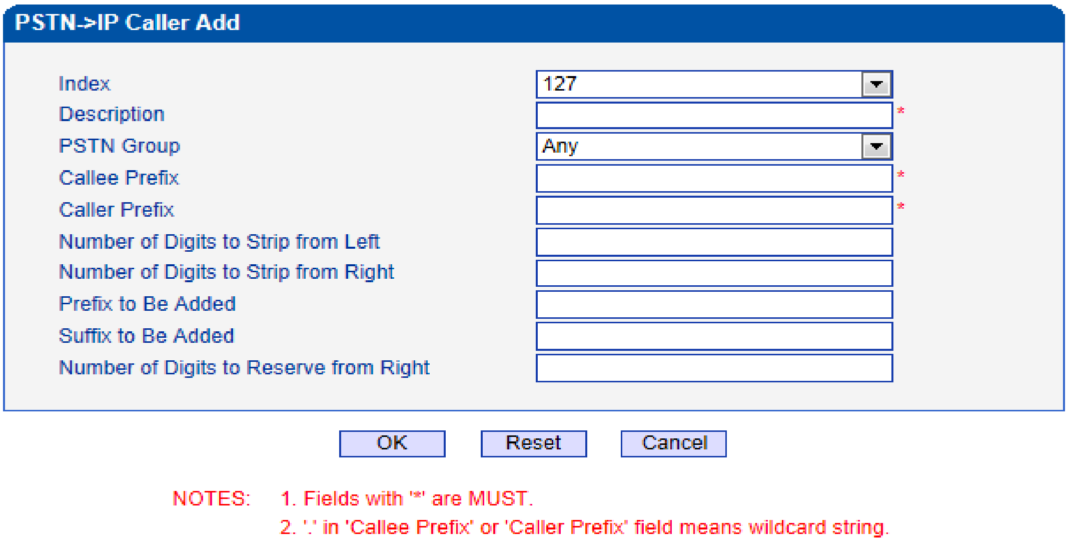


Figure 2-12-5 PSTN->IP Caller Add

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

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.

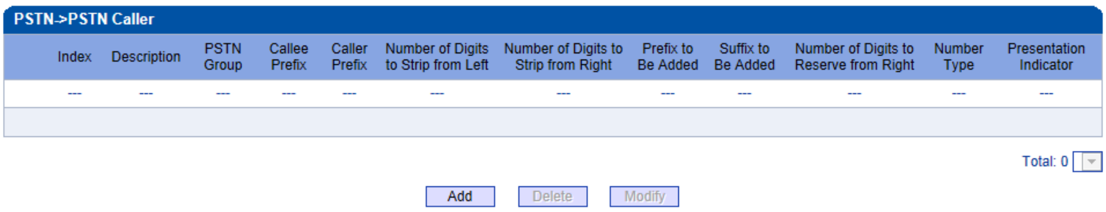


Figure 2-12-7 PSTN->PSTN Caller

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

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

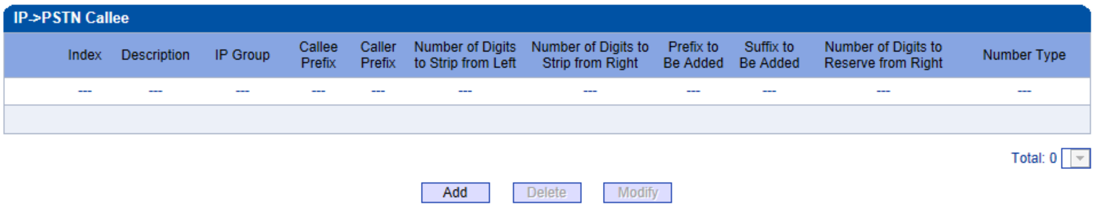


Figure 2-12-8 IP->PSTN Callee

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.

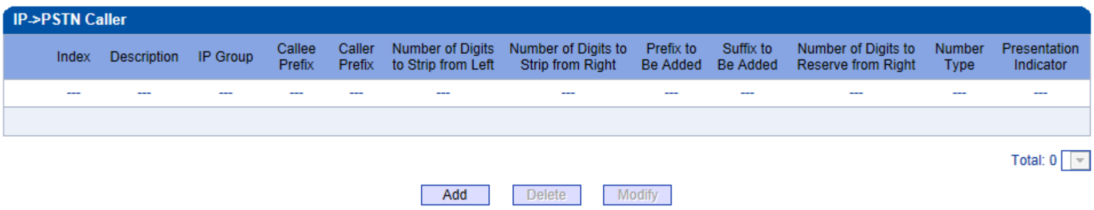


Figure 2-12-9 IP->PSTN Caller

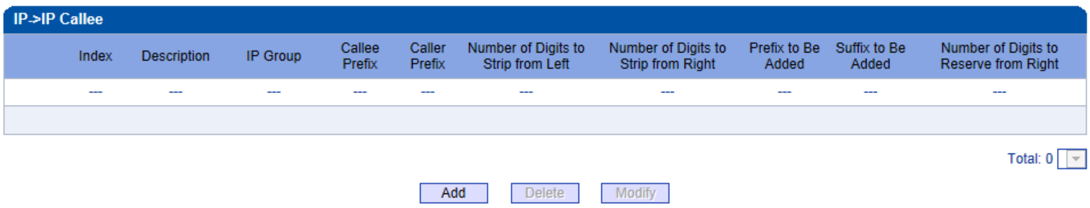


Figure 2-12-10 IP->IP Callee

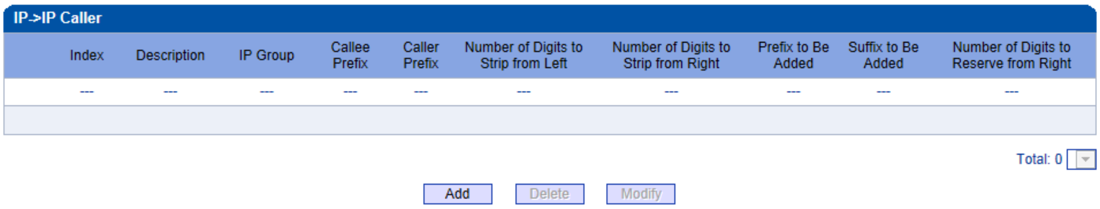


Figure 2-12-11 IP->IP Caller

2.14 Voice & Fax

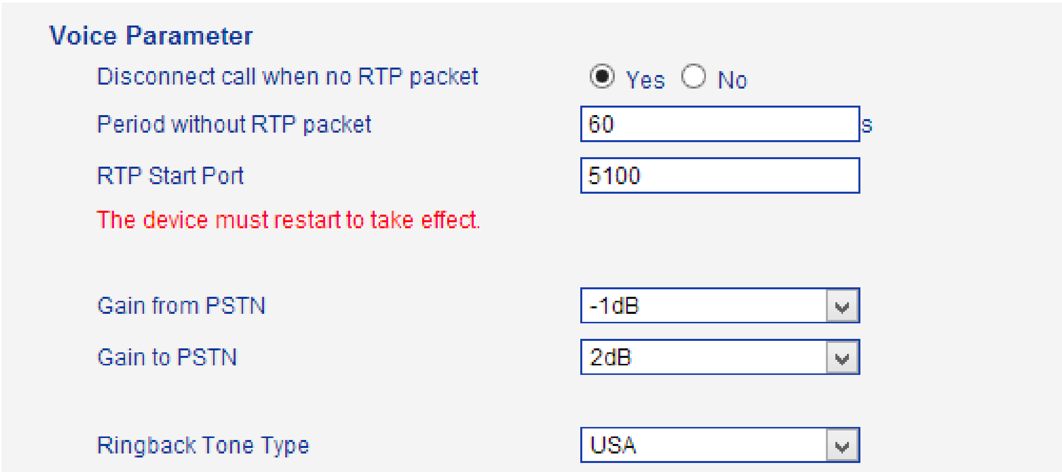


Figure 2-13-1 Voice & Fax

- 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. Default is 60s

RTP Start Port

Default start port is 5100

PSTN in Gain

To improve voice value from PSTN to IP direction.

IP in Gain

To improve voice value from IP to PSTN direction.

Ringback tone Type

It take effective while play local ringback tone.

- Timeout of no answer

No answer timeout timer, default value is 60s on both direction

- FAX Parameter

FAX mode

T.38/Pass-through; default option is T.38.adaptive means auto negotiate with peer side.

FAX TX Gain

Gain of sending a fax

FAX RX Gain

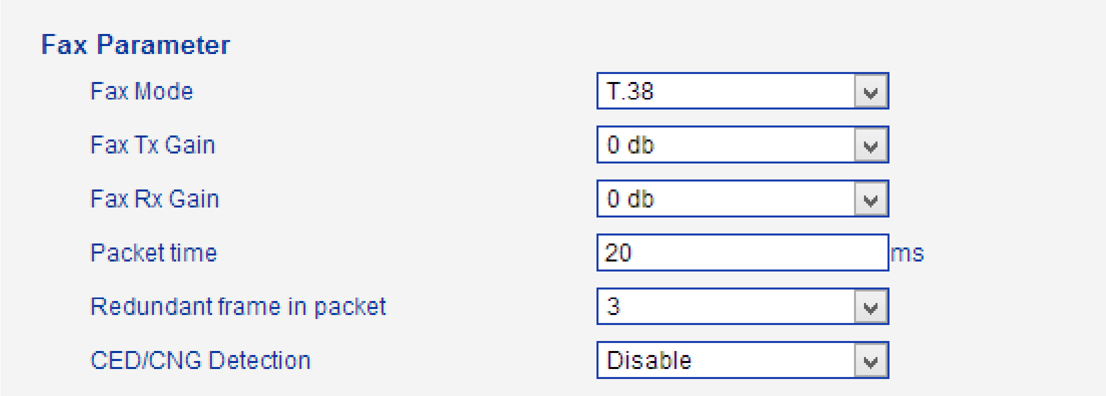
Gain of receiving a fax

Packet time

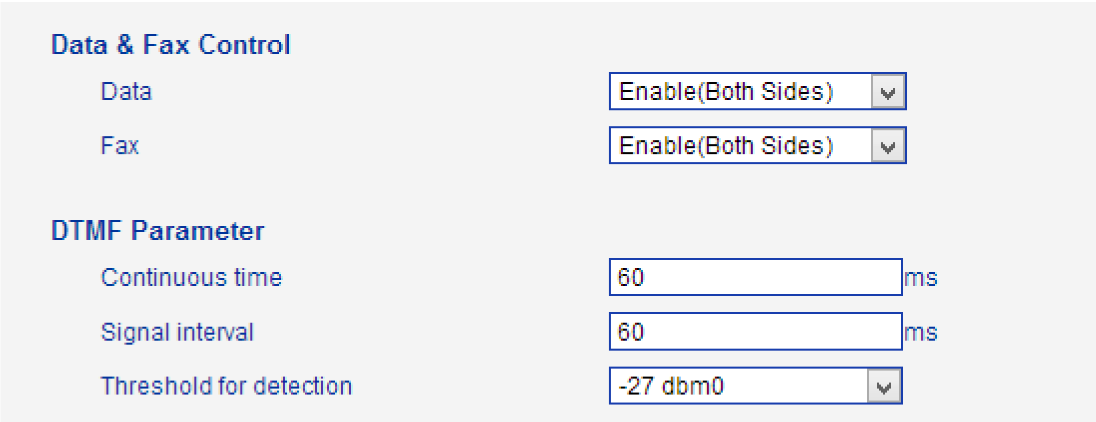
Data packing duration

Redundant frame in packet

The length of frame in RTP packet



- DTMF Parameter



Data & Fax Control

Enable/disable FAX and Data service on the gateway.

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

2.15 Maintenance

2.15.1 Management Parameter

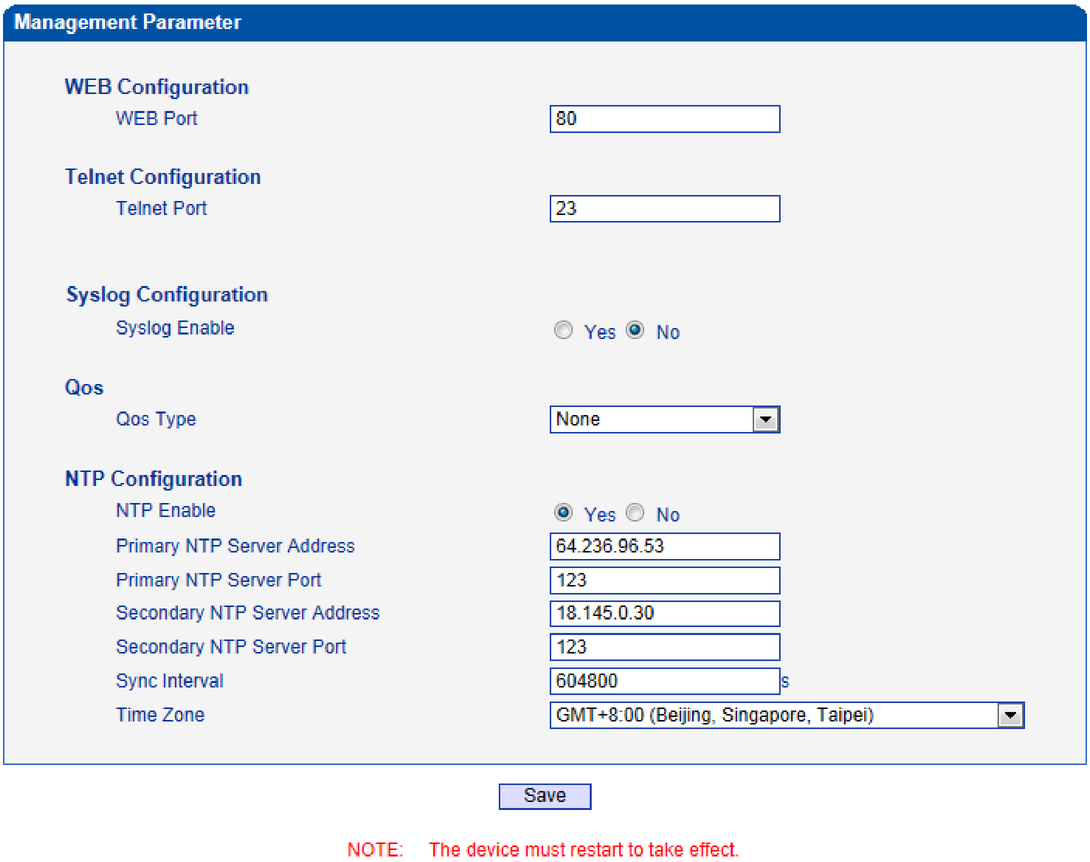


Figure 2-14-1 Management Parameter

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

2.15.2 SNMP Parameter

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

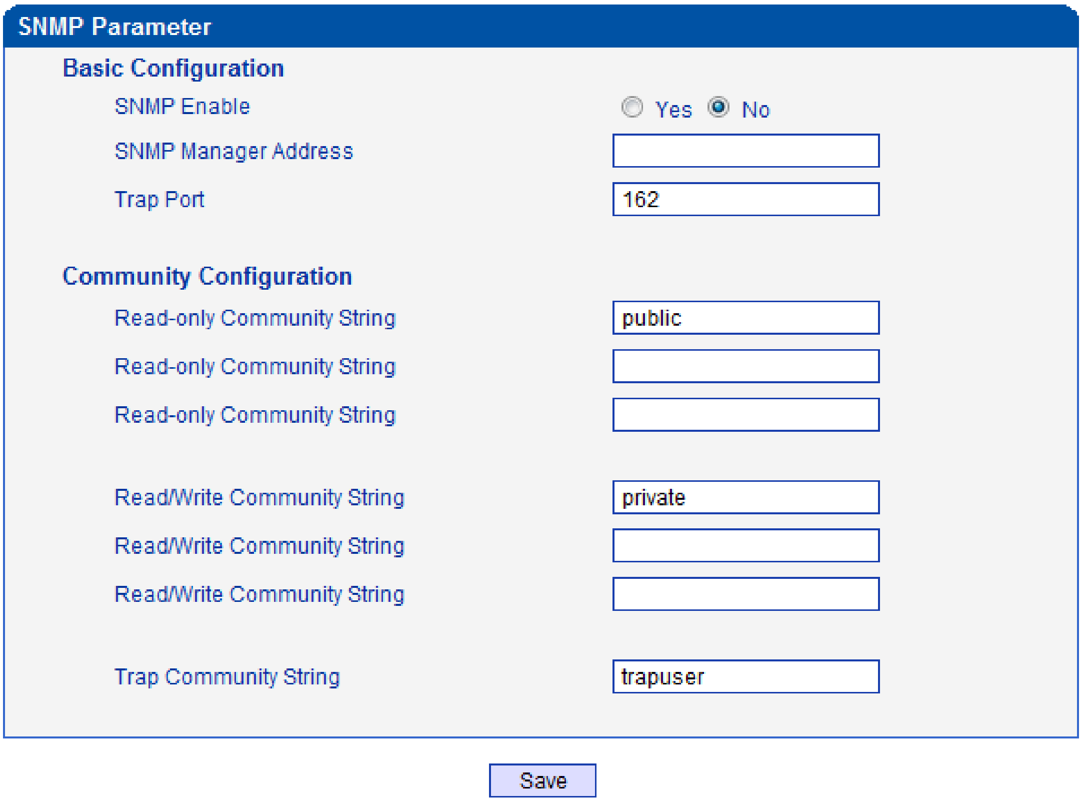


Figure 2-14-3 SNMP Parameter

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.

2.15.3 Data Backup

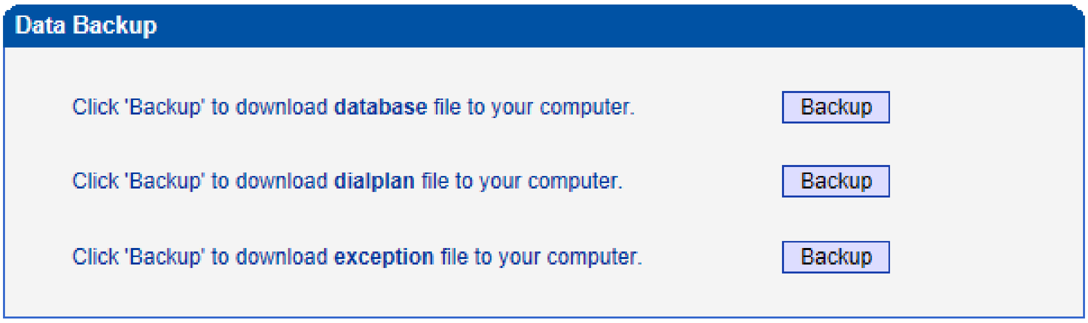


Figure 2-14-4 Data Backup

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

2.15.4 Data Restore

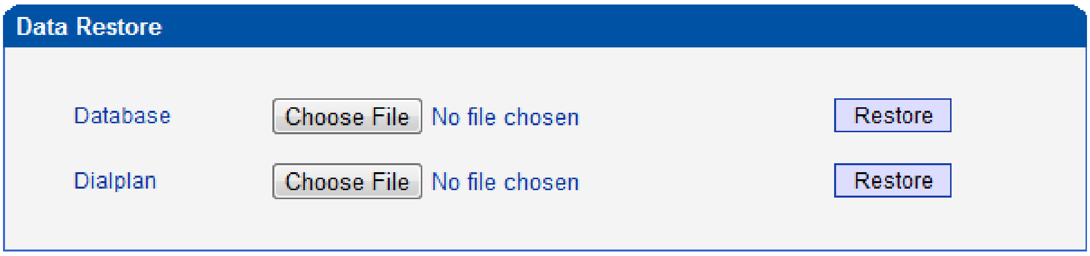
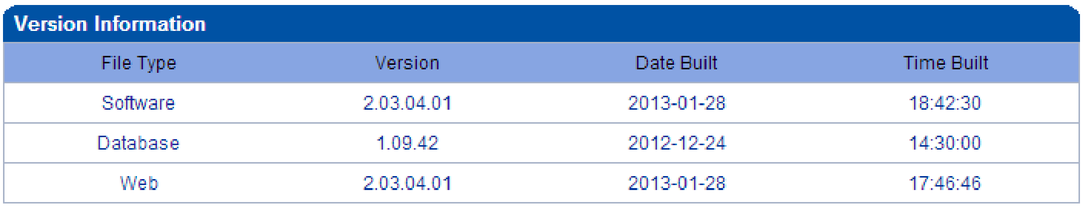


Figure 2-14-5 Data Restore

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".

2.15.5 Version Information



Version Information

File Type	Version	Date Built	Time Built
Software	2.03.04.01	2013-01-28	18:42:30
Database	1.09.42	2012-12-24	14:30:00
Web	2.03.04.01	2013-01-28	17:46:46

Figure 2-14-6 Version Information

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

2.15.6 Firmware Upload

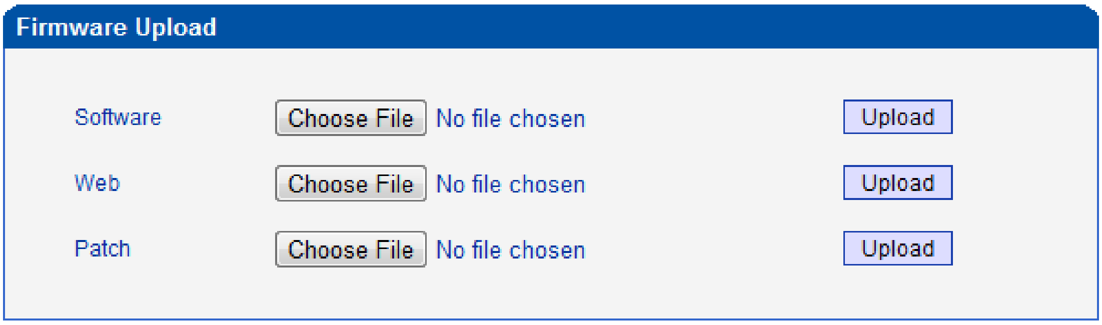


Figure 2-14-7 Firmware Upload

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".

2.15.7 Modify Password

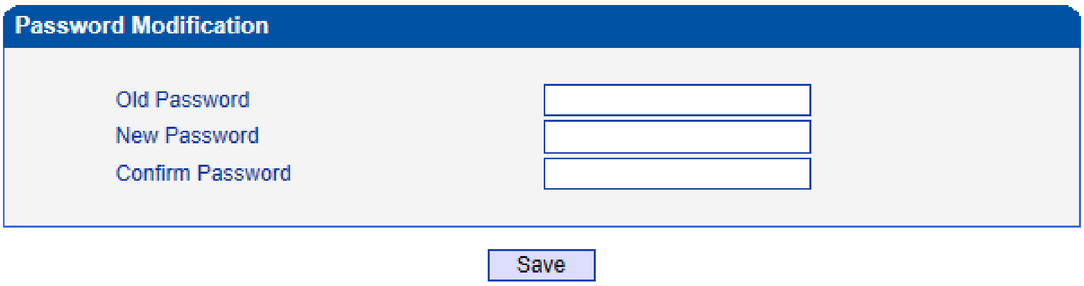


Figure 2-14-8 Modify Password

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

2.15.8 Device Restart

Figure 2-14-9 Restart Device

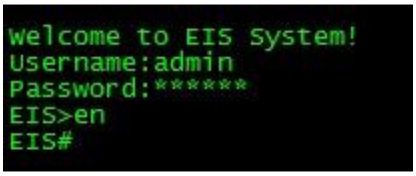
If user click Restart, a message ("Are you sure?") will be popped up, and then click OK.

3 Troubleshooting and Command Line

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

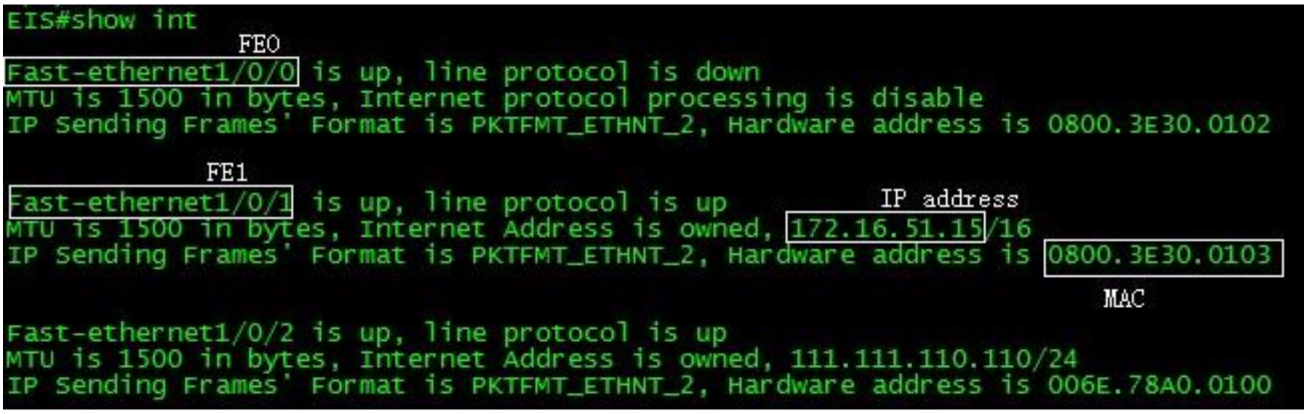
3.1 Basic Command

Run system tool Telnet to login gateway, after entering **username** and **password**, and then run command **en** to active the privileged commands.

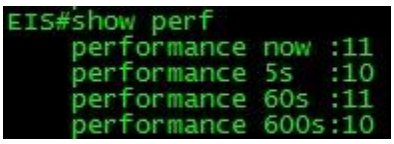


3.2 Show IP address

run the command **show int**, the output shows FE0 and FE1 ports name, IP address and MAC address.



3.3 Show CPU performance



Performance now: cpu load at current time
Performance 5s: cpu load at average 5 seconds
Performance 60s: cpu load at average 60 seconds

Performance 600s: cpu load at average 600 seconds

3.4 Show ss7 status

run the command **show ss7 sta**, the out should like this:

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

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

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

3.5 Show ss7 ts

the system will show the status of each channel in each port**.**

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

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

Note: This is not available for PRI

3.6 Block ss7 ts

enter config mode by running command **^config**

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

Block entire e1

Example:

block port 2, run the command **busy –cic 2**, the system will disable port 2 into a locked status.

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

Unblock entire e1

Example:

unblock port 2, to active the port 2, please run the command **free –cic 2**

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

Block specified ts

Example:

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

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

unBlock specified ts

Example:

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

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

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

3.7 Show ss7/PRI/cc call information

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

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

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

online total calls: 9
```

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

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

3.8 Debug call (call control log analyze):

debug call control(recommend)

set the track condition

debug all the call, run the command **debug cc detail all** to debug all calls.

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

Or debug a call by the called or calling number

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

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

(replace the called/calling number by yours)

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

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

 Trace num:2  All trace:0
```

And then exit config mode, into ada mode and turnon port

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

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

(ex = exit)

Cancel debug cc, turn off the debug mode for cc all.

```
EIS(ada)#turnoff 27
EIS(ada)#ex
EIS#^config
EIS(config)#no debug cc all
 Set successfully! current:0
```

Example 1 : One succ call from IP to PSTN:

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

(note: receive a call from siptrunk)

o=- 12949395404797000 1 IN IP4 172.16.100.172
s=CounterPath X-Lite 4.0
c=IN IP4 172.16.100.172
t=0 0
a=ice-ufrag:2c37f5
a=ice-pwd:a0b4dc8cc787732f66e9e625e69dbbfd
m=audio 50832 RTP/AVP 107 0 8 101
a=rtpmap:107 BV32/16000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=candidate:1 1 UDP 659136 172.16.100.172 50832 typ host

[070-14:18:49:710]ST: <5,Sip-t,2,65535,987654321,idle> ==>> CC_ST_SETUP, ccb:5, user type:0(Norm), calling:987654321, longnum:987654321, trunkGrpId:255, profileId:255, std sdp:v=0
o=- 12949395404797000 1 IN IP4 172.16.100.172
s=CounterPath X-Lite 4.0
c=IN IP4 172.16.100.172
t=0 0
a=ice-ufrag:2c37f5
a=ice-pwd:a0b4dc8cc787732f66e9e625e69dbbfd
m=audio 50832 RTP/AVP 107 0 8 101
a=rtpmap:107 BV32/16000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=candidate:1 1 UDP 65913
[071-14:18:49:710]ST: <Sip-t,2,65535,987654321> ====Processed: SIP_CALL_INVITE
[072-14:18:49:710]ST: cr, no:9, ccb:5, State:1(init), cause:0(CCS_NONE(无原因值)), redirect:0
[073-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> <<== CC_ST_SETUP, cr:9, calling:987654321, longNum:987654321, dial:1234567, num_ok:1,calltype:2(msg), rtsType:0, callType:2(ccb), fax dest<65535,65535>, trunkGrpId:255, profileId:255, sigToneTyp:0, std sdp:v=0

o=- 12949395404797000 1 IN IP4 172.16.100.172
s=CounterPath X-Lite 4.0
c=IN IP4 172.16.100.172
t=0 0
a=ice-ufrag:2c37f5
a=ice-pwd:a0b4dc8cc787732f66e9e625e69dbbfd
m=audio 50832 RTP/AVP 107 0 8 101

a=rtpmap:107 BV32/16000
a=rtpmap:101
[074-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> **predispose start calling :987654321 called:1234567**!
[075-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> predispose end calling :987654321 called:1234567!
[076-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> source user auth:0x6, is fxo call in auth pass:0
[077-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> route type:3(Out route) -- before cc number analysis.
[078-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> **[IP2tel]match route succ! srcIpGrpId:0, dstTrkGrpId:0, ChnSelMode:0, callingProfId:0, srcIpGrpId:0.**

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

[079-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> **[before manipulate number]calling:987654321, called:1234567, longNum:987654321, anl called:1234567, isRouteAfNumManip:0, callerNumTyp:255, calledNumTyp:255, presentId:0.**
[080-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> **[after manipulate number]calling:987654321, called:1234567, longNum:987654321, anl called:1234567, callerNumTyp:0, calledNumTyp:0, presentId:0.**

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

[081-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> analysis successfully, service:0(normal), bill:4(normal), route:3(rts_out), dest_term:8(Ss7), dest_dev:65535, dest_port:65535, dest_grp:65535, called:1234567 !
[082-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> number convert, old calling:987654321, old called:1234567
[083-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> before trans num process! caller:987654321, disNum:, called:1234567, g_ulIsTransOrgCalleeNum:0, g_ulNumTransType:1, g_ulAllowMobileTransfer:0!
[084-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> number convert, new calling:987654321, dis num:, new called:1234567
[085-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> ==>> CC_ST_PROCEEDING, called:1234567
[086-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> calling :0.0.0.0 called:255.255.255.255
[087-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> is need reflect:0, callingProfId:0.
[088-14:18:49:710]CC: <5,Ss7,65535,65535,,idle> ==>> CC_ST_SETUP, cr:10, calling:987654321, longNum:987654321, dial:1234567, OrgCallee:, num_ok:1, trunkGrpId:0, profileId:255, isForceReflect(ccb):0, ringback2IP:1,std sdp:v=0
o=- 12949395404797000 1 IN IP4 172.16.100.172
s=CounterPath X-Lite 4.0
c=IN IP4 172.16.100.172

t=0 0
a=ice-ufrag:2c37f5
a=ice-pwd:a0b4dc8cc787732f66e9e625e69dbbfd
m=audio 50832 RTP/AVP 107 0 8 101
a=rtpmap:107 BV32/16000
a=rtpmap:101 telephone-event/8000
a=fmtp:10
[089-14:18:49:710]CC: <5,Sip-t,2,65535,,idle> ccb state change from 'idle' to 'proceeding', ccb no:5
[090-14:18:49:710]CC: <Sip-t,2,65535>, <Ss7,65535,65535>, ====Processed: CC_ST_SETUP
[091-14:18:49:710]CCB: no:5, cr1:9, cr2:10, State:4(proceeding), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[092-14:18:49:710]ST: <5,Sip-t,2,65535,987654321,init> <<== CC_ST_PROCEEDING, Std Sdp:, Priv Sdp:
[093-14:18:49:710]ST: <Sip-t,2,65535,987654321> ====Processed: CC_ST_PROCEEDING
[094-14:18:49:710]ST: cr, no:9, ccb:5, State:4(out_proc), cause:0(CCS_NONE(无原因值)), redirect:0

[095-14:18:49:710]ST: <5,,65535,65535,,idle> <<== CC_ST_SETUP, calling:987654321, long:987654321, dial:1234567, send_ok:1, Std Sdp:v=0
o=- 12949395404797000 1 IN IP4 172.16.100.172
s=CounterPath X-Lite 4.0
c=IN IP4 172.16.100.172
t=0 0
a=ice-ufrag:2c37f5
a=ice-pwd:a0b4dc8cc787732f66e9e625e69dbbfd
m=audio 50832 RTP/AVP 107 0 8 101
a=rtpmap:107 BV32/16000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=candidate:1 1 UDP 659136 172.16.100.172 50832 typ host
a=candidate:
[096-14:18:49:710]ST: <5,Ss7,65535,65535,,idle> **==>> CC_SETUP_REQ, index:10, if:65535, trunkGrp:0, calling:987654321, called:1234567, callingTyp:0, calledTye:0, presentId:0, trans:**

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

[097-14:18:49:710]ST: <Ss7,65535,65535,> ====Processed: CC_ST_SETUP
[098-14:18:49:710]ST: cr, no:10, ccb:5, State:6(present), cause:0(CCS_NONE(无原因值)), redirect:0

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

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

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

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

[102-14:18:49:760]ST: <5,Ss7,65535,65535,,present> ==>> CC_ST_SETUP_ACK, cause:0(CCS_NONE(无原因值))
[103-14:18:49:760]ST: <5,Ss7,65535,65535,,in_proc> Tm alloc, e1:10, ts:5
[104-14:18:49:760]ST: <5,Ss7,65535,65535,,in_proc> Tm crcx, connid:196758, ip:172.16.100.172, port:50832, algo:0, pkt:20, zip:0, ZipEia:65535, crypt:0, tcp:0, p2pV2:0, telEventPayload:101, dtmfMode:0.
[105-14:18:49:760]ST: <5,Ss7,65535,65535,,in_proc> play ringBack to IP.
[106-14:18:49:760]ST: <5,Ss7,65535,65535,,in_proc> ==>> CC_ST_ALERTING, ccb:5, user type:0(Norm), calling:987654321, longnum:987654321, std sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.10
s=-
c=IN IP4 172.16.51.10
t=0 0
m=audio 5102 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
a=ptime:20
, priv sdp:
[107-14:18:49:760]ST: <Ss7,65535,65535,> ====Processed: CC_ALERTING_IND
[108-14:18:49:760]ST: cr, no:10, ccb:5, State:8(recving), cause:0(CCS_NONE(无原因值)), redirect:0

[109-14:18:49:770]CC: <5,Ss7,65535,65535,,proceeding> <<== CC_ST_SETUP_ACK, cause:0(CCS_NONE(无原因值)), longnum:
[110-14:18:49:770]CC: <5,Sip-t,2,65535,,proceeding> ccb state change from 'proceeding' to 'wait ack', ccb no:5
[111-14:18:49:770]CC: <Sip-t,2,65535>, <Ss7,65535,65535>, ====Processed: CC_ST_SETUP_ACK
[112-14:18:49:770]CCB: no:5, cr1:9, cr2:10, State:5(wait ack), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[113-14:18:49:770]CC: <5,Ss7,65535,65535,,wait ack> <<== CC_ST_ALERTING, std sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.10
s=-
c=IN IP4 172.16.51.10

```
t=0 0
m=audio 5102 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
a=ptime:20
, priv sdp:
[114-14:18:49:770]CC: <5,Sip-t,2,65535,,wait ack> ccb state change from 'wait ack' to 'alerting',
ccb no:5
[115-14:18:49:770]CC: <5,Sip-t,2,65535,,alerting> route type:3(rts_out), called term type:8(Ss7)
[116-14:18:49:770]CC: <5,Ss7,65535,65535,,alerting> ==>> CC_ST_ALERTING, std sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.10
s=-
c=IN IP4 172.16.51.10
t=0 0
m=audio 5102 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
a=ptime:20
, priv sdp:
[117-14:18:49:770]CC: <Sip-t,2,65535>, <Ss7,65535,65535>, ====Processed: CC_ST_ALERTING
[118-14:18:49:770]CCB: no:5, cr1:9, cr2:10, State:6(alerting), SubState:0(idle), serv:0(normal),
serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[119-14:18:49:770]ST: <5,Sip-t,2,65535,987654321,out_proc> <<== CC_ST_ALERTING, Std
Sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.10
s=-
c=IN IP4 172.16.51.10
t=0 0
m=audio 5102 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
a=ptime:20
, Priv Sdp:
[120-14:18:49:770]ST: <5,Sip-t,2,65535,987654321,out_proc> ==>> ST_SIP_CALL_PRE_ACCEPT,
index:9, local:1234567@172.16.51.10, peer:987654321@172.16.51.10, std sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.10
s=-
c=IN IP4 172.16.51.10
t=0 0
m=audio 5102 RTP/AVP 0 101
```

a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
a=ptime:20
, priv sdp:
[121-14:18:49:770]ST: <Sip-t,2,65535,987654321> ====Processed: CC_ST_ALERTING
[122-14:18:49:770]ST: cr, no:9, ccb:5, State:5(deliver), cause:0(CCS_NONE(无原因值)), redirect:0

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

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

[124-14:18:52:470]ST: <5,Ss7,65535,65535,,recving> connId:0x30096, isPlayLocalRingback2IP:1.
[125-14:18:52:470]ST: <5,Ss7,65535,65535,,recving> ==> CC_ST_CONNECT,
[126-14:18:52:470]ST: <Ss7,65535,65535,> ====Processed: CC_SETUP_CFM
[127-14:18:52:470]ST: cr, no:10, ccb:5, State:9(active), cause:0(CCS_NONE(无原因值)), redirect:0

[128-14:18:52:470]CC: <5,Ss7,65535,65535,,alerting> <<== CC_ST_CONNECT, calling:987654321, long:987654321, called:1234567, calling dial num:1234567, Std Sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.10
s=-
c=IN IP4 172.16.51.10
t=0 0
m=audio 5102 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
a=ptime:20
, Priv Sdp:
[129-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> stop queue seat timer!
[130-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> stop queue timer!
[131-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> clear bill end time(cc connect).
[132-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> stop hint at port:65535 ,connid:4294967295
[133-14:18:52:470]CC: <5,Ss7,65535,65535,,alerting> stop hint at port:65535, connid:4294967295
[134-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> vpbx process flag:0, ippbx process flag:0
[135-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> route type:3(rts_out), called term type:8(Ss7)
[136-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> get bill start time:14-18-52
[137-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> payer info(DevNo-2, PortNo-65535, callDirect-1, termType-Sip-t), Is ccb stpayer.pstPort NULL:yes. Service type(ccb):normal, is need settle:no.

[138-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> ==>> CC_ST_CONNECT, called:1234567
[139-14:18:52:470]CC: <5,Sip-t,2,65535,,alerting> ccb state change from 'alerting' to 'active', ccb no:5
[140-14:18:52:470]CC: <Sip-t,2,65535>, <Ss7,65535,65535>, ====Processed: CC_ST_CONNECT
[141-14:18:52:470]CCB: no:5, cr1:9, cr2:10, State:7(active), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[142-14:18:52:470]ST: <5,Sip-t,2,65535,987654321,deliver> <<== CC_ST_CONNECT, Std Sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.10
s=-
c=IN IP4 172.16.51.10
t=0 0
m=audio 5102 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
a=ptime:20
, Priv Sdp:
[143-14:18:52:470]ST: <5,Sip-t,2,65535,987654321,deliver> ==>> SIP_CALL_ACCEPT, index:9, calltype:0 local:1234567@172.16.51.10, peer:987654321@172.16.51.10, std sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.10
s=-
c=IN IP4 172.16.51.10
t=0 0
m=audio 5102 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
a=ptime:20
, priv sdp:, ext:
[144-14:18:52:470]ST: <5,Sip-t,2,65535,987654321,active> start wait peer conn timer, len:15s
[145-14:18:52:470]ST: <Sip-t,2,65535,987654321> ====Processed: CC_ST_CONNECT
[146-14:18:52:470]ST: cr, no:9, ccb:5, State:9(active), cause:0(CCS_NONE(无原因值)), redirect:0

[147-14:18:52:510]ST: <5,Sip-t,2,65535,987654321,active> <<== SIP_ACCEPT_ACK, Index:9, Local:1234567@172.16.51.10, Peer:987654321@172.16.51.10
[148-14:18:52:510]ST: <5,Sip-t,2,65535,987654321,active> ==> CC_ST_CONNECT_ACK
[149-14:18:52:510]ST: <5,Sip-t,2,65535,987654321,active> stop wait peer conn timer
[150-14:18:52:510]CC: <5,Sip-t,2,65535,,active> <<== CC_ST_CONNECT_ACK
[151-14:18:52:510]CC: <Sip-t,2,65535>, <Ss7,65535,65535>, ====Processed: CC_ST_CONNECT_ACK
[152-14:18:52:510]CCB: no:5, cr1:9, cr2:10, State:7(active), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

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

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

[154-14:19:20:680]ST: <5,Ss7,65535,65535,,disconn> Tm dlcx, connid:196758
[155-14:19:20:680]ST: <5,Ss7,65535,65535,,disconn> Release the call, cause:CCS_NORM_CLEAR(正常释放)(1) !
[156-14:19:20:680]ST: <5,Ss7,65535,65535,,disconn> ==>> CC_RELEASE_REQ, index:10, if:2, q931_id:773, cause:16
[157-14:19:20:680]ST: <Ss7,65535,65535,> ====Processed: CC_DISCONNECT_IND
[158-14:19:20:680]ST: cr, no:10, ccb:5, State:11(release), cause:1(CCS_NORM_CLEAR(正常释放)), redirect:0

[159-14:19:20:690]ST: <5,Ss7,65535,65535,,release> <<== CC_RELEASE_CFM, q931id:773, if:2, calling:, called: org_called:, e1:10, ts:5, callingTyp:0, calledTyp:0, presentationInd:0, send_ok:0, cause:16(正常的呼叫清除)
[160-14:19:20:690]ST: <5,Ss7,65535,65535,,release> Release the call, cause:CCS_NORM_CLEAR(正常释放)(1) !
[161-14:19:20:690]ST: <5,Ss7,65535,65535,,release> ==> CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放))
[162-14:19:20:690]ST: <5,Ss7,65535,65535,,release> Free CR 10, cause:1(CCS_NORM_CLEAR(正常释放))
[163-14:19:20:690]ST: <,65535,65535,> ====Processed: CC_RELEASE_CFM
[164-14:19:20:690]ST: cr, no:10, ccb:4294967295, State:0(idle), cause:0(CCS_NONE(无原因值)), redirect:0

[165-14:19:20:690]CC: <5,Sip-t,2,65535,,active> [cc release comp]ccb no:5, sub ccb no:4294967295
[166-14:19:20:690]CC: <-1,Ss7,65535,65535,,idle> <<== CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放)))
[167-14:19:20:690]CC: <5,Sip-t,2,65535,,active> State(active) is not match, refuse resel route!
[168-14:19:20:690]CC: <5,Sip-t,2,65535,,active> bill start time:14-18-52, bill end time: 0- 0- 0.
[169-14:19:20:690]CC: <5,Sip-t,2,65535,,active> [bill end time]bill type:normal, service type(ccb):normal, is need settle:no.redirect flag:0, called term type:Ss7, Is ccb stpayer.pstPort NULL:yes.
[170-14:19:20:690]CC: <5,Sip-t,2,65535,,active> ==>> CC_ST_RELEASE, cause:1(CCS_NORM_CLEAR(正常释放))
[171-14:19:20:690]CC: <5,Sip-t,2,65535,,active> ccb state change from 'active' to 'release', ccb no:5
[172-14:19:20:690]CC: <Sip-t,2,65535>, <Ss7,65535,65535>, ====Processed: CC_ST_REL_COMP

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

[174-14:19:20:690]ST: <5,Sip-t,2,65535,987654321,active> <<== CC_ST_RELEASE, cause:CCS_NORM_CLEAR(正常释放)
[175-14:19:20:690]ST: <5,Sip-t,2,65535,987654321,active> ==>> SIP_CALL_BYE, index:9, local:1234567@172.16.51.10, peer:987654321@172.16.51.10, cause:CCS_NORM_CLEAR(正常释放)
[176-14:19:20:690]ST: <5,Sip-t,2,65535,987654321,active> ==> CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放))
[177-14:19:20:690]ST: <5,Sip-t,2,65535,987654321,active> Free CR 9, cause:1(CCS_NORM_CLEAR(正常释放))
[178-14:19:20:690]CC: <-1,Sip-t,2,65535,,idle> [cc release comp]ccb no:5, sub ccb no:4294967295
[179-14:19:20:690]CC: <-1,Sip-t,2,65535,,idle> stop queue seat timer!
[180-14:19:20:690]CC: <-1,Sip-t,2,65535,,idle> stop queue timer!
[181-14:19:20:690]**CC: <-1,Sip-t,2,65535,,idle> <<== CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放)))**

(**note:** release complete)

[182-14:19:20:690]CC: <-1,Sip-t,2,65535,,idle> Free CCB 5, cause:1(CCS_NORM_CLEAR(正常释放))
[183-14:19:20:690]CC: <-1,Sip-t,2,65535,,idle> stop queue seat timer!
[184-14:19:20:690]CC: <-1,Sip-t,2,65535,,idle> stop queue timer!

Example 2: One succ call from PSTN to IP:

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

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

[032-00:14:01:640]ST: <-1,Ss7,65535,65535,,idle> Can't recognize calling :987654321, with format locolwihtarea:0, longwith0:1
[033-00:14:01:640]ST: <-1,Ss7,2,65535,00000000,idle> Tm alloc succ, e1:10, ts:11, conn id:196782, port:5120
[034-00:14:01:640]ST: <-1,Ss7,2,65535,00000000,idle> @@@ add called:1234567, lines:1
[035-00:14:01:640]ST: <11,Ss7,2,65535,00000000,idle> ==>> CC_ST_SETUP, ccb:11, user type:0(Norm), calling:987654321, longnum:987654321, trunkGrpId:2, profileId:0, std sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.15

s=-
c=IN IP4 172.16.51.15
t=0 0
m=audio 5120 RTP/AVP 4 18 8 0 101
a=rtpmap:4 G723/8000
a=rtpmap:18 G729/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
, priv sdp:a=X-ACrypt
a=X-Tcp
a=X-P2PV2
a=X-P2PDst:67241984.104333337
[036-00:14:01:640]ST: <Ss7,2,65535,00000000> ====Processed: CC_SETUP_IND
[037-00:14:01:640]ST: cr, no:21, ccb:11, State:1(init), cause:0(CCS_NONE(无原因值)), redirect:0

[038-00:14:01:640]CC: <11,Ss7,2,65535,,idle> <<== CC_ST_SETUP, cr:21, calling:987654321, longNum:987654321, dial:1234567, num_ok:1,calltype:7(msg), rtsType:0, callType:7(ccb), fax dest<65535,65535>, trunkGrpId:2, profileId:0, sigToneTyp:0, std sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.15
s=-
c=IN IP4 172.16.51.15
t=0 0
m=audio 5120 RTP/AVP 4 18 8 0 101
a=rtpmap:4 G723/8000
a=rtpmap:18 G729/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0
[039-00:14:01:640]CC: <11,Ss7,2,65535,,idle> predispose start calling :987654321 called:1234567!
[040-00:14:01:640]CC: <11,Ss7,2,65535,,idle> predispose end calling :987654321 called:1234567!
[041-00:14:01:640]CC: <11,Ss7,2,65535,,idle> Invoke cc_pstn_in_proc()!
[042-00:14:01:640]CC: <11,Ss7,2,65535,,idle> PSTN in call process start! called:1234567, pstnInUserGrp:65535, numRecvIsComp:1, isSbnFlow:1.
[043-00:14:01:640]CC: <11,Ss7,2,65535,,idle> search destination port by long number fail!called:1234567, firstCalled:1234567.
[044-00:14:01:640]**CC: <11,Ss7,2,65535,,idle> [before manipulate number]calling:987654321, called:1234567, longNum:987654321, anl called:1234567, isRouteAfNumManip:0, callerNumTyp:0, calledNumTyp:0, presentId:0.**

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

[046-00:14:01:640]CC: <11,Ss7,2,65535,,idle> [tel2IP]match route succ! ipGrpId:3, trkSelMode:0.
[047-00:14:01:640]CC: <11,Ss7,2,65535,,idle> select ip trunk succ! trunkGrpId:3, trunkType:4(Sip trunk), trunkNo:4, trunkPriority:0, calledProfId:0.

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

[048-00:14:01:640]CC: <11,Ss7,2,65535,,idle> analysis successfully, service:0(normal), bill:4(normal), route:3(rts_out), dest_term:4(Sip-t), dest_dev:4, dest_port:65535, dest_grp:65535, called:1234567 !
[049-00:14:01:640]CC: <11,Ss7,2,65535,,idle> ==>> CC_ST_PROCEEDING, called:1234567
[050-00:14:01:640]CC: <11,Ss7,2,65535,,idle> calling :0.0.0.0 called:255.255.255.255
[051-00:14:01:640]CC: <11,Ss7,2,65535,,idle> is need reflect:0, callingProfId:0.
[052-00:14:01:640]CC: <11,Sip-t,4,65535,,idle> ==>> CC_ST_SETUP, cr:22, calling:987654321, longNum:987654321, dial:1234567, OrgCallee:, num_ok:1, trunkGrpId:3, profileId:0, isForceReflect(ccb):0, ringback2IP:0,std sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.15
s=-
c=IN IP4 172.16.51.15
t=0 0
m=audio 5120 RTP/AVP 4 18 8 0 101
a=rtpmap:4 G723/8000
a=rtpmap:18 G729/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
, priv sdp:
[053-00:14:01:640]CC: <11,Ss7,2,65535,,idle> ccb state change from 'idle' to 'proceeding', ccb no:11
[054-00:14:01:640]ST: <11,Ss7,2,65535,00000000,init> <<== CC_ST_PROCEEDING, calling:, long:, dial:1234567, send_ok:1, Std Sdp:, Priv Sdp:, cause:0(CCS_NONE(无原因值))
[055-00:14:01:640]ST: <11,Ss7,2,65535,00000000,init> ==>> CC_PROCEEDING_REQ, index:21, if:2, q931_id:779
[056-00:14:01:640]ST: <11,Ss7,2,65535,00000000,init> [custom ringback] call type:2, called:1234567, call forward flag:0, vpbx flag:0
[057-00:14:01:640]ST: <Ss7,2,65535,00000000> ====Processed: CC_ST_PROCEEDING

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

[059-00:14:01:640]ST: <11,Sip-t,4,65535,,idle> <<== CC_ST_SETUP, presentId:0, Std Sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.15
s=-
c=IN IP4 172.16.51.15
t=0 0
m=audio 5120 RTP/AVP 4 18 8 0 101
a=rtpmap:4 G723/8000
a=rtpmap:18 G729/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
, Priv Sdp:
[060-00:14:01:640]ST: <11,Sip-t,4,65535,00000000,present> ==>> CC_ST_SETUP_ACK, cause:0(CCS_NONE(无原因值))
[061-00:14:01:640]**ST: <11,Sip-t,4,65535,00000000,present> ==>> SIP_CALL_INVITE, index:22, local:sip:987654321@172.16.51.15, peer:sip:1234567@172.16.50.170 (ip:172.16.50.170, port:5060), std sdp:v=0**

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

o=call 10000 20000 IN IP4 172.16.51.15
s=-
c=IN IP4 172.16.51.15
t=0 0
m=audio 5120 RTP/AVP 4 18 8 0 101
a=rtpmap:4 G723/8000
a=rtpmap:18 G729/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
, priv sdp:, ext:
[062-00:14:01:640]ST: <Sip-t,4,65535,00000000> ====Processed: CC_ST_SETUP
[063-00:14:01:640]ST: cr, no:22, ccb:11, State:7(in_proc), cause:0(CCS_NONE(无原因值)), redirect:0

[064-00:14:01:640]CC: <11,Sip-t,4,65535,,proceeding> <<== CC_ST_SETUP_ACK, cause:0(CCS_NONE(无原因值)), longnum:
[065-00:14:01:640]CC: <11,Ss7,2,65535,,proceeding> ccb state change from 'proceeding' to 'wait ack', ccb no:11

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[066-00:14:01:640]CC: <Ss7,2,65535>, <Sip-t,4,65535>, ====Processed: CC_ST_SETUP_ACK
[067-00:14:01:640]CCB: no:11, cr1:21, cr2:22, State:5(wait ack), SubState:0(idle), serv:0(normal),
serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[068-00:14:01:680]ST: <11,Sip-t,4,65535,00000000,in_proc> <<== SIP_CALL_RING,
Local:1234567@172.16.50.170, Peer:987654321@172.16.51.15, Std Sdp:, Priv Sdp:, Ext:
[069-00:14:01:680]ST: <11,Sip-t,4,65535,00000000,in_proc> ==>> CC_ST_ALERTING, ccb:11, user
type:0(Norm), calling:987654321, longnum:987654321, std sdp:, priv sdp:
[070-00:14:01:680]ST: <Sip-t,4,65535,00000000> ====Processed: SIP_CALL_RING
[071-00:14:01:680]ST: cr, no:22, ccb:11, State:8(recving), cause:0(CCS_NONE(无原因值)),
redirect:0

[072-00:14:01:680]CC: <11,Sip-t,4,65535,,wait ack> <<== CC_ST_ALERTING, std sdp:, priv sdp:
[073-00:14:01:680]CC: <11,Ss7,2,65535,,wait ack> ccb state change from 'wait ack' to 'alerting',
ccb no:11
[074-00:14:01:680]CC: <11,Ss7,2,65535,,alerting> route type:3(rts_out), called term type:4(Sip-t)
[075-00:14:01:680]CC: <11,Sip-t,4,65535,,alerting> ==>> CC_ST_ALERTING, std sdp:, priv sdp:
[076-00:14:01:680]CC: <Ss7,2,65535>, <Sip-t,4,65535>, ====Processed: CC_ST_ALERTING
[077-00:14:01:680]CCB: no:11, cr1:21, cr2:22, State:6(alerting), SubState:0(idle), serv:0(normal),
serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[078-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> <<== CC_ST_ALERTING, calling:,
long:, dial:, send_ok:1, Std Sdp:, Priv Sdp:, cause:0(CCS_NONE(无原因值))
[079-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> Tm crcx, connid:196782,
ip:172.16.51.15, port:5121, algo:4, pkt:30, zip:0, ZipEia:65535, crypt:0, tcp:0, p2pV2:0
[080-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> called dev no:4, called term type:4,
called profile id:0, call type:2.
[081-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> [calling] std sdp:v=0
o=call 10000 20000 IN IP4 172.16.51.15
s=-
c=IN IP4 172.16.51.15
t=0 0
m=audio 5120 RTP/AVP 4 18 8 0 101
a=rtpmap:4 G723/8000
a=rtpmap:18 G729/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
, priv sdp:a=X-ACrypt
a=X-Tcp
a=X-P2PV2
a=X-P2PDst:67241984.1043333379.2886742799.4000.20072.65535.65535
```

[082-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> [called] std sdp:, priv sdp:.
[083-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> get ip profile succ!
[084-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> Is need send local ringback tone to tel:yes, call type:2
[085-00:14:01:680]ST: <11,Ss7,2,65535,00000000,out_proc> ==>> CC_ALERTING_REQ, index:21, if:2, q931_id:779
[086-00:14:01:680]ST: <Ss7,2,65535,00000000> ====Processed: CC_ST_ALERTING
[087-00:14:01:680]ST: cr, no:21, ccb:11, State:5(deliver), cause:0(CCS_NONE(无原因值)), redirect:0

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

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

o=Qtech 8723835 8723836 IN IP4 172.16.50.170
s=-
c=IN IP4 172.16.50.170
t=0 0
m=audio 8000 RTP/AVP 4 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
, Priv Sdp:, Ext:
[089-00:14:02:010]ST: <11,Sip-t,4,65535,00000000,recving> ==> CC_ST_CONNECT,
[090-00:14:02:010]ST: <Sip-t,4,65535,00000000> ====Processed: SIP_CALL_ACCEPT
[091-00:14:02:010]ST: cr, no:22, ccb:11, State:9(active), cause:0(CCS_NONE(无原因值)), redirect:0

[092-00:14:02:010]CC: <11,Sip-t,4,65535,,alerting> <<== CC_ST_CONNECT, calling:987654321, long:987654321, called:1234567, calling dial num:1234567, Std Sdp:v=0
o=Qtech 8723835 8723836 IN IP4 172.16.50.170
s=-
c=IN IP4 172.16.50.170
t=0 0
m=audio 8000 RTP/AVP 4 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
, Priv Sdp:
[093-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> stop queue seat timer!
[094-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> stop queue timer!
[095-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> clear bill end time(cc connect).
[096-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> stop hint at port:65535 ,connid:4294967295
[097-00:14:02:010]CC: <11,Sip-t,4,65535,,alerting> stop hint at port:65535, connid:4294967295

[098-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> vpbx process flag:0, ippbx process flag:0
[099-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> route type:3(rts_out), called term type:4(Sip-t)
[100-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> get bill start time:00-14-02
[101-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> payer info(DevNo-2, PortNo-65535, callDirect-1, termType-Ss7), Is ccb stpayer.pstPort NULL:yes. Service type(ccb):normal, is need settle:no.
[102-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> ==>> CC_ST_CONNECT, called:1234567
[103-00:14:02:010]CC: <11,Ss7,2,65535,,alerting> ccb state change from 'alerting' to 'active', ccb no:11
[104-00:14:02:010]CC: <Ss7,2,65535>, <Sip-t,4,65535>, ====Processed: CC_ST_CONNECT
[105-00:14:02:010]CCB: no:11, cr1:21, cr2:22, State:7(active), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

[106-00:14:02:010]ST: <11,Ss7,2,65535,00000000,deliver> <<== CC_ST_CONNECT, calling:, long:, dial:1234567, send_ok:1, Std Sdp:v=0
o=Qtech 8723835 8723836 IN IP4 172.16.50.170
s=-
c=IN IP4 172.16.50.170
t=0 0
m=audio 8000 RTP/AVP 4 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
, Priv Sdp:, cause:0(CCS_NONE(无原因值))
[107-00:14:02:010]ST: <11,Ss7,2,65535,00000000,deliver> Tm mdcx, connid:196782, ip:172.16.50.170, port:8000, algo:4, pkt:30, zip:0, ZipEia:65535, crypt:0, tcp:0, p2pV2:0, faxMode:0
[108-00:14:02:010]ST: <11,Ss7,2,65535,00000000,deliver> [custom ringback] call type:2, called:1234567, call forward flag:0, vpbx flag:0
[109-00:14:02:010]ST: <11,Ss7,2,65535,00000000,deliver> ==>> CC_SETUP_RSP, index:21, if:2, q931_id:779
[110-00:14:02:010]ST: <Ss7,2,65535,00000000> ====Processed: CC_ST_CONNECT
[111-00:14:02:010]ST: cr, no:21, ccb:11, State:9(active), cause:0(CCS_NONE(无原因值)), redirect:0

[112-00:14:02:010]ST: <11,Ss7,2,65535,00000000,active> <<== CC_SETUP_COMPL_IND, q931id:779, if:2, calling:, called: org_called:, e1:10, ts:11, callingTyp:0, calledTyp:0, presentationInd:0, send_ok:0, cause:0(OK)
[113-00:14:02:010]ST: <11,Ss7,2,65535,00000000,active> <<== CC_SETUP_COMPL_IND
[114-00:14:02:010]ST: <11,Ss7,2,65535,00000000,active> ==> CC_ST_CONNECT_ACK
[115-00:14:02:010]ST: <Ss7,2,65535,00000000> ====Processed: CC_SETUP_COMPL_IND
[116-00:14:02:010]ST: cr, no:21, ccb:11, State:9(active), cause:0(CCS_NONE(无原因值)), redirect:0

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

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

[119-00:14:02:010]CC: <11,Ss7,2,65535,,active> <<== CC_ST_CONNECT_ACK
[120-00:14:02:010]CC: <Ss7,2,65535>, <Sip-t,4,65535>, ====Processed: CC_ST_CONNECT_ACK
[121-00:14:02:010]CCB: no:11, cr1:21, cr2:22, State:7(active), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:0(CCS_NONE(无原因值))

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

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

[123-00:14:04:060]ST: <11,Sip-t,4,65535,00000000,active> ==> CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放))
[124-00:14:04:060]ST: <11,Sip-t,4,65535,00000000,active> Free CR 22, cause:1(CCS_NORM_CLEAR(正常释放))
[125-00:14:04:060]CC: <11,Ss7,2,65535,,active> [cc release comp]ccb no:11, sub ccb no:4294967295
[126-00:14:04:060]CC: <-1,Sip-t,4,65535,,idle> <<== CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放)))
[127-00:14:04:060]CC: <11,Ss7,2,65535,,active> State(active) is not match, refuse resel route!
[128-00:14:04:060]CC: <11,Ss7,2,65535,,active> bill start time: 0-14- 2, bill end time: 0- 0- 0.
[129-00:14:04:060]CC: <11,Ss7,2,65535,,active> [bill end time]bill type:normal, service type(ccb):normal, is need settle:no.redirect flag:0, called term type:Sip-t, Is ccb stpayer.pstPort NULL:yes.
[130-00:14:04:060]CC: <11,Ss7,2,65535,,active> ==>> CC_ST_RELEASE, cause:1(CCS_NORM_CLEAR(正常释放))
[131-00:14:04:060]CC: <11,Ss7,2,65535,,active> ccb state change from 'active' to 'release', ccb no:11
[132-00:14:04:060]CC: <Ss7,2,65535>, <Sip-t,4,65535>, ====Processed: CC_ST_REL_COMP
[133-00:14:04:060]CCB: no:11, cr1:21, cr2:22, State:9(release), SubState:0(idle), serv:0(normal), serv_state:20(), route:3(rts_out), cause:1(CCS_NORM_CLEAR(正常释放))

[134-00:14:04:060]ST: <11,Ss7,2,65535,00000000,active> <<== CC_ST_RELEASE, calling:, long:, dial:, send_ok:1, Std Sdp:, Priv Sdp:, cause:1(CCS_NORM_CLEAR(正常释放))
[135-00:14:04:060]ST: <11,Ss7,2,65535,00000000,active> needPlaySigTone2Tel:0, isReflectRoute:0, cause:1.
[136-00:14:04:060]ST: <11,Ss7,2,65535,00000000,active> Tm dlcx, connid:196782
[137-00:14:04:060]ST: <11,Ss7,2,65535,00000000,active> @@@ free called:1234567, lines:0
[138-00:14:04:060]ST: <11,Ss7,2,65535,00000000,active> Release the call, cause:CCS_NORM_CLEAR(正常释放)(1) !

[139-00:14:04:060]ST: <11,Ss7,2,65535,00000000,active> ==>> CC_DISCONNECT_REQ, index:21, if:2, q931_id:779
[140-00:14:04:060]ST: <Ss7,2,65535,00000000> ====Processed: CC_ST_RELEASE
[141-00:14:04:060]ST: cr, no:21, ccb:11, State:11(release), cause:1(CCS_NORM_CLEAR(正常释放)), redirect:0

[142-00:14:04:070]ST: <11,Ss7,2,65535,00000000,release> <<== CC_RELEASE_IND, q931id:779, if:2, calling:, called: org_called:, e1:10, ts:11, callingTyp:0, calledTyp:0, presentationInd:0, send_ok:0, cause:16(正常的呼叫清除)
[143-00:14:04:070]ST: <11,Ss7,2,65535,00000000,release> Release the call, cause:CCS_NORM_CLEAR(正常释放)(1) !
[144-00:14:04:070]ST: <11,Ss7,2,65535,00000000,release> ==> CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放))
[145-00:14:04:070]ST: <11,Ss7,2,65535,00000000,release> Free CR 21, cause:1(CCS_NORM_CLEAR(正常释放))
[146-00:14:04:070]ST: <,65535,65535,> ====Processed: CC_RELEASE_IND
[147-00:14:04:070]ST: cr, no:21, ccb:4294967295, State:0(idle), cause:0(CCS_NONE(无原因值)), redirect:0

[148-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> [cc release comp]ccb no:11, sub ccb no:4294967295
[149-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> stop queue seat timer!
[150-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> stop queue timer!
[151-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> <<== CC_ST_REL_COMP, cause:1(CCS_NORM_CLEAR(正常释放)))
[152-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> Free CCB 11, cause:1(CCS_NORM_CLEAR(正常释放))
[153-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> stop queue seat timer!
[154-00:14:04:070]CC: <-1,Ss7,2,65535,,idle> stop queue timer!