

MTG1000 Trunk Gateway User Manual V2.0



Dinstar Technologies Co., Ltd.

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1. Product Introduction

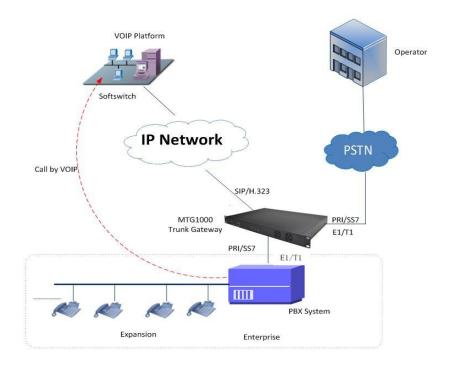
1.1 Overview

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

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

- MTG1000-4E1
- MTG1000-8E1

A typical network diagram shows the function of MTG1000 as below. Figure 1-1-1 Application topology



1.2 Equipment Structure

1.2.1 Rear View



Figure 1-2-1 MTG1000 Rear View

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

1.2.2 Front View



Figure 1-2-2 MTG1000 Front View

LED	Function	Color	Work Status
			Off: Power is off
POWER	Power status indicator	Green	On: Power is on
DUN	De sister in disstant	Creation	Slow blinking: Unregister
RUN	Register indicator	Green	Fast blinking: Register
ALM	The failure of device	Yellow	Off: Normal
ALIVI	indicator	reliow	On: Failed
RST	Reset button, it is used to rest	art the devi	ce
CONSOLE	RS232 console port: it can be	used to deb	ug and configure the device. The baud rate is 115200
CONSOLL	bps.		
			Off: E1/T1 port connection normal
E1/T1	Indicating the connection	Green	On: E1/T1 port connection and sending/ receiving
L-4/11	state of device E1/T1.	Green	message normal
			Flash:E1/T1 port connection failed
	Indicating the connection		Off: Network connection failed
LINK	Indicating the connection state of the network	Green	On: Network connection normal, and 0 indicates FE0
	state of the network		and 1 indicates FE1
SPEED	Indicating the network	Yellow	Off:10Mbps bandwidth
Jr LLD	bandwidth	TEHOW	On:100Mbps bandwidth

Table 1-2-2 MTG1000 Front View Description

1.2.3 RJ-48c Line sequence

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

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

1.3 Functions and Features

1.3.1 Protocol standard supported

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

1.3.2 System Function

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

1.3.3 Industrial standards supported

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

1.3.4 General hardware specification

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

2. Parameter Setting

2.1 Login

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

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

需要进行身份验证		<u> </u>
服务器 172.30.6 器提示:GoAhe	55.25:80 要求用户输入用户名和密码。 ead。	服务
用户名:	admin	
密码:	****	
	登录 取消	ji d

Figure 2-1-1 Login Interfaces

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

Password Modification	
Old Password New Password Confirm Password	
	Save

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.

& Statistics	System Informa	tion			
em Information	General				
1 Status	MAC	ddress	00-02-45-BA-02-01		
ink Status	Servic	e Ethernet Interface(FE1)	172.30.65.10	255.255.0.0	172.30.0.1
tatus	Mana	gement Ethernet Interface(FE0)	0.0.0.0	0.0.00	
Statistics	DNS	Server			
Statistics	Syste	m Time	2012-5-3 17:51:56	5	
Statistics	Syste	m Uptime	7 m 29 s		
Il Statistics	Traffic	Statistics	Received	136.478.333	bytes
			Sent	61,900,087	bytes
	Version				
Config	Devic	e Model	MTG1000		
	Hardv	vare Version	PCB 06		
	DSP	/ersion	5.08.04		
	Web	/ersion	25.03.03.01		
	Softw	are Version	25.03.03.01		
ation	Time	Built	2012-05-03 , 17:23	:16	
			Refresh		

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.

General			
MAC Address	00-02-45-BA-02-01		
Service Ethernet Interface(FE1)	172.30.65.10	255.255.0.0	172.30.0.1
Management Ethernet Interface(FE0)	0.0.0.0	0.0.00	
DNS Server			
System Time	2012-5-3 17:52:25		
System Uptime	7 m 58 s		
Traffic Statistics	Received	146.871.868	bytes
	Sent	68,019,062	bytes
Version			
Device Model	MTG1000		
Hardware Version	PCB 06		
DSP Version	5.08.04		
Web Version	25.03.03.01		
Software Version	25.03.03.01		
Time Built	2012-05-03, 17:23:1	6	

Refresh

Figure 2-2-1 System Information

Users through to traverse the left navigation tree, and can complete view, edit and configuration

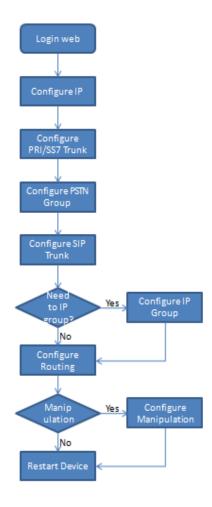
device in the right configuration interface.

Figure 2-2-2 Navigation tree

- Status & Statistics

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

MTG configuration flow chart below:



2.3 Status & Statistics

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

and state system.

Figure 2-3-1 Status & Statistics

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

2.3.1 System Information

System information interface shows the general information and version information.

Figure 2-3-1 System Information

eneral			
MAC Address	00-02-45-BA-02-01		
Service Ethernet Interface(FE1)	172.30.65.10	255.255.0.0	172.30.0.1
Management Ethernet Interface(FE0) DNS Server	0.0.0.0	0.0.0.0	
System Time	2012-5-3 17:53:16		
System Uptime	8 m 49 s		
Traffic Statistics	Received	160.104.055	bytes
	Sent	73,671,353	bytes
ersion			
Device Model	MTG1000		
Hardware Version	PCB 06		
DSP Version	5.08.04		
Web Version	25.03.03.01		
Software Version	25.03.03.01		
Time Built	2012-05-03, 17:23:1	6	

Refresh

Table 2-3-1 System Information

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

2.3.2 E1/T1 Status



Status	Frame-Sync	Idle	Signal	Busy	Fault	Disable	L-blocked	R-blocked	B-blocked	
Color										
Totalize	0	0	0	0	0	128	0	0	0	1

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

Figure 2-3-2 E1/T1 Status

Table 2-3-2 Description of E1/T1 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
E1/T1 Port Status	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.
	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
	ОК.
	3.Signal: Signal channel
F4/T4 Channel Chature	4.Busy: Means this channel is occupied by voice
E1/T1Channel Status	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
0.07111.0.0			
SS7 Link Status			
SS7 Link Status SS7 Trunk No.	Trunk Name	E1/T1 Port No.	Link Status
	Trunk Name	E1/T1 Port No.	Link Status
SS7 Trunk No.			

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 Statu	IS				
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

PRI Trunk Call Statisti	CS			
PRI Trunk No.	Trunk Name	Current Calls	Accumulated Calls	ASR

Release Cause Statist	ics	
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	

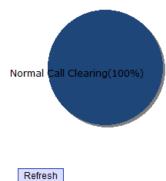


Figure 2-3-5 PRI Call Statistics description

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.			

PRI call statistics description

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 Statisti	ics			
SS7 Trunk No.	Trunk Name	Current Calls	Accumulated Calls	ASR



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

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

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

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

2.4 Network

Network Configuration							
Service Ethernet Interface(FE1)							
IP Address	172.30.65.10						
Subnet Mask	255.255.0.0						
Default Gateway	172.30.0.1						
Management Ethernet Interface(FE0)	Management Ethernet Interface(FE0)						
IP Address 0.0.0.0							
Subnet Mask	0.0.0.0						
DNS Server	DNS Server						
Primary DNS Server							
Secondary DNS Server							



NOTE: The device must restart to take effect.

Figure 2-4-1 Network Configuration

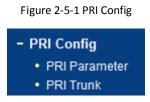
Network Configuration

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

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

2.5 PRI Config

PRI configuration includes PRI parameter and PRI trunk configuration



2.5.1 PRI Parameter

PRI F	Parameter		
0	Calling Party Numbering Plan	ISDN/Telephony numbering plan	-
C	Calling Party Number Type	Unknown	
s	Screening Indicator for Displaying Caller Number	User provide,no shield	•
s	Screening Indicator for No Displaying Caller Number	User provide,no shield	-
0	Called Party Numbering Plan	ISDN/Telephony numbering plan	-
0	Called Party Number Type	Unknown	-
I	nformation Transfer Capability	Speech	-
F	Reset to default configuration	Reset	

Save

PRI parameter description

Calling Party	Provide six plans: Unknown, ISDN/Telephony numbering plan, data numbering plan,			
Numbering Plan	telegraph numbering plan, national standard numbering plan, private numbering plan.			
Numbering Flan	The default is ISDN/Telephony numbering plan.			
Calling Dorty Number	Six optional types are provided for calling party: Unknown, International number,			
Calling Party Number	National number, Network special number, User number, Short code dialing. The			
Туре	default option is Unknown.			
Screening Indicator for	Four options available: User provider, no shield; User provide, check and send; User			
Displaying Caller	provide, check and having failure; Network provide. The default option is: User			
Number	provider, no shield.			
Screening Indicator for	Four options available: User provider, no shield; User provide, check and send; User			
No Displaying Caller	provide, check and having failure; Network provide. The default option is: User			
Number	provider, no shield.			
Called Party Numbering	Provide six plans: Unknown, ISDN/Telephony numbering plan, data numbering plan,			
Called Party Numbering	telegraph numbering plan, national standard numbering plan, private numbering plan.			
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	Support speech and 3.1khz audio. The default option is speech.		
Capability	Support speech and 3.1khz audio. The default option is speech.		

2.5.2 PRI Trunk

Figure 2-5-3 PRI Trunk

PRI T	runk							
	Trunk No.	Trunk Name	Channel ID	D-Channel	E1/T1 Port No.	Protocol	Switch Side	Alerting Indication
Add Delete Modify								

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

Figure 2-5-4 Add PRI Trunk

Trunk No.	3
Trunk Name	
Channel ID	
D-Channel	Enable
E1/T1 Port No.	3
Protocol	ISDN 💌
Switch Side	User Side 💌
Alerting Indication	ALERTING

PRI trunk description

	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				
Trunk No	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				
FIOLOCOI	ISDN.				

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

2.6 SS7 Config

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

Figure 2-6-1 Add PRI Trunk
- SS7 Config
SS7 Trunk
SS7 MTP Link
SS7 CIC
 SS7 CIC Maintain
 Slave TG Management

2.6.1 SS7 Trunk

Figure 2-6-2 SS7 Trunk

SS7	Trunk								
	Trunk No.	Trunk Name	Protocol	Protocol Type	SPC Format	OPC	DPC	Network Indicator	Sending SLTM
	Add Delete Modify								

Figure 2-6-3 SS7 Trunk Add

Select Trunk No.	3	•
runk Name		
Protocol	ITU	•
Protocol Type	ISUP	-
PC Format	Hex	•
PC		
PC		
letwork Indicator	National Network	•
Sending SLTM	Enable	•
	Linable	•

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).

The	The number of SS7 trunk. Generally, a DPC will establish a SS7 trunk number			
No res	respectively, SS7 trunk number cannot be conflict with PRI trunk number. After			
SS7	SS7 trunk is established, assign E1/T1 to SS7 trunk in "SS7 Circuit" option.			
Na	ne of trunk, it can be edited to any name user want.			
SPO	types: ITU-T (14 bit), ANSI (24 bit), ITU-CHINA (24 bit)			
e Sup	ported two protocol types: ISUP and TUP			
Sig	naling Point Code format includes hexadecimal system and ITU pointcode			
str	icture (decimal system)			
Ori	ginal Point Code			
De	tination Point Code			
SS7	service types: ISUP (ISDN User Part) and TUP (Telephone User Part).			
Ind	cate the network property of SS7, including International Network,			
Int	International Spare, National Network, National Spare; the default is "National			
icator Ne	Network" (this type is used in China, USA, and Japan), "International Network"			
is ę	enerally used in inter-office switch room; others will be selected according			
to	hysical circumstances.			
e Sup Sig stri Ori Des SS7 Ind Inte icator Ne is g	ported two protocol types: ISUP and TUP naling Point Code format includes hexadecimal system and ITU point acture (decimal system) ginal Point Code tination Point Code service types: ISUP (ISDN User Part) and TUP (Telephone User Part). icate the network property of SS7, including International Network ernational Spare, National Network, National Spare; the default is "National work" (this type is used in China, USA, and Japan), "International Network enerally used in inter-office switch room; others will be selected acco			

SS7 trunk add

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; $x_y x_z$ 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_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

Trunk No.	Link No.	Signaling Link Code	E1/T1 Port No.	Channel No.
		(111)		

Figure 2-6-5 SS7 MTP Link Add

SS7 MTP Link Add			
Trunk No.			•
Link No.		0	•
Signaling Link Code			
E1/T1 Port No.		0	-
Channel No.		16	
	OK Re	Cancel	

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

SS7 MTP link description

Trunk No	It is consistent with foregoing "Trunk No" of SS7 trunk.
	Equipment maximum support 2 signaling links, these two links share workload, when
Link No	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.
	Indicate which E1/T1 this link is established on, it is stipulated that such numbering is
E1/T1 Port No	carried out according to the physical position of E1/T1.
	Indicate time slot that link is established on. It is assigned to 1 or 16 for time slot, the
Channel No	default is 16 time slot.

2.6.3 SS7 Circuit

Figure 2-6-5 SS7 Circuit

SS7 Circuit					
	Trunk No.	E1/T1 Port No.	Start Channel	Start CIC No.	Count
		Add	Delete Modify	/	

	Add		Delete		Modify	
--	-----	--	--------	--	--------	--

Figure 2-6-6 SS7 Circuit description

SS7 Circuit Add			
			_
Trunk No.			•
E1/T1 port No.		0	• 1
Start Channel			
Start CIC No.			
Count			
	OK R	eset Cancel	

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

2. When option 'ANSI' has been selected in 'Protocol' of sub-menu SS7 Trunk, the 'Start CIC No.' must be less than 16384.

CIC (circuit identification code) is an important parameter of SS7 circuit. It should be confirmed

with service provider. If the CIC is mismatched, it will result in one-way voice communication.

SS7 Circuit Add

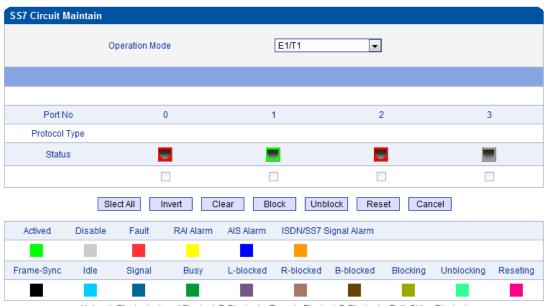
Trunk No	The "Trunk No." here corresponds to the "Trunk No." of SS7 trunk.
E1/T1 port No	Fill in the port number of E1/T1. Assign E1/T1 to selected SS7 trunk.
Start Channel	The start of SS7 channel trunk

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

2.6.4 SS7 Circuit Maintain

According to the different operating modes, 7 circuit maintenance objects into two categories:

ports and channel.



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

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

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.

SS7 Circui	t Maiı	ntain														
Operation Mode							(Char	nnel		•					
Current	Port			•		Statu	IS				Pro	tocol	Туре	und	lefine	d
L																
Channel	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
CIC No.																
Status																
Channel	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
CIC No.																
Slect Invert Clear Block Unblo Reset Cance																
Actived	Dis	F	ault	RAI	A	AIS A	Ы I	ISDN	/SS7	Sig						
Frame	Idle	Si	gnal	Bu	isy	L-blo)	R-blo) E	B-blo.	В	lock	. Un	bl	Res	se

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

Slave TG						
		Local TG Fla	Maste	r	•	
	Trunk No.	Decribes	IP Auur	ETINUM	Start No.	Status
	0	65.27	172.30.65.27			Available
		Add	Delete	Modify		

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.

E1/T1 Pa	rameter					
		E1/T1 (Clock Source	Remote		
	Port No.	Work Mode	PCM Mode	Frame Mode	Line Code	Line Built Out
	0	E1	A LAW	DF	HDB3	Short Haul,(-10DB)
	1	E1	A LAW	CRC-4	HDB3	Short Haul,(-10DB)
	2	E1	A LAW	CRC-4	HDB3	Short Haul,(-10DB)
	3	E1	A LAW	CRC-4	HDB3	Short Haul,(-10DB)
	3	E1	A LAW	CRC-4	HDB3	Short Haul,(-10

Modify

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

2.7.2 Coder Group

		Co	der Group ID	0(default	setting)	•		
	Coder		Payload Type Value	Packetizati (ms		Rate (kbps)	Silence Supp	ression
1st	G711A	-	8	20	-	64	Disable	-
2nd	G711U	-	0	20	-	64	Disable	-
3rd	G729	-	18	20	-	8	Disable	-
4th	G723	-	4	30	-	6.3	Disable	-
5th		-			-			-
6th		-			-			-



Figure 2-7-2 Coder Group

Coder group description

	ID standard for Voice ability, total with 8 groups, where 0 is the default group ID
Coder Group ID	number, the codec that equipment supports in the grouping will be displayed in O
	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.
Ciloneo Cupprossion	It is disabled by default. During talking, the bandwidth occupied by voice transmission
Silence Suppression	will be released automatically for silence party or when talk is paused.
	ID standard for Voice ability, total with 8 groups, where 0 is the default group ID
Coder Group ID	number, the codec that equipment supports in the grouping will be displayed in 0
	group. Default value cannot be modified.

2.7.3 Dial Plan

Dial Plan				
		Dial Plan ID 0	•	
	Index	Prefix	Min Length	Max Length
	0		0	30
				Total: 1 Page 1 💌
		Add Delete	Modify	

Figure 2-7-3 Dial Plan

Dial plan used for configuring the receiving number, user can configure different prefix number,

these rules can be divided into 5 groups with a dial plan ID, where 0 is the default setting.

Notes:

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

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

Dial Plan Add			
D:-1 D: 1D			
Dial Plan ID		1	
Index		1999	
Prefix			
Min Length			
Max Length			
	OK Re	cancel	

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

Dial Plan ID	The number to identify a dial plan
laday	Dial plan priority rules take effect in accordance with dial plan index size, and not
Index	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)

Figure 2-7-4 Dial Plan Add

special version:

Dial Plan description

1. Dial plan can be backup and restore in management configuration.

2. "Min Length" and "Max Length" are equal to the total number of possible length minus the prefix length.

3. When overlap dialing, called number length sure, and then the "Min Length" and "Max Length" will

be set to the same value to accelerate connection rate.

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

2.7.4 Dial Timeout

Dial Tir	neout				
	Dial Timeout ID	Description	Max Time for Collecting Prefix(s)	Time to Reach Min Length (s)	Time to Reach Max Length (s)
	0	Default	20	10	10
					Total: 1 Page 1 💌
		[Add Delete	Modify	

Figure 2-7-5 Dial Timeout

Dial Timeout Add	
Dial Timeout ID	1
Description	
Max Time for Collecting Prefix	s
Time to Reach Min Length(after Prefix)	s
Time to Reach Max Length(after Min Length)	s
OK R	Reset Cancel

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

Figure 2-7-6 Dial Timeout Add

Dial timeout description

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

2.7.5 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	0	101	RFC2	SIP IN	Inband	Enable	0	0 <default></default>	Not remove	No
										т	otal: 1 Page 1 🗖

Figure 2-7-7 PSTN Profile

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

OK

STN Profile ID	1	-
	1	
cription		
der Group ID	0	
FC2833 Payload Type	101	
TMF Tx Priority 1st	RFC2833	-
TMF Tx Priority 2nd	SIP INFO	-
TMF Tx Priority 3rd	Inband	-
verlap Receiving	Disable	-
emove CLI	Not remove	-
lay Busy Tone to PSTN	No	-

Reset

Cancel

Figure 2-7-8 PSTN Profile Add

·	
PSTN Profile ID	The number to the PSTN Profile
Description	Description of the PSTN Profile
Code Group ID	Refer to "Coder Group"
RFC2833 Payload Type	The item is 101 by default.
1 st /2 nd /3 rd 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

PSTN profile add description

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.

PSTN Group			
	Group ID	Name	Channel Selection
	0	r2-0	Cyclic Ascending
	1	r2-12	Cyclic Ascending
			Total: 2 Page 1 💌
		Add Delete Modify	

Figure 2-7-9 PSTN Group

STN Group Add	
Trunk Group ID Name	2
Channel Selection	Cyclic Ascending
	OK Res Cyclic Descending Descending

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

PSTN Group Management								
	Group ID	Start E1/T1	End E1/T1	Start Channel	End Channel	PSTN Profile ID		
	0 <r2-0></r2-0>	0	0	1	31	0 <default></default>		
	0 <r2-0></r2-0>	1	2			0 <default></default>		

Total: 2 Page 1 💌

Delete Figure 2-7-11 PSTN Group Management

Modify

Add

Group ID	0 <r2-0></r2-0>	•
Start E1	0	-
End E1	0	-
Start Channel	1	-
End Channel	31	-
PSTN Profile ID	0 <default></default>	-



Figure 2-7-12 PSTN Group Management Add

PSTN group management add

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

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

2.8 SIP Config

2.8.1 SIP Parameter

Local SIP Port	5060	
Local Domain		

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

SIP Tr				Romoto	Local			Pegister to	Outgoing	Incoming	Detect Trunk	Enable Of
Tru No		Trunk Name	Remote Address	Remote Port	Local Domain	Support SIP-T	Get Callee from	Register to Remote	Outgoing Call Mode	Incoming Authentication Type	Status	Trunk
] (172.30.66.16	172.30.66.16	5060	Disable	Disable	Request-line	No	Peer	IP Address	Yes	Yes
											Total:	1 Page 1
						Add	Delete	Modify				
						Figure	2-8-2 SIP T	runk				
S	P	Trunk A	dd									
		Trunk No					[1		•		
		Trunk Na	me				Ì					
		Remote A	\ddress				Ì					
	Remote Port						Ì	5060				
		Outbound	l Proxy									
		Outbound	Porxy Port					5060				
		Local Dor	main				[Disable		.		
		Support S	SIP-T				[Disable		-		
		Get Calle	e from				[Request-li	ne	-		
		Register t	to Remote				[No		-		
		Incoming	SIP Authen	tication	Туре		[IP Address	5	-		
		IP to PST	N Calls Res	triction			[No		-		
		PSTN to	IP Calls Res	triction			[No		-		
		IP to PST	N Time Res	triction			[Disable		-		
		Detect Tr	unk Status				[Yes		-		
		Detect Pe	eriod (3s ~ 6	3s)			[3				
		Enable S	IP Trunk				[Yes		-		

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.				
Damasta Dant	Q.931 port of SIP of remote platform interfacing with this equipment, the default is				
Remote Port	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				
	Defined by IETF work group RFC3372, it is a standard used to establish remote				
Register to Remote	communication between SIP and ISUP; the default is "Yes"; if SIP trunk does not				
	support, then set it to "No".				
Incoming SIP	There are two modes: IP address and Password. If user selects "password", then				
Authentication Type	password will be filled.				
IP to PSTN Calls	IP to PSTN side of the limitation on the number of calls; the range is 0^{65535} , the				
Restriction	default is no limitation; If Yes is selected, then input limitation number of calls in the				
Restriction	edit box appeared.				
PSTN to IP Calls	PSTN to IP side of the limitation on the number of calls; the range is 0^{-65535} , the				
Restriction	default is no limitation; If Yes is selected, then input limitation number of calls in the				
Restriction	edit box appeared.				
IP to PSTN Time	The default setting is disabled. If Enabled is selected, then user can edit the start				
Restriction	and stop time of prohibition time interval. Within this time interval, all calls from IP				
	to PSTN are prohibited. (Calls from PSTN to IP are not limited)				
Detect Trunk Status	Detect the status of SIP trunk. If select it, the equipment will send HEARTBEAT				
	message to peer to make sure the link status is OK.				
Enable SIP Trunk	A switch used to enable this SIP trunk or not; user can select "Yes" or "No",				
	when "No" is selected, this SIP trunk is invalid.				

2.8.3 SIP Account

SIP Account							
	SIP Account ID	Description	Binding PSTN Group	SIP Trunk No.	Username	Expire Time	
						Total: 0 💌	
			Add Delete	Modify			

Figure 2-8-4 SIP Account

SIP Account ID	0
Description	
Binding PSTN Group	None
SIP Trunk No.	0 <172.30.66.16>
Username	
Password	
Confirm Password	
Expire Time	1800 s

Figure 2-8-5 SIP Account Add

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

SIP trunk account

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

2.9 H323 Config

2.9.1 H.323 Parameter

H.323 Parameter		
Call Mode	FastStart	
Call Signal Port	1720	
Enable H.245 Tunneling	Yes	
DTMF Transfer Mode	H.245 Alphabet	
Start H.245 on Fast Call	Enable	
Start H.245 on	CONNET	
Respond to FastStart on	PROCEEDING	
Start H.245 Negotiation Actively	Enable	
Reset to default configuration	Reset	
	Save	

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

Figure 2-9-1 H.323 Parameter

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	Whether establish H.245, terminal equipment will be sent H.245 negotiation news
Actively	consult.
Reset to default	Click the button to recover factory configuration.
configuration	

H.323 Parameter description

2.9.2 H.323 Trunk

Trunk No.	Trunk Name	Remote IP	Remote Port	Enable H.323 Trun
				Total: 0
	Add	Delete Modify		
	Eigure 2	9-2 H.323 Trunk		
	i igule 2-	5 2 11.323 HUIR		
323 Trunk Add				
Trunk No.		0	•	
Trunk Name				
Remote IP				
Remote Port		1720		
IP to PSTN Calls Restriction		No	•	
PSTN to IP Calls Restriction		No	•	
IP to PSTN Time Restriction		Disable	•	
Enable H.323 Trunk		Yes	•	

Figure 2-9-3 H.323 Trunk Add

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.

H.323 trunk description

2.10 IP Group Config

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

2.10.1 IP Profile

	IP Profile ID	Description	Declare RFC2833 in SDP	Support Early Media	Ringback Tone to PSTN Originated from	Ringback Tone to IP Originated from	Wait for RTP Packet from Peer	T.30 Expanded Type in SDP
	0	Default	Yes	Yes	Local	Local	No	X-Fax
								Total: 1 Page 1
				Add	Delete Mo	dify		
				Figu	ro 2 10 1 ID Drofil	•		
				Figu	re 2-10-1 IP Profil	e		
ID	Profile	Add						
P	Prome	Add						
	IP Pro	file ID			1		•	
	Descri	ption						
	Declar	e RFC283	3 in SDP		No		-	
	Suppo	rt Early Me	edia		Yes		•	
	Ringba	ack Tone t	o PSTN Origina	ted from	Local		•	
	Ringba	ack Tone to	o IP Originated	from	Local		-	
	Wait fo	or RTP Pa	cket from Peer		No		-	
	T.30 E	xpanded T	ype in SDP		X-Fax		-	
		· ·						

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 Forly Modia	Whether support Early Media(183). If select "Yes", the called side to the early
Support Early Media	media to provide ring back tone to the caller.
Ring back Tone to PSTN	IP-> PSTN call ring back tone player side, if setting to local, it will play from the
Originated from	equipment. If setting to IP, it will play by the called
Ring back Tone to IP Originated	PSTN->IP call ring back tone player side, if setting to local, it will play from the
from	equipment and set to PSTN, it will play by the called
Wait for RTP Packet from Peer	If set to No, it will auto send RTP packets during the call and if set to Yes, it will
	wait the RTP packet was sent by the back side first ,then send out RTP packets
T.30 Expanded Type in SDP	T30 extended types in SDP: x-fax or fax

2.10.2 IP Group

IP Group			
	Group ID	Name	IP Trunk Selection
	0	66.16	Cyclic Ascending
			Total: 1 Page 1 💌
		Add Delete Modify	
		Figure 2-10-3 IP Group	
IP Group Ad	ld		
IP Group Name IP Trunk		1 Cyclic Ascending	
		OK Reset Cancel	

Figure 2-10-4 IP Group Add

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

2.10.3 IP Group Management

IP Trunk G	roup				
	Group ID	Index	Trunk Type	Trunk No.	IP Profile ID
	0 <66.16>	0	SIP	0 <172.30.66.16>	0 <default></default>
					Total: 1 Page 1 💌

Add Delete Modify

Figure 2-10-5 IP Trunk Group

IP trunk group description

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

2.11 Call Routing

2.11.1 Routing Parameter

iting 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.

2.11.2 PSTN->IP Routing

PST	N->IP Ro	outing							
	Index	Description	Trunk No.	PSTN Group	Callee Prefix	Caller Prefix	Trunk Type	Trunk No.	Destination IP Group
	255	any		Any		-	Any		0 <66.16>
									Total: 1 Page 1
				Add	Delete	Modify			

Figure 2-11-2 PSTN->IP Routing

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



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

Figure 2-11-3 PSTN->IP Add

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

PSTN->IP routing description

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

2.11.3 PSTN->PSTN Routing

Figure 2-11-4 PSTN->PSTN Routing

PST	N->PST	N Routing						
	Index	Description	Trunk No.	PSTN Group	Callee Prefix	Caller Prefix	Dst Trunk No.	Dst PSTN Group
								Total: 0 💌
				Add	Delete	lodify		

Figure 2-11-5 PSTN->PSTN Add

oute PSTN->PSTN Add		
Index	255	
Description	200	
Source Type	Group	•
PSTN Group	Any	•
Callee Prefix		
Caller Prefix		
Destination Type	Group	•
Destination PSTN Group	0 <r2-0></r2-0>	•

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

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

PSTN->PSTN Routing

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

2.11.4 IP->PSTN Routing

Figure 2-11-6 IP->PSTN Routing

IP->	PSTN R	outing							
	Index	Description	Trunk Type	Trunk No.	IP Group	Callee Prefix	Caller Prefix	PSTN Trunk	Dst PSTN Group
	255	all	Any	Any	0 <66.1				0 <r2-0></r2-0>
								т	otal: 1 Page 1 💌
				Add	Delete	e Modify	1		

Figure 2-11-7 IP->PSTN Routing

dex	254	•
escription		
ource Type	Group	-
unk Type	Any	•
Group	0 <66.16>	-
allee Prefix		
ller Prefix		
estination Type	Group	-
estination PSTN Group	0 <r2-0></r2-0>	-

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

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

IP->PSTN routing configuration and PSTN->PSTN routing configuration are similar, the only

difference is PSTN destination group.

2.11.5 IP->IP Routing

IP->	IP Routi	ing								
	Index	Description	Trunk Type	Trunk No.	IP Group	Callee Prefix	Caller Prefix	Trunk Type	Trunk No.	Dst IP Group
										Total: 0 🔽
				Ad	ld	Delete	Modify			

Figure 2-11-8 IP->IP Routing

Figure 2-11-9 IP->IP Add

ndex	255	-
Description		
Source Type	Group	-
Trunk Type	Any	-
IP Group	0 <66.16>	•
Callee Prefix		
Caller Prefix		
Destination Type	Group	-
Destination IP Group	0 <66.16>	-

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

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

2.12 Number Manipulation

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

 Number Manipulation
 PSTN->IP Callee
 PSTN->IP Caller
 PSTN->PSTN Callee
 PSTN->PSTN Caller
 IP->PSTN Callee
 IP->PSTN Caller
 IP->IP Callee
 IP->IP Caller

Figure 2-12-1 Number Manipulation

"Number Manipulation" is used to replace numbers. User can replace and remove the inbound

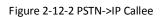
and outbound calling / called number.

Notes:

- 1. The more precise configuration, index values should be smaller.
- 2. Suggesting configure "Any" and ".", avoid missing the call for the replace number $_{\circ}$
- 3. When configuring data, it is suggested that index starts from large index value, to avoid adding an exact match data, not directly use the data.
- 4. When configuring data, it is suggested that keep using index value.

2.12.1 PSTN->IP Callee

Index	Description	PSTN Group	Callee Prefix	caller Prefix	Number of Digits to Strip from Left	Number of Digits to Strip from Right	Prefix to Be Added	Suffix to Be Added	Number of Digits to Reserve from Right
									Total: 0



PSTN->IP Callee Add	
Index	127 💌
Description	*
PSTN Group	Any 💌
Callee Prefix	*
Caller Prefix	*
Number of Digits to Strip from Left	
Number of Digits to Strip from Right	
Prefix to Be Added	
Suffix to Be Added	
Number of Digits to Reserve from Right	
	·

NOTES: 1. Fields with '*' are MUST.

2. '.' in 'Callee Prefix' or 'Caller Prefix' field means wildcard string.

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

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

PSTN->IP destination number

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

2.12.2 PSTN->IP Caller

Index	Description	PSTN Group	Callee Prefix	Caller Prefix	Number of Digits to Strip from Left	Number of Digits to Strip from Right	Prefix to Be Added	Suffix to Be Added	Number of Dig Reserve from
									Tot
				1	Add Delete	Modify			
				Figu	re 2-12-4 PSTN	I->IP Caller			
TN->II	P Caller A	dd							
Index					127	,		-	
Desc	ription							*	
PSTN	I Group				Any	1		-	
Calle	e Prefix							*	
Calle	r Prefix							*	
Numb	per of Digits	to Strip	from Let	ft					
Numb	per of Digits	to Strip	from Rig	ght					
Prefix	to Be Add	ed							
Suffix	to Be Add	ed							
Numb	per of Digits	to Rese	erve from	Right					
				ОК	Reset	Cancel			
					are MUST.				

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

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

same.

PSTN	->PSTN	Callee								
	Index	Description	PSTN Group	Callee Prefix	Number of Digits to Strip from Left			Suffix to Be Added	Number of Digits to Reserve from Right	Number Type
										Total: 0 💌
					Add	Delete Modify	V			

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.

Index	Description	PSTN Group	Callee Prefix	Caller Prefix	Number of Digits to Strip from Left	Prefix to Be Added	Suffix to Be Added	Number of Digits to Reserve from Right	Number Type	Presentation Indicator
										Total: 0

Figure 2-12-7 PSTN->PSTN Caller

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

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

IP->PS	STN Cal	lee									
	Index	Description	IP Group	Callee Prefix	Caller Prefix		Number of Digits to Strip from Right	Prefix to Be Added	Suffix to Be Added	Number of Digits to Reserve from Right	Number Type
											Total: 0
						Add	Delete Modify	V -			

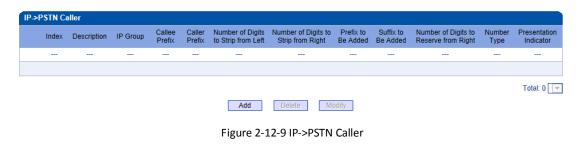
Figure 2-12-8 IP->PSTN Callee

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	Remove the called number digits from the left
left	
Number of Digits to Strip from	Remove the called number digits from the right
right	
Prefix to be Add	Add a called number prefix
Suffix to be Add	Add a called number suffix
Number of Digits to Reserve	Starting from the right to retain the called number digits
from Right	
Number Type	Common number types are: Not Configured, Unknown, International, National,
	Network Specific, Subscriber and Abbreviated.

"IP->PSTN Caller", "IP->IP Callee", "IP->IP Caller" configuration parameters in the previous

number manipulation rules have been mentioned, please refer that section.



IP->IP (Callee									
	Index	Description	IP Group	Callee Prefix	Caller Prefix	Number of Digits to Strip from Left	Number of Digits to Strip from Right	Prefix to Be Added	Suffix to Be Added	Number of Digits to Reserve from Right
										Total: 0
					A	d Delete	Modify			

Figure 2-12-10 IP->IP Callee

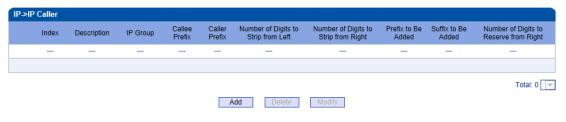


Figure 2-12-11 IP->IP Caller

2.13 Voice & Fax

ce & Fax Configuration	
Maine Demonster	
Voice Parameter	
Disconnect call when no RTP packet	Yes No
Period without RTP packet	60 s
Gain from PSTN	-1dB
Gain to PSTN	2dB
Timeout of No Answer	
Call from PSTN	60 s
Call from IP	60 s
Fax Parameter	
Fax Mode	T.38
Fax Tx Gain	0 db
Fax Rx Gain	0 db
Packet time	20 ms
Redundant frame in packet	3
Data & Fax Control	
Data	Disable
Fax	Disable
DTMF Parameter	
Continuous time	60 ms
Signal interval	60 ms
Threshold for detection	-27 dbm0

Save

Figure 2-13-1 Voice & Fax

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

Voice & Fax description

2.14 Management Parameter

nagement Parameter	
WEB Configuration	
WEB Port	80
Talant Configuration	
Telnet Configuration	
Telnet Port	23
Syslog Configuration	
Syslog Enable	© Yes ◉ No
Qos	
Qos Type	None
NTP Configuration	
NTP Enable	Yes No
Primary NTP Server Address	64.236.96.53
Primary NTP Server Port	123
Secondary NTP Server Address	18.145.0.30
Secondary NTP Server Port	123
Sync Interval	604800 s
Time Zone	GMT+8:00 (Beijing, Singapore, Taipei)

Save

NOTE: The device must restart to take effect.

Figure 2-14-1 Management Parameter

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 There are three options: none, TOS and DS. TOS only supports IPv4. Qos Type **NTP Enable** Simple Network Management Protocol is enabled or not; the default is Yes. Primary NTP server The Primary IP address of SNMP management host computer. The host computer Address of the IP address will carry out monitoring and management to equipment. The port that managed device provides trap message (it is generally alarm Primary NTP server Port message) to SNMP management host computer, the default is 123. The Secondary IP address of SNMP Secondary NTP server Address Time interval of check Sync Interval Time Zone The time zone of local

Management parameter description

2.14.2 SNMP Parameter

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

SNMP Parameter	
Basic Configuration	
SNMP Enable	🛇 Yes 🖲 No
SNMP Manager Address	
Trap Port	162
Community Configuration	
Read-only Community String	public
Read-only Community String	
Read-only Community String	
Read/Write Community String	private
Read/Write Community String	
Read/Write Community String	
Trap Community String	trapuser

Save

Figure 2-14-3 SNMP Parameter

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

SNMP Parameter description

Note: After configuration, please restart equipment to take effect.

Users can manage and configure gateway on remote NM server through SNMP configuration. But

in order to security, recommend this option to open when needed.

2.14.3 Data Backup

Data Backup	
Click 'Backup' to download database file to your computer.	Backup
Click 'Backup' to download dialplan file to your computer.	Backup
Click 'Backup' to download exception file to your computer.	Backup

Figure 2-14-4 Data Backup

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

2.14.4 Data Restore

Data Restore		
Database	〔浏览.	Restore
Dialplan	[浏览.	Restore

Figure 2-14-5 Data Restore

Data restore description

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

2.14.5 Version Information

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



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

2.14.6 Firmware Upload

Firmware Upload		
Software Web	浏览 浏览	Upload Upload

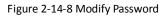
NOTE: The device must restart to take effect after uploading.

Figure 2-14-7 Firmware Upload

Firmware upload description

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

2.14.7 Modify Password



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

2.14.8 Restart Device

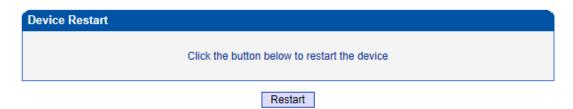


Figure 2-14-9 Restart Device

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

3. FAQ

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

There are one way to get the IP address:

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

Please refer to http://www.dinstar.com/service/fag_145.aspx

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

4. Glossary

PRI: Primary rate interface

- DND: Do-not-Disturb
- FMC: Fixed Mobile Convergence
- SIP: Session Initiation Protocol
- DTMF: Dual Tone Multi Frequency
- USSD: Unstructured Supplementary Service Data
- PSTN: Public Switched Telephone Network
- STUN: Simple Traversal of UDP over NAT
- IVR: Interactive Voice Response
- IMSI: International Mobile Subscriber Identification Number
- IMEI: International Mobile Equipment Identity
- DMZ: Demilitarized Zone