

MTG1000B Trunk Gateway User Manual V2.0



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1. Product Introduction

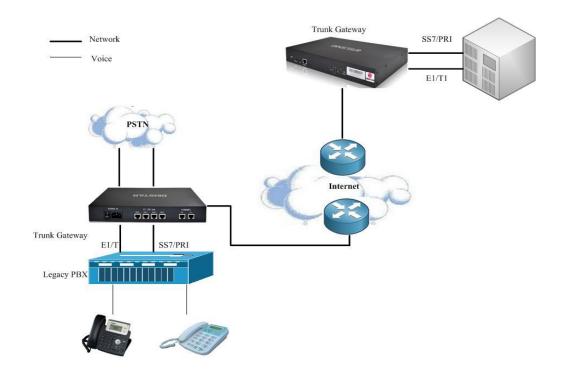
1.1 Overview

MTG1000B is a trunk gateway aimed at small and medium enterprise, and used to help enterprise to realize the evolution from the traditional PBX to voice IP. On the one hand, it supports PRI/SS7/R2 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. MTG1000B 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.

MTG1000B has good call processing ability, and provides 1/2 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:

- MTG1000B-1E1
- MTG1000B-2E1

A typical network diagram shows the function of MTG1000B as below. Figure 1-1-1 Application topology



- **1.2 Equipment Structure**
- 1.2.1 Rear View



Figure 1-2-1 MTG1000B Rear View

Table 1-2-1 MTG1000B Rear View Description

PWR	The power interface, 200VAC, 50~60HZ
Port0-Port3	E1/T1 port, 1/2 E1 ports
FE0	The Service Ethernet Interface, standard 10/100BASE-TX Ethernet interfaces. Default IP address is 192.168.1.111, default subnet mask is 255.255.255.0
FE1	Management Ethernet Interface. Default IP address is 192.168.11.1, default subnet mask is 255.255.255.0

1.2.2 Front View



Figure 1-2-2 MTG1000B Front View

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

Table 1-2-2 MTG1000B 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

MTG1000B 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 /PRI protocol
- NAT Traversing (STUN)
- Hypertext Transfer Protocol (HTTP)
- Domain Name System (DNS)
- 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: 10W
- Dimension (mm): 436*300*44
- Net Weight: 1.9 kg

2. Parameter Setting

2.1 Login

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

If customer modified the default IP or forgot the IP, that can't enter the configuration page. Please connect PC and device serial with the serial line. Enter the CLI to view or modify the equipment IP. Here IP is set to 172.16.88.12. 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.

Authentication Requ	ired 🛛 🕅
The server 172.16.8 password. The serv	8.12:80 requires a username and /er says: GoAhead.
User Name:	admin
Password:	****
	Log In Cancel

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.

General			
MAC Address	00-88-34-11-12-00)	
Service Ethernet Interface(FE0)	172.16.88.12	255.255.0.0	172.16.0.155
Management Ethernet Interface(FE1)	192.168.11.1	255.255.255.0	
DNS Server			
System Time	2012-10-23 17:44	:10	
System Uptime	1 m 17 s		
Traffic Statistics	Received	177.850	bytes
	Sent	365,157	bytes
/ersion			
Device Model	MTG1000B		
Hardware Version	PCB 01		
DSP Version	v7_22_03_16_HW	_12	
Web Version	2.04.03.01		
Software Version	2.04.03.01		
Time Built	2012-08-22, 11:58	3:32	

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	00-88-34-11-12-0	0	
Service Ethernet Interface(FE0)	172.16.88.12	255.255.0.0	172.16.0.155
Management Ethernet Interface(FE1)	192.168.11.1	255.255.255.0	
DNS Server			
System Time	2012-10-23 17:4	4:10	
System Uptime	1 m 17 s		
Traffic Statistics	Received	177.850	bytes
	Sent	365,157	bytes
Version			
Device Model	MTG1000B		
Hardware Version	PCB 01		
DSP Version	v7_22_03_16_HV	/_12	
Web Version	2.04.03.01		
Software Version	2.04.03.01		
Time Built	2012-08-22, 11:5	8:32	

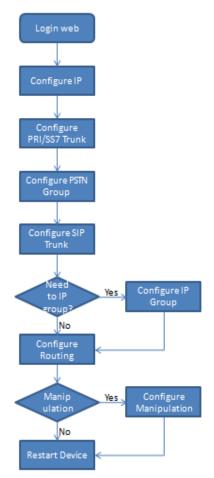
Figure 2-2-1 System Information

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

configuration device in the right configuration interface.

- Status & Statistics
System Information
 E1/T1 Status
 PSTN Trunk Status
 IP Trunk Status
 PRI Call Statistics
 SS7 Call Statistics
 SIP Call Statistics
Network
+ PRI Config
+ SS7 Config
+ R2 Config
+ PSTN Group Config
+ SIP 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			

2.3.1 System Information

System information interface shows the general information and version information.

m Information			
General			
MAC Address	00-88-34-11-12-0	D	
Service Ethernet Interface(FE0)	172.16.88.12	255.255.0.0	172.16.0.155
Management Ethernet Interface(FE1)	192.168.11.1	255.255.255.0	
DNS Server			
System Time	2012-10-23 17:44	4:10	
System Uptime	1 m 17 s		
Traffic Statistics	Received	177.850	bytes
	Sent	365,157	bytes
Version			
Device Model	MTG1000B		
Hardware Version	PCB 01		
DSP Version	v7_22_03_16_HV	/_12	
Web Version	2.04.03.01		
Software Version	2.04.03.01		
Time Built	2012-08-22, 11:5	8:32	

MAC address	Hardware address of FE0 port
Service Ethernet Mode	Network mode of FE0, include: static and DHCP.
Service Ethernet Interface	Include: IP address, subnet mask, FE0 port default gateway
Management Ethernet Interface	Include IP address, subnetmask of FE1
DNS	DNS server IP address
System Up Time	Time elapsed from device power on to now
Traffic Statics	Total bytes of message received and sent by FE0 port
EquipmentType	Equipment type; this equipment is: MTG1000B
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
BuiltTime	The build time of current software version

Table 2-3-1	System	Information

2.3.2 E1/T1 Status



Figure 2-3-2 E1/T1 Status

	Table 2-3-2 Description of E1/T1 status
	LOS Alarm: Signal loss alarm, this alarm is created when receiving is lost; please
	check the physical connection whether disconnected.
	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.
	AIS Alarm: The Alarm Indication Signal (AIS) failure is dedared when an AIS
E1/T1 Port Status	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.
	Disable: Means that this E1/T1 is not used.
	ISDN/SS7 Signal Alarm: Means physical connection is normal, signaling link has
	problem.
	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 OK.
	Signal: Signal channel
E1/T1Channel	Busy: Means this channel is occupied by voice
Status	Fault: The channel is enabled but the cable is not connected.
	Disable: Have not use this E1/T1 trunk
	L-blocked:
	Local blocked, means that communication can onlybe initiated from
	local

. -

R-blocked:
Remote blocked, means that communication can onlybe initiated from
remote
B-blocked:
Both Sides blocked, means that the two sides cannot communication

2.3.3 PSTN Trunk Status

PRI Trunk No.	Trunk Name	E1/T1 Port No.	Link Status
ink Status			
ink Status SS7 Trunk No.	Trunk Name	E1/T1 Port No.	Link Status
	Trunk Name	E1/T1 Port No.	Link Stat

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	

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

IP Trunk Status Trunk No	Trunk Name	Trunk Mode	Username	Incoming Authentication Type	Link Status
0	172.30.66.16	Peer		IP Address	Established

Figure 2-3-4 SIP Trunk Status

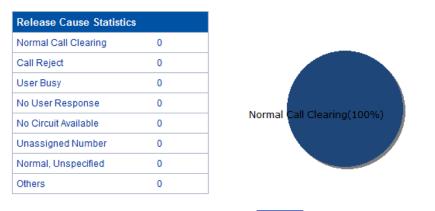
IP trunk status

SIP Trunk No	The number of SIP trunk			
Username	When SIP trunk is under registered mode, change the value in the			

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

2.3.5 PRI Call Statistics

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



Refresh

图 2-3-5 PRI Call Statistics description

PRI call statistics description

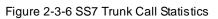
PRI Trunk No	The number of PRI trunk
Trunk Name	The name used to describe the PRI trunk
Current Calls	Number of lines that are being called currently
Accumulated Calls	Total number of calls from running start of system to current time.
ASR	The percent of calls completed in total calls.

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





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

2.4 Network

Service Ethernet Interface(FE0)		
IP Address	172.30.65.25	
Subnet Mask	255.255.0.0	
Default Gateway	172.30.0.1	
Management Ethernet Interface(FE1)	
IP Address	192.168.11.1	
Subnet Mask	255.255.255.0	
DNS Server		
Primary DNS Server		
Secondary DNS Server		

Save

NOTE: The device must restart to take effect.

Figure 2-4-1 Network Configuration

Network Configuration

Service Ethernet	IP address	Set FE0 port static IP address.
Interface (FE0)	SubnetMask	Fill in subnetmask
	Default Gateway	Fill in default gateway
Management	IP address	Set FE1 port static IP address
Ethernet Interface	CubratMaak	Fill in automatik
(FE1)	SubnetMask	Fill in subnetmask
	Primary DNS	Fill in DNS Server IP address.
DNS Server	Secondary DNS	The secondary DNS server is option.

Ntoe: FE0 port IP and FE1 port IP should be set in different segments. After configure the

network address, and restart the gateway configuration to take effect.

2.5 PRI Config

PRI configuration includes PRI parameter and PRI trunk configuration

Figure 2-5-1 PRI Config



2.5.1 PRI Parameter

RI Parameter		
Calling Party Numbering Plan	ISDN/Telephony numbering plan	-
Calling Party Number Type	Unknown	•
Screening Indicator for Displaying Caller Number	User provide,no shield	•
Screening Indicator for No Displaying Caller Number	User provide,no shield	-
Called Party Numbering Plan	ISDN/Telephony numbering plan	•
Called Party Number Type	Unknown	•
Information Transfer Capability	Speech	•
Reset to default configuration	Reset	

Figure 2-5-2 PRI Parameter

Save

PRI parameter description

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.
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.
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.
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.
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.
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.
Support speech and 3.1 khz audio. The default option is speech.
Support speech and S. INIZ audio. The delaut option is speech.

2.5.2 PRI Trunk

Figure 2-5-3 PRI Trunk PRI Trunk Trunk No. Trunk Name Channel ID D-Channel E1/T1 Port No. Protocol Switch Side Alerting Indication --------____ ----____ ____ --------Add Delete Modify

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

Figure 2-5-4 Add PRI Trunk

Trunk Add		
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 number is numbered according to the physical position of E1/T1, it
E1/T1 Port No	generallystarts from 0.
Droto col	Interface type of PRI. There are two types are available: ISDN and QSIG; the
Protocol	default is ISDN.
	Indicate PRI network property of E1/T1, it is divided into: "User side" and "Network
Switch Side	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 and SS7 CIC Maintain.

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

2.6.1 SS7 Trunk

Figure 2-6-2 SS7 Trunk

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

Figure 2-6-3 SS7 Trunk Add

67 Trunk Add		
Select Trunk No.	3	•
Trunk Name		
Protocol	ITU	•
Protocol Type	ISUP	•
SPC Format	Hex	•
OPC		
DPC		
Network Indicator	National Networ	rk 💌
Sending SLTM	Enable	•

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 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.
Name of trunk, it can be edited to any name user want.
SPC types: ITU-T (14 bit), ANSI (24 bit), ITU-CHINA (24 bit)
Supported two protocol types: ISUP and TUP
Signaling Point Code format includes hexadecimal system and ITU
pointcode structure (decimal system)
Original Point Code
Destination Point Code
SS7 service types: ISUP (ISDN User Part) and TUP (Telephone User
Part).
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.

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, 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

Trunk No.	Link No.	Signaling Link Code	E1/T1 Port No.	Channel No
		· (1223
	Add	Delete Modify		

Figure 2-6-4 SS7 MTP Link

runk No.		-
Link No.	0	-
Signaling Link Code		
E1/T1 Port No.	0	•
Channel No.	16	

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

SS7	MTP	link	description	
-----	-----	------	-------------	--

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

Figure 2-6-6 SS7 Circuit description

SS7 Circuit Add	
Trunk No.	■
E1/T1 port No.	0
Start Channel	
Start CIC No.	
Count	
	OK Reset 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.

SS7 Circuit N	laintain								
	Operation Mode				E1/T1	•			
Port N	lo	0		1	l	2	2	3	
Protocol	Туре								
Statu	S								
]]]
	Sle	ct All In	vert C	lear Blo	ock Unb	lock Res	et Can	cel	
Actived	Disable	Fault	RAI Alarm	AIS Alarm	ISDN/SS7 S	Signal Alarm			
Frame-Sync	Idle	Signal	Busy	L-blocked	R-blocked	B-blocked	Blocking	Unblocking	Reseting

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

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	Displaythe 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 Circuit	t Mair	ntain														
Operation Mode							(Char	nnel		•					
Current F	Port			-		Statu	IS				Pro	tocol	Туре	und	efine	d
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.																
														_		
	SI	ect/	Inv	ert	Cle	ar	Blo	ck	Unb	lo	Rese	et	Canc	e		
Actived I	Dis	F	ault	RAI	A	AIS A	N I	ISDN	/SS7	Sig						
Frame	Idle	Si	gnal	Bu	isy	L-blo) I	R-blo	E	B-blo.	В	lock.	. Ur	ıbl	Res	se
							1									

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

2.7 R2 Config

2.7.1 R2 Param

R2	Param									
	Param ID	Description	CDbits	Req Next DNIS	Request Next ANI	Request Category	DNIS End	ANI End	Adress Complete	Answer Signal
	0	ITU	01	A-1	A-5	A-5	I-15	I-15	A-3	Call with charge
	1	Argentina	01	A-1	A-5	A-5	INVALID	I-12	A-3	Call with charge
	2	Brazil	01	A-1	A-5	A-5	INVALID	I-15	A-3	Call with charge
	3	China	11	A-1	A-1	A-6	INVALID	I-15	A-3	Call with charge
	4	Czech	01	A-1	A-5	A-5	I-15	I-15	A-3	Call with charge
	5	Colombia	01	A-1	A-5	A-5	I-15	I-15	A-3	Call with charge
	7	Mexico	01	A-1	INVALID	INVALID	I-15	I-15	INVALID	Call with charge
	8	Philippines	01	A-1	A-5	A-5	INVALID	I-15	A-3	Call with charge
	9	Venezuela	01	A-1	A-9	A-5	INVALID	I-15	A-3	Call with charge
	11	Bolivia	01	A-1	A-5	A-5	I-15	I-15	A-3	Call with charge
	14	India	01	A-1	A-4	A-5	INVALID	I-10	A-3	Call with charge
	15	Indonesia	01	A-1	A-6	A-6	I-15	I-15	A-3	Call with charge
	16	Korea	01	A-1	A-5	A-5	INVALID	I-15	A-3	Call with charge
	17	Malaysia	01	A-1	A-6	A-6	I-15	I-15	A-3	Call with charge
	18	Panama	01	A-1	A-5	A-5	INVALID	I-15	A-3	Call with charge
	19	Singapore	01	A-1	A-6	A-6	I-15	I-15	A-3	Call with charge
	20	Thailand	01	A-1	A-1	A-6	I-15	I-15	A-3	Call with charge

Figure 2-7-1 R2 Parameter

It is the default configuration for MTG1000B. Description says the state name, means the different countries supported R2 parameters standards. According to demands add R2 parameters of user countries.

aram Add		
Config Mode	Typical	-
Param ID	6	•
Description		
CDbits	00	-
Calling Party Category	National subscriber	-
Answer tone	Call with charge	-
Seize Timer (ms)	5000	
Protect Timer (ms)	300000	
Receive Timer (ms)	5000	
Wait Response Timer (ms)	3000	
MF Off Timer (ms)	3000	
Wait Release Timer (ms)	3000	
Group I:		
DNIS end flag	I-15	-
ANI end flag	I-15	-
Group A:		
Address Complete	A-3	•
Request next DNIS	A-1	•
Request next ANI	A-5	•
Request category	A-5	•
Request Change to Group C	INVALID	•
Request Last Digit Again	A-8	•
Repeat All DNIS Digit	A-8	•
Group B:		
Unallocated number	B-5	T
User busy	B-3	-
Line out of order	B-2	•
Group C (for Mexico):		
Request Next ANI	C-1	-
Request All DNIS and change to Group	C-2	-
A Address Complete	C-3	-

Figure 2-7-2 R2 Parameter Add

C-4

Network Congestion

Group A

Request next DNIS and change back to C-5

Request Last DNIS and change back to C-6 Group A

•

•

•

Param ID	Identification parameter group				
Description	Description parameter information, Points out which countries standard the				
Description	parameters are.				
CDbits	C, Dbit value of A, B, C,Dbit in R2 lines of signaling.				
RequestNext	The rear party notices the front party ahead called number has received, and each				
DNIS	other can send a next number.				
RequestNext	The rear party notices the front party ahead callee number has received, and each				
ANI	other can send a next number.				
Request					
category	Means KA request code of R2 lines signaling				
DNIS end flag	The front party notices the rear party that the called numbers send completely.				
ANI end flag	The front party notices the rear party that the callee numbers send completely.				
Address	The rear party notices the front party that the called and the callee num bers received				
Complete	completely.				
AnswerTone	The general calls is free of charge or not.				

Parameter Description

2.7.2 R2 Trunk

R2 Trun	k			
	Trunk No.	Trunk Name	E1 Port No	Paramld
	0	R2	0	3 <china></china>
	1	R2	1	0 <itu></itu>
	2	R2	2	3 <china></china>

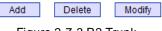


Figure 2-7-3 R2 Trunk

R2 Trunk Add	
Trunk No	3
Trunk Name	
E1 Port No	3
Protocol Param	0 <itu></itu>
	OK Reset Cancel

Figure 2-7-4 R2 Trunk Add

PRI trunk description

Trunk No	The unique identifiers of R2 trunk; system customs eight relay index number.
Trunk Name	Used to identify and describe R2 trunk

E1 Port No	According to T1 / E1 port position sequence sort, usually starting from 0
Protocol Param	Select R2 parameter group.

2.8 PSTN Group Config

2.8.1 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)

Modify

Figure 2-8-1 E1/T1 Parameter

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 Mode	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				
Line Code	of T1 are: NRZ, CMI, AMI, B8ZS, the default is B8ZS.				
	Cable length. E1 lines docking, the environment will affect the E1 line signal				
Line Built Out	strength, signal strength according to (DB value) to select the long-term or				
	short-term.				

E1/T1 parameter description

2.8.2 Coder Group

		Сос	der Group ID	0(default	setting)	•		
	Coder		Payload Type Value	Packetizat (ms		Rate (kbps)	Silence Supp	pression
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		-			-			T
6th		-			-			-



Figure 2-8-2 Coder Group

<u> </u>	
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.
	group. Delaut value cannot be mounieu.
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	Voice Codec packetization time, user can define different kinds of coding
Time(ms)	and decoding minimum packetization time
Rate(kbps)	Show the rate.
	It is disabled by default. During talking, the bandwidth occupied by voice
Silence Suppression	transmission 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.

Coder group description

2.8.3 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 2-8-3 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.

efix	
Prefix	
Prefix IIII IIIII IIIII IIIIIIIIIIIIIIIIIII	
Max Length	
OK Reset Cancel	

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

NOTES: 1. '.' in 'Prefix' field means wildcard string.

2. 'Max Length' and 'Min Length' do not include the 'prefix'.

3. The value of 'Max Length' plusing the length of 'Prefix' should less than 30.

Figure 2-8-4 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
Index	according to the maximum number received.
Prefix	Match number, "." representative of any number
	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
Min Longth	will be used to continue the call. If the maximum length determine the number to
Min Length	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.

2.8.4 Dial Timeout

I	Dial Timeout ID	Description	Max Time for Collecting Prefix(s)	Time to Reach Min Length (s)	Time to Reach Max Lengt (s)
	0	Default	20	10	10
					Total: 1 Page 1
			Add Delete	Modify	
			Figure 2-8-5 D	al Timeout	
			Figure 2-8-5 D	al Timeout	
al Tin	neout Add		Figure 2-8-5 D	al Timeout	
	neout Add		Figure 2-8-5 D	al Timeout	
Dia			Figure 2-8-5 D		
Dial Des	I Timeout ID	ecting Prefix	Figure 2-8-5 D		
Dial Des Max	I Timeout ID scription	-	[1		

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

Figure 2-8-6 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

2.8.5 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 Ton to PSTN
0	Default	0	101	RFC2	SIP IN	Inband	Enable	0	0 <default></default>	Not remove	No
										т	otal: 1 P

Figure 2-8-7 PSTN Profile

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

PSTN Profile ID	1	-
Description		
Coder Group ID	0	-
RFC2833 Payload Type	101	
DTMF Tx Priority 1st	RFC2833	-
DTMF Tx Priority 2nd	SIP INFO	-
DTMF Tx Priority 3rd	Inband	-
Overlap Receiving	Disable	-
Remove CLI	Not remove	-
Play Busy Tone to PSTN	No	-

OK Reset

Figure 2-8-8 PSTN Profile Add

Cancel

PSTN profile add description

The number to the PSTN Profile
Description of the PSTN Profile
Refer to "Coder Group"
The item is 101 by default.
There are three ways to send DTMF: RFC2833/SIP INFO/ INBAND,
in accordance with the priority choice to send the configuration mode
Not enabled by default, only user enables this feature, "Dial plan" and
"Dial timeout" would work.
Default does not remove CLI
Equipment will playbusy tone from IP to PSTN
The number to the PSTN Profile
Description of the PSTN Profile

2.8.6 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 Grou	р		
	Group ID	Name	Channel Selection
	0	r2-0	Cyclic Ascending
	1	r2-12	Cyclic Ascending
			Total: 2 Page 1 💌

Add Delete Modify
Figure 2-8-9 PSTN Group

PSTN Group Add			
Trunk Group ID		2	-
Name Channel Selection		Cyclic Ascending	-
		Cyclic Ascending Ascending	
	ОК	Res Cyclic Descending Descending	



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.

2.8.7 PSTN Group Management

STN Grou	p 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

Figure 2-8-11	PSTN Group	Management

Modify

Delete

Add

TN Group Management 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 2-8-12 PSTN Group Management Add

<u> </u>				
Group ID	PSTN group ID			
Start E1	E1/T1 trunk group port number in the initial			
End E1	Lasta E1/T1 trunk group port number			
Start Channel	The beginning of time s lot, 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			

PSTN group management add

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

2.9 SIP Config

2.9.1 SIP Parameter

SIP Parameter		
Local SIP Port	5060	
Local Domain		



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

2.9.2 SIP Trunk

SI	SIP Trunk											
	Trunk No.	Trunk Name	Remote Address	Remote Port	Local Domain	Support SIP-T	Get Callee from	Register to Remote	Outgoing Call Mode	Incoming Authentication Type	Detect Trunk Status	Enable SIP Trunk
	0	172.30.66.16	172.30.66.16	5060	Disable	Disable	Request-line	No	Peer	IP Address	Yes	Yes
											Total:	1 Page 1 🖣
						Add	Delete	lodify				

Figure 2-9-2 SIP Trunk

runk Add		
Trunk No.	4	-
		•
Trunk Name		
Remote Address		
Remote Port	5060	
Outbound Proxy		
Outbound Porxy Port	5060	
Local Domain	Disable	$\overline{\mathbf{v}}$
Support SIP-T	Disable	•
Get Callee from	Request-line	•
Register to Remote	No	•
Incoming SIP Authentication Type	IP Address	•
IP to PSTN Calls Restriction	No	•
PSTN to IP Calls Restriction	No	•
IP to PSTN Time Restriction	Disable	•
Detect Trunk Status	Yes	•
Detect Period (3s ~ 63s)	3	
Enable SIP Trunk	Yes	•

OK Reset Cancel

Figure 2-9-3 SIP Trunk Add

SIP trunk description

SIF trunk description	
Trunk No	The range of number is 1~99
Trunk Name	Description the trunk
Remote Address	IP address of remote platform interfacing with this equipment.
Remote Port	Q.931 port of SIP of remote platform interfacing with this equipment, the
Kemole Fold	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
	Defined by IETF work group RFC3372, it is a standard used to establish
Register to Remote	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",
Authentication Type	then 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,
Restriction	the default is no limitation; If Yes is selected, then input limitation number of
Restriction	calls in the edit box appeared.
PSTN to IP Calls	PSTN to IP side of the limitation on the number of calls; the range is 0~65535,
Restriction	the default is no limitation; If Yes is selected, then input limitation number of
	calls in the edit box appeared.
IP to PSTN Time	The default setting is disabled. If Enabled is selected, then user can edit the
Restriction	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 the status of SIP trunk. If select it, the equipment will send				
Detect Trunk Status	HEARTBEAT message to peer to make sure the link status is OK.				
	A switch used to enable this SIP trunk or not; user can select "Yes" or "No",				
Enable SIP Trunk	when "No" is selected, this SIP trunk is invalid.				

2.9.3 SIP Account

SIP Account ID	Description	Binding PSTN Group	SIP Trunk No.	Username	Expire Tir
					Total: 0
	[Add Delete	Modify		
		Figure 2-9-4 SIP Ac	count		
Account Add					
SIP Account ID		0		-	
Description					
Binding PSTN Gro	up	None		-	
SIP Trunk No.		0 <172.	30.66.16>	•	
Username					
Password					
Confirm Password					
		1800			

Figure 2-9-5 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 anytrunk 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		

2.10 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
							1	Fotal: 1 Page 1
				Add	Delete Mod	dify		
				Figur	e 2-10-1 IP Prof	file		
IP	Profile	Add						
	IP Pro	file ID			1		-	
	Descri	ption						
	Declar	e RFC283	3 in SDP		No		-	
	Suppo	rt Early Me	edia		Yes		-	
	Ringba	ack Tone to	o PSTN Origina	ited from	Local		-	
	Ringba	ack Tone to	o IP Originated	from	Local		-	
			cket from Peer		No		-	
	T.30 E	xpanded T	Type in SDP		X-Fax		-	
		- C						

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	Defaultsupport
Support Forly Madia	Whether support Early Media(183). If select "Yes", the called side to the
Support Early Media	early 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
Originated from	from the equipment. If setting to IP, it will play by the called
Ring back Tone to IP	PSTN->IP call ring back tone player side, if setting to local, it will play
Originated from	from the equipment and set to PSTN, it will play by the called
Wait for RTP Packet from	If set to No, it will auto send RTP packets during the call and if set to Yes,
Peer	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 A	dd		
IP Group Name IP Trunk	DID Selection	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 Modif	fy	

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

ting 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

PSTN->IP Routing											
	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>	-

OK Reset Cancel

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
	callernumber
Destination Type	Destination type is IP group or SIP/H.323 trunk.
Destination IP Group	Refer to "IP Group"
Trunk Type	Trunk type means IP side trunk type-SIP/H.323.
Trunk No.	Trunk number

2.11.3 PSTN->PSTN Routing

Figure 2-11-4 PSTN->PSTN Routing

PSTN-	>PSTN	Routing						
1	ndex	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

e PSTN->PSTN Add		
ndex	255	•
Description		
Source Type	Group	•
PSTN Group	Any	•
Callee Prefix		
Caller Prefix		
Destination Type	Group	•
Destination PSTN Group	0 <r2-0></r2-0>	•

el

NOTE: 11 in 'Callee Prefix' or 'Caller Prefix' field means wildcard string.

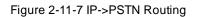
"PSTN->PSTN Routing": Routing Call from PSTN to PSTN

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 anytrunk 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
	callernumber
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->	IP->PSTN Routing										
	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	/				



dex	254	-
escription		
ource Type	Group	-
runk Type	Any	-
^o Group	0 <66.16>	-
allee Prefix		
aller 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

Figure 2-11-8 IP->IP Routing

1	IP->IP Rout	ing								
	Index	Description	Trunk Type	Trunk No.	IP Group	Callee Prefix	Caller Prefix	Trunk Type	Trunk No.	Dst IP Group
										Total: 0 💌
				Ad	d	Delete	Modify			

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: Figure 2-12-1 Number Manipulation

 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

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

Notes:

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

2.12.1 PSTN->IP Callee

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

Figure 2-12-2 PSTN->IP Callee

PSTN->IP Callee Add	
Index	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 anytrunk 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	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 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 D Reserve from
									То
				-	Add Delete	Modify			
				Figure	2-12-4 PSTN	I->IP Caller			
TN->II	P Caller A	dd							
Index	:				127			-	
Desc	ription							*	
PSTN	I Group				Any			-	
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							
Numt	per of Digits	s to Rese	rve from	n Right					
				ОК	Reset	Cancel			
		ES: 1			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	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
										Total: 0
					Add	Delete Modify	y .			

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	N Caller Description	PSTN Group	Callee Prefix	Caller	Number of Digits to Strip from Right	Suffix to	Number of Digits to	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 Left	Number of Digits to Strip from Right		Suffix to Be Added	Number of Digits to Reserve from Right	Number Type
											Total: 0 💌
						Add	Delete Modify	У			

Figure 2-12-8 IP->PSTN Callee

IP->PSTN c	allee de	scription
------------	----------	-----------

-	
Index	Index number (0 ~ 127)
Description	Describe the transformation of the number
IP Group	Refer to "IP Group", "any" means anytrunk group
Callee Prefix	Called number prefix, "." means any called number
Caller Prefix	Caller number prefix, "." Means any caller number
Number of Digits to Strip	Remove the called number digits from the left
from left	
Number of Digits to Strip	Remove the called number digits from the right
from 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->PSTN Caller												
	Index	Description	IP Group	Callee Prefix	Caller Prefix	Number of Digits to Strip from Left	Number of Digits to Strip from Right		Suffix to Be Added	Number of Digits to Reserve from Right	Number Type	Presentation Indicator
												Total: 0 🔽
						Add	Delete	odify				

Figure 2-12-9 IP->PSTN Caller

Index	Description	IP Group	Callee Prefix	Caller Prefix	Number of Digits to Strip from Left	Number of Digits to Strip from Right	Prefix to Be Added	Suffix to Be Added	Number of Digits to Reserve from Right
									Total: 0
				Ad	ld Delete	Modify			

Figure 2-12-10 IP->IP Callee

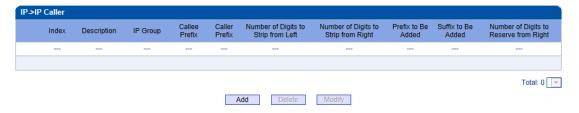


Figure 2-12-11 IP->IP Caller

2.13 Voice & Fax

e & Fax Configuration		
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	

Figure 2-13-1 Voice & Fax

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

Voice & Fax description

2.14 Management Parameter

agement Parameter	
WEB Configuration	
WEB Port	80
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

Management parameter description

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

2.14.2 SNMP Parameter

Simple Network Management Protocol (SNMP) is application layer protocol, and used to manage communication line. This equipment supported three versions: V1, V2C and V3. In addition to V3 version, the other two versions do not support encryption. However, the service is usually located on the edge of the network devices, security risk, it is best to disable, to be used again.

Parame	eter								
	nable		() Yes (5 No					
			0.000	- 110					
SNMP V	ersion		v1	•					
_									
Commu	nity Configuratio	on Commu	nity				Source		
lst d	instar				default				
2nd									
Brd									
Votice:de	afault value of source	e is default,	if other valu	ue,please input	t IP!(eg:192.16	8.1.1)			
Group (Configuration	_							
_		Grou	р				Community		
									-
Ist 2nd									
2nd									v
2nd Srd									
2nd Srd	onfiguration					ferr On bland		16-mAte	
2nd	ViewName			iewType		/iewSubtree		ViewMa	
2nd and and and and and and and and and a	ViewName		Vincluded		.1	/iewSubtree		ViewMa	
2nd Brd /iew Co Ist 2nd	ViewName			v	.1	/iewSubtree		ViewMa	
View Co Ist a Ind and and and and and and and and and a	ViewName		ncluded	▼ ▼	.1	∕iewSubtree		ViewMa	
View Co Ist a Ind and and and and and and and and and a	ViewName		ncluded	▼ ▼	.1	∕iewSubtree		ViewMa	
And	ViewName	x.x.x.x.if just	ncluded	▼ ▼	.1	/iewSubtree		ViewMa	
And	ViewName II fiewSubtree style:x.	x.x.x.x.if just	ncluded	▼ ▼		/iewSubtree		ViewMa	sk
Access	ViewName II fiewSubtree style:x. Configuration(v1	x.x.x.x.if just	ncluded t one,style:	V V X	.1 		all		sk
Ard Srd Srd Srd Srd Srd Srd Srd Srd Srd S	ViewName II FiewSubtree style:x. Configuration(v1 Group	x.x.x.if jusi	ncluded t one,style:	x Read	.1	Write	all		sk
Access	ViewName II FiewSubtree style:x. Configuration(v1 Group	x.x.x.if jusi I/v2c)	ncluded t one,style:	Read	.1	Write	all		sk
Access Ist P Red Red Red Red Red Red Red Red	ViewName II FiewSubtree style:x. Configuration(v1 Group	x.x.x.if jusi	t one,style:	Read	.1	Write		Notify	sk
Access Ist P Acces	ViewName II FiewSubtree style:x: Configuration(v1 Group articularGroup ead/Write/Notify val	x.x.x.if jusi	t one,style:	Read	.1	Write		Notify	sk
Access Ist P Acces	ViewName II FiewSubtree style:x: Configuration(v1 Group articularGroup ead/Write/Notify val onfiguration	x.x.x.if jusi	i one,style: II to ViewNai	Read Mead Mead Mead/Wr	.1	Write		Notify	sk
Access Ist P Acces	ViewName II FiewSubtree style:x: Configuration(v1 Group articularGroup ead/Write/Notify val	x.x.x.if jusi	i one,style: II to ViewNai	Read	.1	Write		Notify	sk

Save

Figure 2-14-3 SNMP Parameter

SNMP Parameter description

CommunityConfiguration	Community	The name of network management server
		managed equipment
	Source	Networkmanagementserveraddress
Group Configuration	Group	Name of community group, different versions can
		use a same group name
	Community	Communityjoin the group
View Configuration	View name	The name of description mib tree

	View type	There are Included and excluded options
	View subtree	Displayed OID of access parameters
	View mask	The same with equipment subnet mask. Generally
		don't configure
Access Configuration(V1,	Group	Joined communitygroups
V2c)	Read	Read parameters of mib view
	Write	Write parameters of mib view
	Notify	Equipment send notify parameters to NM server
Trap Configuration	Trap Flag	Version of SNMP
	Trap IP	Device to inform the NM server's IP address. The
		IP can be configured the same with source IP in
		community, also be different.
	Trap Port	Default service port is 162
	Trap Community	The same with "community" in community
		configuration

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

Version Information				
File Type	Version	Date Built	Time Built	
Software	2.04.03.01	2012-08-22	11:58:32	
Database	2.01.01	2012-05-23	18:53:00	
Web	2.04.03.01	2012-08-22	11:45:09	

Figure 2-14-6 Version Information

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

2.14.6 Firmware Upload

Firmware Upload	
Software Web	 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

Old Password	
New Password	
Confirm Password	

Figure 2-14-8 Modify Password

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

2.14.8 Restart Device

Device Restart	
	Click the button below to restart the 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. 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:

 Connect the CONSOLE with your PC Serial Port. The baud rate is 115200 bps. The user name and password is "admin". When users logged in system, and then run command "show int" for getting the IP.

Please refer to http://www.dinstar.com/service/faq 145.aspx

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

4. Glossary

PRI: Primary rate interface

ISUP: ISDN User Part. It is part of the Signaling System #7 which is used to set up telephone calls in Public Switched Telephone Networks;

TUP: Telephone User Part (TUP) provides the signaling backbone between switching elements for basic call establishment, supervision, and release of circuit switched network connections for telecommunications services;

FMC: Fixed Mobile Convergence
SIP: Session Initiation Protocol
DTMF: Dual Tone Multi Frequency
SNMP: simple network management protocol
PSTN: Public Switched Telephone Network
STUN: Simple Traversal of UDP over NAT
IVR: Interactive Voice Response
RTP: Real-Time Transport Protocol
ISDN: Integrated Services Digital Network
NTP: Network Time Protocol