



MMTG200 Trunk Gateway User Manual

V2.0



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1. Product Introduction

1.1 Overview

MTG200 T1/E1 trunk gateway provide the ability to terminate (or originate) T1/E1 circuits directly into IP telephony system or call center, Asterisk-based distributions such as Elastix, FreePBX, etc., or Freeswitch.

MTG200 trunk gateway provides 1/2/4 T1/E1 interfaces in a small box and also easy to install and configure. It provides the ability to handle 120 concurrent calls maximum with G711 codec and 64 concurrent calls with G729. Moreover, it's support ISDN PRI, SS7 and R2 signaling so that provide good connectivity to different PBXs system. MTG200 includes following models:

- MMTG200-1E1: single port E1/T1 trunk gateway
- MMTG200-2E1: two ports E1/T1 trunk gateway
- MMTG200-4E1: four ports E1/T1 trunk gateway

Notes: MMTG200 support ISDN PRI by default, need to apply new license to enable SS7/R2 if necessary.

1.2 Equipment Structure

1.2.1 Rear View

Figure 1-2-1 MTG200 Rear View



Table 1-2-1 MTG200 Rear View Description

Interface	Description
PWR	The power interface. DC12V.1A

Port0-Port3	E1/T1 Port. There are 4E1 options.
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 MTG200 Front View



Table 1-2 -2 MTG200 Front View Description

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

1.2.3 RJ-48c Line sequence

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

MTG200 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
- Dynamic Host Configuration Protocol (DHCP)
- Point-to-Point Protocol over Ethernet (PPPoE)
- 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: 12VDC, 1A
- Temperature: 0~40°C (operational), -20~70°C (storage)
- Humidity: 10%~90%, no condensation
- Max power consumption: 15W
- Dimension (mm): 210*150*38
- Net weight: 0.75kg

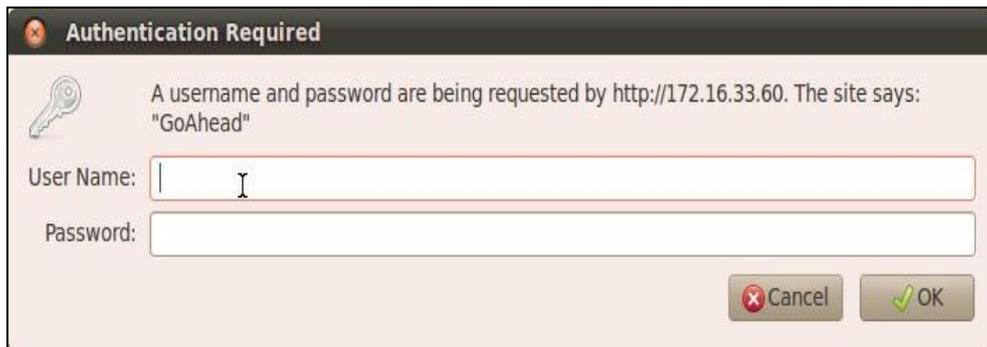
2. Parameter setting

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 172.16.33.60. In addition, hold down the RST button to restart the device, customer can regain the port's default IP. Then enter the IP address of device in the browser address bar. Customer will see the following page.

Figure 2-1-1 Login Interfaces



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

Figure 2-1-2 Modify Password

Password Modification

Old Password	
New Password	
Confirm Password	

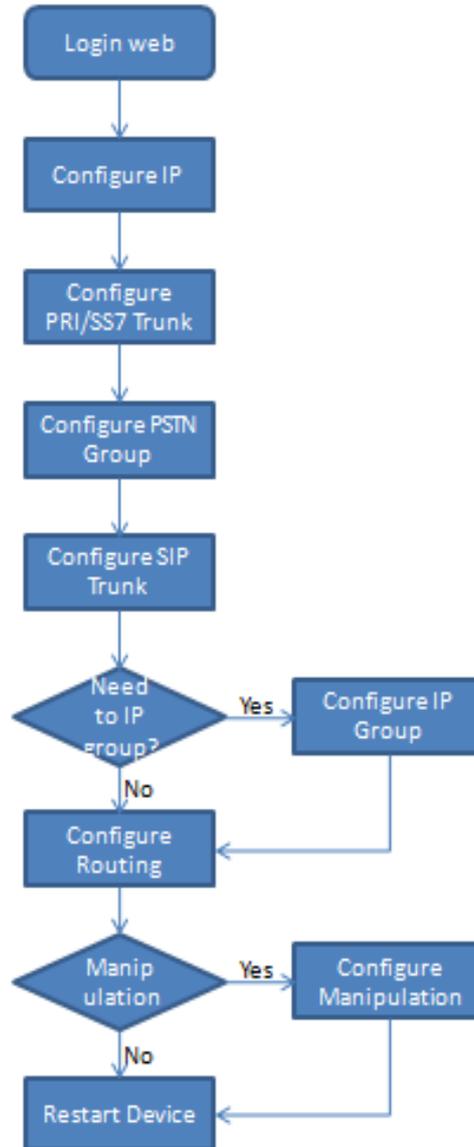
Save

After inputting the old password, input a new password and confirm it by inputting it again.

2.2 Status & Statistics

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

TG configuration flow chart below:



2.2.1 System Information

This configuration page includes general information and version information.

Figure 2-2-1 System Information

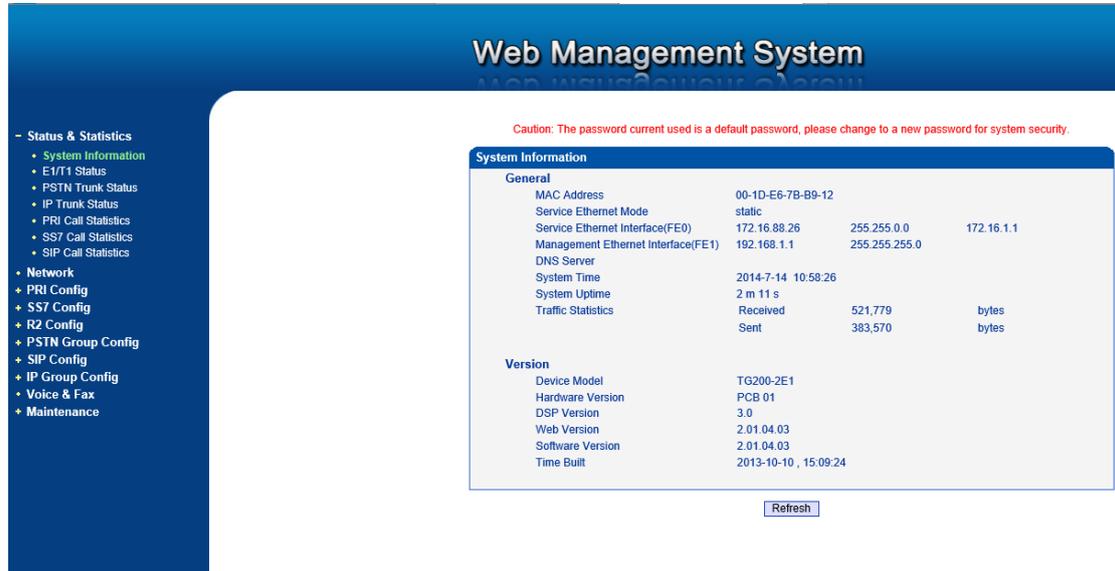


Table 2-2-1 Description of System Information

MAC address	Hardware address of FE0 port
Service Ethernet Mode	Network mode of FE0, include: static and DHCP.
Service Ethernet Interface	Include: IP address, subnet mask, FE0 port default gateway
Management Ethernet Interface	Include IP address、 subnet mask of FE1
DNS	DNS server IP address
System Up Time	Time elapsed from device power on to now
Traffic Statics	Total bytes of message received and sent by FE0 port
Equipment Type	Equipment type; this equipment is: MTG200
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

2.2.2 E1/T1 Status

Figure 2-2-2 E1/T1 Status

E1/T1 Port Status				
Port No.	0	1	2	3
Physical Status				

NOTES: Active Disable LOS Alarm
 RAI Alarm AIS Alarm ISDN/SS7 Signal Alarm

E1/T1 Channel Status																																
Channel No.	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
Port 0																																
Port 1																																
Port 2																																
Port 3																																

Status	Frame-Sync	Idle	Signal	Busy	Fault	Disable	L-blocked	R-blocked	B-blocked
Color									
Totalize	2	60	2	0	0	64	0	0	0

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

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

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

2.2.3 PSTN Trunk Status

Figure 2-2-3 PSTN Trunk Status

PRI Link Status			
PRI Trunk No.	Trunk Name	E1/T1 Port No.	Link Status
0	pri0	0	Established
1	pri1	1	Established

Table 2-2-3 Description of PSTN Trunk 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.

2.2.4 IP Trunk Status

Figure 2-2-4 IP Trunk Status

SIP Trunk Status					
Trunk No	Trunk Name	Trunk Mode	Username	Incoming Authentication Type	Link Status
0	3cx.sip	Access	333	IP Address	Established
1	elastix.sip	Peer	---	IP Address	Established
2	dag.sip	Peer	---	IP Address	Established

Refresh

Table 2-2-4 Description of 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.

2.2.5 PRI Call Statistics

Figure 2-2-5 PRI Trunk Call Statistics

PRI Trunk Call Statistics				
PRI Trunk No.	Trunk Name	Current Calls	Accumulated Calls	ASR
0	pri0	0	0	100%
1	pri1	0	0	100%

Release Cause Statistics	
Normal Call Clearing	0
Call Reject	0
User Busy	0
No User Response	0
No Circuit Available	0
Unassigned Number	0
Normal, Unspecified	0
Others	0

Normal Call Clearing(100%)

Refresh

Table 2-2-5 PRI Description of PRI call statistics

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.

2.2.6 SIP Call Statistics

Figure 2-2-6 SIP Trunk Call Statistics

SIP Trunk Call Statistics		
SIP Trunk No.	Trunk Name	Current Calls
0	3cx.sip	0
1	elastix.sip	0
2	dag.sip	0

Refresh

Table 2-2-6 Description of SIP Call Statistics

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 Network

Figure 2-3-1 Network Configuration

Network Configuration

Service Ethernet Interface(FE0)

Obtain IP address automatically

Use the following IP address

IP Address

Subnet Mask

Default Gateway

PPPoE

Account

Password

Service Name

Management Ethernet Interface(FE1)

IP Address

Subnet Mask

DNS Server

Obtain DNS server address automatically

DNS Server

Primary DNS Server

Secondary DNS Server

Table 2-3-1 Description of Network Configuration

Service Ethernet Interface (FE0)	Obtain IP address automatically	If Selected, the TG will obtain IP address via DHCP
	Use the following IP address	If Selected ,Set a static IP for Service Ethernet Interface .Need to fill the IP address, Subnet Mask, and Default Gateway
	PPPoE	If users approach the net via PPPoE, please Select it and fill your account and password.
Management Ethernet Interface	IP address	Fill the IP address of FE1
	Subnet mask	Fill the subnet mask of FE1
DNS Server	Obtain DNS server address automatically	If selected, the TG will obtain DNS server IP address via DHCP
	Use the following DNS server addresses	If selected, you need fill Primary DNS server addresses, the secondary DNS Server is optional.

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.

2.4 PRI Config

2.4.1 PRI Parameter

Figure 2-4-1 PRI Parameter

PRI Parameter

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

Reset to default configuration Reset

Save

Table 2-4-1 Description of PRI Parameter

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

2.4.2 PRI Trunk

Figure 2-4-2 PRI Trunk

PRI Trunk								
	Trunk No.	Trunk Name	Channel ID	D-Channel	E1/T1 Port No.	Protocol	Switch Side	Alerting Indication
<input type="checkbox"/>	0	pri0	0	Enable	0	ISDN	User Side	ALERTING
<input type="checkbox"/>	1	pri1	0	Enable	1	ISDN	Network Side	ALERTING

Click “Add” to add a PRI Trunk. If user want to delete or modify a PRI Trunk, please select the PRI Trunk user want to do.

Figure 2-4-3 PRI Trunk Add

PRI Trunk Add

Trunk No.	<input type="text" value="0"/>
Trunk Name	<input type="text"/>
Channel ID	<input type="text"/>
D-Channel	<input type="text" value="Enable"/>
E1/T1 Port No.	<input type="text" value="0"/>
Protocol	<input type="text" value="ISDN"/>
Switch Side	<input type="text" value="User Side"/>
Alerting Indication	<input type="text" value="ALERTING"/>
PSTN Profile ID	<input type="text" value="0 <Default>"/>

Table 2-4-2 Description of Add PRI Trunk

Trunk No	The number of PRI trunk; when user add PRI trunk, 0~7 number will appear in the pull-down menu to be selected (the number here depends on E1/T1 physical port number actually existed in equipment). After trunk number is established, filling in corresponding port number in “E1/T1 Port No.,” so as to assign E1/T1 to designated trunk; Each PRI trunk corresponds to a E1/T1 port.
Trunk Name	Description of PRI trunk
Channel ID	Channel ID of E1/T1 ports, this number definition generally starts from 0.
D-channel	Indicate whether E1/T1 supports D channel, the default is Yes.
E1/T1 Port No	E1/T1 port number is numbered according to the physical position of E1/T1, it generally starts from 0.
Protocol	Interface type of PRI. There are two types are available: ISDN and QSIG; the default is ISDN.
Switch Side	Indicate PRI network property of E1/T1, it is divided into: “User side” and “Network side”. When PRI loopback is carried out, the network properties of E1/T1 port at both receiving and sending sides must be different.

Alerting Indication	The ring signal include Alerting and Progress
---------------------	---

2.6 SS7 Config (optional)

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

Figure 2-6-1 Add PRI Trunk



2.6.1 SS7 Trunk

Figure 2-6-2 SS7 Trunk

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

Figure 2-6-3 SS7 Trunk Add

SS7 Trunk Add

Select Trunk No.

Trunk Name

Protocol

Protocol Type

SPC Format

OPC

DPC

Network Indicator

Sending SLTM

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

Notes:

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

SS7 trunk add

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

Note:

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

2. If protocol standard chose 'ITU', and then the SPC length is 14 bits.
3. SPC length performance on the OPC/DPC structure; SPC pattern instructions of the different structure OPC/DPC input formats.
4. When the SPC length is 24 bits, and chosen ITU, OPC/DPC structure format is :x-y-z; x、y、z is a number of 0-255, such as: 22-222-77
5. When the SPC length is 24 bits, and chosen Hex, OPC/DPC structure format is :xyz; x、 y、 z must be Hex number of 00-FF, such as: 33AA55
6. When the SPC length is 24 bits, and chosen ITU, OPC/DPC structure format is : x-y-z; x、 z must be decimal value; y is decimal number 0-255, such as: 6-222-3
7. When the SPC length is 24 bits, and chosen Hex, OPC/DPC structure format is :xyz; x、 z must be three bits hex value; y is 8 bits hex value, such as: (202E) 100 00000101 110

2.6.2 SS7 MTP Link

Figure 2-6-4 SS7 MTP Link

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

Figure 2-6-5 SS7 MTP Link Add

SS7 MTP Link Add

Trunk No.	<input type="text"/>
Link No.	<input type="text" value="0"/>
Signaling Link Code	<input type="text"/>
E1/T1 Port No.	<input type="text" value="0"/>
Channel No.	<input type="text" value="16"/>

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

SS7 MTP link description

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

2.6.3 SS7 Circuit

Figure 2-6-5 SS7 Circuit

SS7 Circuit				
Trunk No.	E1/T1 Port No.	Start Channel	Start CIC No.	Count
---	---	---	---	---

Figure 2-6-6 SS7 Circuit description

SS7 Circuit Add

Trunk No.

E1/T1 port No.

Start Channel

Start CIC No.

Count

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

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

SS7 Circuit Add

Trunk No	The “Trunk No.” here corresponds to the “Trunk No.” of SS7 trunk.
E1/T1 port No	Fill in the port number of E1/T1. Assign E1/T1 to selected SS7 trunk.

Start Channel	The start of SS7 channel trunk
Start CIC No	An initial circuit number to this E1/T1 matches by both parties
Count	A total of 32 channels

2.6.4 SS7 Circuit Maintain

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

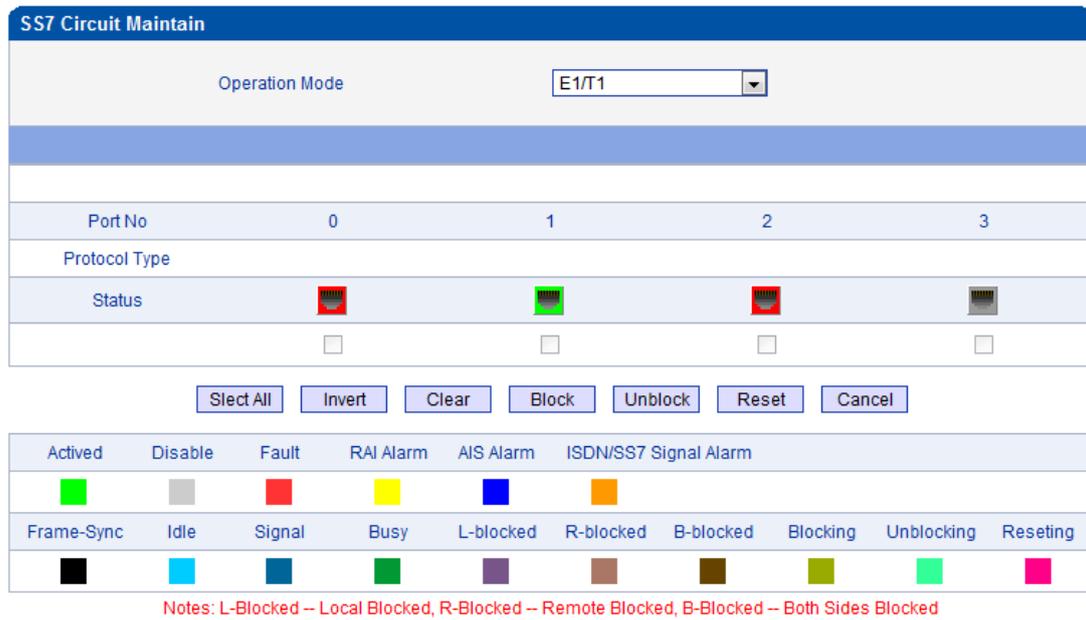


Figure 2-6-7 SS7 Circuit Maintain-E1/T1

SS7 Circuit Maintain-E1/T1 description

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

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

SS7 Circuit Maintain

Operation Mode Channel

Current Port Status Protocol Type undefined

Channel	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
CIC No.																
Status																
	<input type="checkbox"/>															

Channel	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
CIC No.																

Actived	Dis...	Fault	RAI A...	AIS AI...	ISDN/SS7 Sig...				
■	■	■	■	■	■				
Frame-...	Idle	Signal	Busy	L-blo...	R-blo...	B-blo...	Block...	Unbl...	Rese...
■	■	■	■	■	■	■	■	■	■

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(optional)

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	
<input type="checkbox"/>	0	ITU	01	A-1	A-5	A-5	I-15	I-15	A-3	Call with charge
<input type="checkbox"/>	1	Argentina	01	A-1	A-5	A-5	INVALID	I-12	A-3	Call with charge
<input type="checkbox"/>	2	Brazil	01	A-1	A-5	A-5	INVALID	I-15	A-3	Call with charge
<input type="checkbox"/>	3	China	11	A-1	A-1	A-6	INVALID	I-15	A-3	Call with charge
<input type="checkbox"/>	4	Czech	01	A-1	A-5	A-5	I-15	I-15	A-3	Call with charge
<input type="checkbox"/>	5	Colombia	01	A-1	A-5	A-5	I-15	I-15	A-3	Call with charge
<input type="checkbox"/>	7	Mexico	01	A-1	INVALID	INVALID	I-15	I-15	INVALID	Call with charge
<input type="checkbox"/>	8	Philippines	01	A-1	A-5	A-5	INVALID	I-15	A-3	Call with charge
<input type="checkbox"/>	9	Venezuela	01	A-1	A-9	A-5	INVALID	I-15	A-3	Call with charge
<input type="checkbox"/>	11	Bolivia	01	A-1	A-5	A-5	I-15	I-15	A-3	Call with charge
<input type="checkbox"/>	14	India	01	A-1	A-4	A-5	INVALID	I-10	A-3	Call with charge
<input type="checkbox"/>	15	Indonesia	01	A-1	A-6	A-6	I-15	I-15	A-3	Call with charge
<input type="checkbox"/>	16	Korea	01	A-1	A-5	A-5	INVALID	I-15	A-3	Call with charge
<input type="checkbox"/>	17	Malaysia	01	A-1	A-6	A-6	I-15	I-15	A-3	Call with charge
<input type="checkbox"/>	18	Panama	01	A-1	A-5	A-5	INVALID	I-15	A-3	Call with charge
<input type="checkbox"/>	19	Singapore	01	A-1	A-6	A-6	I-15	I-15	A-3	Call with charge
<input type="checkbox"/>	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 MTG200. Description says the state name, means the different countries supported R2 parameters standards. According to demands add R2 parameters of user countries.

R2 Param 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
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 A	C-2
Address Complete	C-3
Network Congestion	C-4
Request next DNIS and change back to Group A	C-5
Request Last DNIS and change back to Group A	C-6

Figure 2-7-2 R2 Parameter Add

Parameter Description

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

2.7.2 R2 Trunk

R2 Trunk				
	Trunk No.	Trunk Name	E1 Port No	ParamId
<input type="checkbox"/>	0	R2	0	3 <China>
<input type="checkbox"/>	1	R2	1	0 <ITU>
<input type="checkbox"/>	2	R2	2	3 <China>

Figure 2-7-3 R2 Trunk

R2 Trunk Add	
Trunk No	<input type="text" value="3"/>
Trunk Name	<input type="text"/>
E1 Port No	<input type="text" value="3"/>
Protocol Param	<input type="text" value="0 <ITU>"/>

Figure 2-7-4 R2 Trunk Add

R2 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

Figure 2-8-1 E1/T1 Parameter

Port No.	Work Mode	PCM Mode	Frame Mode	Line Code	Line Built Out
0	E1	A LAW	CRC-4	HDB3	Short Haul(-10DB)
1	E1	A LAW	CRC-4	HDB3	Short Haul(-10DB)

Modify the E1/T1 parameters:

Figure 2-8-2 E1/T1 Parameter

NOTE: The device must restart to take effect.

Table 2-8-1 Description of Modify E1/T1 Parameter

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

2.8.2 Coder Group

Figure 2-8-3 Coder Group

Coder Group

Coder Group ID
0(default setting)

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

Save

Table 2-8-2 Description of Coder Group

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

2.8.3 Dial Plan

Figure 2-8-4 Dial Plan

Dial Plan

Dial Plan ID
0

	Index	Prefix	Min Length	Max Length
<input type="checkbox"/>	0	.	0	30

Total: 1 Page 1

Add
Delete
Modify

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:

Table 2-8-3 Description of Dial Plan

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

2.8.4 Dial Timeout

Figure 2-8-4 Dial Timeout

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

Total: 1 Page 1

Figure 2-8-5 Dial Timeout Add

Dial Timeout Add	
Dial Timeout ID	<input type="text" value="1"/>
Description	<input type="text"/>
Max Time for Collecting Prefix	<input type="text"/> s
Time to Reach Min Length(after Prefix)	<input type="text"/> s
Time to Reach Max Length(after Min Length)	<input type="text"/> s

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

Table 2-8-4 Description Dial Timeout Add

Dial Time ID	The number to identify a dial timeout rule
Description	Description of dial timeout
Max Time for Collecting Prefix	Generally refer to the time from user dial first digit to harvest in prefix number.

Time to Reach Min Length(after Prefix)	After receiving prefix number, the number has not yet reached the length of the minimum receiving number, the length of timeout
Time to Reach Max Length(after Min Length)	After receiving number, the number has reached the minimum length, but not reached the maximum length of the dial timeout

2.8.5 PSTN Profile

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

Figure 2-8-6 PSTN Profile

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

Total: 1 | Page 1 |

Figure 2-8-7 PSTN Profile Add

PSTN Profile Add

PSTN Profile ID:

Description:

Coder Group ID:

RFC2833 Payload Type:

DTMF Tx Priority 1st:

DTMF Tx Priority 2nd:

DTMF Tx Priority 3rd:

Overlap Receiving:

Remove CLI:

Play Busy Tone to PSTN:

Table 2-8-5 Description of Add PSTN Profile

PSTN Profile ID	The number to 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 enable this feature, "Dial plan" and "Dial timeout" have the meaning
Remove CLI	Default does not remove CLI
Play busy tone to PSTN	Equipment will play busy tone from IP to PSTN

2.9 SIP Config

2.9.1 SIP Parameter

Figure 2-9-1 SIP Parameter

SIP Parameter

Local SIP Port

Local Domain

SIP port number and domain name would be allowed to set to different ports and domain name.

2.9.2 SIP Trunk

Figure 2-9-2 SIP Trunk

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	IP Profile ID
---	---	---	---	---	---	---	---	---	---	---	---	---

Total: 0

Figure 2-9-3 SIP Trunk Add

SIP Trunk Add

Trunk No.

Trunk Name

Remote Address

Remote Port

Local Domain

Get Callee from

Register to Remote

IP Profile ID

Incoming SIP Authentication Type

IP to PSTN Calls Restriction

PSTN to IP Calls Restriction

IP to PSTN Time Restriction

Detect Trunk Status

Enable SIP Trunk

Table 2-9-1 Description of Add SIP Trunk

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

2.10 IP Group Config

2.10.1 IP Profile

Figure 2-10-1 IP Profile

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	
<input type="checkbox"/>	0	Default	Yes	Yes	Local	Local	No	X-Fax

Total: 1 Page 1

Figure 2-10-2 IP Profile Add

IP Profile Add

IP Profile ID	<input style="width: 90%;" type="text" value="1"/>
Description	<input style="width: 90%;" type="text"/>
Declare RFC2833 in SDP	<input style="width: 90%;" type="text" value="No"/>
Support Early Media	<input style="width: 90%;" type="text" value="Yes"/>
Ringback Tone to PSTN Originated from	<input style="width: 90%;" type="text" value="Local"/>
Ringback Tone to IP Originated from	<input style="width: 90%;" type="text" value="Local"/>
Wait for RTP Packet from Peer	<input style="width: 90%;" type="text" value="No"/>
T.30 Expanded Type in SDP	<input style="width: 90%;" type="text" value="X-Fax"/>

Table 2-10-1 Description of Add IP Profile

IP Profile ID	The number to mark the IP Profile
Description	Description of the PSTN Profile
Declare RFC2833 in SDP	Support by default
Support Early Media	Whether support Early Media(183)
Ringback Tone to PSTN originated from	IP-> PSTN call ring back tone player side, if set to local, it will play from the equipment and set to IP , it will play by the called
Ringback Tone to IP originated from	PSTN->IP call ring back tone player side, if set to local, it will play from the equipment and set to PSTN, it will play by the called
Wait for RTP Packet from Peer	If set to No, will auto send RTP packets during the call, if set to Yes, will wait the RTP packet was sent by the opposite end first ,then send out RTP packets
T.30 Expanded Type in SDP	T30 extended types in SDP: Huawei/ZTE

2.11 Voice & Fax

Figure 2-11-1 Voice & Fax Configuration

Voice & Fax Configuration

Voice Parameter

Disconnect call when no RTP packet Yes No

Period without RTP packet s

Gain from PSTN

Gain to PSTN

Timeout of No Answer

Call from PSTN s

Call from IP s

Fax Parameter

Fax Mode

Fax Tx Gain

Fax Rx Gain

Packet time ms

Redundant frame in packet

Data & Fax Control

Data

Fax

DTMF Parameter

Continuous time ms

Signal interval ms

Threshold for detection

Save

Table 2-11-1 Description of Voice & Fax

Voice Parameter	Disconnect Call when no RTP packet	When selected "Yes", detected call's silence time longer than silence timeout that for a long time not received RTP packets, then hangup the call.
	Period without RTP packet	The maximum time length of silence
	PSTN in Gain	Incoming PSNT gain
	IP in Gain	Incoming IP gain
Timeout of no answer	Call from PSTN	Call timeout of no answer from PSTN
	Call from IP	Call timeout of no answer from IP
Fax Parameter	Fax Mode	Two modes are provided: T.38/Pass-through; default option is T.38.
	Fax Tx Gain	Gain of sending a fax

	Fax Rx Gain	Gain of receiving a fax
	Packet time	Data packing duration
	Redundant frame in packet	The length of frame in RTP packet
Data & Fax Control	Data	Whether to allow the control of voice data
	Fax	Whether to allow the control of fax
DTMF Parameter	Continuous time	The level of a frequency duration
	Signal interval	The time interval between two different frequency signals
	Threshold for detection	Frequency detection threshold

2.12 Maintenance

2.12.1 Management Parameter

Figure 2-12-1 Management Parameter

Table 2-12-1 Description of 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”. If select “Yes”, users will set syslog server address and syslog level.
Server Address	Address for saving system log

Syslog Level	None, Debug, Notice, Warning, Error. Please choose the file you want to output information level.
Send CDR	Whether send Call Detail Record through syslog
Qos Type	There are three options: none, TOS and DS. TOS only supports IPv4.
NTP Enable	Simple Network Management Protocol is enabled or not; the default is Yes.
Primary NTP server Address	The Primary IP address of SNMP management host computer. The host computer of the IP address will carry out monitoring and management to equipment.
Primary NTP server Port	The port that managed device provides trap message (it is generally alarm message) to SNMP management host computer, the default is 123.
Secondary NTP server Address	The Secondary IP address of SNMP
Secondary NTP server Port	The port of the Secondary IP address of SNMP
Sync Interval	Time interval of check
Time Zone	The time zone of local

2.12.2 Data Restore

Figure 2-9-2 Data Backup

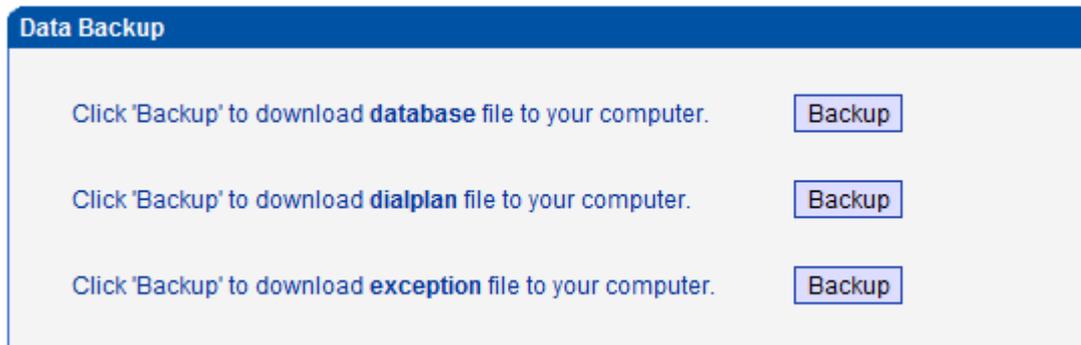


Table 2-12-2 Description of Data Backup

database	Click the Backup , and save the database in your PC
dialplan	Click the Backup , and save the dialplan in your PC
exception	Click the Backup , and save the exception in your PC

2.12.3 Data Restore

Figure 2-12-3 Data Restore

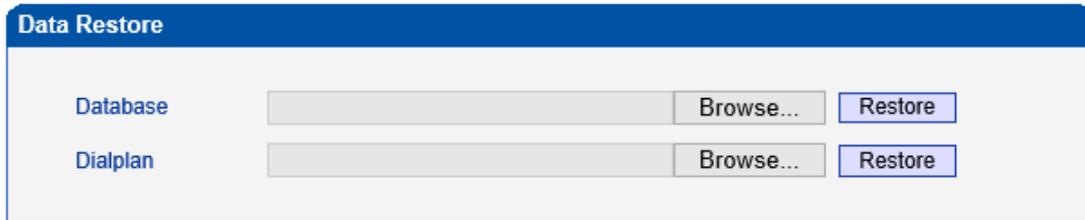


Table 2-12-3 Description of Data Restore

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

2.12.4 Version Information

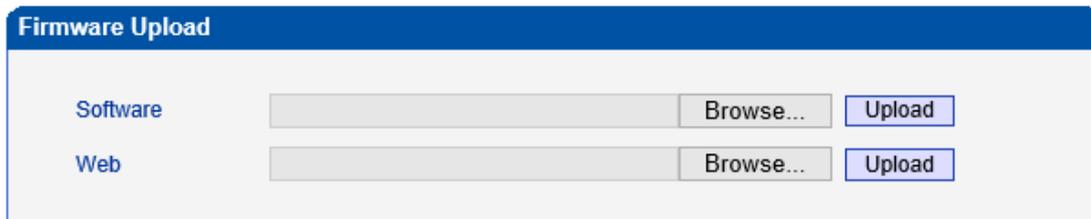
Figure 2-12-4 Version Information

Version Information			
File Type	Version	Date Built	Time Built
Software	2.01.04.03	2013-10-10	15:09:24
Database	2.03.01	2013-06-17	19:32:00
Web	2.01.04.03	2013-10-08	15:56:21

Here users can view software, database and web version information.

2.12.5 Firmware Upload

Figure 2-12-5 Firmware Upload



The process of firmware upload:

- 1) Click "Firmware Upload"
- 2) Browse files and choose the loading program (Name the file extension. ldf)
- 3) Click "Upload", the upload process will last about 60s and device can automatically restart after uploading. (The firmware update process don't shut off the power) .

2.12.6 Password Modification

Figure 2-9-6 Password Modification

The screenshot shows a web interface titled "Password Modification" with a blue header. Below the header, there are three text input fields labeled "Old Password", "New Password", and "Confirm Password". A "Save" button is located below the input fields.

The configuration items are used to change the login password of web configuration.

2.12.7 Device Restart

Figure 2-9-7 Device Restart

The screenshot shows a web interface titled "Device Restart" with a blue header. Below the header, there is a text prompt: "Click the button below to restart the device". A "Restart" button is located below the text.

Some configuration need to restart device to take effect. Click "Restart" to restart the device.

3. Glossary

PRI: Primary rate interface

FMC: Fixed Mobile Convergence

SIP: Session Initiation Protocol

DTMF: Dual Tone Multi Frequency

PSTN: Public Switched Telephone Network

STUN: Simple Traversal of UDP over NAT

DMZ: Demilitarized Zone

SS7: Signaling System No. 7

ISDN: Integrated Services for Digital Network

SNMP: Simple Network Management Protocol

DSCP: Differentiated Services Code Point

OPC: Original Signaling Point Code

DPC: Destination Signaling Point Code

NGN: Next Generation Network

PBX: Private Branch Exchange

RTP: Real-time Transport Protocol

RTCP: Real-time Transport Control Protocol

STP: Signaling Transfer Point

TDM: Time Division Multiplex