MTG200 User Manual V1.0

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Amendment Records

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Contents

1. Overview

1.1 Product Introduction

MTG200 series is a kind of digital trunk gateway based on embedded operating system. It supports standard SIP protocol, with large-capacity carrier class telephone trunk gateway functions. Currently it supports 1/2/4 E1/T1 interfaces and can realize intercommunication with mainstream manufacturers soft switch system, and interwork with carrier's local Telephone exchange by PRI interface. A typical network diagram shows the function of MTG200 as below.

Figure 1-1-1Application topology

1.2 Equipment Structure

1.2.1 Rear View

Figure 1-2-1 MTG200 Rear View

Table 1-2-1 Description of MTG200 Rear View

PWR	The power interface. DC12V.1A
Port0-Port3	E1/T1 Port. There are 4E1 if you have buy a MTG200-4E1/T1
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

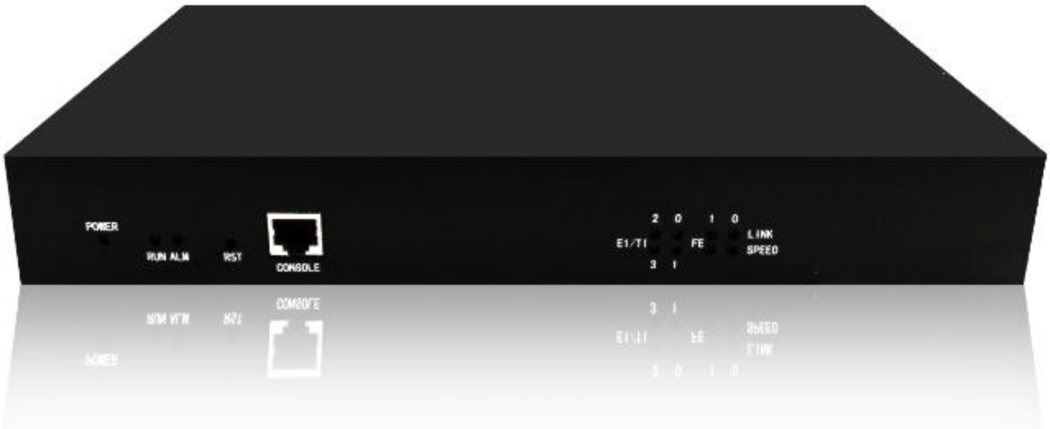


Table 1-2 -2 Description of MTG200 Front View

LED	Color	Name	Status	Description
POWER	Green	Power status indicator	Off	Power is off
			On	Power is on
RUN	Green	Register indicator	Fast blinking	Register
			Slow blinking	Unregister
ALM	Yellow	The failure of device indicator	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. 0, 1,2,3indicates the connection state of E1/T1 interfaces respectively			
LINK	Indicating the connection state of the network . 0 indicates FE0 and 1 indicates FE1			
SPEED	Yellow	Indicating the network bandwidth	Off	10Mbps
			On	100Mbps

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

- PLC,VAD and CNG
- 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℃(operational),-20~70℃(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

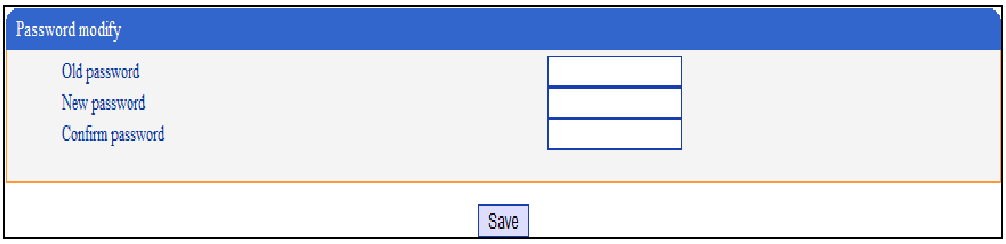
Enter the IP of FE1 or FE0 in customer's browser. FE1 default IP address is 192.168.11.1, FE0 default IP address is 192.168.1.111. It will request customer to input user name and password. Default user name and password are "admin".

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 172.16.33.62. 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 interface

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



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

2.2 Status &Statistics

Show the status of trunk and statistics of call.

2.2.1 System Information

This configuration page includes general information and version information.

Figure 2-2-1 System Information

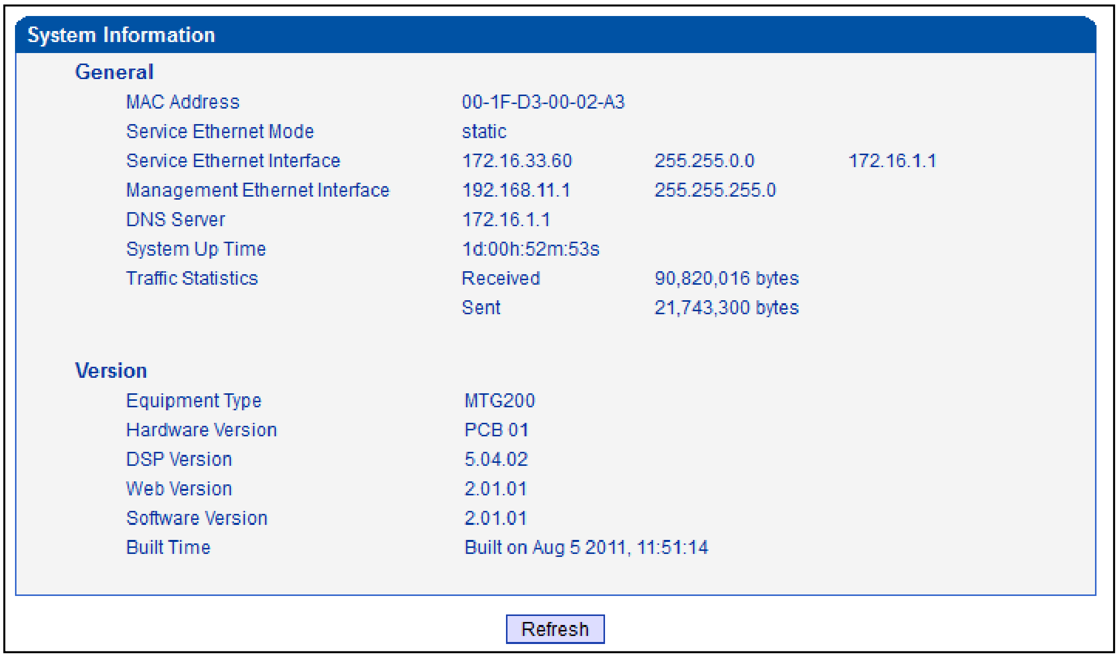


Table 2-2-1 Description of System Information

MAC address	MAC address of FE1 port.
Service Ethernet Mode	The network mode of FE1
Service Ethernet Interface	Include IP address、subnet mask、default gateway of FE1
Management Ethernet Interface	Include IP address、subnet mask of FE0
DNS Server	IP addresses of primary DNS server
System Up Time	Time elapsed from device power on to now
Traffic Statics	Total bytes of message received and sent by FE1 port
Equipment Type	Equipment type; this equipment is: MTG200
Hardware Version	Hardware version of device
DSP Version	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

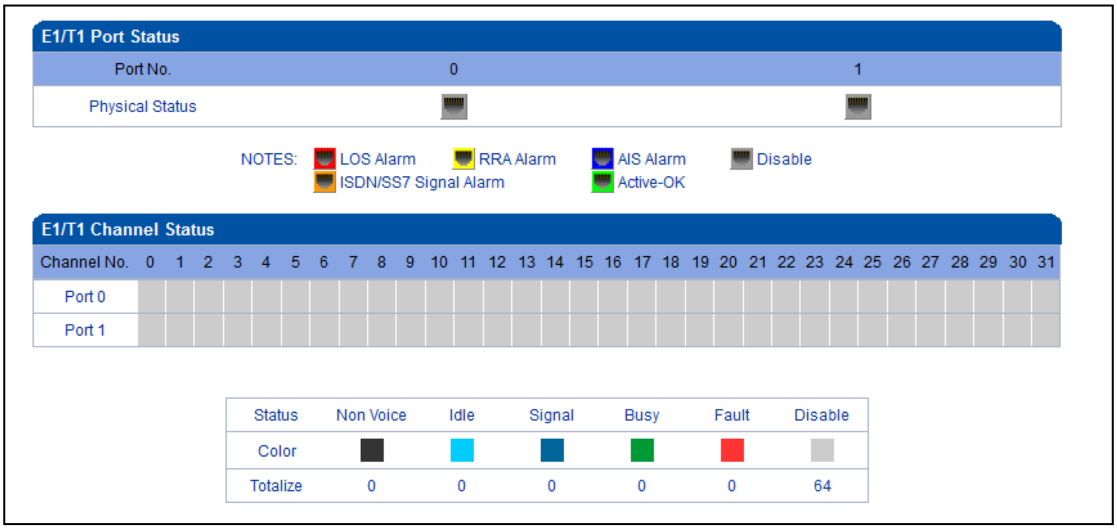


Table 2-2-2 Description of E1/T1 status

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.**RRA Alarm:** Receive Remote Alarm, when distant end detects LOS alarm or LFA alarm, it will insert an alarm message to near end device in transmit data, check the device of opposite terminal to see if it is perfect. 3.**AIS Alarm**: Alarm indicating; during a time interval, when received data is detected to have only 2 or less than 2 zeros, then AIS alarm is created, check line device. 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/T1 Channel Status	1.**Non Voice:** 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 5.**Fault:** when the channel is enabled and the cable is not connected. 6.**Disable:** Have not use this E1/T1 trunk

2.2.3 PSTN Trunk Status

Figure 2-2-3 PSTN Trunk Status

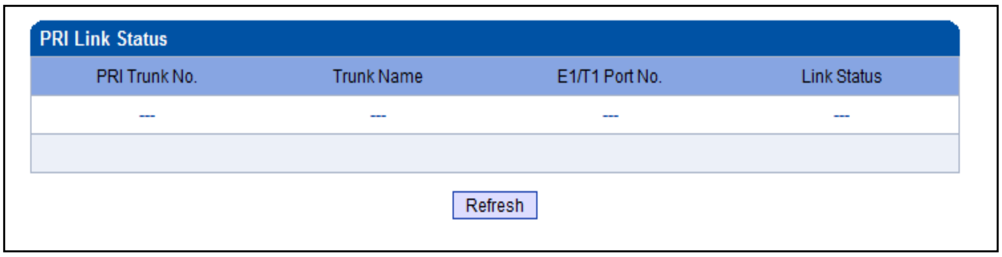


Table 2-2-3 Description of PSTN Trunk Status

PRI Trunk No	The number of PRI trunk, each trunk corresponds to a PRI link
Trunk Name	Identification of the trunk can be remembered easily.
E1/T1 Port 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 IPTrunk Status

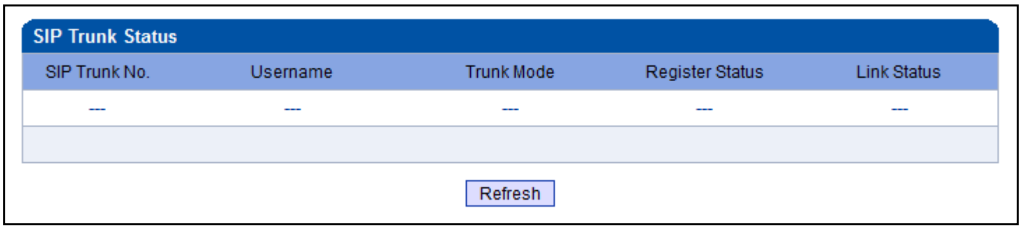


Table 2-2-4 Description of IP Trunk Status

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, peer is peer to peer mode, access is access mode
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 call statistics

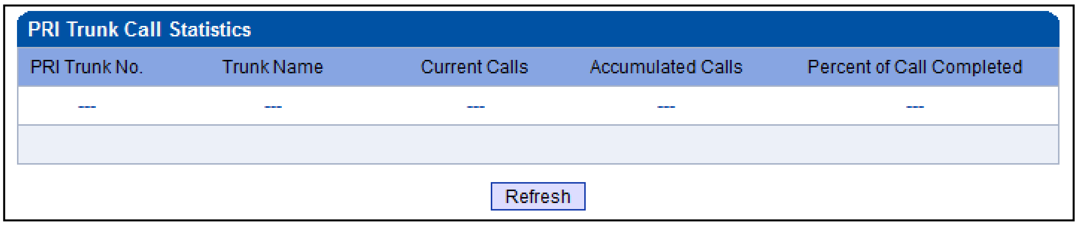


Table 2-2-5 Description of PRI call statistics

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.
Percent of Call Completed	The percent of calls completed in all calls.

2.2.6 SIP Call Statistics

Figure 2-2-6 SIP Call Statistics

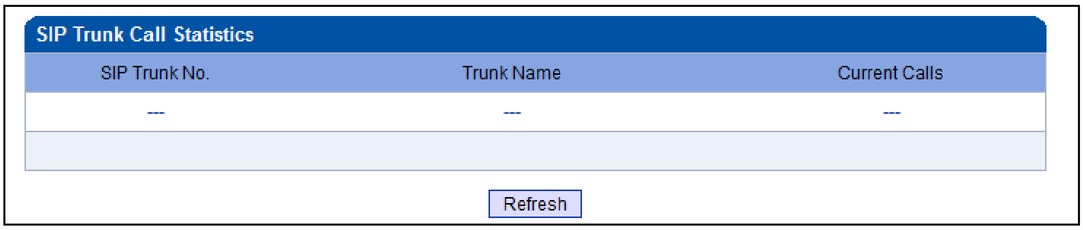


Table 2-2-6 Description of SIP Call Statistics

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 Configuration

Figure 2-3-1 Network Configuration

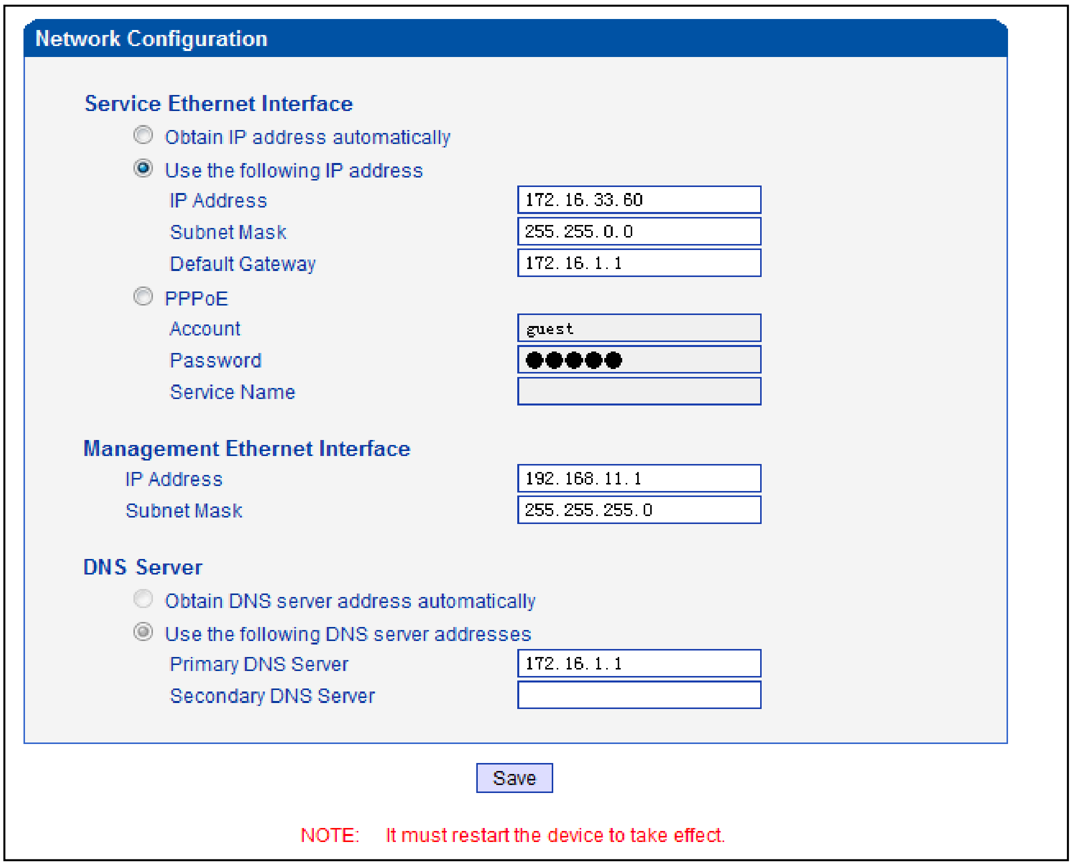


Table 2-3-1 Description of Network Configuration

Service Ethernet Interface	Obtain IP address automatically	If Selected, the MTG will obtain IP address via DHCP
	Use the following IP address	If Selected ,Set a static IP for Service Ethernet Interface .You 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 of Management Ethernet Interface
	Subnet Mask	Fill the Subnet Mask of Management Ethernet Interface
DNS Server	Obtain DNS server address automatically	If selected, the MTG 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

Note: FE0 port IP and FE1 port IP should be set in different segments. After configure the network address, and restart the gateway configuration to take effect.

2.4Voice & Fax Configuration

Figure 2-4-1 Voice & Fax Configuration

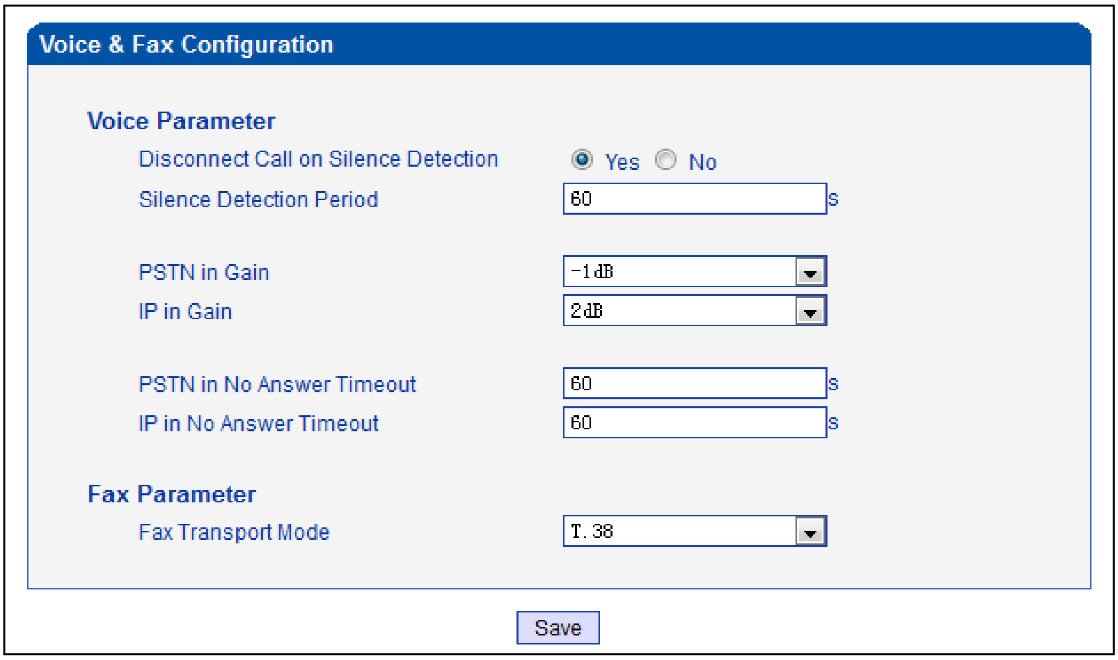


Table 2-4-1 Description of Voice & Fax Configuration

Disconnect Call on Silence Detection	Silence time out is detected, end of the call when selected "Yes"
Silence Detection Period	The interval time of silence detection
PSTN in Gain	Incoming PSNT gain
IP in Gain	Incoming IP gain
PSTN in No Answer Timeout	The no answer timeout of a call which make from PSTN
IP in No Answer Timeout	The no answer timeout of a call which make from VoIP
Fax Transport Mode	Two modes are provided: T.38 and Pass-through, default option is T.38

2.5 Protocol Configuration

2.5.1 PRI Parameter

Figure 2-5-1 PRI Parameter

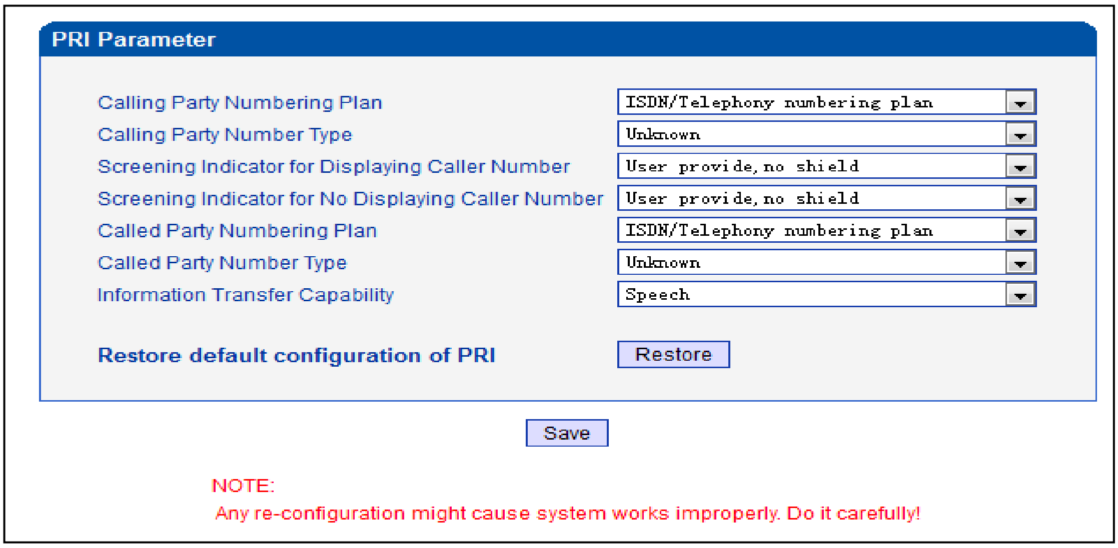


Table 2-5-1 Description of PRI Parameter

Calling Party Numbering Plan	Provide 6 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	6 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	4 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	4 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 6 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	6 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

2.5.2 SIP Parameter

Figure 2-5-2 SIP Parameter

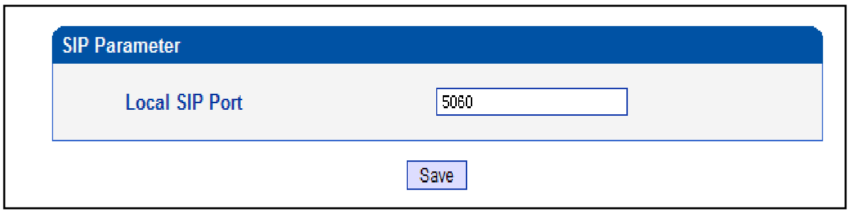


Table 2-5-2 Description of SIP Parameter

Local SIP Port	Local SIP monitoring port, the default is 5060

2.6 Profile Definitions

2.6.1 Coder Group

Figure 2-6-1 Coder Group

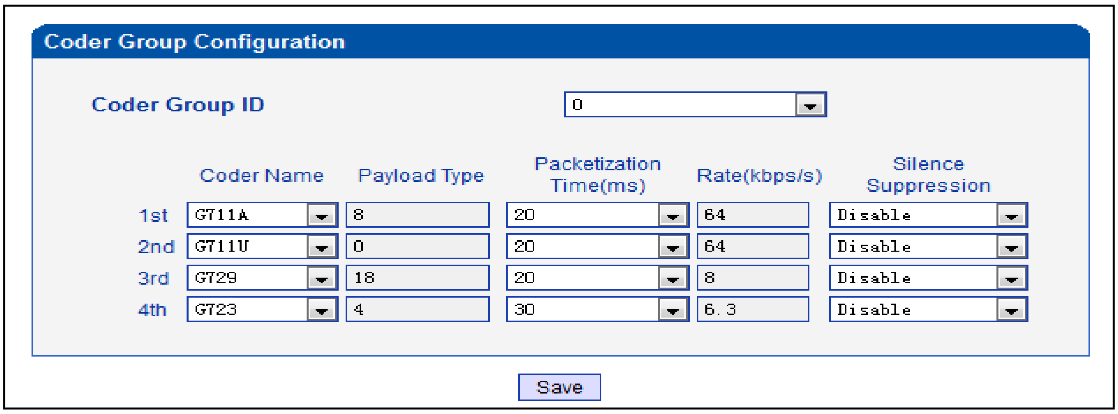
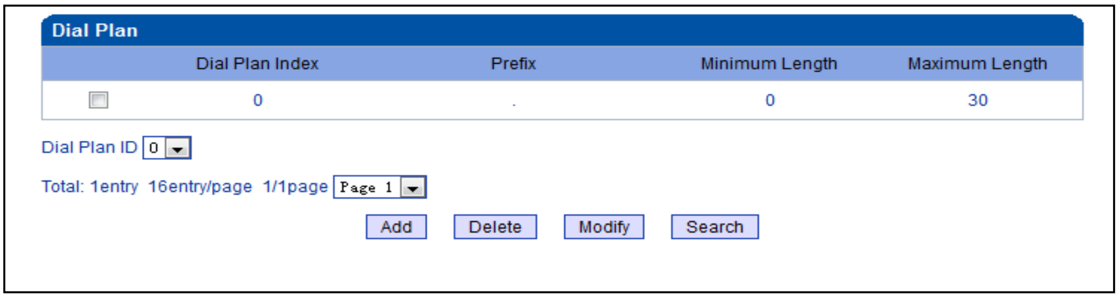


Table 2-6-1 Description of Coder Group

Coder Group	Used for configure the voice codec parameters, through it can configured voice capabilities into 8 groups, each group can have different audio capabilities, such as the priority of voice codec, packaging length and whether to support silence suppression
Coder Group ID	ID standard for Voice Ability, total with 8 groups, where 0 is the default group ID number, the codec that MTG equipment support in the grouping will be displayed in 0 group ,on the map only shows 4 kinds , mean MTG equipment only support this 4 codecs.
Coder Name	Support 4 kinds of audio codec. G711A/G711U/G729/G723/iLBC
Payload Type	Coder name is the interpretation of the field, each codec has a unique value, refer to RFC3551
Packetization Time(ms)	Voice Codec packetization time, you can define different kinds of coding and decoding, minimum packetization time
Rate(kbps/s)	The proportion of the data-stream that is useful
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.

2.6.2 Dial Plan

Figure 2-6-2 Dial Plan



Dial plan configuration used to receive numbers, you can configure different prefix number, these rules can be divided into 5 groups, separate 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, so when through the file into the rules, need put t long matching numbers (prefix) rule before the file
2. No maximum length is 30, this value is the number of the total length, including the prefix length, such prefix is 0755, the maximum value of Maximum Length is only 26, and "." Wildcard is not included in the number length

Figure 2-6-3 Add Dial Plan

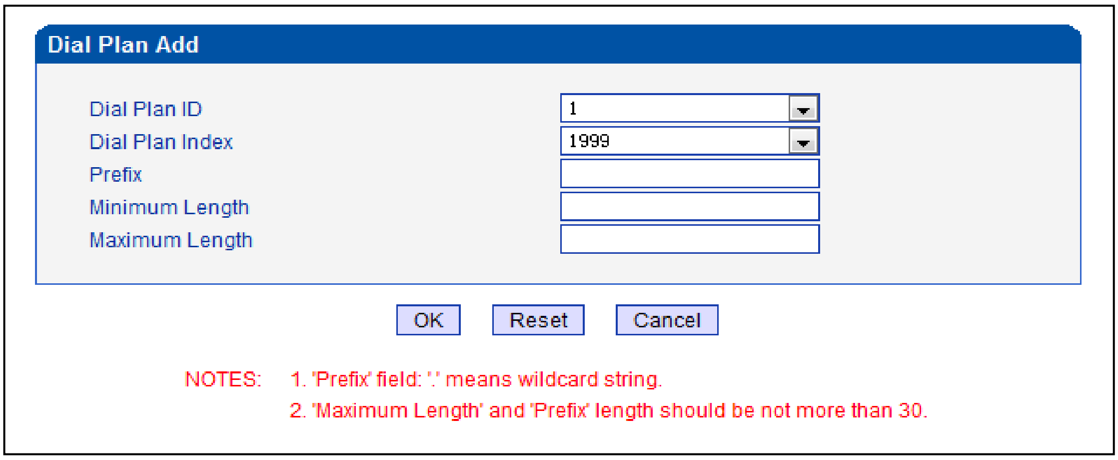


Table 2-6-2 Description of Add Dial Plan

Dial Plan ID	The number to identify a dial plan
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
Minimum Length	The minimum receiving Number length (0 to 30). If receive 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.
Maximum Length	The largest Received number length (0 to 30), the maximum number length that can be received, if the received number in this length, the system will determine the receiving number is completed, no longer continue to receive numbers, immediately begin numbers analyzed, If there are numbers keep to send over, the system will drop the new numbers dial plan rules can through management configuration-> Data Restore into dial plan, the file is a txt format.

2.6.3 Dial Timeout

Figure 2-6-4 Dial Timeout

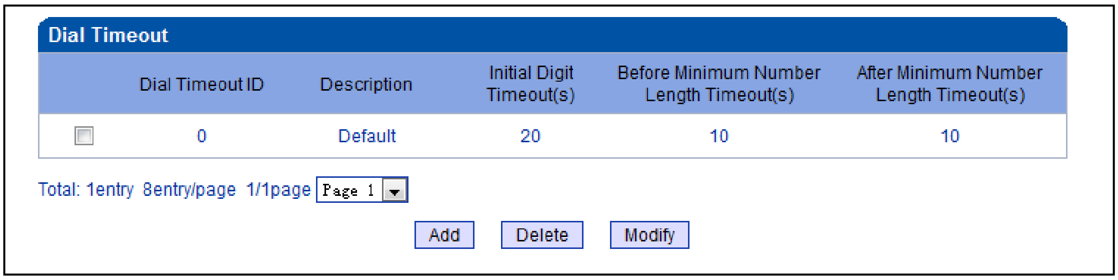


Figure 2-6-5 Add Dial Timeout

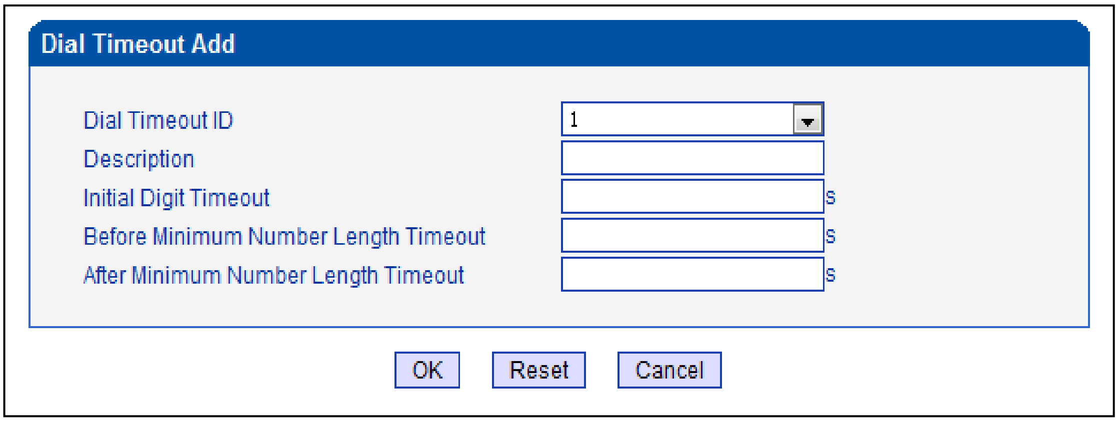
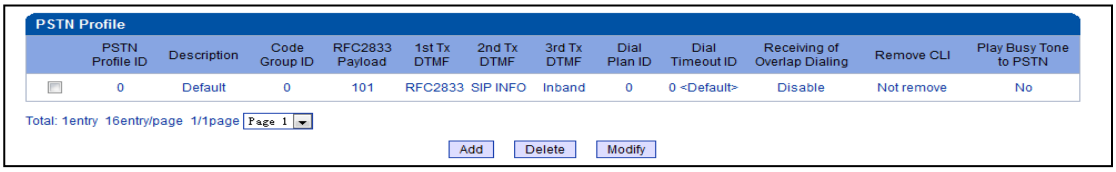


Table 2-6-3 Description of Add Dial Timeout

Dial timeout ID	The number to identify a dial timeout rule
description	Description of dial timeout
Initial Digit Timeout	Generally refer to the time from user dial first digit to harvest in prefix number
Before Minimum Number Length Timeout(S)	After receiving prefix number, the number has not yet reached the length of the minimum receiving number, the length of timeout
After Minimum Number Length Timeout(S)	After receiving number, the number has reached the minimum length, but not reached the maximum length of the dial timeout

2.6.4 PSTN Profile

Figure 2-6-6 PSTN Profile



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

Figure 2-6-7Add PSTN Profile

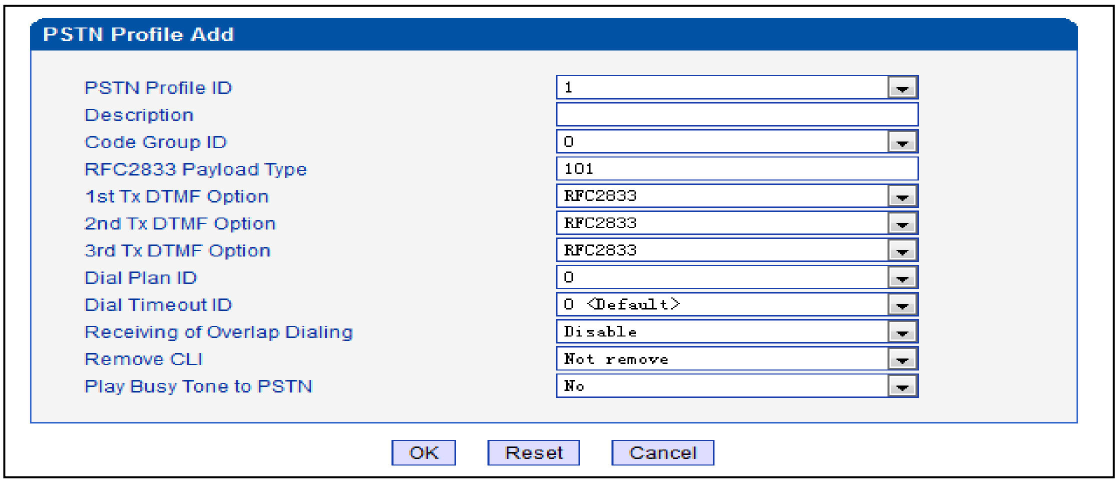


Table 2-6-4 Description of Add PSTN Profile

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
Dial Plan ID	Refer to "Dial Plan"
Dial Timeout ID	Refer to "Dial Timeout"
Receiving of Overlap Dialing	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

2.6.5 IP Profile

Figure 2-6-8 IP Profile

Figure 2-6-9 Add IP Profile

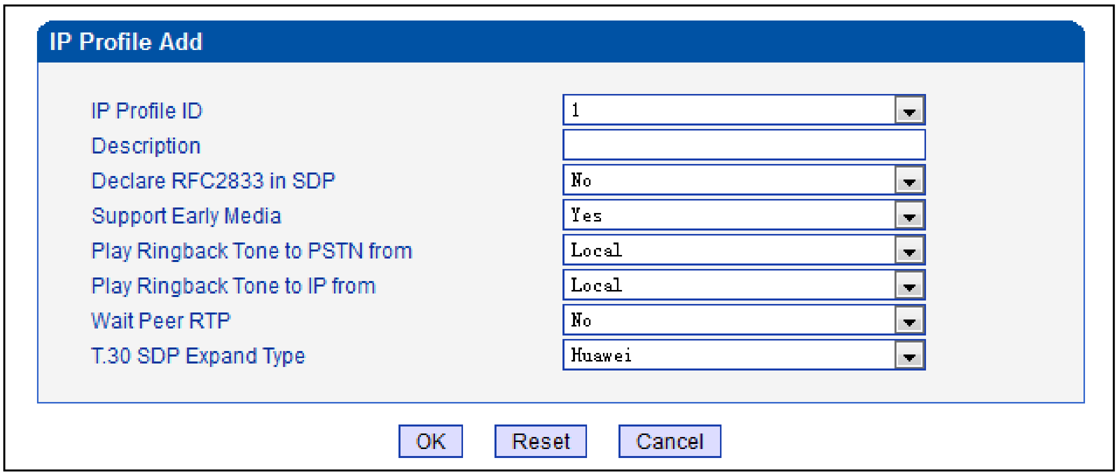


Table 2-6-5 Description of Add IP Profile

IP Profile ID	The number to mart the IP Profile
Description	Description of the PSTN Profile
Declare RFC2833 in SDP	Support by default
Support Early Media	Whether support Early Media(183)
Play Ringback Tone to PSTN from	I 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
Play Ringback Tone to IP 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 Peer RTP	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 SDP Expand Type	T30 extended types in SDP: Huawei/ZTE

2.7 Trunk Configuration

2.7.1 E1/T1 Parameter

Figure 2-7-1 E1/T1 Parameter

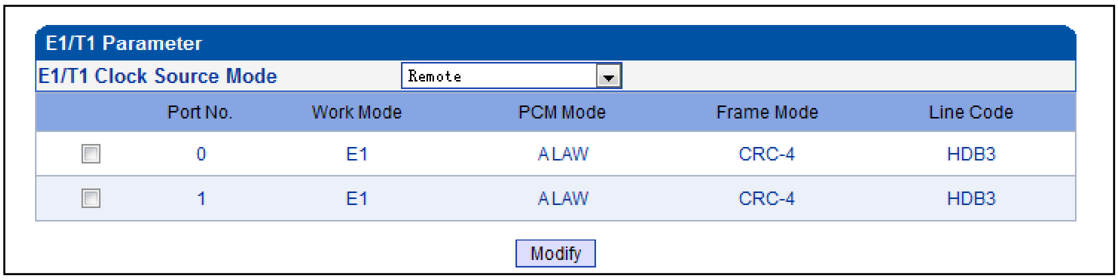


Figure 2-7-2 Modify E1/T1 Parameter

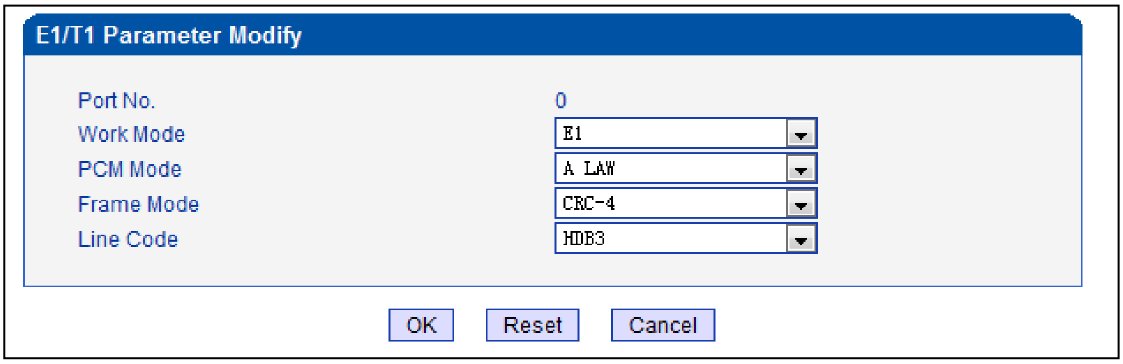
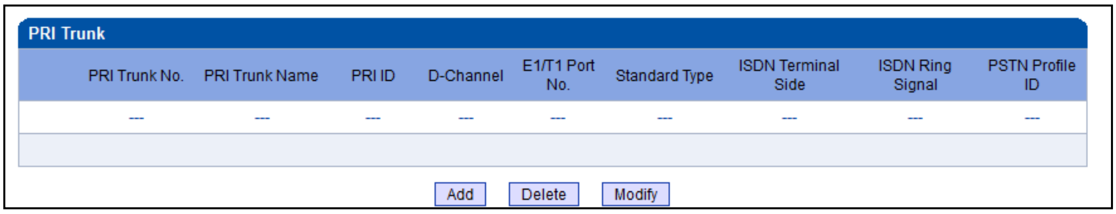


Table 2-7-1 Description of Modify E1/T1 Parameter

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

2.7.2 PRI Trunk

Figure 2-7-3 PRI Trunk



In this configuration page, users can “Add”, “Delete”, “Modify” PRI trunk.

Figure 2-7-4 Add PRI Trunk

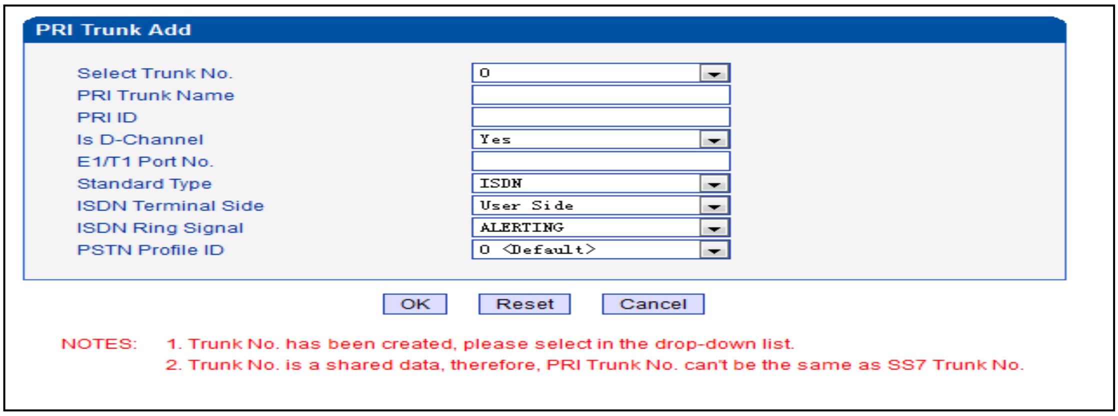
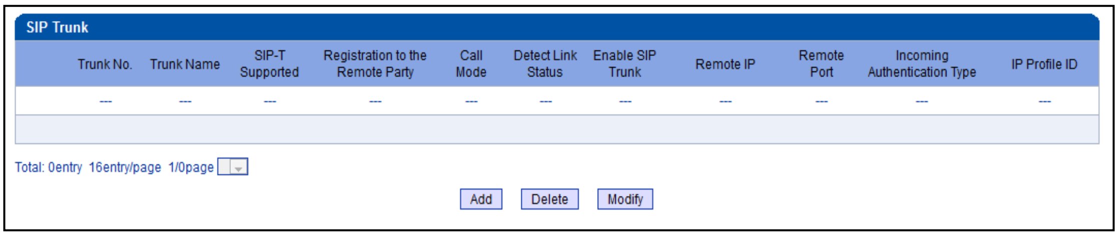


Table 2-7-1 Description of Add PRI Trunk

Select Trunk No	The number of PRI trunk; when you add PRI trunk, 0~7 number will appear in the pull-down box to be selected (the number here depends on E1/T1 physical port number actually existed in MTG). After trunk number is established, fill 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, if it is required to share D channel by several E1/T1, please refer to “PSTN trunk binding” in route configuration.
PRI Trunk Name	Identification of PRI trunk, which can be remembered easily
PRI ID	Identification of PRI trunk number to outside (switch side), this number definition generally begin from 0
Is D channel	Indicate whether this E1/T1 has D channel, the default is YES, which means it has D channel.
E1/T1 Port No	E1/T1 port number is numbered according to the physical position sequence of E1/T1, it generally begins from 0.
Standard Type	Interface type of PRI, two types available: ISDN and QSIG; the default is ISDN.
ISDN Terminal 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.
ISDN Ring signal	The ring signal include Alerting and progress
PSTN profile ID	Refer to PSTN profile

2.7.3 SIP Trunk

Figure 2-7-5 SIP Trunk



Click “Add” to add a SIP Trunk. If costomer want to delete or modify a SIP Trunk, please select the SIP Trunk you want to operation.

Figure 2-7-6 Add SIP Trunk

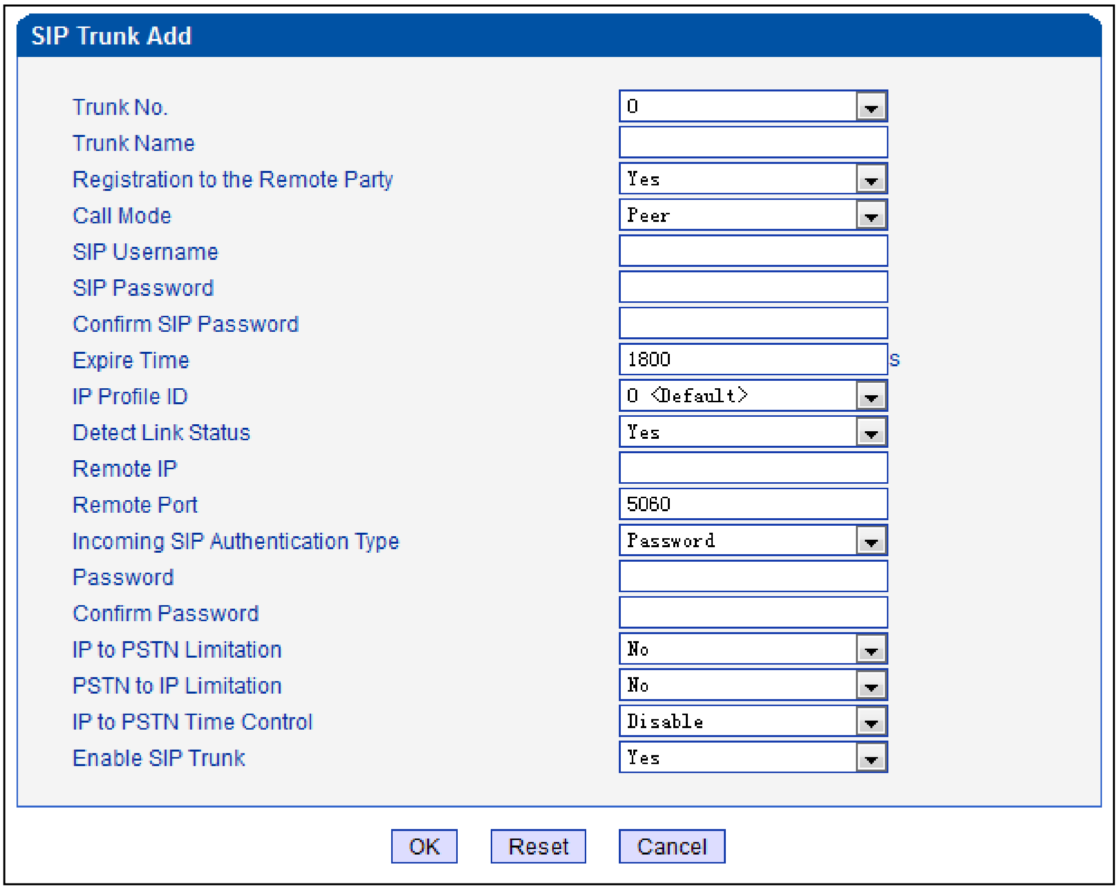


Table 2-7-3 Description of Add SIP Trunk

Trunk No	The range of number is 1~50
Trunk Name	It can be edited freely, which can be identified and remembered easily.
Registration to Remote Party	Defined by IETF work group RFC3372, it is a standard used to establish communication between SIP and ISUP; the default is "Yes"; if SIP trunk does not support, then set it to "No".
Call mode	Whether register request message to far-side equipment will be sent or not, you can select "Yes" or "No". There are two modes: peer and access
SIP Username	SIP user name which registers to soft switch/SIP server
SIP Password	SIP password which registers to soft switch/SIP server
Confirm SIP Password	Make sure the password matches the password entered above
Expire Time	Time interval of sending register request message to opposite equipment each time; the range is from 1-3600 seconds.
IP Profile ID	Refer to IP Profile
Detect Link Status	If select it, the MTG200 will send HEARTBEAT message to peer to make sure the link status is OK.
Remote IP	IP address of remote platform interfacing with this MTG.
Remote Port	Q.931 port of SIP of remote platform interfacing with this MTG, the default is 5060
Incoming SIP Authentication Type	You can select IP address authentication or password authentication, when "IP Address" authentication is selected, the calling initiated from remote will not subject to domain name or password authentication, only judge whether the IP address is legal or not; if "No" is selected, authentication realm/password authentication will be carried out.
Password	It constitutes SIP protocol safety authentication together with domain name of authentication.
Confirm Password	Input password again to verify password
IP to PSTN Limitation	IP to PSTN calls; the range is 0~65535, the default is no limitation; If Yes is selected, then input limited calls in the edit box appeared.
PSTN to IP Limitation	PSTN to IP calls, the range is 0~65535; the default is no limitation; If Yes is selected, then input limited calls in the edit box appeared.
IP to PSTN Time Control	The default setting is disabled. If Enabled is selected, then you 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)
Enable SIP Trunk	A switch used to enable this SIP trunk or not; you can select "Yes" or "No", when "No" is selected, this SIP trunk is invalid.

2.8 Management Configuration

2.8.1 Management Parameter

Figure 2-8-1 Management Parameter

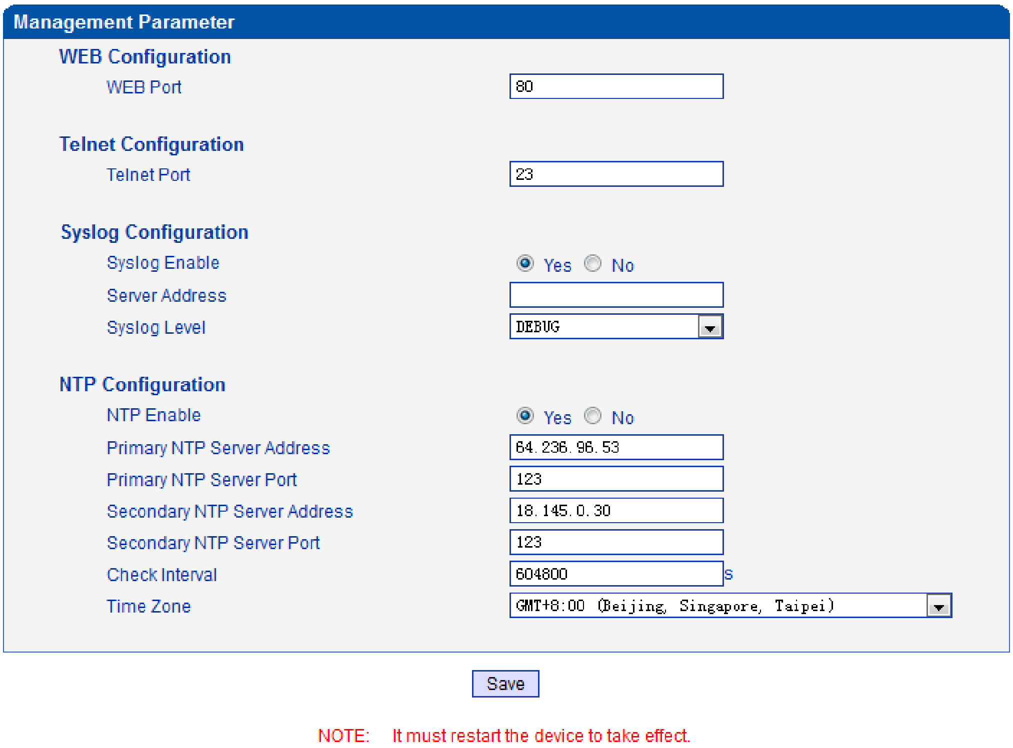


Table 2-8-1Description of Management Parameter

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”.
Server Address	Address for saving system log.
Syslog Level	None, Debug, Notice, Warning and Error
NTP Enable	Simple Network Management Protocol is enabled or not; the default is No.
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 MTG.
Primary NTP server Port	The port where managed device (MTG) 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
Check Interval	Time interval of check
Time Zone	The time zone of local

2.8.2 Data Backup

Figure 2-8-2 Data Backup

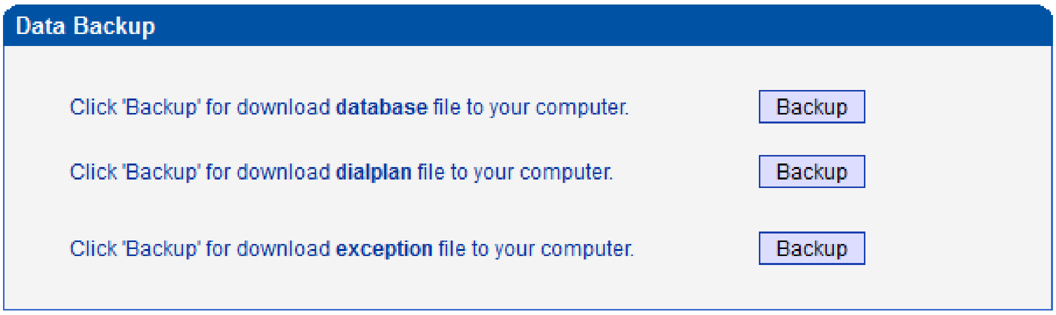


Table 2-8-2 Description of Data Backup

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

2.8.3 Data Restore

Figure 2-8-3 Data Restore

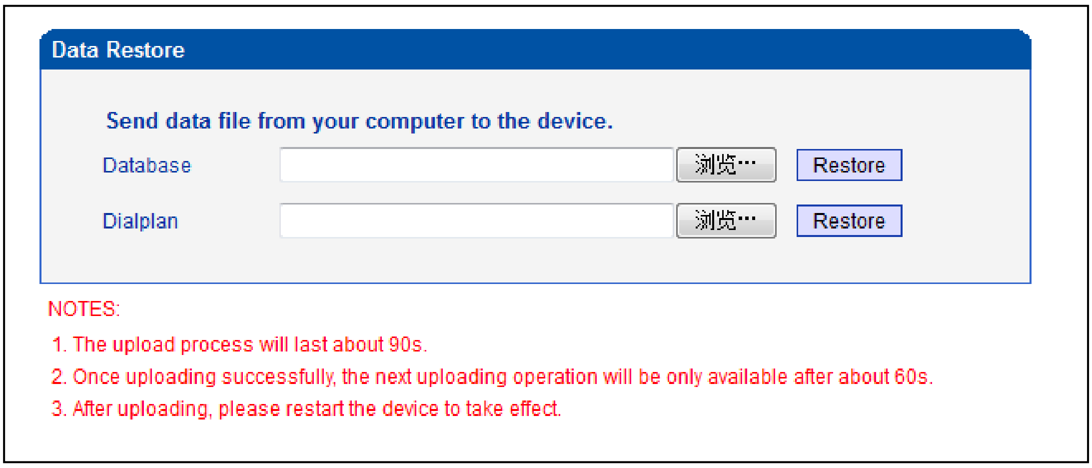


Table 2-8-3 Description of Data Restore

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

2.8.4 Version Information

Figure 2-8-4 Version Information

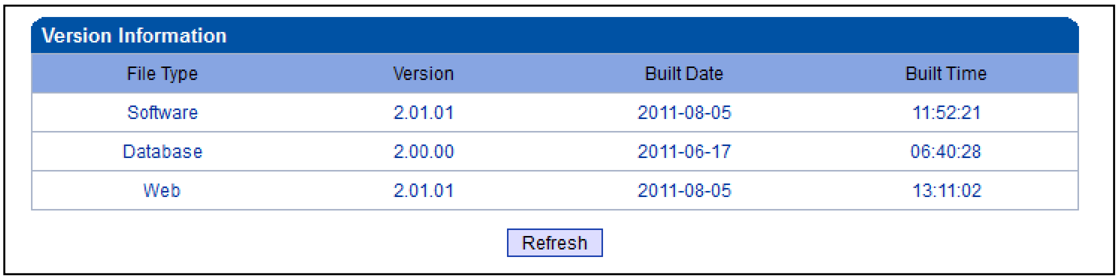


Table 2-8-4 Description of Version Information

Software	The information of firmware
Database	The information of database
Web	The information of Web software

2.8.5 Firmware Upload

Figure 2-8-5 Firmware Upload

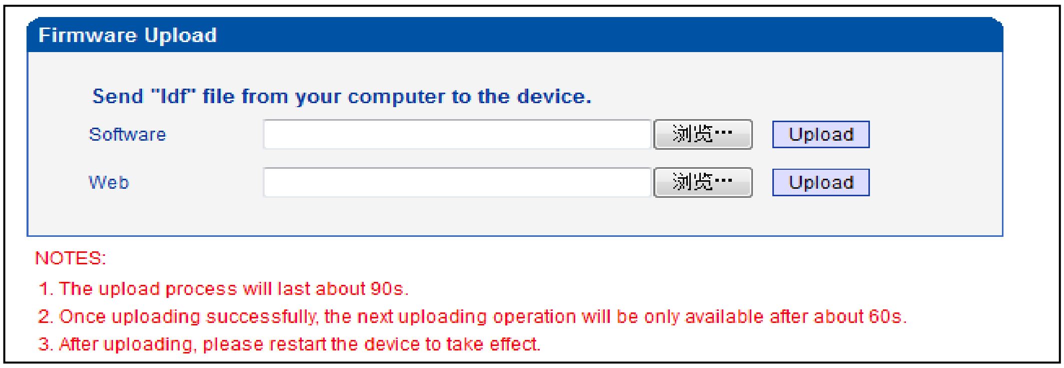


Table 2-8-5 Description of Firmware Upload

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

2.8.6 Modify Password

Figure 2-8-6 Modify Password

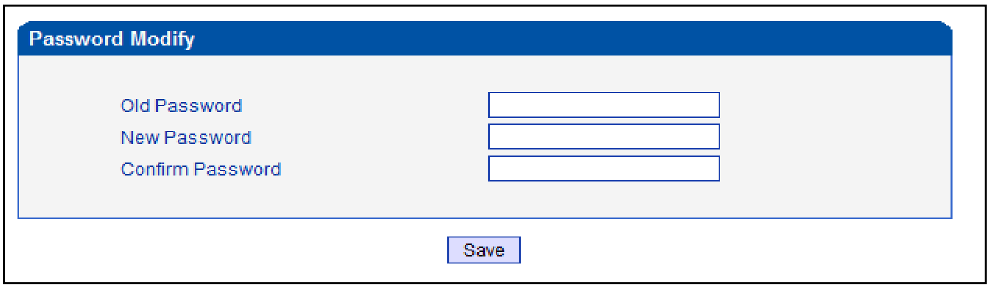


Table 2-8-6 Description of Modify Password

Old Password	Current password
New Password	The new password
Confirm Password	verify password

2.8.7 Restart Device

Figure 2-8-7 Restart Device

If you 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 I have modified the default IP or forgot it ?

Customers have two ways to get the IP address:

1) Press the RST button, then users can regain default IP. Refer to 1.2.1 Front View
2) Connect the CONSOLE with your PC Serial Port. The baud rate is 115200 bps. The user name is "admin"，password is telnet/web login password. If password is reset, the default password is “admin”. When customers have accessed it and input the command "show int" to get the IP.

3.2 Equipment physical connection to normal, but the network cannot be connected or network communication is not normal

1) Make sure the network cable is ok or not , can through view the device WAN port or LAN port indicator light to determine the work states of physical connection
2) Makeing sure the connected network devices (router, switch or hub) support 10M/100M adaptive.
   Else, connecting the Equipment directly to PC and landing Web , then in the "local connection" .Selecting the correct Ethernet Work Mode
3) Check whether there is a LAN port conflict with the existing IP address
4) Login using the serial port, in the enable mode to view the correct IP and mask, and ping the same segment of the PC or device to see if can pass.

3.3 Equipment can’t register

If the Run LED flashes slowly ,it means unregistered.

1) Check the network connection is working (see above section), whether the Configuration is correct
2) Check whether the LAN firewall setting is inappropriate (such whether limit the network Communication); If it is, there are two ways to try to resolve:
   (2.1) Ask network administrators to open limitation with the equipment’s network communications (it is a special equipment, not afraid of virus attacks);
   (2.2) Try to enable the equipment tunnel (Through the WEB for Configuration, Also, please NOTE, open the tunnel will impact voice quality, Please do not enable the tunnel as far as possible, reference WEB Configuration Interface, Description section).
4) Check whether the Local Network to the SIP PROXY platform network environment is relatively poor or not, and if so, please check Local Network or contact the service provider.
5) If go through those steps, the device still be in trouble, please contact the equipment provider.

4. Glossary

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