MTG200 Trunk Gateway User Manual V2.0

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Revision Records

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Content

1. Product Introduction

1.1 Overview

MTG200 is a kind of digital trunk gateway based on embedded operating system, and designed for IPPBX and call center. MTG200 provides 1/2/4*E1/T1 interfaces for users. MTG200 mainly provide E1/T1 interface for small and medium enterprise. It supports rich GUI configuration and 120 road traffic, and the user setting and maintaining system easily. A typical network diagram shows the function of MTG200 as below.

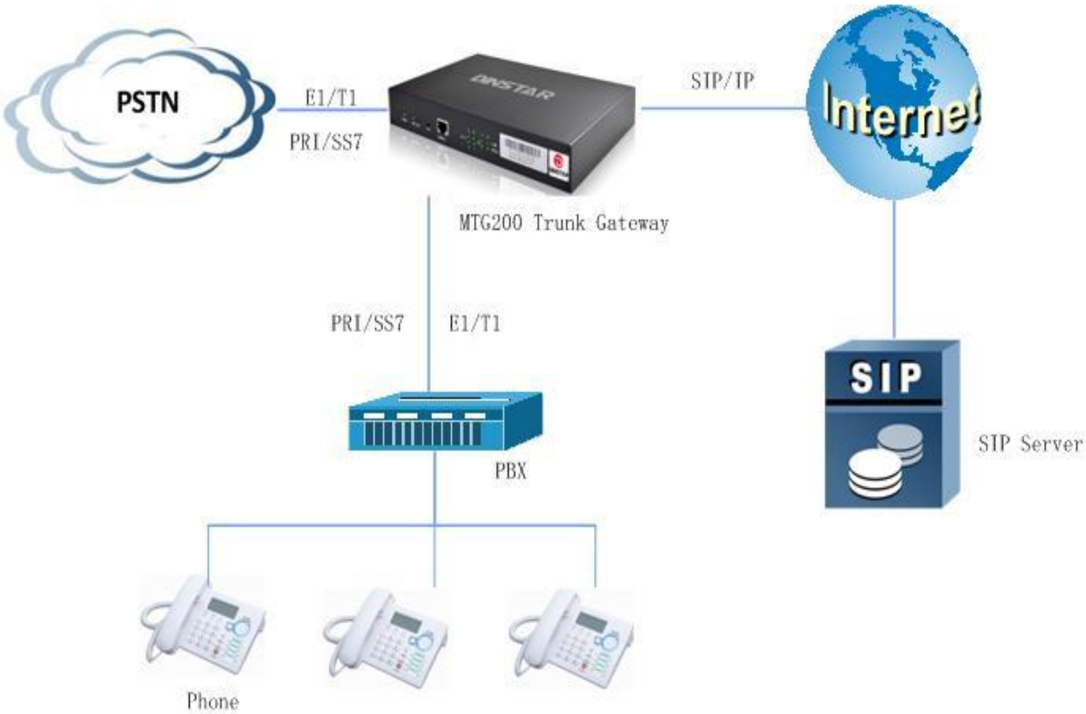


Figure 1-1-1 Application topology

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

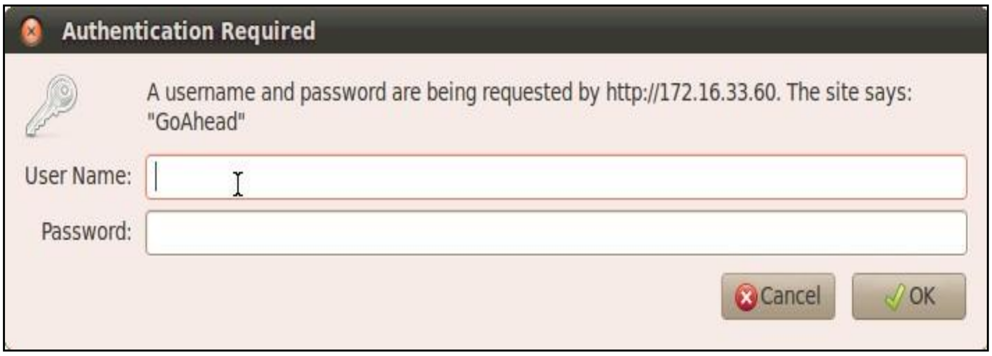
- Power supply: 12VDC, 1A
- Temperature: 0~40℃ (operational),-20~70℃(storage)
- Humidity: 10%~90%, no condensation
- Max power consumption: 15W
- Dimension (mm): 210*150*38
- Net weight: 0.75kg

2. Parameter setting

2.1 Login

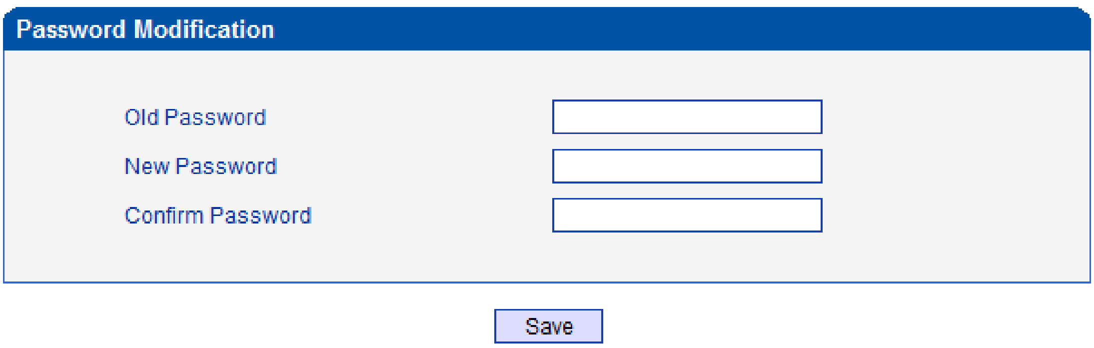
First, device FE0 port connect PC with string, and then fill FE0 IP address in browser, FE0 default IP address is 192.168.1.111. It will request customer to input user name and password. Default user name and password are “admin”.
If customer modified the default IP or forgot the IP, that can’t enter 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

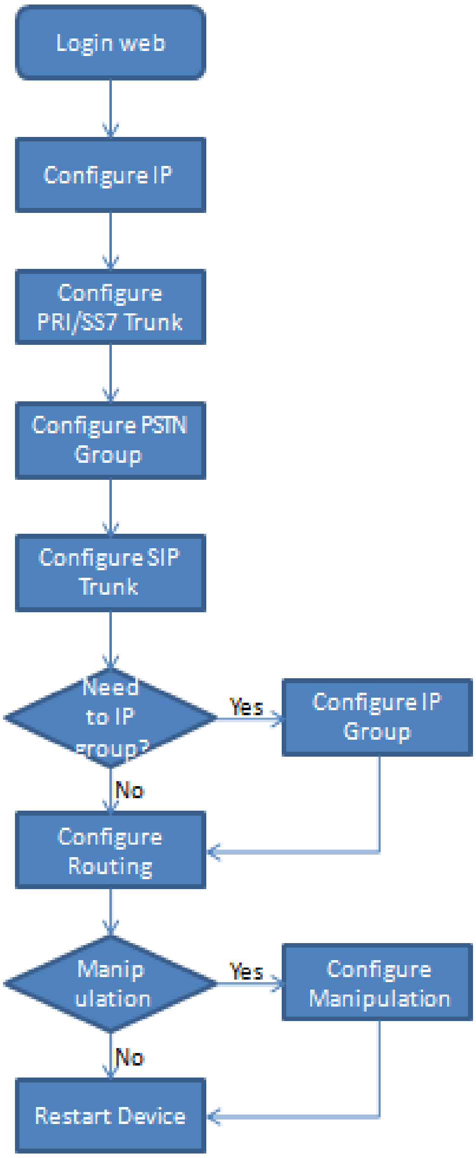


After inputting the old password, input a new password and confirm it by inputting it again.

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.

MTG configuration flow chart below:



2.2.1 System Information

This configuration page includes general information and version information.

Figure 2-2-1 System Information

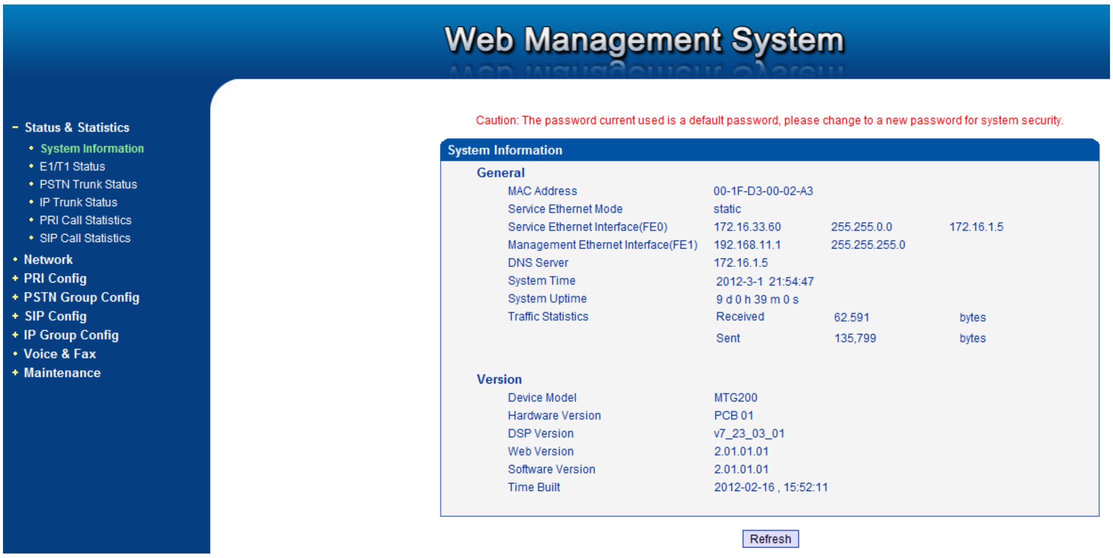


Table 2-2-1 Description of System Information

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

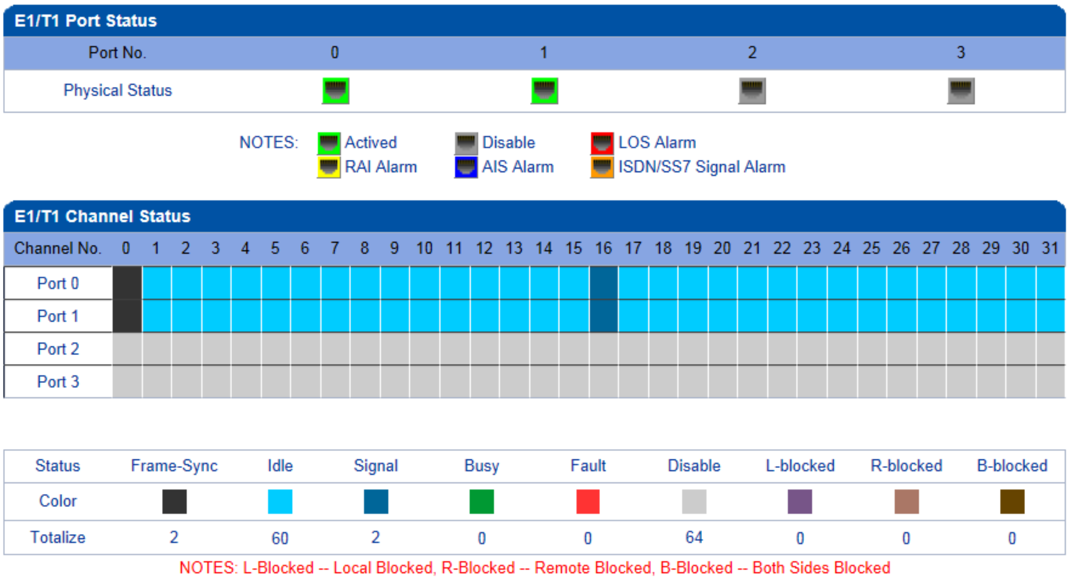


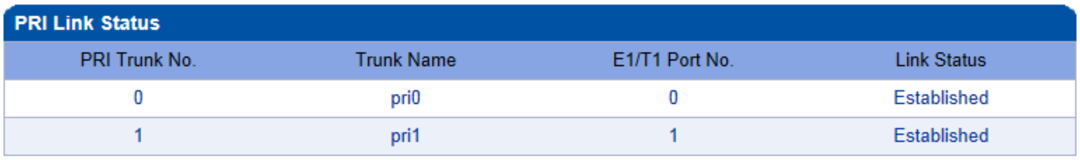
Table 2-2-2 Description of E1/T1 status

<table>
<tr><td rowspan="6">E1/T1 Port Status</td><td>1. LOS Alarm: Signal loss alarm, this alarm is created when receiving is lost; please check the physical connection whether disconnected.</td></tr>
<tr><td>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.</td></tr>
<tr><td>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.</td></tr>
<tr><td>4. Disable: Means that this E1/T1 is not used.</td></tr>
<tr><td>5. ISDN/SS7 Signal Alarm: Means physical connection is normal, signaling link has problem.</td></tr>
<tr><td>6. Active-OK: Means that physical connection and signaling link are normal.</td></tr>
<tr><td rowspan="8">E1/T1Channel Status</td><td>1.Frame-Sync: Non voice channel, which used as a synchronization channel</td></tr>
<tr><td>2.Idle: Means this channel is idle, when the channel is enabled and the cable is connected OK.</td></tr>
<tr><td>3.Signal: Signal channel</td></tr>
<tr><td>4.Busy: Means this channel is occupied by voice</td></tr>
<tr><td>5. Fault: The channel is enabled but the cable is not connected.</td></tr>
<tr><td>6.Disable: Have not use this E1/T1 trunk</td></tr>
<tr><td>7.L-blocked:
Local blocked, means that communication can only be initiated from local</td></tr>
<tr><td>8.R-blocked:</td></tr>
</table>

	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

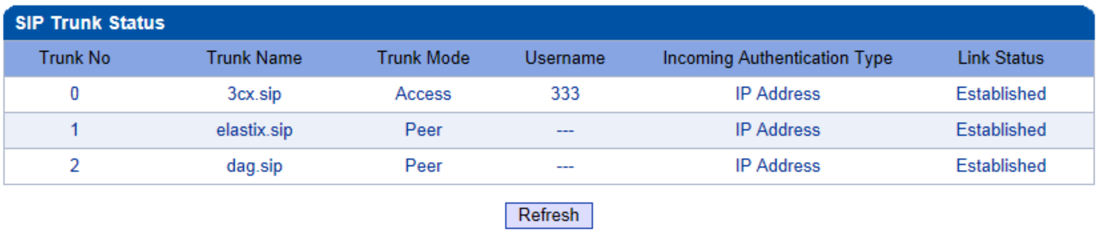
PRI Trunk No.	Trunk Name	E1/T1 Port No.	Link Status
0	pri0	0	Established
1	pri1	1	Established

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

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

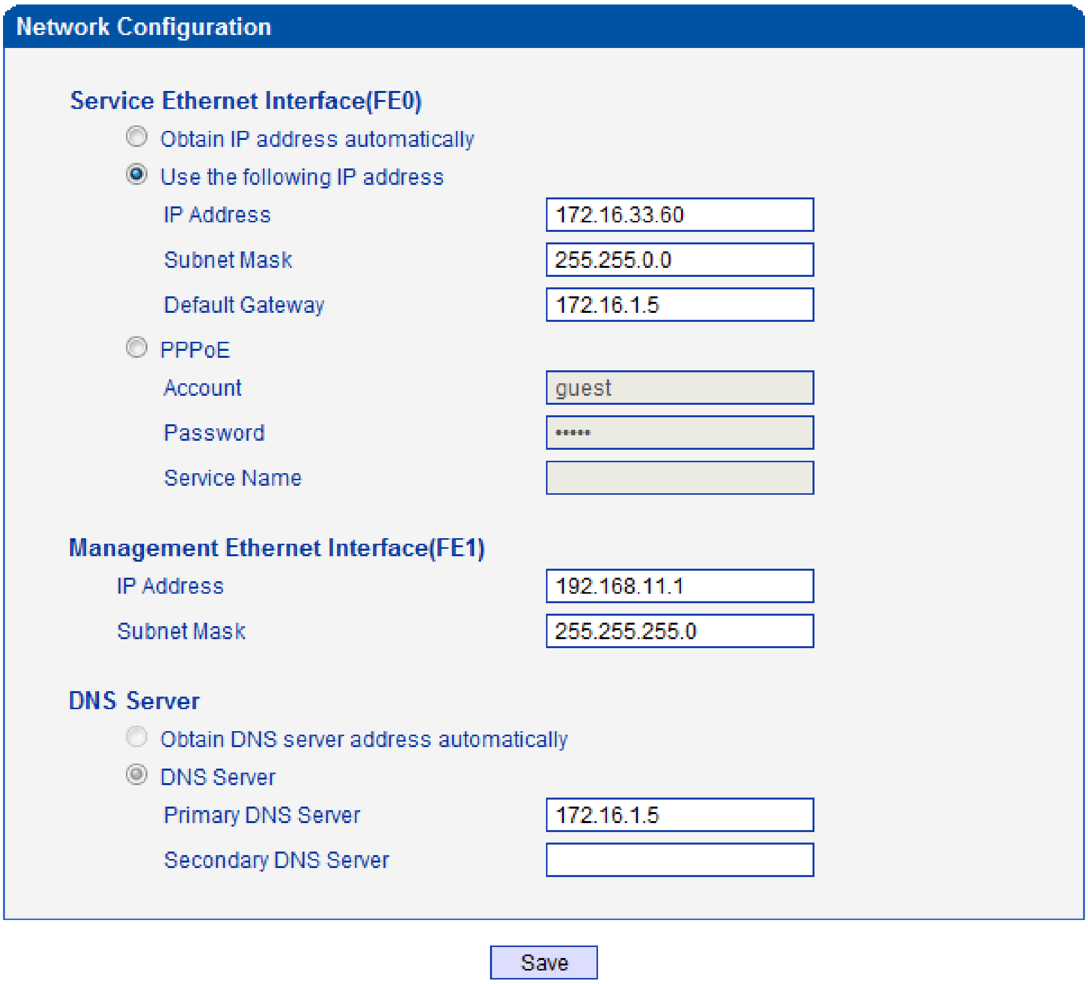


Table 2-3-1 Description of Network Configuration

Service Ethernet Interface (FE0)	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 .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 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.

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

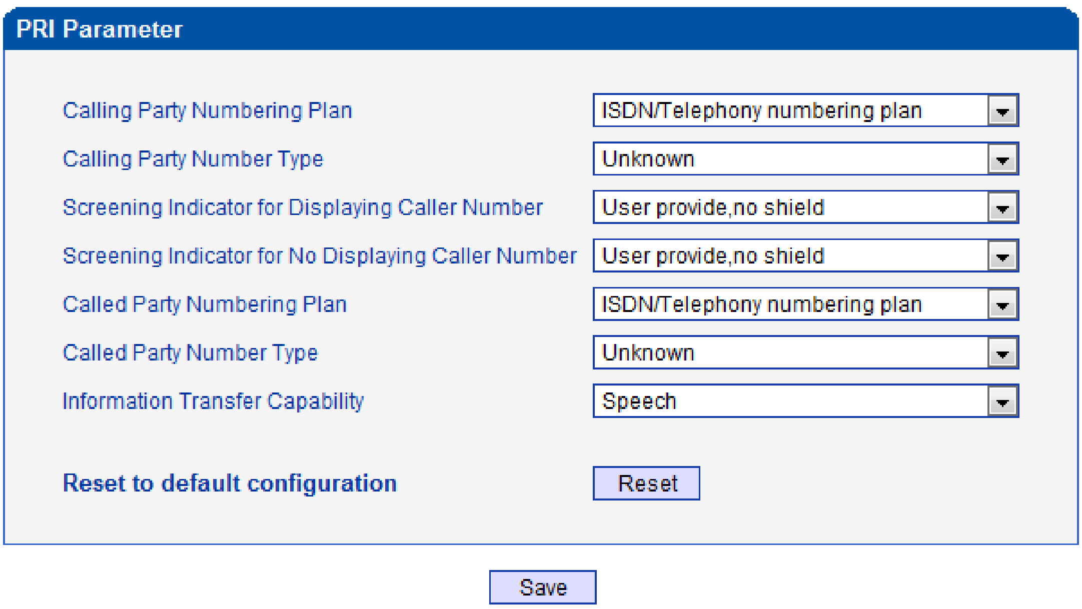


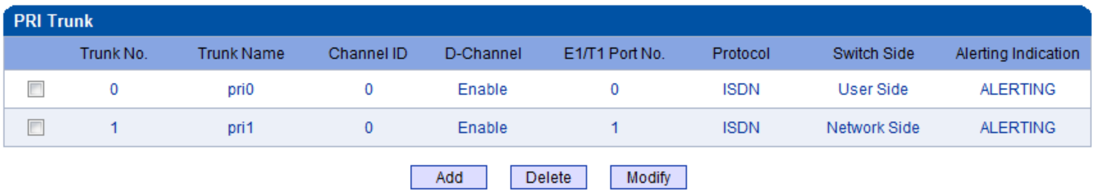
Table 2-4-1 Description of PRI Parameter

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



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

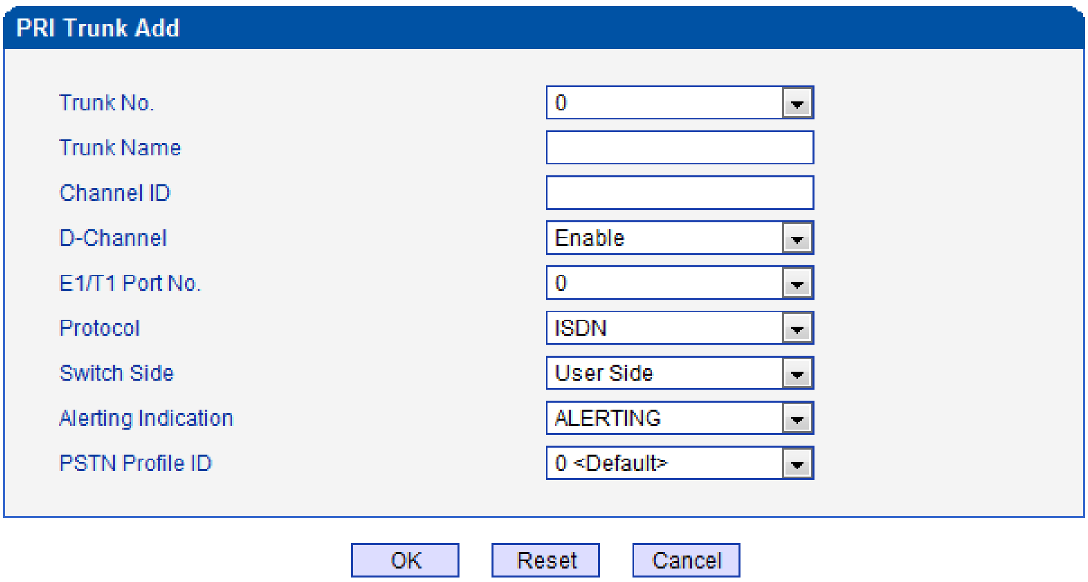


Table 2-4-2 Description of Add PRI Trunk

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

2.5 PSTN Group Config

2.5.1 E1/T1 Parameter

Figure 2-5-1 E1/T1 Parameter

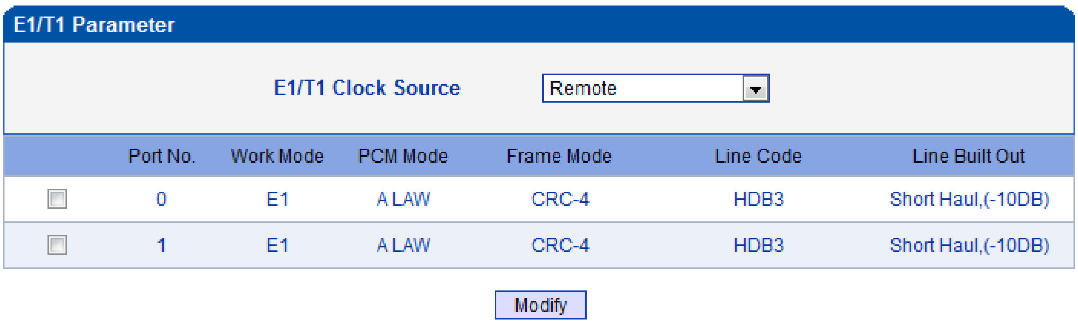


Figure 2-5-2 E1/T1 Parameter

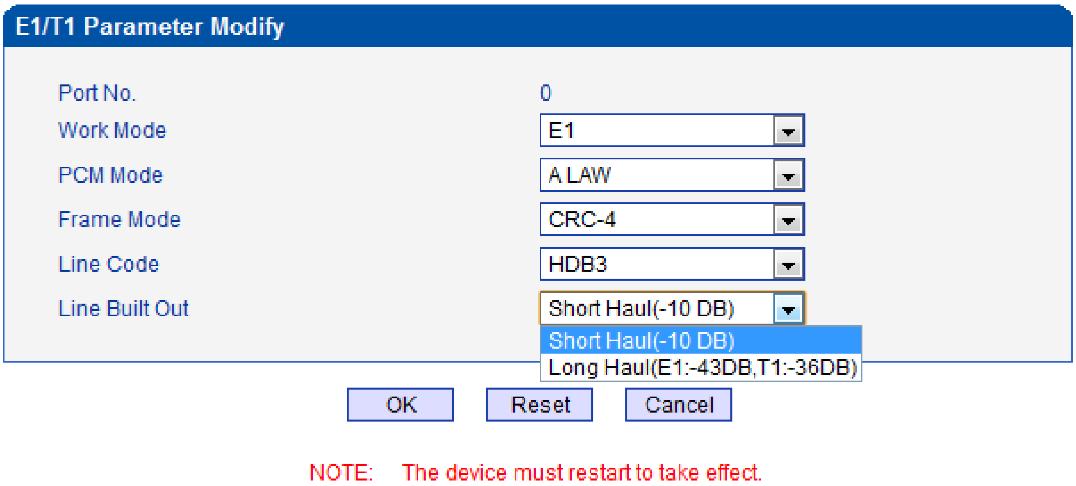


Table 2-5-1 Description of Modify E1/T1 Parameter

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.

2.5.2 Coder Group

Figure 2-5-3 Coder Group

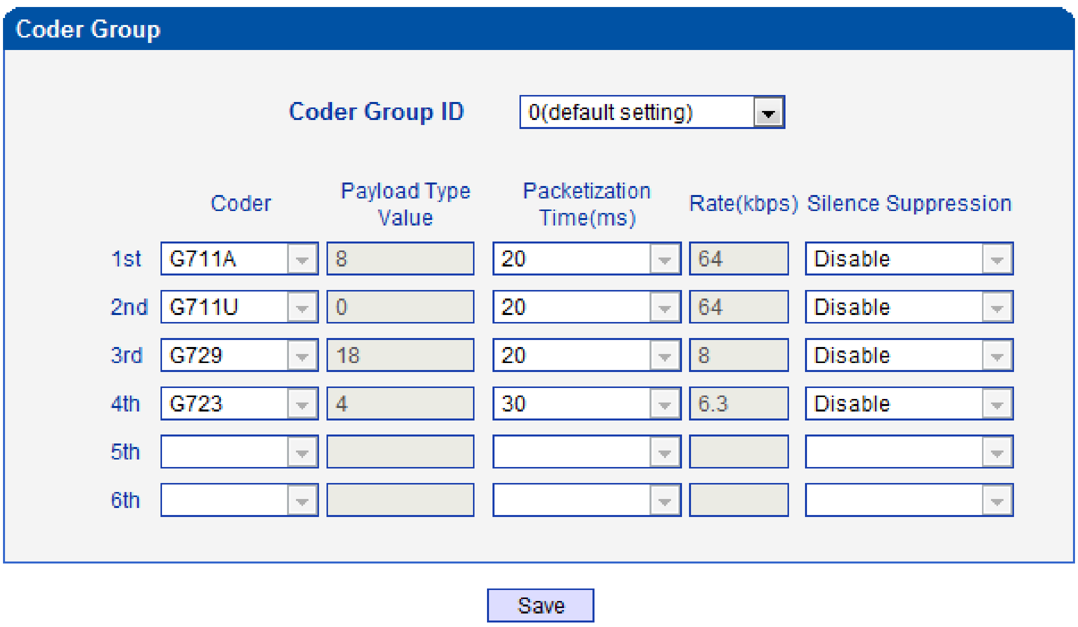
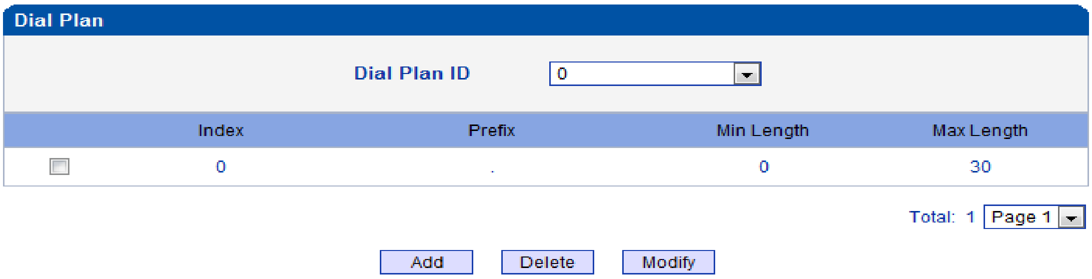


Table 2-5-2 Description of Coder Group

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.

2.5.3 Dial Plan

Figure 2-5-4 Dial Plan



Dial plan used for configuring the receiving number, user can configure different prefix number, these rules can be divided into 5 groups with a dial plan ID, where 0 is the default setting.

Notes:

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

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

Table 2-5-3 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)

2.5.4 Dial Timeout

Figure 2-5-4 Dial Timeout

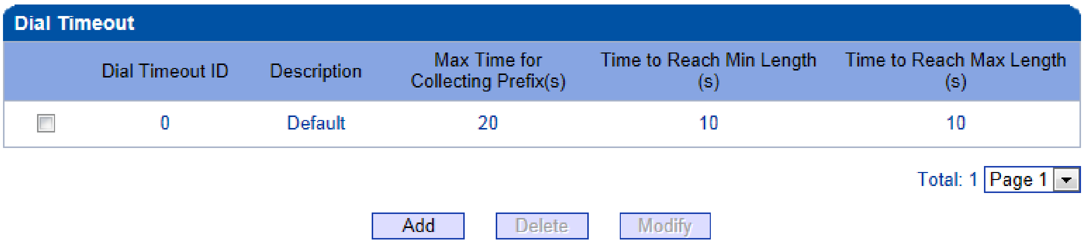
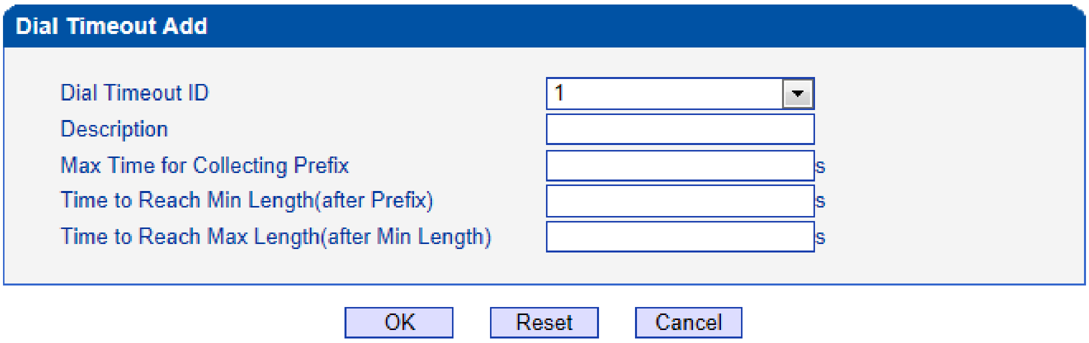


Figure 2-5-5 Dial Timeout Add



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

Table 2-5-4 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

2.5.5 PSTN Profile

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

Figure 2-5-6 PSTN Profile

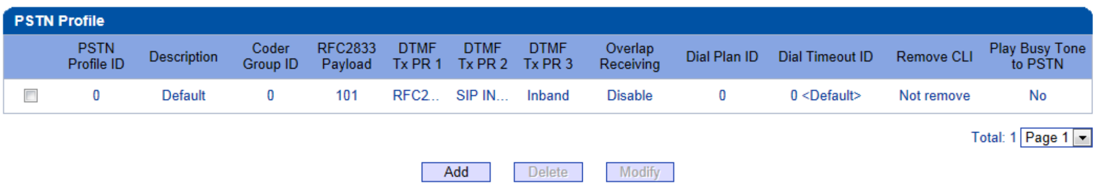


Figure 2-5-7 PSTN Profile Add

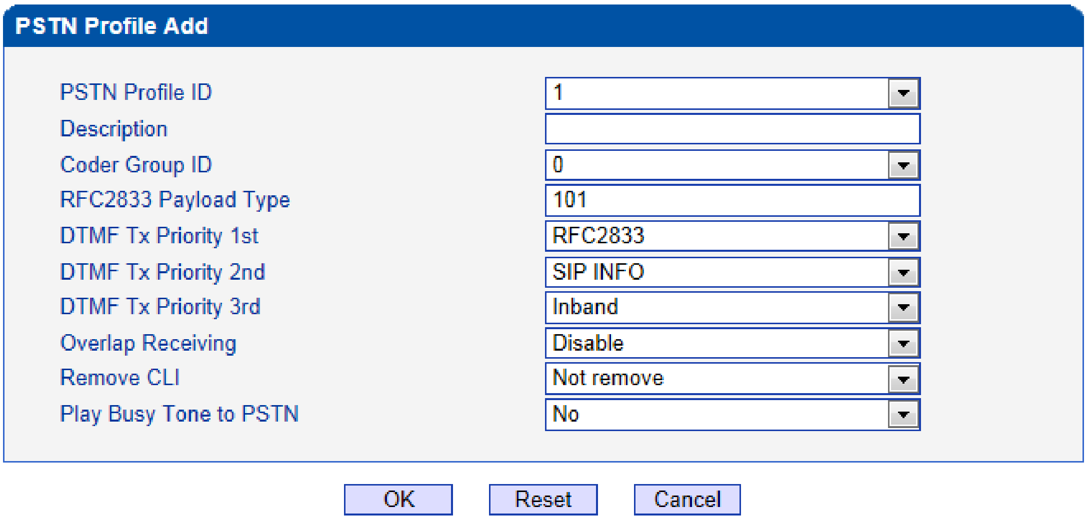


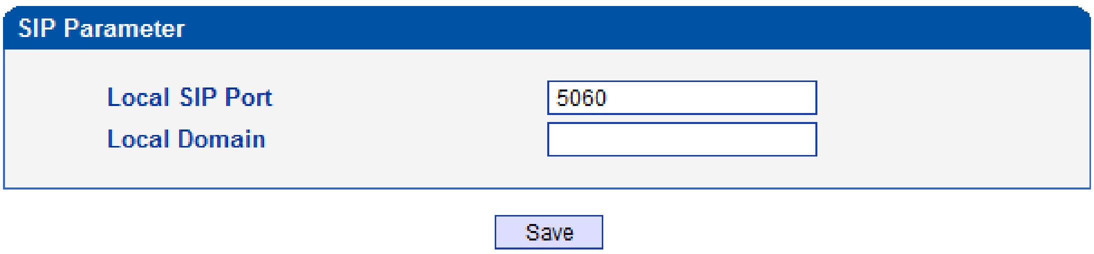
Table 2-5-5 Description of Add PSTN Profile

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
1st/2nd/3rd Tx DTMF Option	There are three ways to send DTMF: RFC2833/SIP INFO/ INBAND, in accordance with the priority choice to send the configuration mode
Overlap Receiving	Not enabled by default, only 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 SIP Config

2.6.1 SIP Parameter

Figure 2-6-1 SIP Parameter



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

2.6.2 SIP Trunk

Figure 2-6-2 SIP Trunk

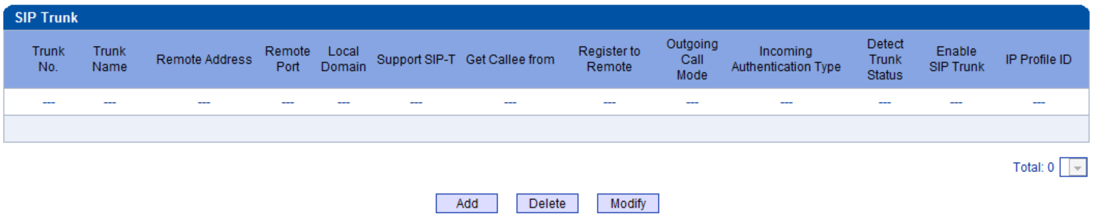


Figure 2-6-3 SIP Trunk Add

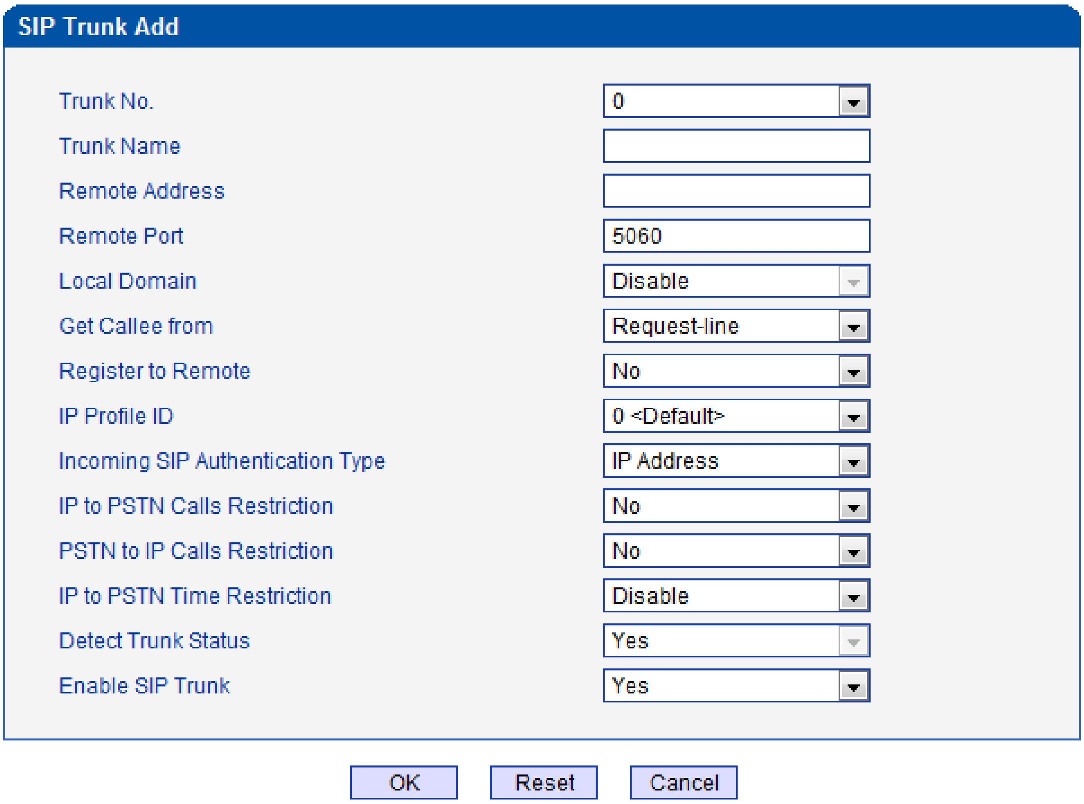


Table 2-6-1 Description of Add SIP Trunk

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 MTG, 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.

2.7 IP Group Config

2.7.1 IP Profile

Figure 2-7-1 IP Profile

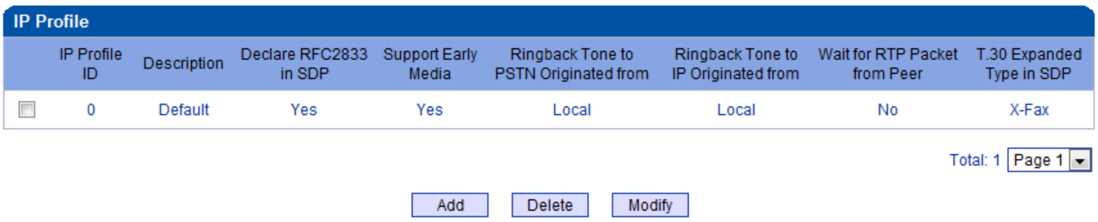


Figure 2-7-2 IP Profile Add

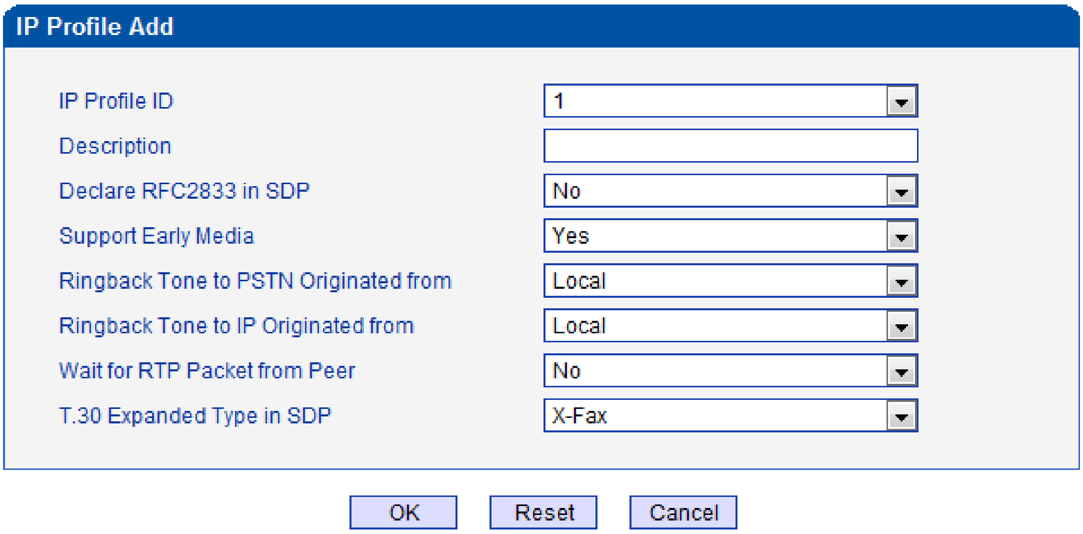


Table 2-7-1 Description of Add IP Profile

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)
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

2.8 Voice & Fax

Figure 2-8-1 Voice & Fax Configuration

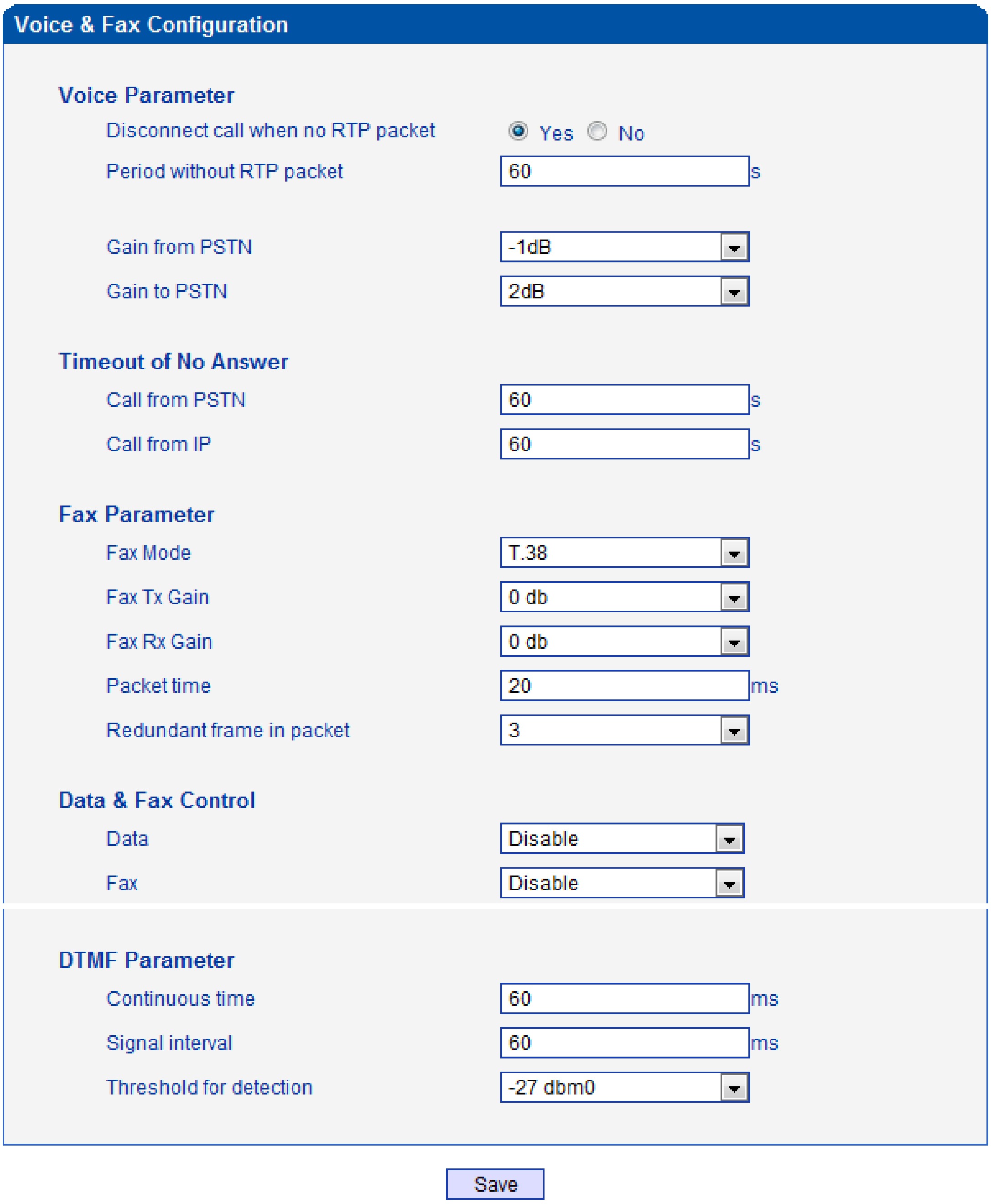


Table 2-8-1 Description of Voice & Fax

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	Call from PSTN	Call timeout of no answer from PSTN

answer	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

2.9 Maintenance

2.9.1 Management Parameter

Figure 2-9-1 Management Parameter

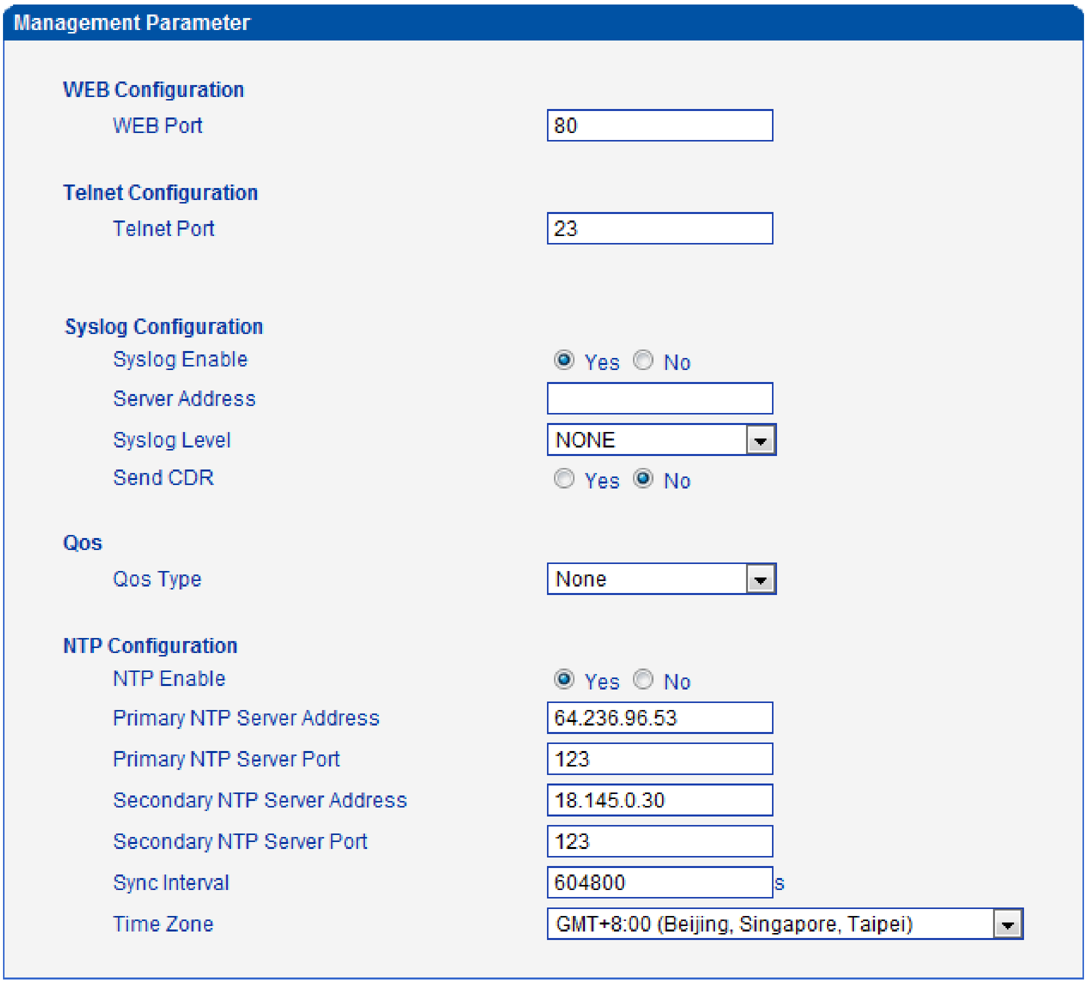


Table 2-9-1 Description of Management Parameter

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

2.9.2 Data Restore

Figure 2-9-2 Data Backup

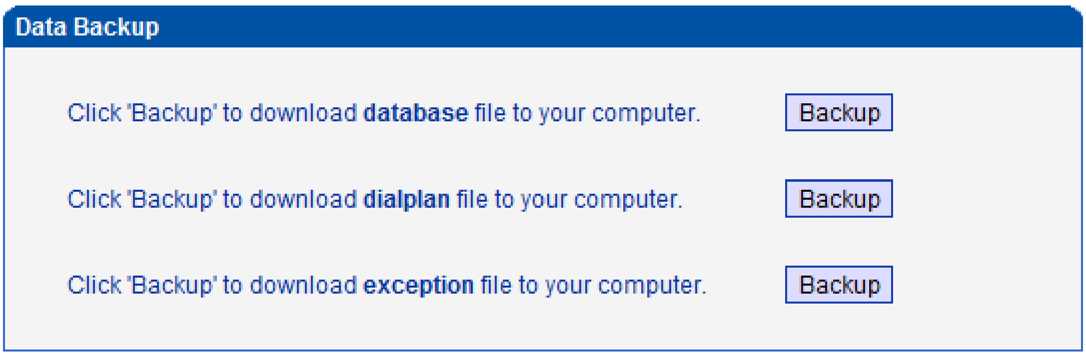


Table 2-9-2 Description of Data Backup

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

2.9.3 Data Restore

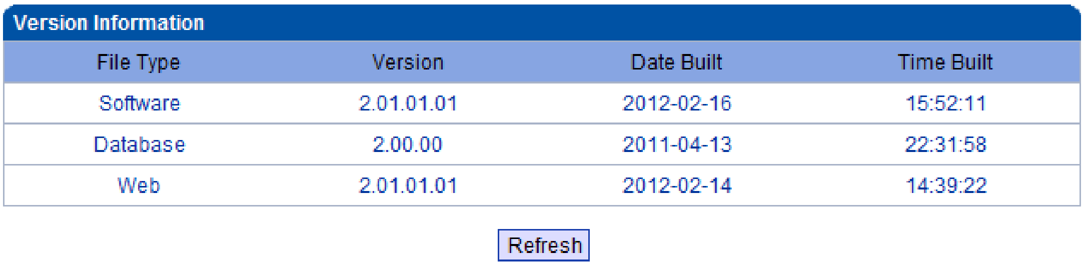
Figure 2-9-3 Data Restore

Table 2-9-3 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".

2.9.4 Version Information

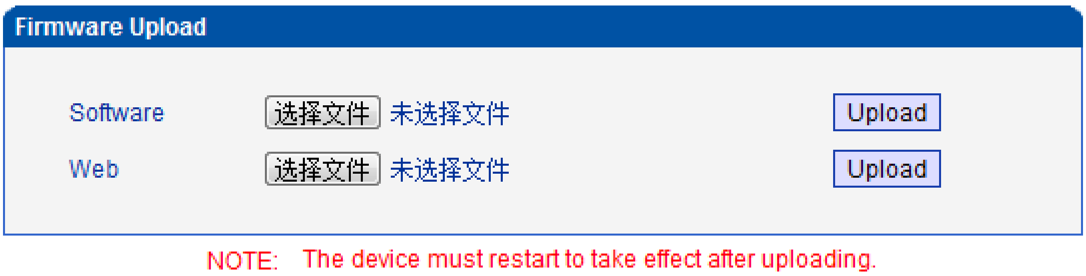
Figure 2-9-4 Version Information



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

2.9.5 Firmware Upload

Figure 2-9-5 Firmware Upload



The process of firmware upload：

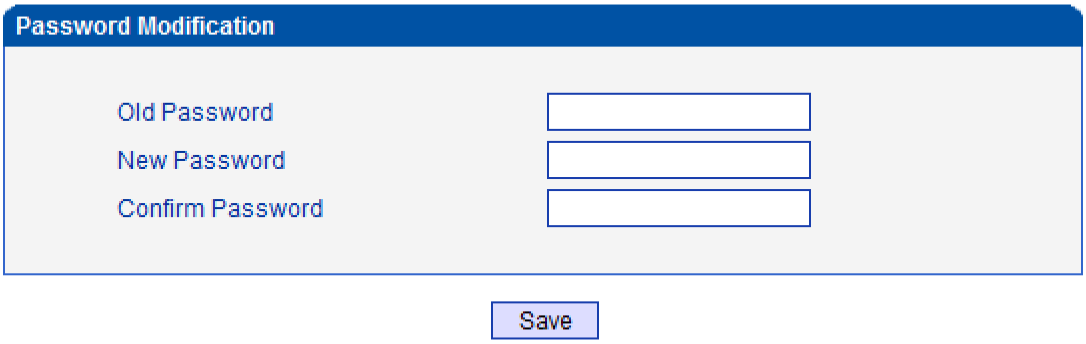
1） Click “Firmware Upload”

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）.

2.9.6 Password Modification

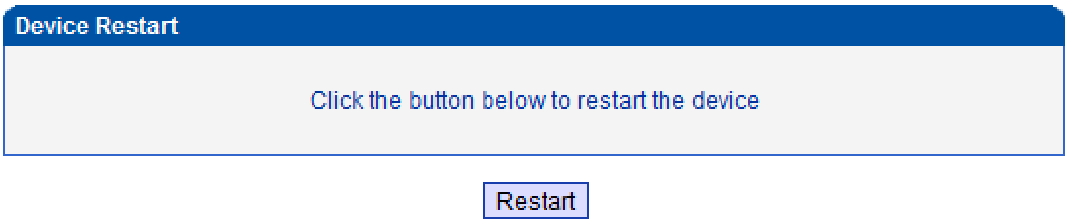
Figure 2-9-6 Password Modification



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

2.9.7 Device Restart

Figure 2-9-7 Device Restart



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

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