MWG1000 SERIES

VoIP GSM Gateway

User Manual

Version 3.0.0.18

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1. Introduction

1.1 Overview

MWG1000 serials GSM VoIP Gateway is a full functional VoIP gateway based on IP and Mobile network, which provides a flexible network configuration,powerful features, and good voice quality . It works for carrier grade, enterprise, SOHO, residential users for cost-effective solution.
MWG1000 serials consists of 2 chanels (MWG1002) and 4 chanels (MWG1004)

1.2 Product Features

➢ Calling: Termination (VoIP to GSM), Origination (GSM to VoIP)
➢ SIP Registration
➢ SIP Trunk
➢ Incoming call routing
➢ Outgoing call routing
➢ SMS sending and receiving
➢ Support USSD
➢ Call Forwarding, Waiting
➢ LCR (Least Cost Routing)
➢ Top voice quality
➢ Simple Web based configuration
➢ Easy to install

1.3 Product Appearance

The appearance of MWG1004 shows as follow

Figure 1-3-1 Front view of MWG1004

Table 1-3-1 Description of Front view

Index	Indicators	Description
1	RUN	On: Starting Off: Abnormal Blinking every 0.5s: Normal status
2	POWER	On: Power on Off: Power off
3	Channel	Use/Unuse indicator with Green color , ON is used, Off is unused
4	SIM Slots	SIM card slot

Figure 1-3-2 Rear view of MWG1004

Table 1-3-2 Description of Rear view

Index	Interface	Description
1	RST	Reset button to restore default IP and password or restore factory setting. Hold RST button 8 seconds, RUN LED being ON during this time
2	Power Connector	Power connector of DC power. Input: DC12V
3	Network	Its default IP address 192.168.6.200

1.4 Scenario of Application

Application 1

Figure 1-4-1

Application 2

Figure 1-4-2

2. Installation Guide

2.1 Installation Notice

MWG1004 adapts 12VDC Power adapter , make sure AC power supply grounded well to ensure the reliability and stability;

Notes: incorrect power connection may damage power adapter and device.

MWG1004 provides standard RJ45 with 10Mbps or 100Mbps interfaces.

For Wireless part, make sure antennas connecting well on device. Inserting SIM cards and GSM channels should work properly .

2.2 Installation Procedure

2.2.1 Install SIM Card

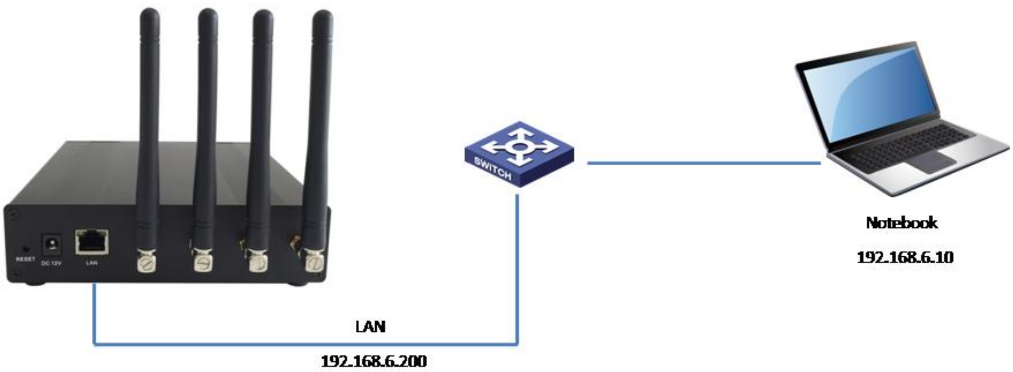
Figure 2-2-1 SIM Card installation

2.2.2 Antenna Installation

Figure 2-2-2 Antenna Installation

2.2.3 Network Cable Connection of Equipment

Figure 2-2-3 MWG1004 network connection



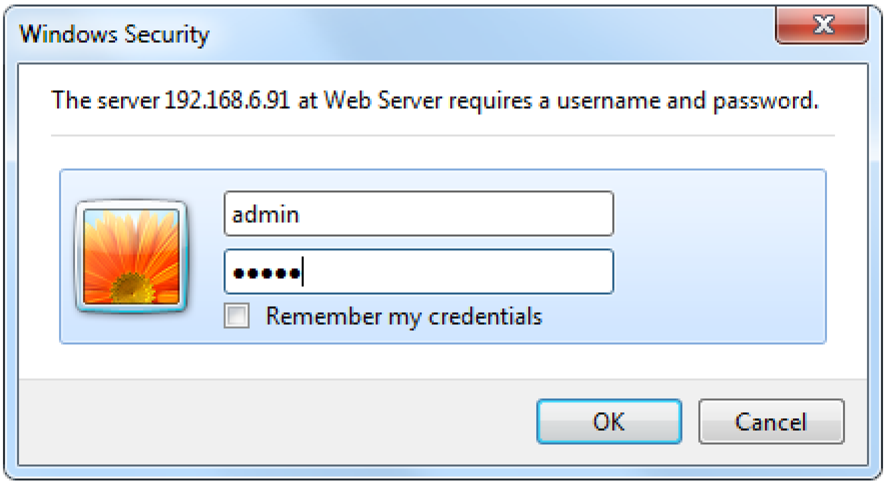
3. WEB Interface Configuration

MWG1000 serials gateway has the same web interface. This charpter describes web configuration of MWG1004. The MWG1004 contains an embedded web server to set parameters by using the HTTP protocol. We are strongly recommend to access device with Google Chrome or Firefox Browser. The configuration introduction also suitable for MWG1002.

3.1 Access MWG1004 unit

Enter IP address of MWG1004 in IE/Google Chrome/Firefox Browser. The default IP of LAN port is 192.168.6.200. and the GUI shows as below:

Figure 3-1-1 WEB log interface



Enter username and password and then click "OK" in configuration interface. The default username and password are "admin/admin". It is strongly recommended, change the default password to a new password for system security .

3.2 Parameters Configuration

MWG1004 WEB configuration interface consists of the navigation tree and the detail configuration interfaces.

Figure 3-2-1 WEB introduction

Go through navigation tree, user can check, view, modify , and set the device configuration on the right of configuration interface.

3.3 System Information

System information interface shows the basic information of status information, mobile information and SIP information.

3.3.1 System Information

Figure 3-3-1 system Information

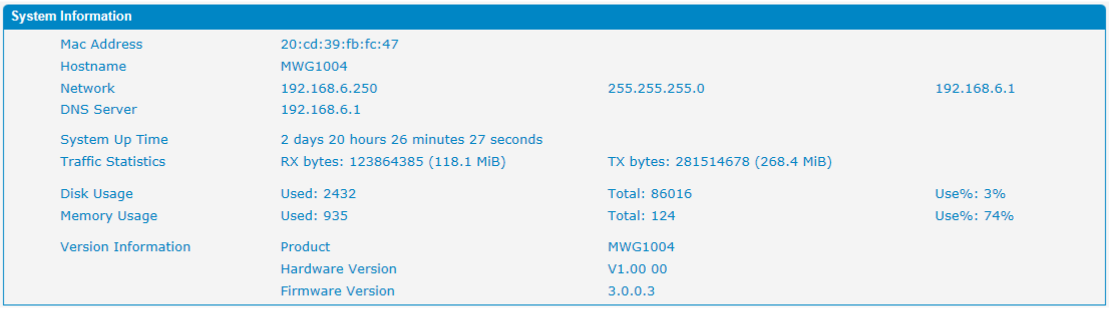


Table 3.3-1 System Information

Parameters	Description
MAC Address	Displays the current MAC of the gateway, for example: 70-B3-D5-1B-3D-02
Network	Current IP address and subnet mask of gateway
DNS Server	Displays DNS server IP address in the same network with the gateway
System Up Time	Shows the time period of the device running. For example, :1h : 20m : 24s
Traffic Statistics	Calculates the net flow, including the total bytes of message received and sent。
Version info	Shows the current firmware version

3.3.2 Mobile Information

Figure 3.3-2 Mobile Information

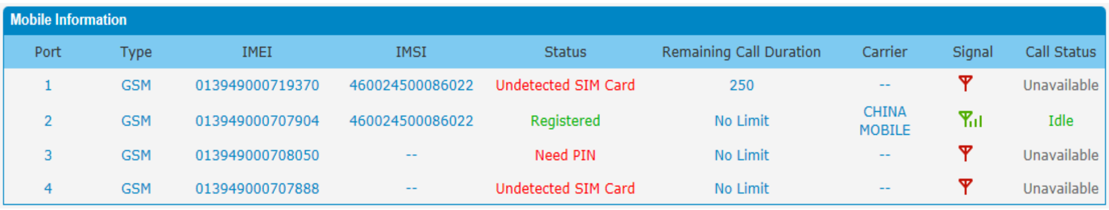


Table 3.3-2 Mobile Information

Parameters	Description
Port	Number of GSM/CDMA ports.

Type	Indicates the current type of network. Such as CDMA or GSM
IMEI	International Mobile Equipment Identity
IMSI	International Mobile Subscriber Identity, it is the uniquely identifies of SIM card
Status	Indicates the connection status of current GSM/CDMA module
Remaining Call Duration	It shows available total call minutes of SIM card while call limitation is enabled.
Carrier	Displays the network carrier of current SIM card.
Signal	Displays the signal strength in each channel of GSM / CDMA.
Call Status	Show the Status of port, including idle, active, alert and processing.

3.3.3 SIP Information

Figure 3-3-3 SIP Information

SIP Information

Status	Trunk Type	Name	SIP/IAX	Transport	User Name	Hostname/IP Address
Request Sent	Trunk	voip-sip31	SIP	udp	78	192.168.6.9
Unreachable	Service Povider	sps31	SIP	udp	--	192.168.6.5
Unregistered	Account	2001	SIP	udp	2001	(Unspecified)

Displays registration status information with Softswitch platform or SIP Server

Table 3-3-3 SIP information

Parameters	**Description**
Status	Shows the registration status of VoIP channel, including registered and unregistered.
Trunk Type	Trunk mode will allow IP phone or IPPBX to register or trunk mode to register to provider
Name	It describes this VoIP channel for the ease of identification. Its value is character string
SIP/IAX	Choose the type of this trunk, SIP or IAX
Transfer Protocol	This will be the transport method used by the trunk. The options are UDP (default) or TCP or TLS.
User Name	The number for this VoIP channel
Hostname/IP Address	Hostname or IP Address of this VoIP channel

3.4 Network Configuration

3.4.1 Local Network

Figure 3-4-1 Local Network

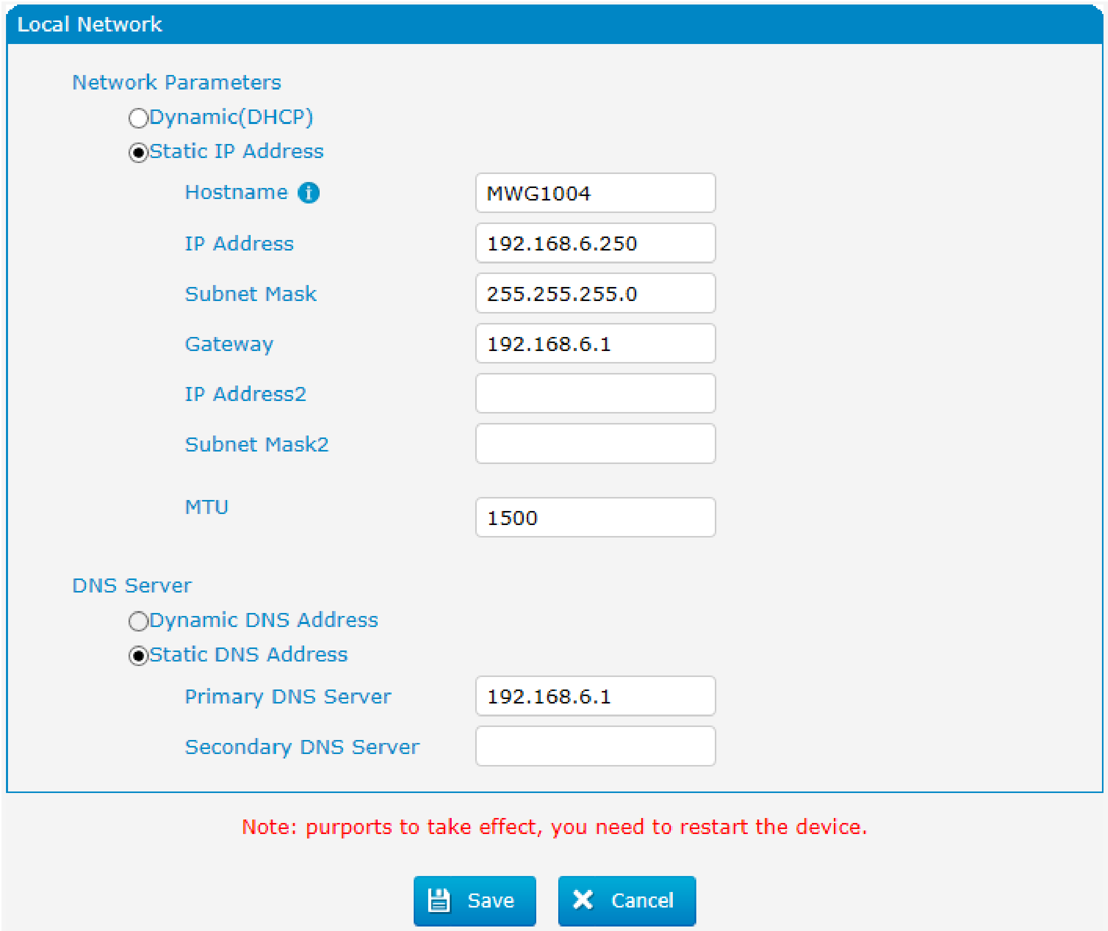


Table 3-4-1 Description of Local network

Parameter	**Description**
Dynamic (DHCP)	Enable the device obtain IP Address automatically
Static IP Address	Configure the "IP Address", "Subnet Mask" and "Default Gateway" by manual
Hostname	Set the host name for MWG1004
IP Address	Set the IP Address for MWG1004, It is recommended to configure a static IP address for MWG1004
Subnet Mask	Set the subnet mask for MWG1004
Gateway	Set the gateway for MWG1004
IP Address 2	Set the second IP Address for MWG1004
Subnet Mask2	Set the second subnet mask for MWG1004
MTU	Message transmit unit, default is 1500
Dynamic DNS Address	Obtain DNS Server Address Automatically
Static DNS Address	Obtain Primary DNS Server by manual
Primary DNS Server	Set the primary DNS Server for MWG1004.
Secondary DNS Server	Set the Secondary DNS Server for MWG1004.

3.4.2 VLAN Configuration

A VLAN (Virtual LAN) is a logical local area network (or LAN) that extends beyond a single traditional LAN to a group of LAN segments, given specific configurations.

Note: MWG1004 is not the VLAN server, a 3-layer switch is still needed, please configure the VLAN information there first, then input the details in MWG1004, so that the packages via MWG1004 will be added the VLAN label before sending to that switch.

Figure 3-4-2 VLAN Configuration

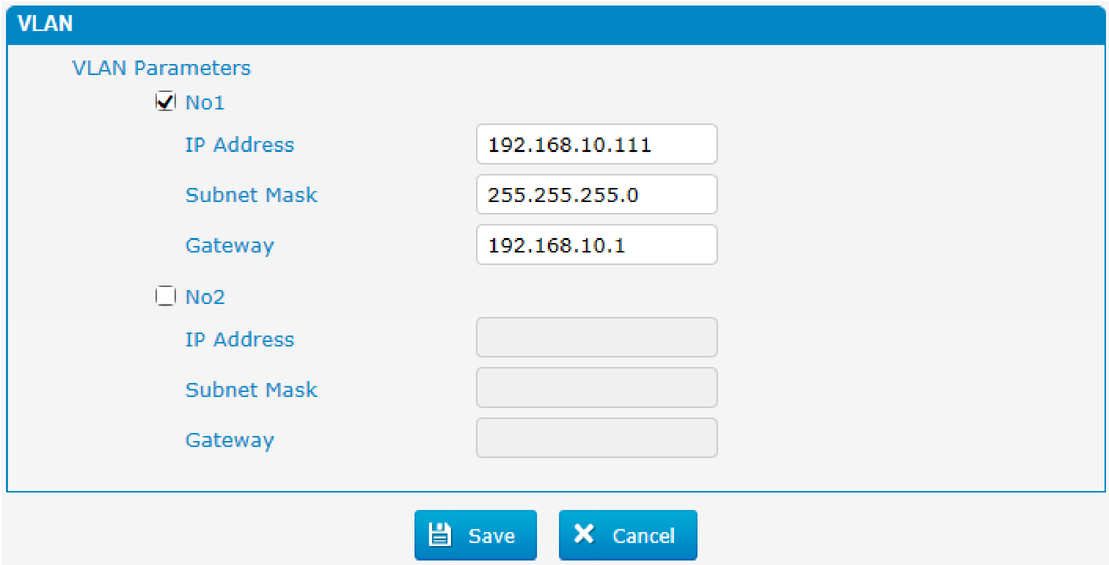


Table 3-4-2 Description of VLAN Configuration

Parameter	Description
NO.1	Click the NO.1 you can edit the first VLAN over LAN
IP Address	Set the IP Address for MWG1004 VLAN over LAN.
Subnet Mask	Set the Subnet Mask for MWG1004 VLAN over LAN.
Gateway	Set the Default Gateway for MWG1004 VLAN over LAN

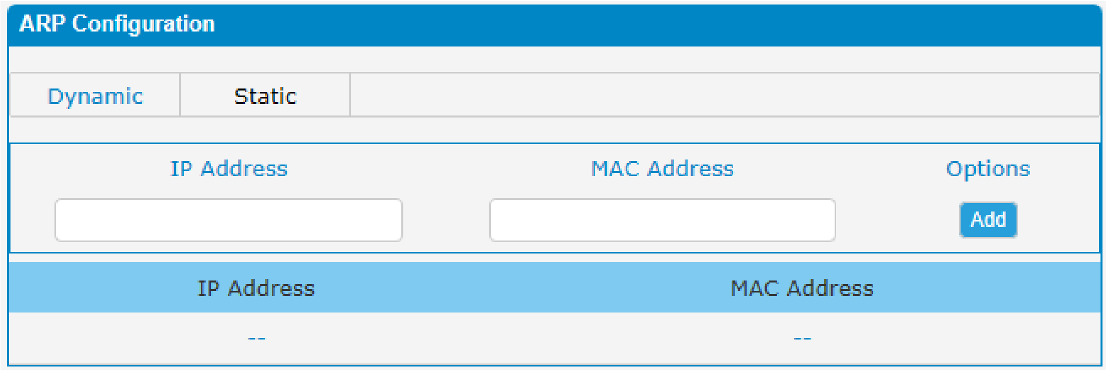
3.4.3 ARP

The ARP function is mainly used to query and add the map of IP and MAC. There are static or dynamic ARP entries.

Like other routers, the gateway can automatically find the network device on the same segment. But, sometimes you don't want to use this automatic mapping, you'd rather have fixed (static) associations between an IP address and a MAC address. Gateway provides you the ability to add static ARP entries to:

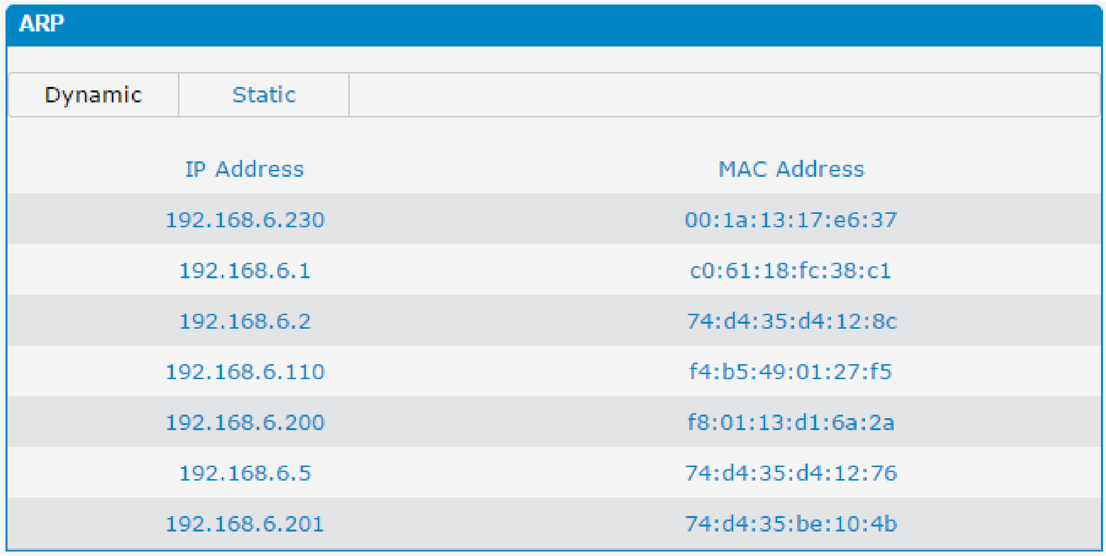
● Protect your network against ARP spoofing
● Prevent network confusion as a result of misconfigured network device

Figure 3-4-3 Add ARP



Click "Dynamic ARP" to check ARP buffer

Figure 3-4-3a Dynamic ARP



3.4.4 VPN Configuration

A virtual private network (VPN) is a method of computer networking--typically using the public internet--that allows users to privately share information between remote locations, or between a remote location and a business' home network. A VPN can provide secure information transport by authenticating users, and encrypting data to prevent unauthorized persons from reading the information transmitted. The VPN can be used to send any kind of network traffic securely. MWG1004 supports OpenVPN.

Figure 3-4-4 VPN Configuration

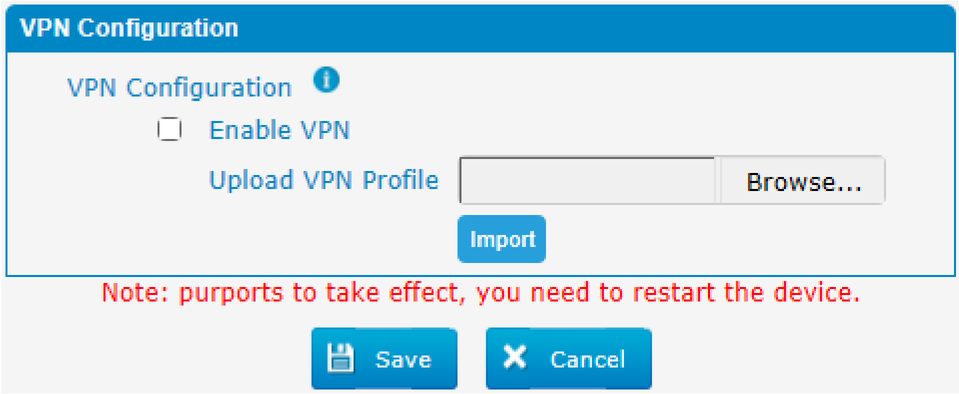


Table 3-4-4 Description of VPN Parameter

Parameters	**Description**
Import VPN Configuration Files	Import configuration file of OpenVPN.

Notes:
1. Don't configure “user” and “group” in the “config” file. You can get the config package from the OpenVPN provider.
2. MWG1004 works as VPN client mode only.

3.4.5 DDNS Server

DDNS(Dynamic DNS) is a method / protocol / network service that provides the capability for a networked device, such as a router or computer system using the Internet Protocol Suite, to notify a Domain Name System (DNS) name server to change, in real time, the active DNS configuration of its configured hostnames, addresses or other information.

Figure 3-4-5 DDNS Server

Table 3-4-5 Description of DDNS Server

Parameters	Description
DDNS Server	Select the DDNS server IP or domain name you sign up for service.
User Name	User name the DDNS server provides you.
Password	User account's password.
Domain Name	The domain name you have got from the DDNS server

Note: DDNS allows you to access your network using domain names instead of IP address. The service manages changing IP address and updates your domain information dynamically. You must sign up for service through dyndns.org, freedns.afraid.org, www.no-ip.com, www.zoneedit.com

3.4.6 Static Routing

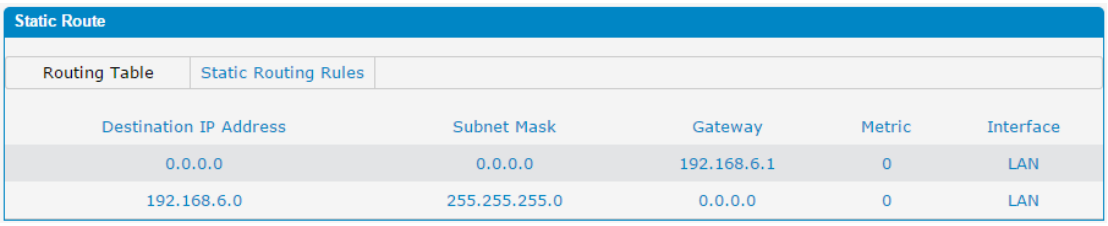
MWG1004 will have more than one internet connection in some situations but it has only one default gateway. You will need to set some Static Route for MWG1004 to force it to go out through different gateway when access to different internet.

The default gateway priority of MWG1004 from high to low is VPN/VLAN-> LAN port.

1) Route Table

The current route rules of MWG1004.

Figure 3-4-6 Static Routing Table



2) Static Route Rules

You can add new static route rules here.

Figure 3-4-6a Static Routing Rules

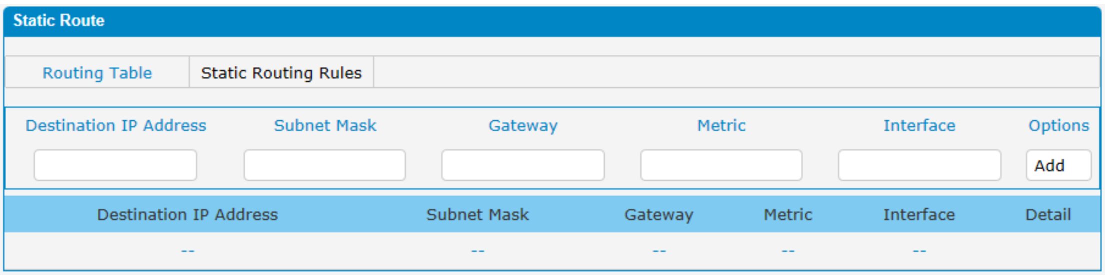


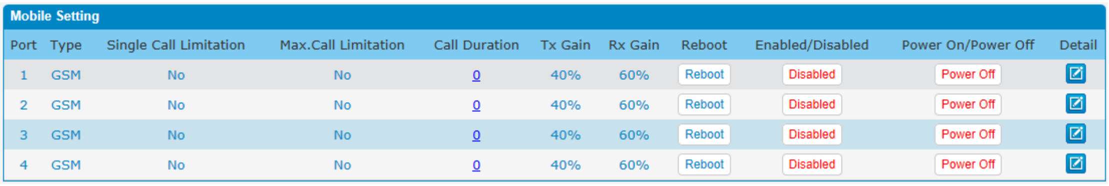
Table 3-4-6 Description of Static Routing

Parameters	Description
Destination IP Address	The destination network to be accessed to by MWG1004.
Subnet Mask	Specify the destination network portion.
Gateway	Define which gateway MWG1004 will go through when access to the destination network.
Metric	The cost of a route is calculated by using what are called routing metric. Routing metrics are assigned to routes by routing protocols to provide measurable statistic which can be used to judge how useful (how low cost) a route is.
Interface	Define which internet port to go through.

3.5 Mobile Configuration

3.5.1 Mobile Settings

Figure 3-5-1 Mobile Settings



Mobile Setting

Port	Type	Single Call Limitation	Max.Call Limitation	Call Duration	Tx Gain	Rx Gain	Reboot	Enabled/Disabled	Power On/Power Off	Detail
1	GSM	No	No	0	40%	60%	Reboot	Disabled	Power Off	
2	GSM	No	No	0	40%	60%	Reboot	Disabled	Power Off	
3	GSM	No	No	0	40%	60%	Reboot	Disabled	Power Off	
4	GSM	No	No	0	40%	60%	Reboot	Disabled	Power Off	

Figure 3-5-1a Mobile Settings Detail

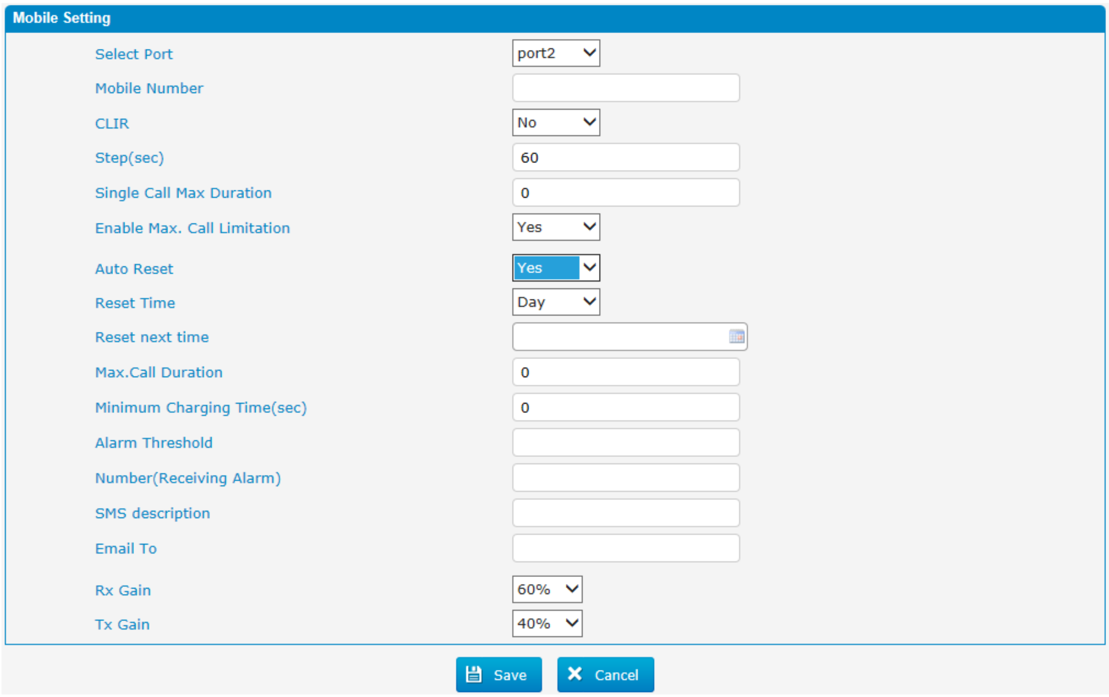


Table 3-5-1 Description of Mobile Settings

Parameters	**Description**
Select Port	Select GSM channel, default Port 1
Mobile Number	SIM card number of the channel. That must be configured when "Forward" function enable.
Step(sec)	Step length value range is 1 -120 s, step length multiplied by time of single call just said a single call duration time allowed.
Single call Max Duration	The value of limitation of a single call, this value range is 1 -65535. Step length multiplied by time of single call just said a single call duration time allowed. Single Call duration is not limitation if this value is 0.
Enable Max Call Limitation	This function is to limit the max call duration of channel. The max call duration is between 1 to 65535 steps.
Auto reset	Set a time make device reboot.
Maximum Call Duration	Defines a value by users. That will limit the SIM/UIM card's total call duration. After the call duration excesses this value, no call will be made from this channel. The value range is1-65535. If user doesn't configure this value, Default is no max call duration limits for this channel.
Minimum Charging Time(sec)	A minimum charging time (in seconds) is defined during which no charging is done at carrier side. If the conversation time is even shorter, the total call duration will not decrease.
Alarm Threshold	When the SIM remain time is or less than this value, MWG will send the alarm SMS to remind the users of the SIM remain time.
Mobile Number (Receiving Alarm)	The mobile phone number which is used to receive the alarm SMS. Users can get SMS report of SIM/UIM card status (SIM Remain Time) in MWG.
SMS Description	Description of the alarm SMS.
Email To	Email Address of receiver.
CLIR	Caller ID display restrict. This function is used to restrict the mobile phone number by adding "#31#" before the mobile phone ID, this function should be supported by carrier.
Mobile Rx Gain	Receive gain of the mobile module, from PSTN side to IP side.
Mobile Tx Gain	Transit gain of the mobile module, from IP side to PSTN side

How to configure maximum call limitation

Preset: 1800 minutes (Ct) for each SIM

Preset: The SIM card billing every 60s (Cu)

So we have to configure maximum call duration as below:

Step = Cu = 60s;

Maximum Call Duration =total call minutes of SIM (minutes) * 60s / step = Ct * 60 / Cu

= 1800 * 60 / 60 = 1800 step

3.5.2 Band Settings

Figure 3-5-2 Band Settings

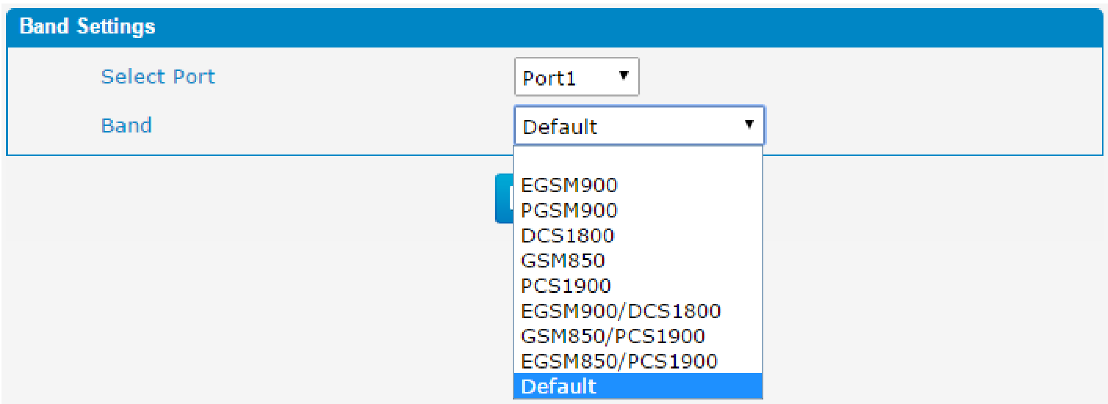


Table 3-5-2 Description of Band Settings

Parameters	Description
Select Ports	Select GSM channel, default Port 1
Band	According to carrier's band standards. Standards are as bellow: GSM: 850/900/1800/1900 MHz The band of this SIM card, you can choose GSM850, EGSM900,PGSM900, DCS1800,PCS1900, EGSM900/DCS1800, GSM850/PCS1900, EGSM850/PCS1900

3.5.3 Carrier

Figure 3-5-3 Carrier

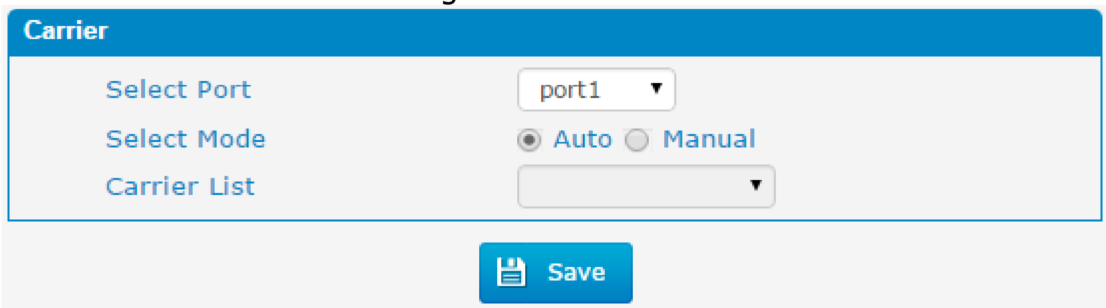


Table 3-5-3 Description of Carrier

Parameters	Description
Select Port	Select GSM channel, default Port 1
Select Mode	There are two modes to select carrier, automatic and manual. Automatic mode can automatically search operators. Manual mode can choose operators from the carrier list.
Carrier List	If you select manual mode, you can select carrier from carrier list.

3.5.4 IMEI

Figure 3-5-4 IMEI

Table 3-5-4 Description of IMEI

Parameters	Description
Port	GSM/CDMA channel
IMEI	International Mobile Equipment Identity of this module, it's not changeable.

3.5.5 PIN Management

Figure 3-5-5 PIN Management

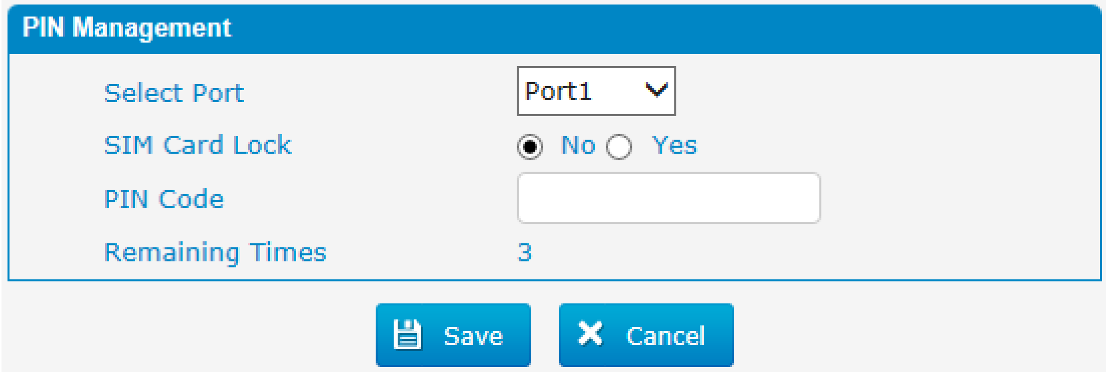


Table 3-5-5 Description of PIN Management

Parameters	Description
Port	Selects the GSM/CDMA channel number.
PIN Code	Personal identification number of SIM card. In the status of SIM card locked, PIN can be modified to prevent SIM card from being stolen.

3.5.6 Call Waiting

Call waiting is the same as mobile phone which to activate/deactivate supplementary service of SIM card. For more details of these services, please contact local providers.

Notes:

1. It takes several seconds to contact SIM carrier to get the call waiting status.
2. When call waiting is enabled, follow me will not work.

Figure 3-5-6 Call Waiting

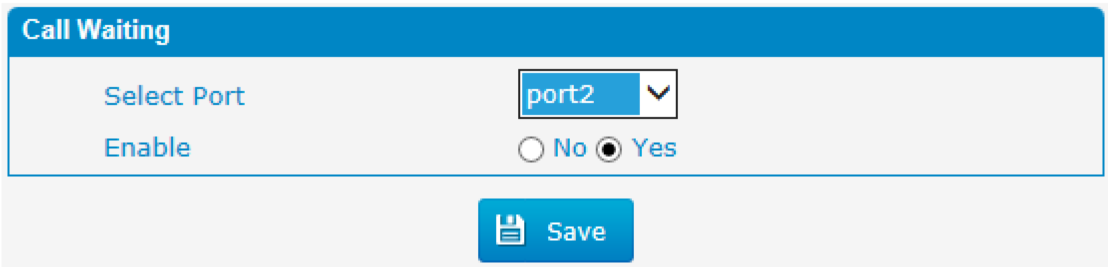


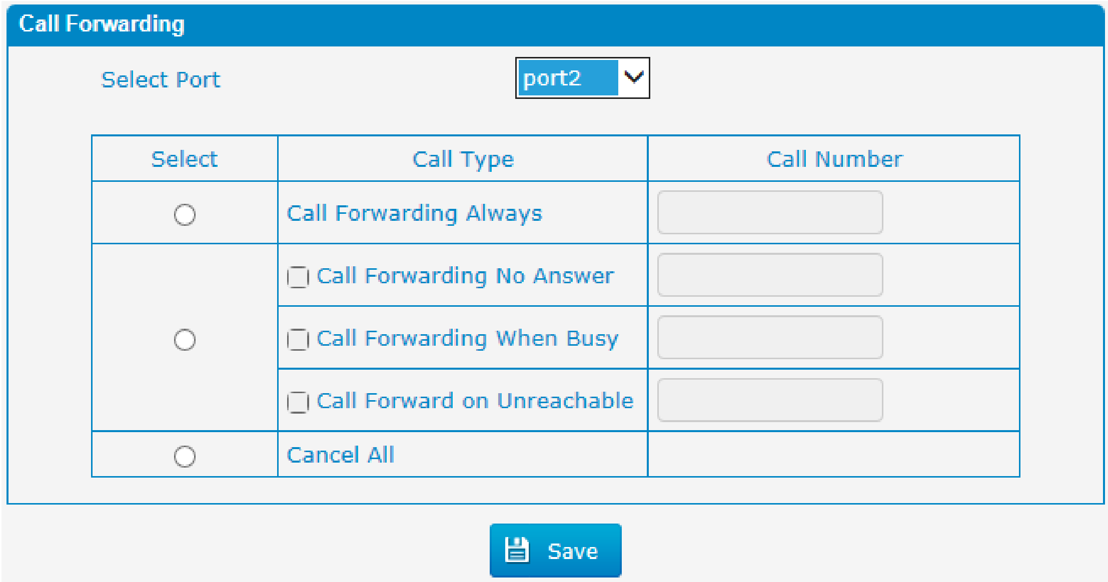
Table 3-5-6 Description of Call Waiting

Parameters	Description
Select Port	Selects the GSM/CDMA channel number

3.5.7 Call Forwarding

Call forwarding is the same as mobile phone which to activate/deactivate supplementary service of SIM card. For more details of these services, please contact local providers.

Figure 3-5-7 Call Forwarding

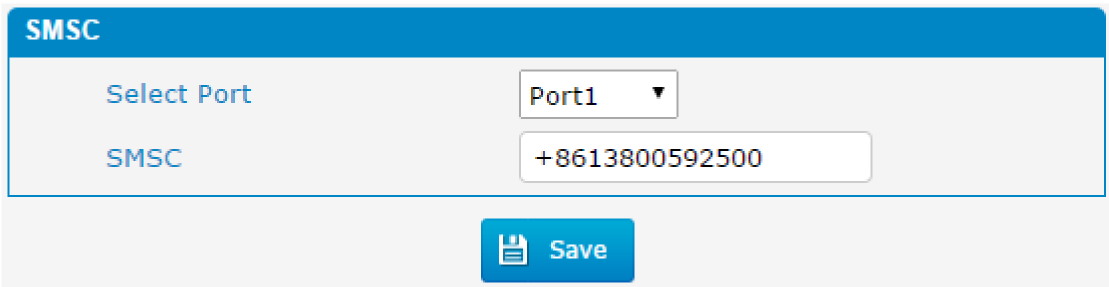


Notes:
1. It takes several seconds to contact SIM carrier to get the status of follow me feature.
2. The Call forwarding feature needs the support of SIM carrier.

3.5.8 SMSC

SMS center of mobile, in most places, the cellular module will automatically detect the SMSC number. This configurable option is used in a situation that the SMSC number could not detected by cellular module. When such case happens, please contact with mobile service provider to identify the SMSC number and then add SMSC number in SMSC configurable web interface.

Figure 3-5-8 SMSC



3.5.9 Send Message

Figure 3-5-9 Send Message

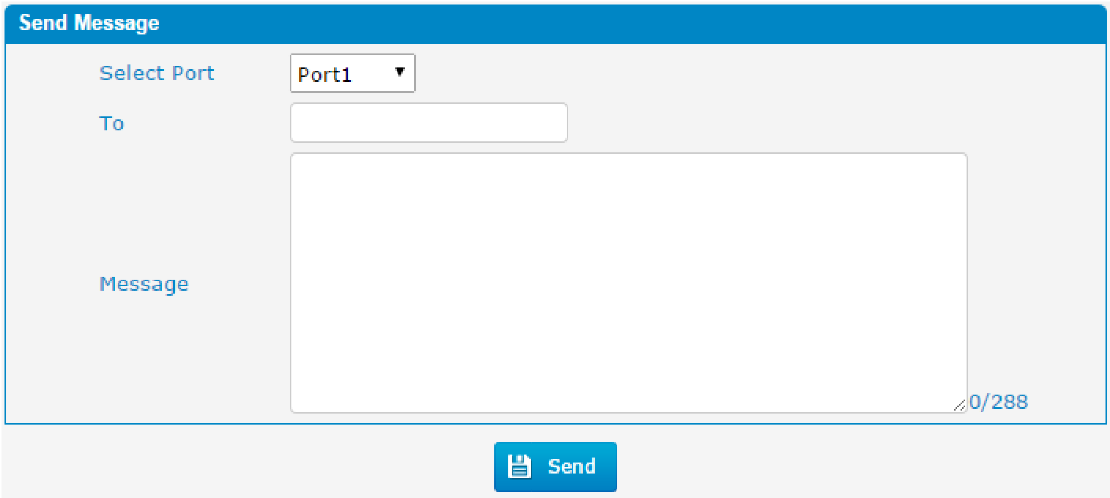


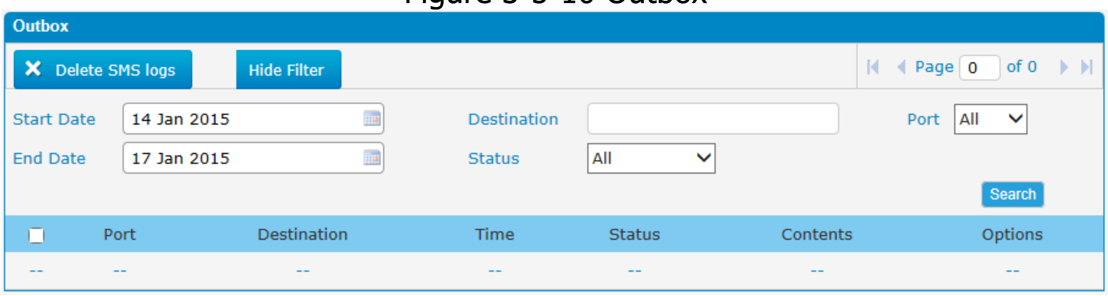
Table 3-5-9 Description of Send Message

Parameters	Description
Select Port	Users can select a defined channel or random channel to send SMS. Input the receiver's mobile phone number to send SMS. Choose the channel to send the SMS
Send To	Mobile phone number of the receiver
Message	Content of the SMS. The length is limited to 288 characters.

3.5.10 Outbox

To check the SMS we sent, we can check it in outbox page, there are some filters for searching the SMS we want. We can also check the status of email below.

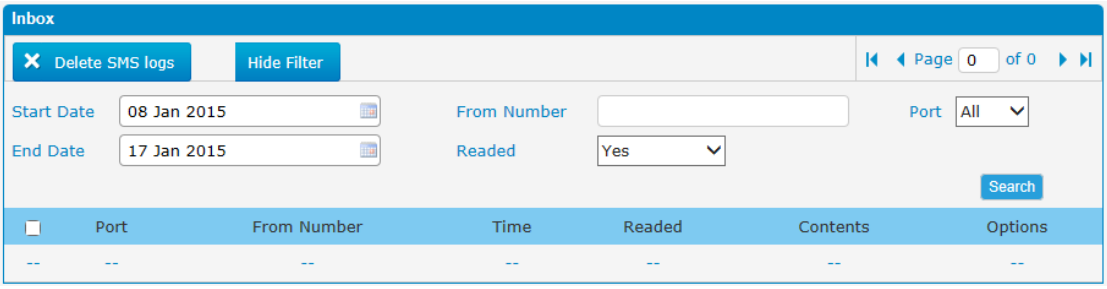
Figure 3-5-10 Outbox



3.5.11 Inbox

MWG1004 can check the incoming SMS also in this page; we can search SMS via filters like date, port and read status etc. We can also reply this SMS directly in this page via the same port.

Figure 3-5-11 Inbox



3.5.12 USSD

USSD (Unstructured Supplementary Service Data) is a Global System for Mobile(GSM) communication technology that is used to send text between a mobile phone and an application program in the network. Applications may include prepaid roaming or mobile chatting.

Figure 3-5-12 USSD

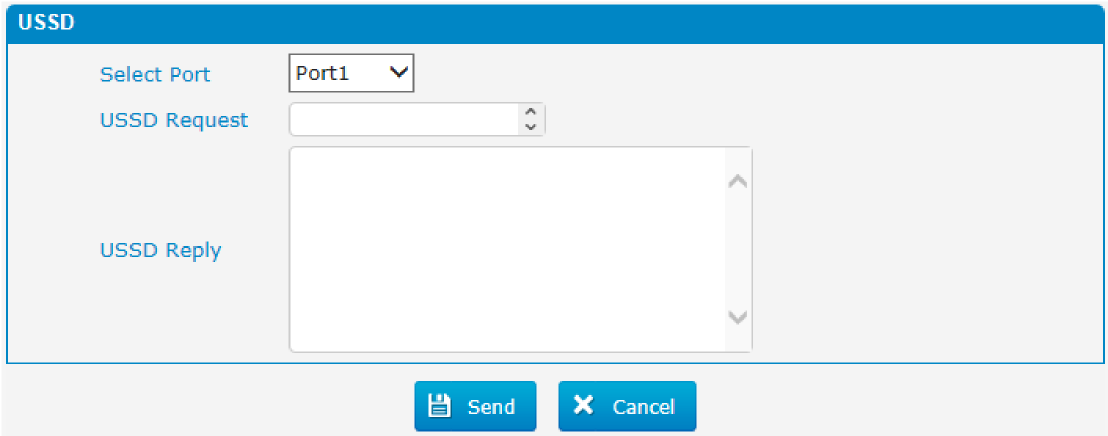


Table 3-5-12 Description of USSD

Parameters	Description
Port	Select the GSM channel to send USSD
USSD Request	Display the result of sending USSD
USSD Reply	Display results of USSD

3.6 Routing Configuration

3.6.1 Routing Parameter

Figure 3-6-1 Routing Parameter

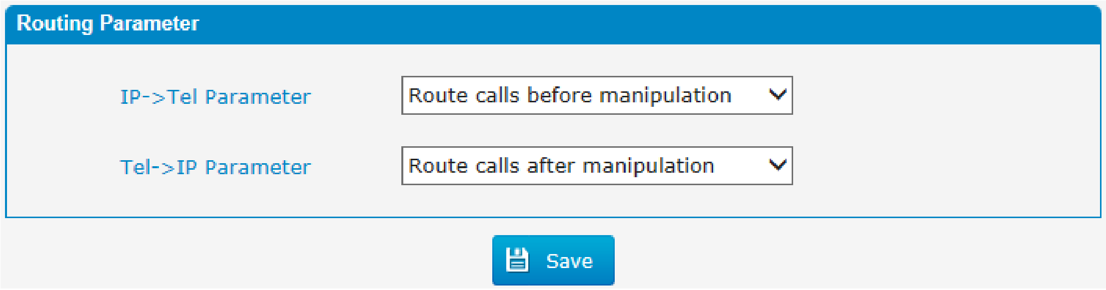


Table 3-6-1 Description of Routing Parameter

Parameters	**Description**
Tel->IP Parameter	Global parameters, it will take effect while number manipulation configured.
Route calls after manipulation	The parameters indicate that the gateway will select Tel->IP routes after number manipulation completed.
Route calls before manipulation	The parameters indicate that the gateway will select Tel->IP routes before number manipulation completed.

3.6.2 IP ->Tel Routing

Figure 3-6-2 IP ->Tel Routing

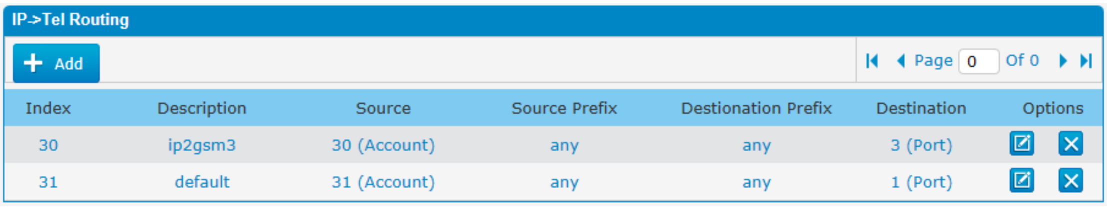


Figure 3-6-2a Add IP ->Tel Routing

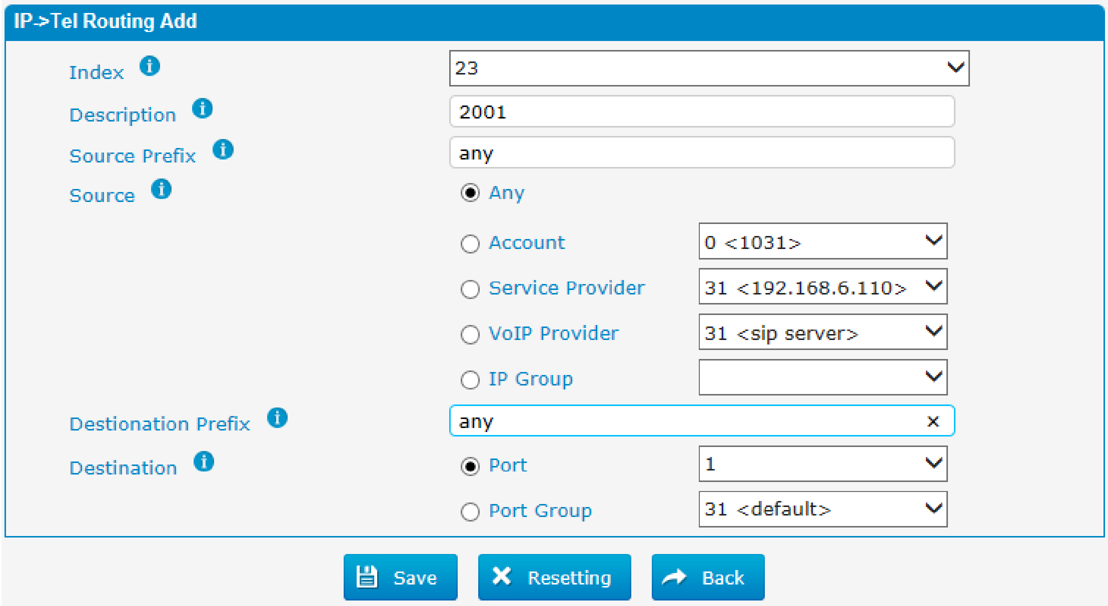


Table 3-6-2 Description of IP ->Tel Routing

Parameters	**Description**
IP ->Tel Routing	This item is used to configure outgoing call routes which can be used for receive the calls from the IP side
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31. The route preferentially match the rules which the value of index is smaller. Index 31 is default route on gateway which to be match all prefixes.
Description	It describes the route for the ease of identification. Its value is character string.
Source	It specifies the IP of the caller.
Source Prefix	All the caller number must match the source prefix. It specifies the source prefix allow to send call out ●Any: include anonymous,0XXXX,1[2-9]XXXX etc. ●X: Any digit from 0-9. ●Z: Any digit from 1-9. ●N: Any digit from 2-9. ●[1235-9]: Any digit in the brackets (in this example, 1,2,3,5,6,7,8,9). Example 1: NXXXXXX would match normal 7 digit dialings. Example 2: 1[5-8]6:consist of some prefix, include 156,166,176,186.
Destination Prefix	All the called number must match the destination prefix, the call prefix indicates the connected number ●Any: include anonymous,0XXXX,1[2-9]XXXX etc. ●X: Any digit from 0-9. ●Z: Any digit from 1-9. ●N: Any digit from 2-9. ●[1235-9]: Any digit in the brackets (in this example, 1,2,3,5,6,7,8,9).

	Example 1: NXXXXXX would match normal 7 digit dialings. Example 2: 1[5-8]6:consist of some prefix, include 156,166,176,186.
Destination	It specifies destination Port or Port Group

3.6.3 Tel->IP Routing

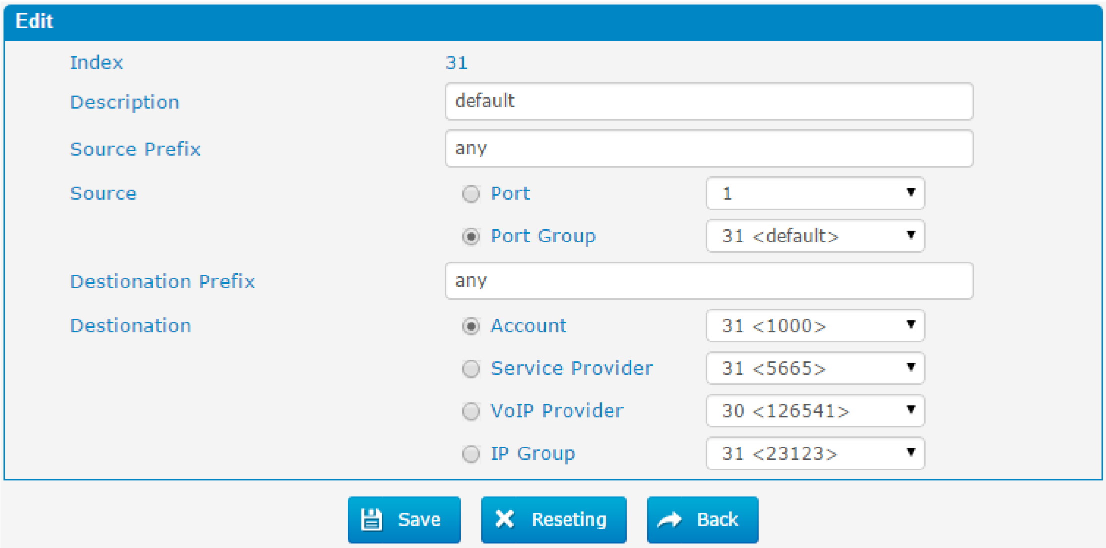
Figure 3-6-3 Tel->IP Routing

Table 3-6-3 Description of Tel->IP Routing

Parameters	Description
Tel -> IP Routing	This item is used to configure incoming call routes which can be used for receive the calls from the mobile.
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31. The route preferentially match the rules which the value of index is smaller.
Description	It describes the route for the ease of identification. Its value is character string.
Source	It specifies the Port or Port Group which will receive the calls from mobile.
Source Prefix	All the caller number must match the source prefix. It specifies the source prefix allow to send call out ●Any: include anonymous,0XXXX,1[2-9]XXXX etc. ●X: Any digit from 0-9. ●Z: Any digit from 1-9. ●N: Any digit from 2-9. ●[1235-9]: Any digit in the brackets (in this example, 1,2,3,5,6,7,8,9). Example 1: NXXXXXX would match normal 7 digit dialings. Example 2: 1[5-8]6:consist of some prefix, include 156,166,176,186.
Destination Prefix	All the called number must match the destination prefix, the call prefix indicates the connected number ●Any: include anonymous,0XXXX,1[2-9]XXXX etc. ●X: Any digit from 0-9. ●Z: Any digit from 1-9. ●N: Any digit from 2-9. ●[1235-9]: Any digit in the brackets (in this example, 1,2,3,5,6,7,8,9). Example 1: NXXXXXX would match normal 7 digit dialings. Example 2: 1[5-8]6:consist of some prefix, include 156,166,176,186.

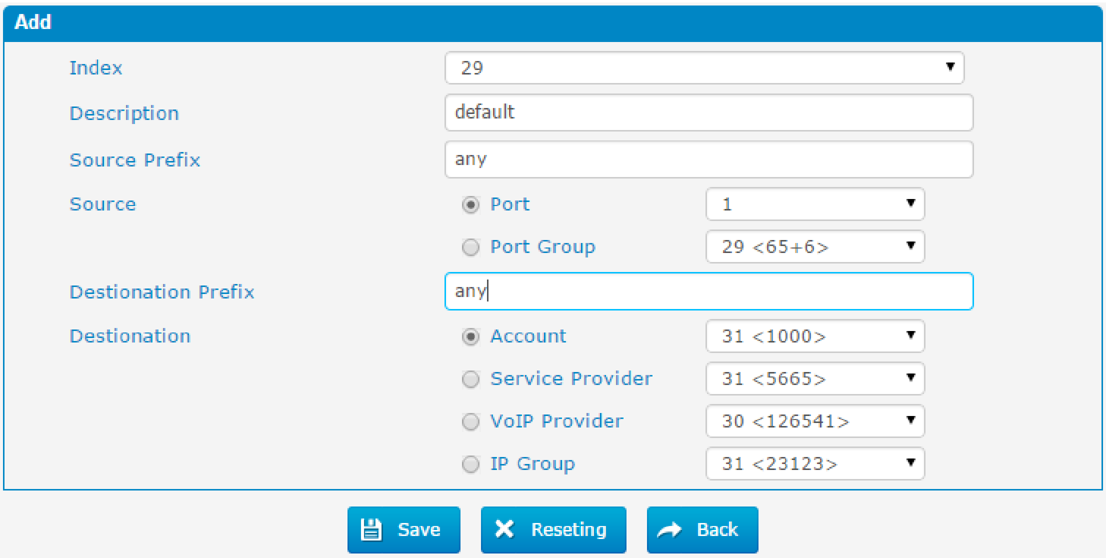
Destination	It specifies destination IP trunk or SIP server

Figure 3-6-3a Tel to IP routing Modify



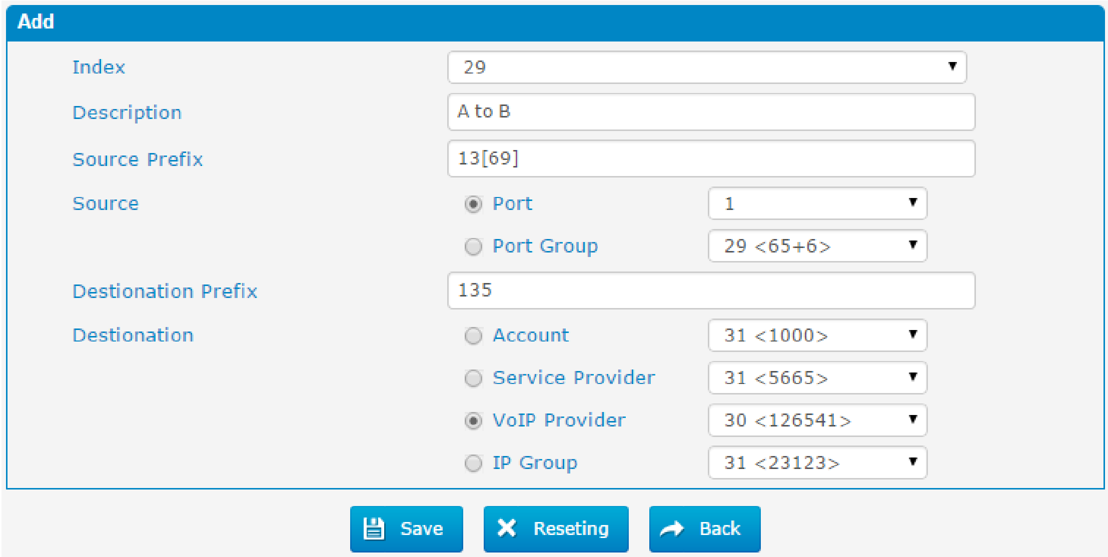
It's a default route configured in gateway . It allows any number from source Port Group send call to SIP account with any prefix.

Figure 3-6-3b Add Tel to IP routing



Add a mobile to VoIP route.It indicates that the calls coming from Port 1 will match the prefix "X.", "X." is a wildcard string which will match any prefix except "anonymous" calls. Meanwhile sending the calls destination SIP account if called number match with destination prefix "X.".

Figure 3-6-3c Tel to IP routing Modify

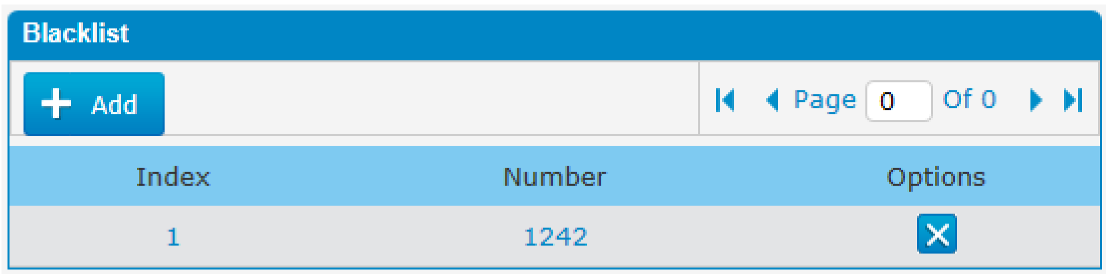


Add mobile to mobile route,it's mainly used for saving the cost between two carriers. It indicates that calls coming from Port 1 will match the prefix 13[69], "13[69]" include prefix 136 and 139, caller number can't match prefix 136 and 139 will be rejected by gateway . Meanwhile sending the calls to VoIP Provider 30 if called number match with prefix 135.

3.6.4 Blacklist

Blacklist is used to block an incoming/outgoing call. If the number of incoming/outgoing call is listed in the number blacklist, the caller will hear the following prompt: "The number you have dialed is not in service. Please check the number and try again". The system will then disconnect the call.

Figure 3-6-4 Blacklist



3.7 Manipulation Configuration

3.7.1 IP->Tel destination numbers manipulation

Figure 3-7-1 IP->Tel destination numbers manipulation

Table 3-7-1 Description of IP->Tel destination numbers manipulation

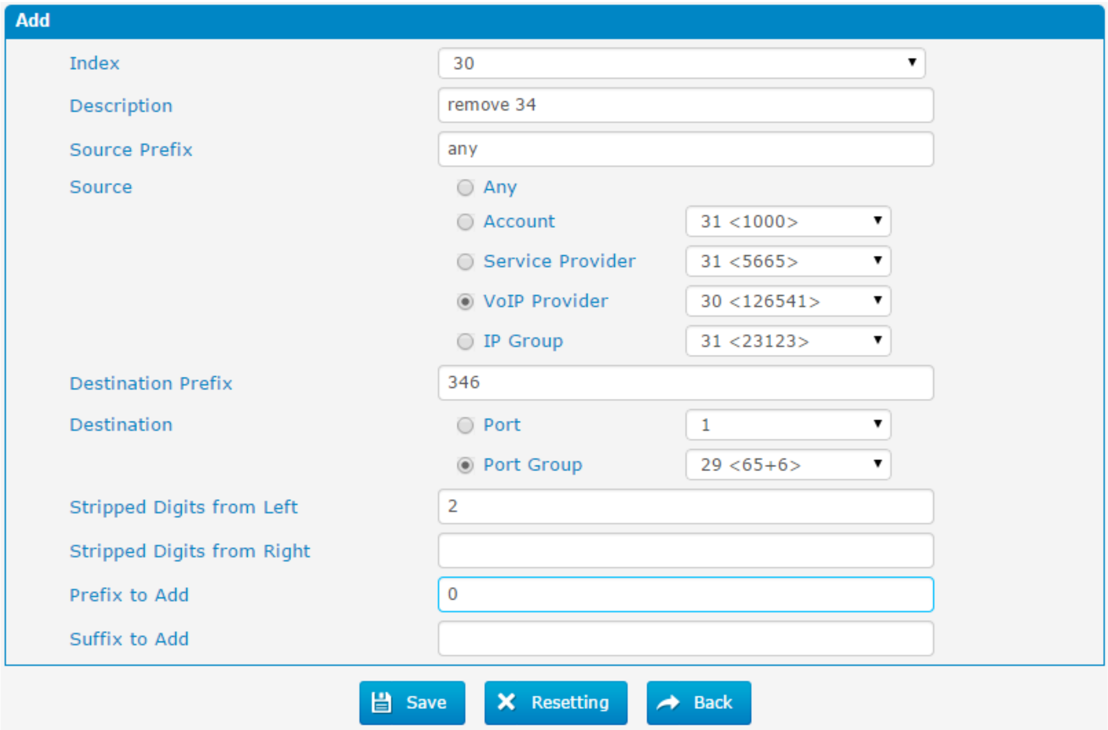
Parameters	**Description**
IP->Tel destination numbers manipulation	It is an optional configuration item, and is used to add a rule for changing number.
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31. The route preferentially match the rules which the value of index is smaller.
Description	It describes the rule for the ease of identification. Its value is character string.
Source	It specifies the source IP which will send the calls to gateway ● Any: any IP address ● Account ● IP: specific an IP address ● IP Group: specific an IP group ● SIP Server
Source Prefix	All the caller number must match the source prefix. It specifies the source prefix allow to send call out ●Any: include anonymous,0XXXX,1[2-9]XXXX etc. ●X: Any digit from 0-9. ●Z: Any digit from 1-9. ●N: Any digit from 2-9. ●[1235-9]: Any digit in the brackets (in this example, 1,2,3,5,6,7,8,9). Example 1: NXXXXXX would match normal 7 digit dialings. Example 2: 1[5-8]6:consist of some prefix, include 156,166,176,186.
Destination Prefix	All the called number must match the destination prefix, the call prefix indicates the connected number ●Any: include anonymous,0XXXX,1[2-9]XXXX etc. ●X: Any digit from 0-9. ●Z: Any digit from 1-9. ●N: Any digit from 2-9. ●[1235-9]: Any digit in the brackets (in this example, 1,2,3,5,6,7,8,9).

	Example 1: NXXXXXX would match normal 7 digit dialings. Example 2: 1[5-8]6:consist of some prefix, include 156,166,176,186.
Destination	It specifies destination Port or Port Group
Stripped Digits from Left	It specifies the length of the digits to be deleted from left.
Stripped Digits from Right	It specifies the length of the digits to be deleted from Right
Prefix to Add	Add the new digits in front of the original number.
Suffix to Add	Add the new digits at the end of the original number.

Example :

Add an IP->Tel Manipulation, to change the called number from 346888888 to 06888888

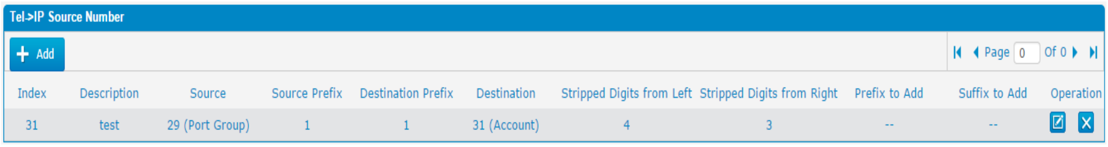
Figure 3-7-1a IP->Tel destination numbers manipulation



It indicates that calls coming from VoIP Provider will match the prefix ”any”, and the called number which match with the prefix ”346” will delete 2 digits in front of it and replace it by digit ”0”.

3.7.2 Tel->IP destination numbers manipulation

Figure 3-7-2 Tel->IP destination numbers manipulation


Tel->IP Source Number

+ Add | Page 0 Of 0

Index	Description	Source	Source Prefix	Destination Prefix	Destination	Stripped Digits from Left	Stripped Digits from Right	Prefix to Add	Suffix to Add	Operation
31	test	29 (Port Group)	1	1	31 (Account)	4	3	--	--	

Table 3-7-2 Description of Tel->IP destination numbers manipulation

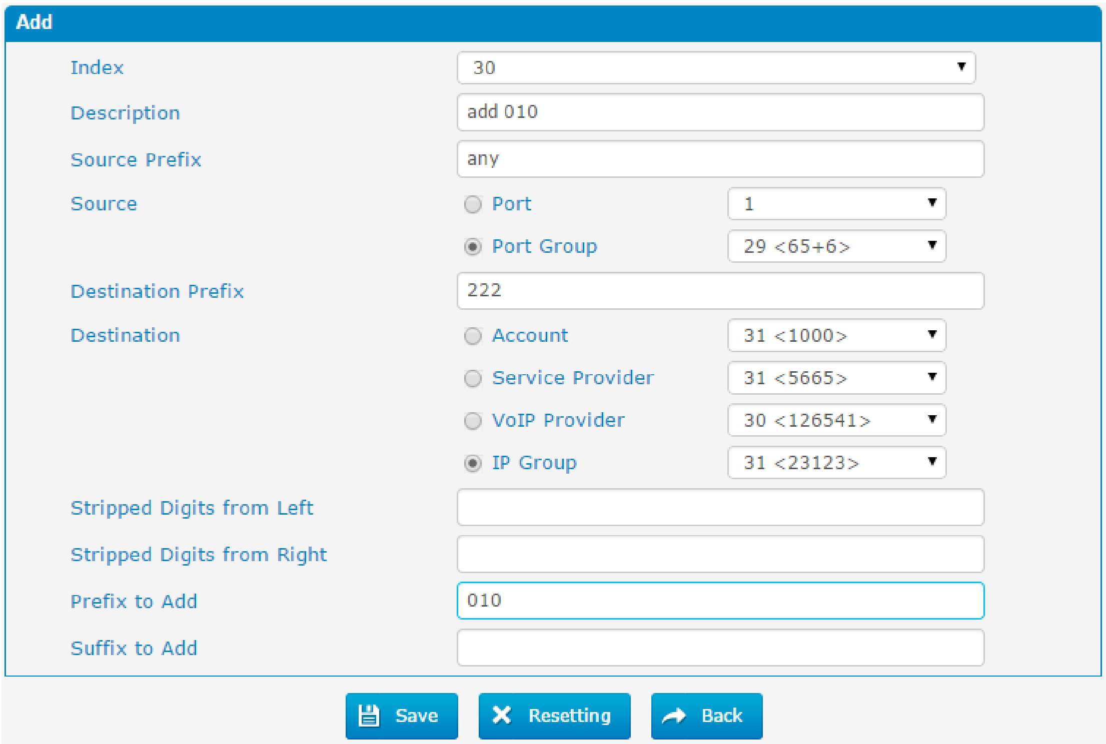
Parameters	Description
Tel->IP destination numbers manipulation	It is an optional configuration item which is used to add Tel-> IP destination number manipulation rules. The Tel-IP manipulation defines the rules of add and deletion of called numbers, which are referenced by Tel->IP routing.
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31 .
Description	It describes the route for the ease of identification. Its value is character string.
Source	It specifies the source port or port group which will send the calls to gateway
Source Prefix	All the caller number must match the source prefix. It specifies the source prefix allow to send call out ●Any: include anonymous,0XXXX,1[2-9]XXXX etc. ●X: Any digit from 0-9. ●Z: Any digit from 1-9. ●N: Any digit from 2-9. ●[1235-9]: Any digit in the brackets (in this example, 1,2,3,5,6,7,8,9). Example 1: NXXXXXX would match normal 7 digit dialings. Example 2: 1[5-8]6:consist of some prefix, include 156,166,176,186.
Destination Prefix	All the called number must match the destination prefix, the call prefix indicates the connected number ●Any: include anonymous,0XXXX,1[2-9]XXXX etc. ●X: Any digit from 0-9. ●Z: Any digit from 1-9. ●N: Any digit from 2-9. ●[1235-9]: Any digit in the brackets (in this example, 1,2,3,5,6,7,8,9). Example 1: NXXXXXX would match normal 7 digit dialings. Example 2: 1[5-8]6:consist of some prefix, include 156,166,176,186.
Destination	Its specifies destinations: SIP Account, IPs , IP Group, SIP Server
Stripped Digits from Left	It specifies the length of the digits to be deleted from left

Stripped Digits from Right	It specifies the length of the digits to be deleted from Right
Prefix to Add	Add the new digits in front of the original number
Suffix to Add	Add the new digits at the end of the original number

Example :

Add a Tel->IP Manipulation rule, to change the called number from 222 to 010222

Figure 3-7-2a Tel->IP destination numbers manipulation



It indicates that calls incoming call from mobile will match the prefix "any", and the called number which match with the prefix "222 " will be added 010 in front of called number.

3.7.3 Tel->IP source numbers manipulation

Figure 3-7-3 Tel->IP source numbers manipulation

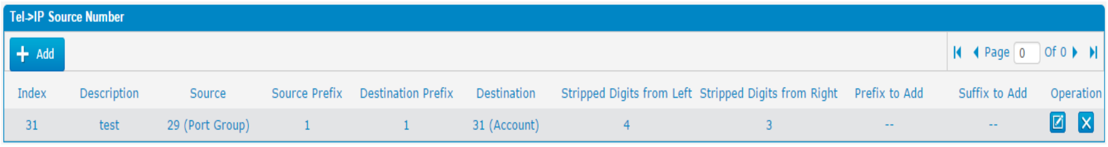


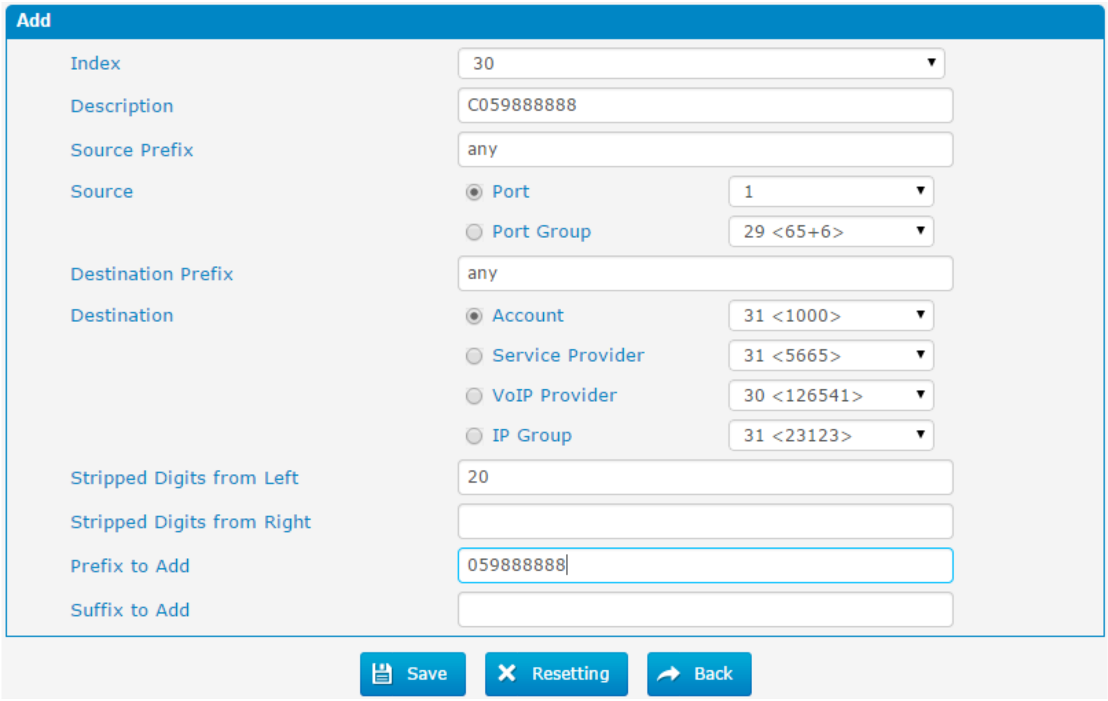
Table 3-7-3 Description of Tel->IP source numbers manipulation

Parameters	**Description**
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31 .
Description	It describes the route for the ease of identification. Its value is character string.
Source	It specifies the source port or port group which will send the calls to gateway
Source Prefix	All the caller number must match the source prefix. It specifies the source prefix allow to send call out ●Any: include anonymous,0XXXX,1[2-9]XXXX etc. ●X: Any digit from 0-9. ●Z: Any digit from 1-9. ●N: Any digit from 2-9. ●[1235-9]: Any digit in the brackets (in this example, 1,2,3,5,6,7,8,9). Example 1: NXXXXXX would match normal 7 digit dialings. Example 2: 1[5-8]6:consist of some prefix, include 156,166,176,186.
Destination Prefix	All the called number must match the destination prefix, the call prefix indicates the connected number ●Any: include anonymous,0XXXX,1[2-9]XXXX etc. ●X: Any digit from 0-9. ●Z: Any digit from 1-9. ●N: Any digit from 2-9. ●[1235-9]: Any digit in the brackets (in this example, 1,2,3,5,6,7,8,9). Example 1: NXXXXXX would match normal 7 digit dialings. Example 2: 1[5-8]6:consist of some prefix, include 156,166,176,186.
Destination	Its specifies destinations: SIP Account, IPs , IP Group, SIP Server
Stripped Digits from Left	It specifies the length of the digits to be deleted from left
Stripped Digits from Right	It specifies the length of the digits to be deleted from Right
Prefix to Add	Add the new digits in front of the original number
Suffix to Add	Add the new digits at the end of the original number

Example：

Add a Tel->IP manipulation, to change the caller number to 059888888

Figure 3-7-3a Tel->IP source numbers manipulation



It indicates that all incoming calls which matched with source & destination prefix ”any”, to delete original caller number and replace by 059888888.

3.8 VoIP Configuration

3.8.1 SIP Account

Figure 3-8-1 SIP Account

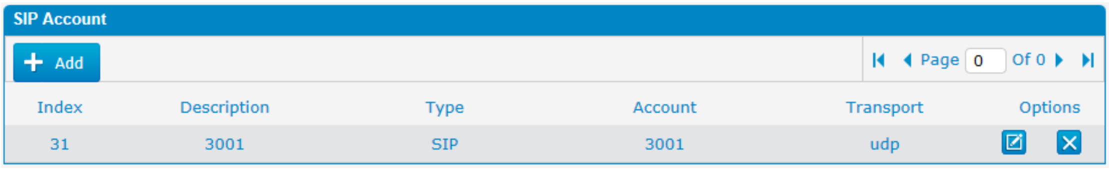


Figure 3-8-1a Add SIP Account

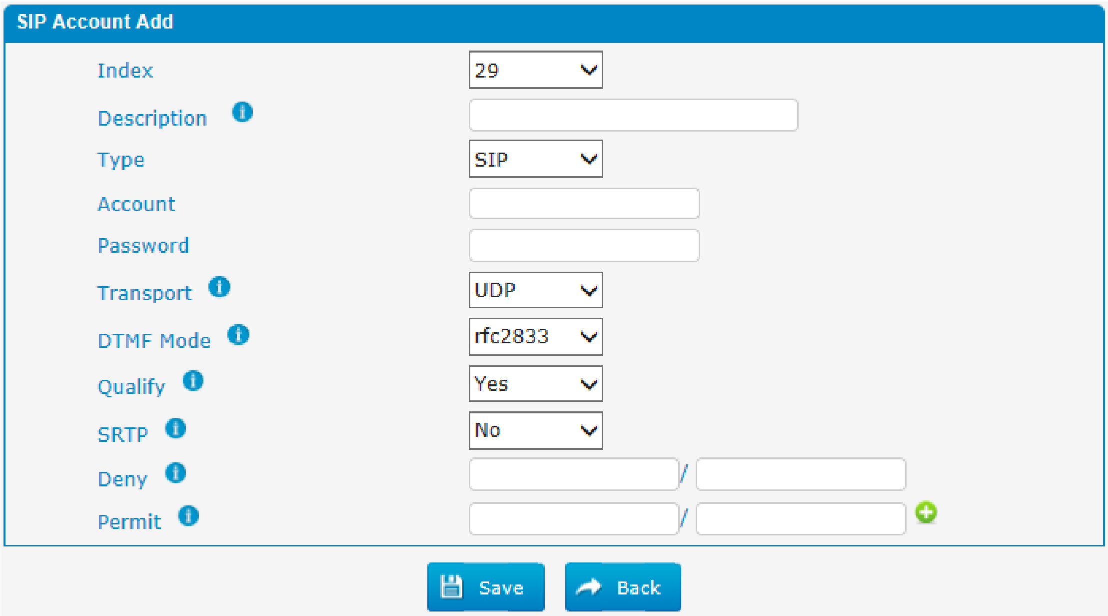


Table 3-8-1 Description of SIP Account

Parameters	**Description**
Index	It uniquely identifies a trunk. Its value is assigned globally, ranging from 0 to 31 .
Description	Define the name for this account.
Type	Choose the type of this trunk, SIP or IAX.
Account	Define the number for this account.
Password	Define the password for this account.
Transport	This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider. The options are UDP (default) or TCP or TLS.
DTMF Mode	Set default mode for sending DTMF of this trunk. Default setting: rfc2833, Info, Shortinfo,Inband, Auto
Qualify	Send checking alive packets to the SIP provider. when it's disabled, MWG1004 will ignore the reachability and the status of this account will be unmonitored.

Enable SRTP	Secure Real-time Transport Protocol, if it's enabled, the same setting should be enabled in IP phone side.
Deny	Control access to this account based on IP address.
Permit	We can also use CIDR notation for subnet masks.

3.8.2 IP Trunk (peer to peer mode)

Figure 3-8-2 IP Trunk

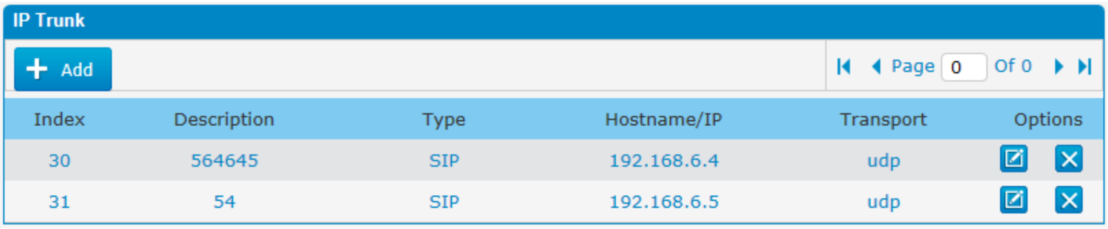


Figure 3-8-2a Add IP Trunk

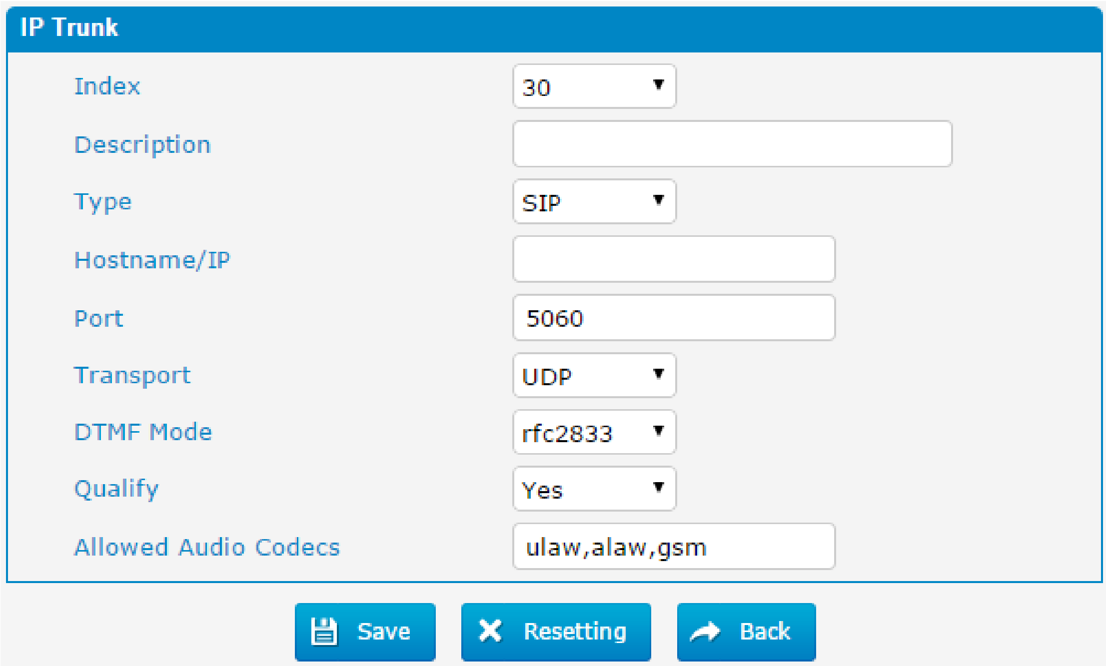


Table 3-8-2 Description of IP Trunk

Parameters	**Description**
IP Trunk	Add remote IP of Softswitch, SIP server which will send call traffics to gateway.
Index	It uniquely identifies a trunk. Its value is assigned globally, ranging from 0 to 31 .
Description	It describes the trunk for the ease of identification.
Type	Choose the type of this trunk, SIP or IAX
Hostname/IP Address	Service provider's hostname or IP address,5060 is the standard port number used by SIP protocol. Don't change this part if it is not required.

Transport	This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider. The options are UDP (default) or TCP or TLS.
DTMF Mode	Set default mode for sending DTMF of this trunk. Default setting: rfc2833, Info, Shortinfo,Inband, Auto
Qualify	Send checking alive packets to the SIP provider. when it's disabled, MWG1004 will ignore the reachability and the status of this account will be unmonitored.
Allow codecs	ulaw,alaw,gsm

3.8.3 VoIP Trunk

Figure 3-8-3 VoIP Trunk

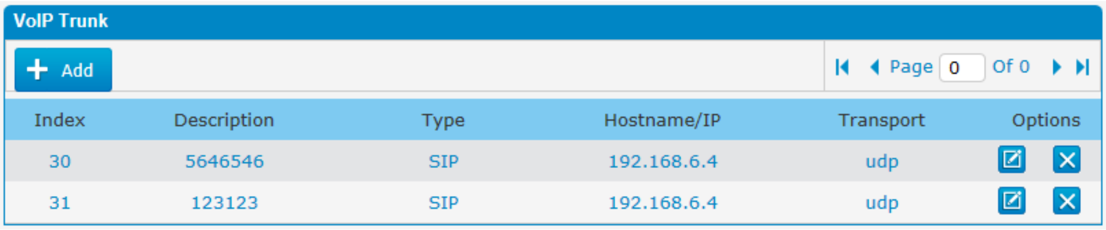


Figure 3-8-3a Add VoIP Trunk

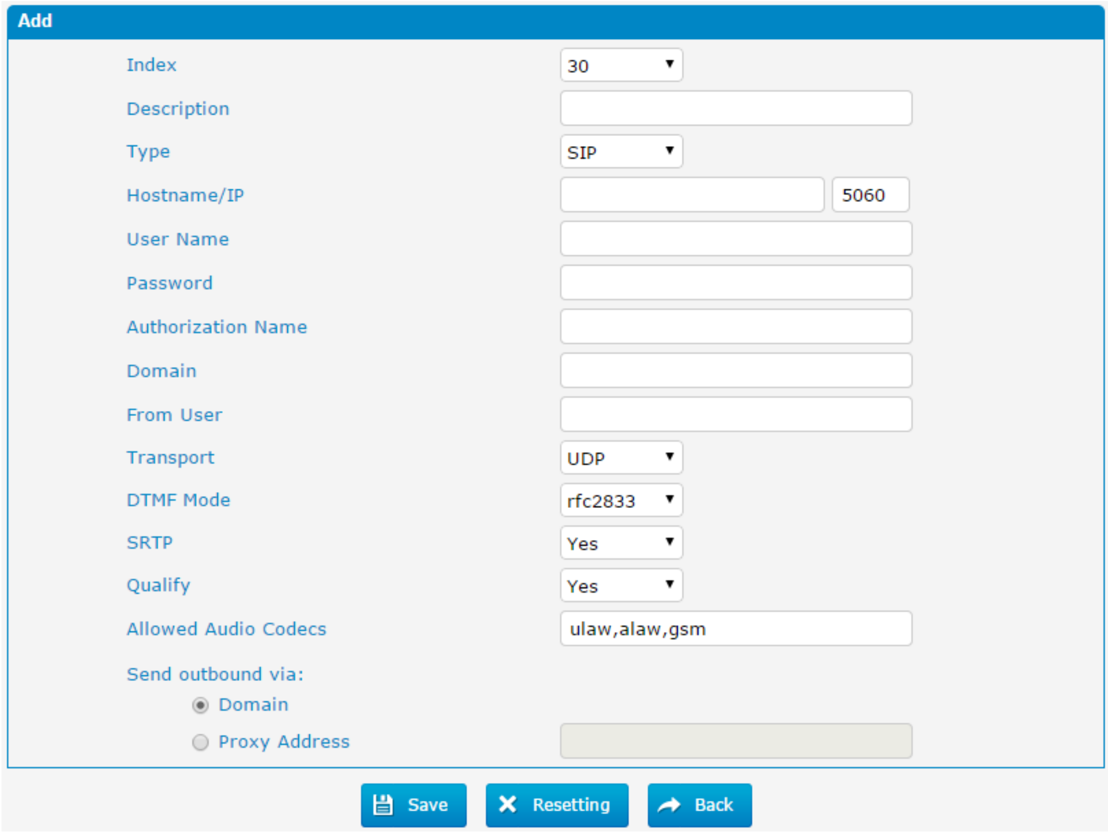


Table 3-8-3 Description of VoIP Trunk

Parameters	Description
Index	It uniquely identifies a trunk. Its value is assigned globally, ranging from 0 to 31 .
Description	It describes the trunk for the ease of identification.
Type	Choose the type of this trunk, SIP or IAX
Hostname/IP Address	Service provider's hostname or IP address, 5060 is the standard port number used by SIP protocol. Don't change this part if it is not required.
User Name	User name of SIP account.
Password	Password of SIP account.
Authorization Name	Used for SIP authentication, it's the same as user name generally.
Domain	VoIP provider's server domain name
From User	All outgoing calls from this SIP Trunk will use the From User in From Header of the SIP Invite package. Keep this field blank if it's not needed.
Transport	This will be the transport method used by the extension. The options are UDP (default) or TCP or TLS.
SRTP	Define if SRTP is enabled for this trunk, it depends on provider's configuration.
DTMF Mode	RFC2833, Info, Shortinfo, Inband, Auto.
Qualify	Send check alive packets to IP phones, when it's disabled, MWG1004 will ignore the reachability and the status of this account will be unmonitored.
Allow codecs	ulaw,alaw,gsm
Domain	VoIP provider's server domain name
Proxy Address	A proxy that receives requests from a client, even though it may not be the server resolved by the Request-URI.

3.9 Group Configuration

3.9.1 IP Trunk Group

Figure 3-9-1 IP Trunk Group

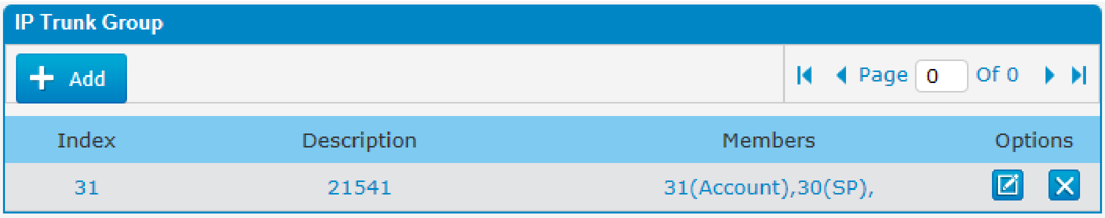
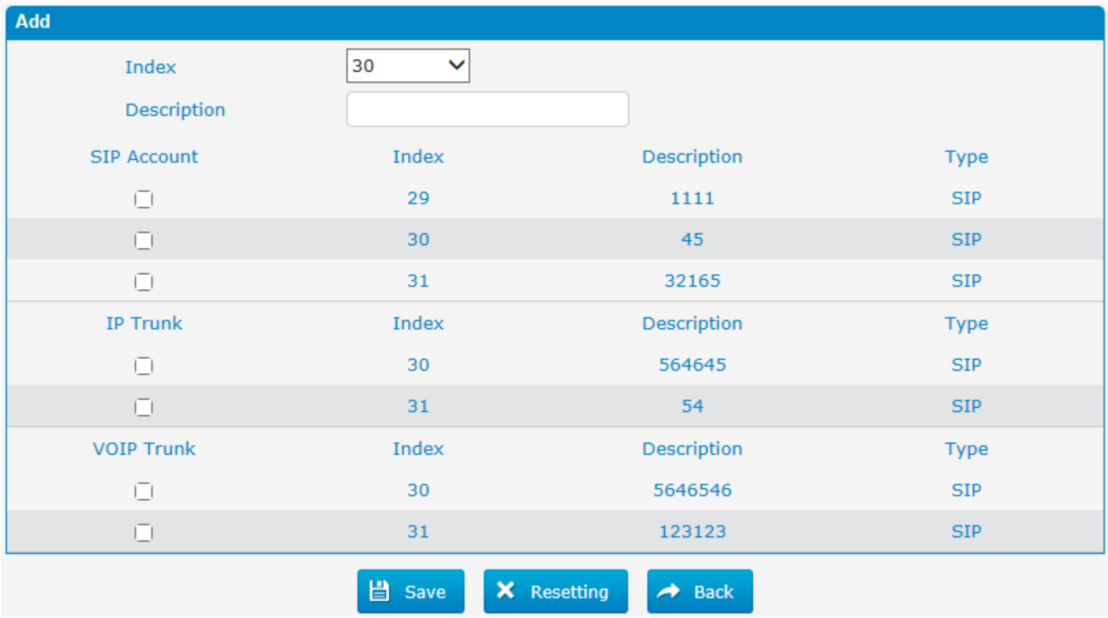


Table 3-9-1 Description of IP Trunk Group

Parameters	**Description**
IP Trunk Group	This configuration is optional, and is used to add the IP that have the same attributes to an IP group. The IP group will referenced by IP->Tel routing and number manipulation.
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31 .
Description	It describes the route for the ease of identification. Its value is character string.
Member	We can choose IP trunk/SIP account/VoIP trunk

Figure 3-9-1a IP Trunk Group Add



Click “Add ” to add a new one, or edit the default one. All the VoIP trunk will be listed here, we can choose the desired trunks as a group.

3.9.2 Port Group

To route the call to a GSM channels group, and dial out by the "Select Mode" we chose, MWG1004 can route the call in advanced method depending on your needs.

Figure 3-9-2 Port Group

Figure 3-9-2a Add Port Group

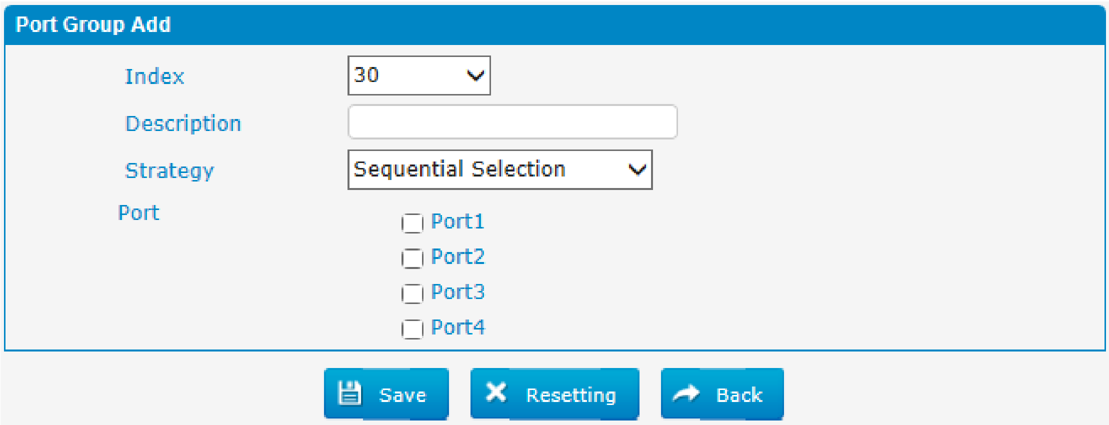


Table 3-9-2 Description of Port Group

Parameters	**Description**
Index	It uniquely identifies a Group. Its value is assigned globally, ranging from 0 to 31 .
Description	It describes the Port Group for the ease of identification. Its value is character string.
Select Mode	Choose the strategy of how to use these GSM channels. Default: The first channel will be used first always, when it's busy, MWG1004 will choose the next one. Sequence: The whole channels will be used one by one.
Port	The channels selected to right side will be a member of this port group.

3.10 system configuration

3.10.1 SIP settings

This is the SIP settings in MWG1004, including General settings, NAT, Codecs, Qos, Response code and Advanced settings.
This section describes how to configure SIP server and SIP parameters

3.10.1.1 SIP General setting

Figure 3.10.1.1 general setting

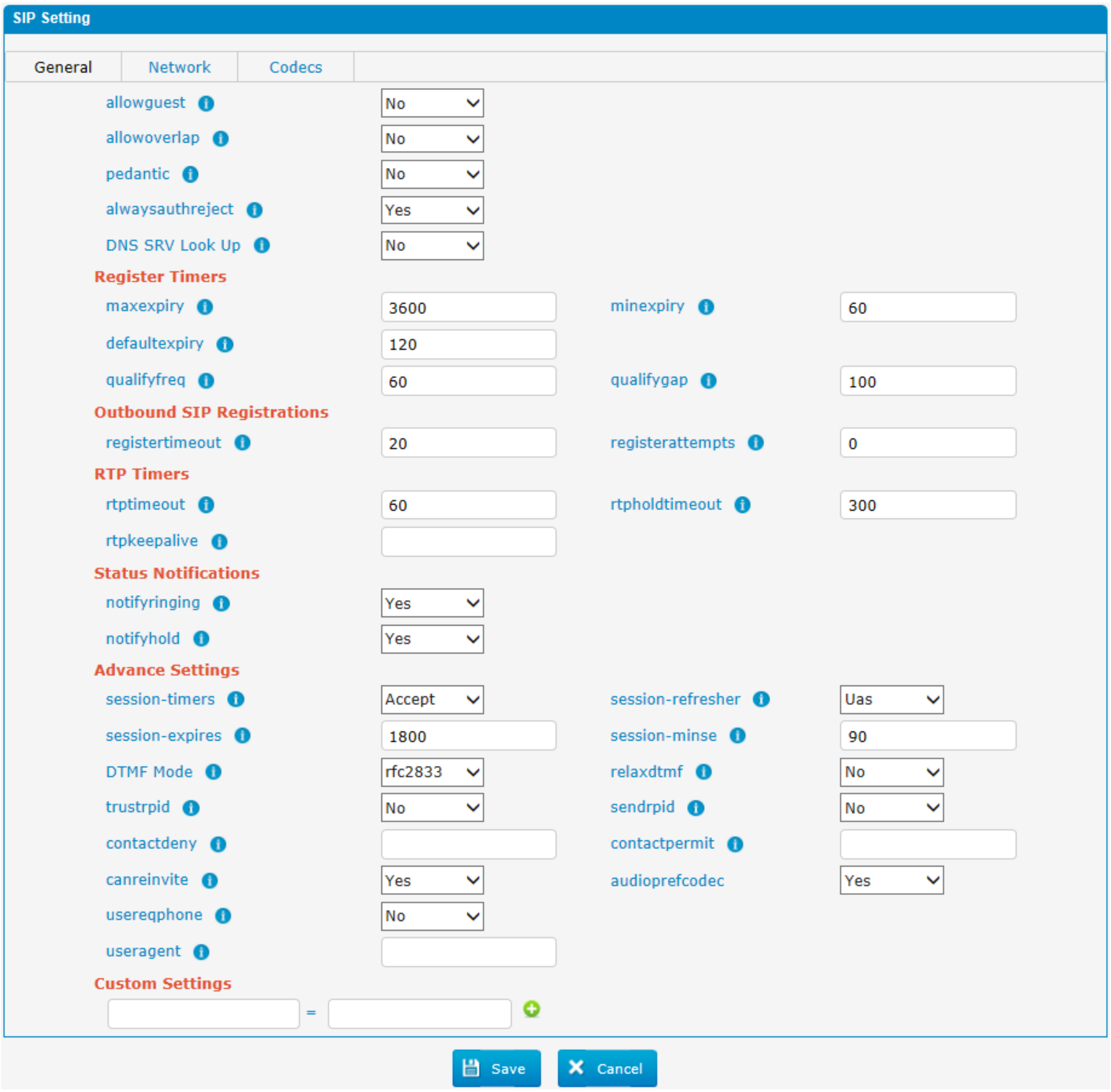


Table 3.10.1.1

Parameters	**Description**
Allowguest	Whether allow anonymous registration extension. Default: no. It's recommended to be disabled for security.
Allowoverlap	Disable overlap dialing support.(Default is yes)

Pedantic	Enable pedantic parameter. Default: no.
Always authreject	If enabled, when MWG1004 rejects "Register"or "Invite" packets, MWG1004 always respond the packets using "SIP404 NOT FOUND". It's recommended to be enabled for security.
DNS SRV Look Up	Please enable this option when your SIP trunk contains more than one IP address.
Maxexpiry	Maximum duration (in seconds) of a SIP registration.Default is 3600 seconds.
Minexpiry	Minimum duration (in seconds) of a SIP registration. Default is 60 seconds.
Defaultexpiry	Default Incoming/Outgoing Registration Time: Default duration (in seconds) of incoming/outgoing registration.
Qualifyfreq	How ofen to check for the host to be up in seconds and reported in milliseconds with sip show settings.
Qualifygap	Number of milliseconds between each group of peers being qualified.
Register Timeout	Number of seconds to wait for a response from a SIP registrar before timed out. Default is 20 seconds.
Register Attempts	The number of SIP REGISTER messages to send to a SIP Registrar before giving up. Default is 0 (no limit).
RTPtimeout	Terminate call if set # seconds of no RTP or RTCP activity on the audio channel when we're not on hold.
RTPholdtimeout	Both ends of the call time
RTPkeepalive	Time of packaging
Notifyringing	Control whether subscriptions already INUSE get send RINGING when another call is sent.
Notifyhold	Notify subscriptions on HOLD state.(default:no)
Session -timers	Enable session-timer mode, default: yes. If you found the call is cut off every 15 minutes every time, please disable this.
Session-refresher	Choose session-refresher, the default is Uas
Session-expires	The max refresh interval
Session-minse	The min refresh interval, which mustn't be shorter than 90s.
DTMF mode	Set default mode for sending DTMF. Default setting: rfc2833
Relaxdtmf	Relax dtmf handing
Trustrpid	If Remote-Party-ID should be trusted
Sendrpid	If Remote-Party-ID should be sent
Contactdeny Contactpermit	Use contactpermit and contactdeny to restrict at what IPs your users may register their phones.
Canreinvite	Asterisk by default tries to redirect the RTP media stream to go directly from the caller to the callee.Some

	devices do not support this (especially if one of them is behind a NAT). The default setting is YES
Audioprefcodec	Once enabled,When the caller call out via SIP/SPS trunks,the audio codec of calling channel whould be selected in preference.
usereqphone	This provider requires,User=phone on URI
User agent	To change the user agent parameter of asterisk, the default is “MWG1004”, you can change it if needed.

3.10.1.2 Network Configuration

Note: Configuration of this section is required when using remote extensions generally.

Figure 3.10.1.2 network configuration

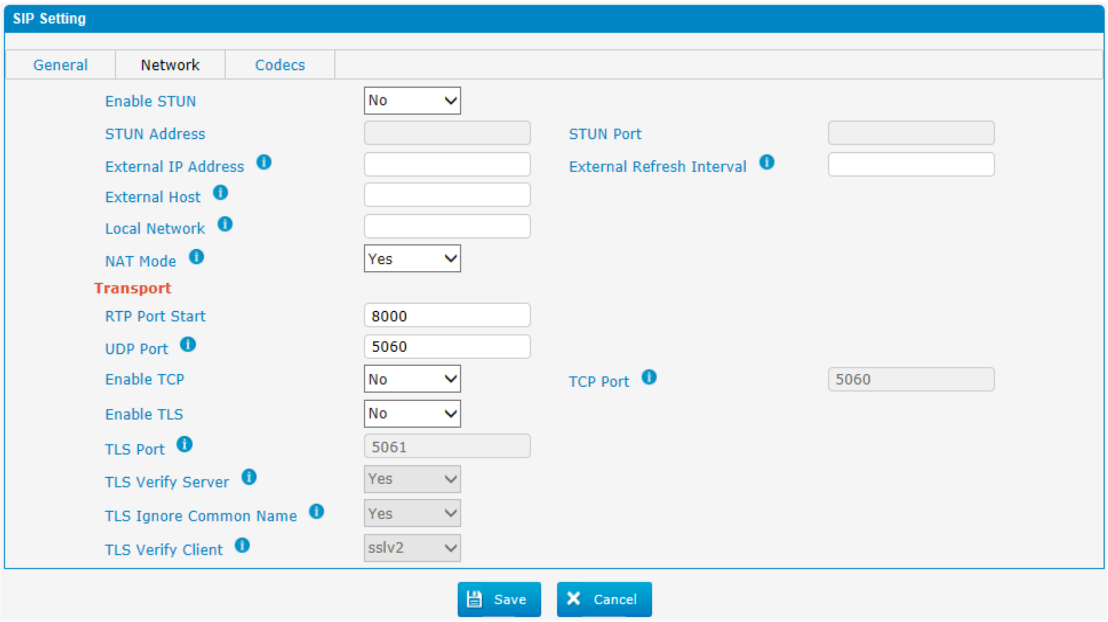


Table 3.10.1.2

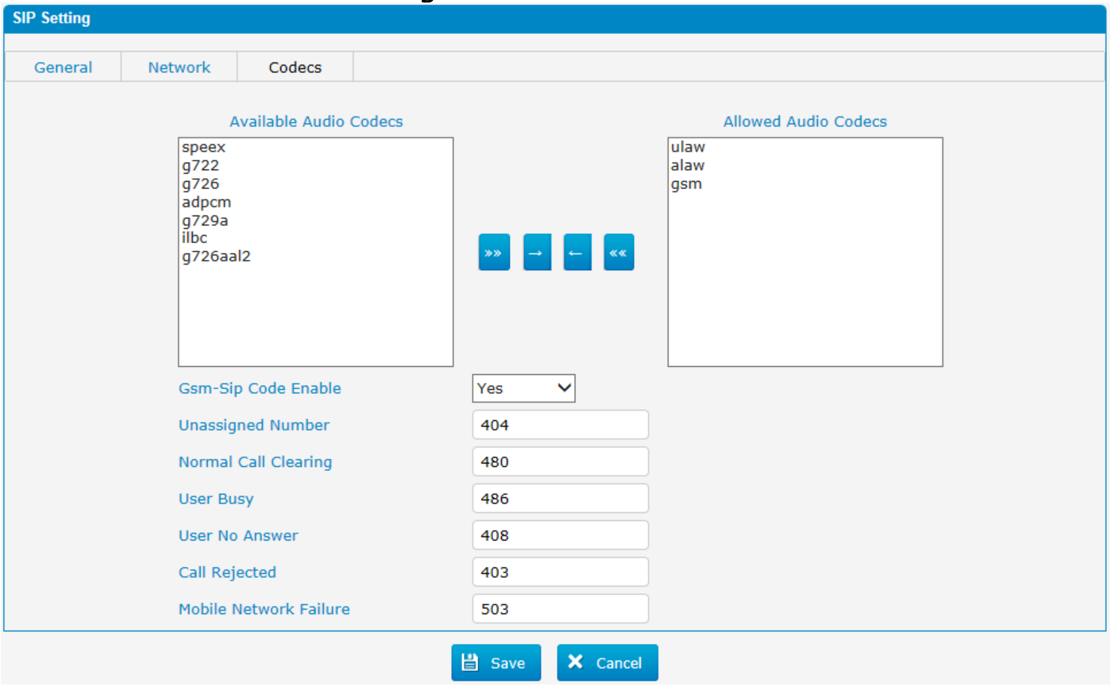
Parameters	**Description**
Enable STUN	STUN (Simple Traversal of UDP through NATs) is a protocol for assisting devices behind a NAT firewall or router with their packet routing.
STUN Address	The STUN server allows clients to find out their public address, the type of NAT they are behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between the client and the VOIP provider and so establish a call.
External IP Address	The IP address that will be associated with outbound SIP messages if the system is in a NAT environment.
External Refresh Interval	Used to identify the local network using a network number/subnet mask pair when the system is behind a

	NAT or firewall. Some examples of this are as follows: "192.168.0.0/255.255.0.0": All RFC 1918 addresses are local networks; "10.0.0.0/255.0.0.0": Also RFC1918; "172.16.0.0/12":Another RFC1918 with CIDR notation; "169.254.0.0/255.255.0.0": Zero conf local network. Please refer to RFC1918 for more information.
External host	Alternatively you can specify an external host, and the system will perform DNS queries periodically. This setting is only required when your public IP address is not static. It is recommended that a static public IP address is used with this system. Please contact your ISP for more information.
NAT mode	Global NAT configuration for the system; the options for this setting are as follows: Yes = Use NAT. Ignore address information in the SIP/SDP headers and reply to the sender's IP address/port. No = Use NAT mode only according to RFC3581. Never = Never attempt NAT mode or RFC3581 support. Route = Use NAT but do not include report in headers.
RTP Port Start	Beginning of RTP port range
UDP port	Port used for SIP registrations, Default is 5060
TCP port	Port used for SIP registrations, Default is 5060
TLS port	Port used for SIP registrations, Default is 5061
TLS Verify Server	When using MWG1004 as a TLS client, whether or not to verify server's certificate. It is "No" by default.
TLS Ignore Common Name	Set this parameter as "No", then common name must be the same with IP or domain name.
TLS Verify Client	When using MWG1004 as a TLS server, whether or not to verify client's certificate. It is "No" by default.
TLS Client Method	When using MWG1004 as TLS client, specify the protocol for outbound TLS connections. You can select it as tlsv1, sslv2 or sslv3.

3.10.1.3 codecs

We can choose the allowed codec in MWG1004, a codec is a compression or decompression algorithm that used in the transmission of voice packets over a network or the Internet. More information about codec, you can refer to this page: http://en.wikipedia.org/wiki/List_of_codecs

Figure 3.10.1.3 codecs



If you want to use codec G729, we recommend buying a license key and input it here.

3.10.2 IAX setting

IAX is the Internal Asterisk Exchange protocol, you can connect to MWG1004 or register IAX trunk to another IAX server. It's supported by the asterisk-based IPPBX.

Figure 3.10.2 IAX setting

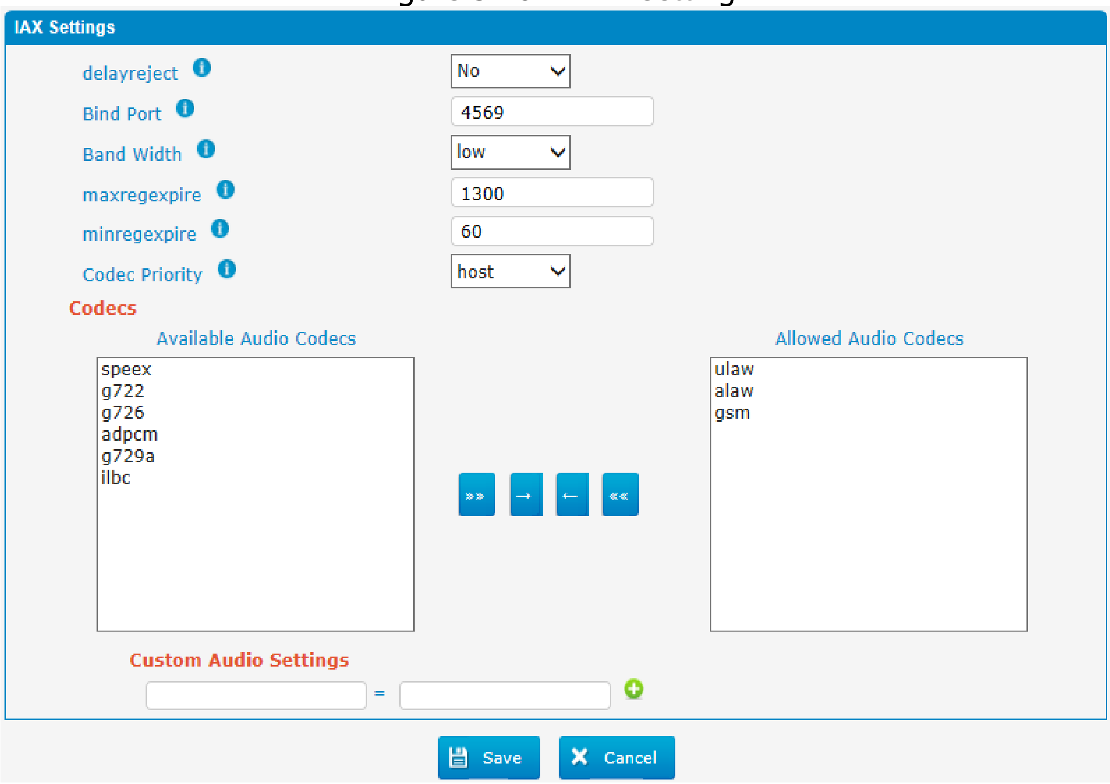
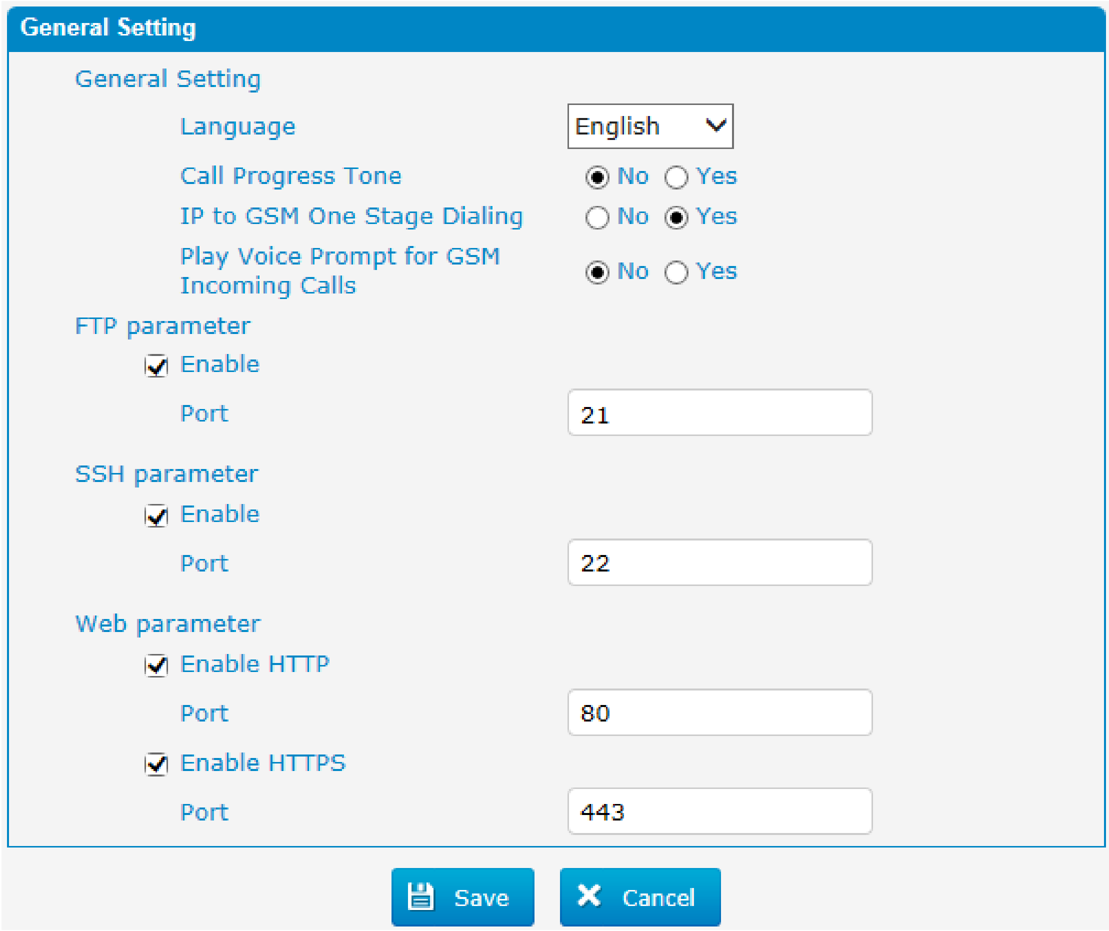


Table 3.10.2

Parameters	Description
Delayreject	Which will delay the sending of authentication reject for REGREQ or AUTHREP if there is a password
Bind port	Port used for IAX2 registrations. Default is 4569.
Bandwidth	Low/medium/high with this option you can control which codec to be used.
Max Registration Time	Maximum duration (in seconds) of an IAX2 registration. Default is 1300 seconds.
Min Registration Time	Minimum duration (in seconds) of an IAX2 registration. Default is 60 seconds.
Codec priority	Codec priority controls the codec negotiation of an inbound IAX call.This option is inherited to all user entities
Codec	Enable the codec you want for IAX communication.

3.10.3 General setting

Figure 3.10.3 general setting



3.11 Reports

3.11.1 CDR Report

The call log captures all call details, including call time, caller number, callee number, call type, call duration, etc. An administrator can search and filter call data by call date, caller/callee, trunk, duration, billing duration, status, or communication type.

Figure 3.11.1 CDR Report

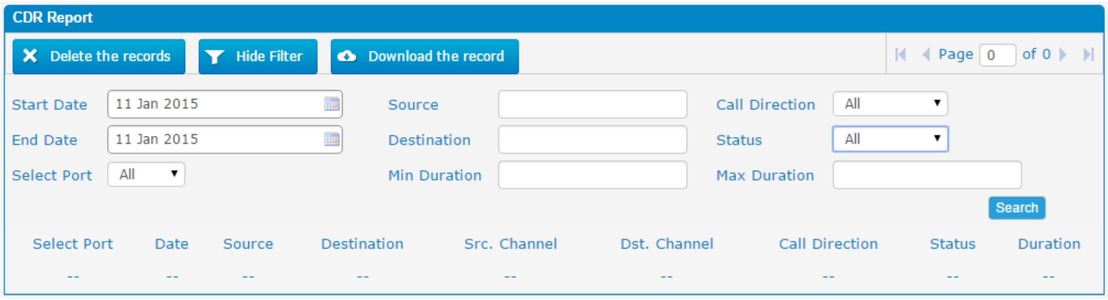


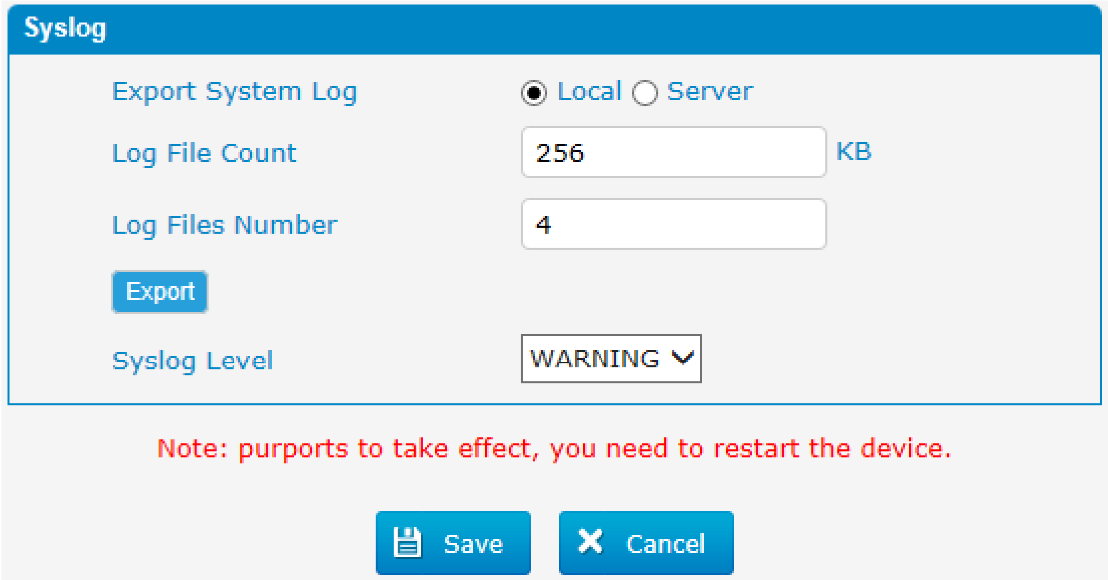
Table 3.11.1 CDR Report

Parameters	Description
Port	GSM port number
Date	start and end time of calls
Source	Call number
Destination	Called number
Src channel	Source channel
Dst channel	Destination channel
Call direction	IP to GSM: outbound calls from softswitch/IPPBX to mobile network GSM to IP: incoming calls from mobile network to IPPBX/Softswitch
Status	Answered: the call was established successful Canceled: the call was canceled by calling party No Carrier: the call was rejected by mobile network Not Answered: no body to answer the call Busy: user busy
Duration	Call duration of the call.

3.11.2 System logs

Syslog is a standard for network device data logging. It allows separation of the software that generates messages from the system that stores them and the software that reports and analyzes them. It also provides devices which would otherwise be unable to communicate a means to notify administrators of problems or performance. There are 6 levels of syslog, including DEBUG, NOTICE, WARNING and ERROR, EMERG,ALERT,CRIT,INFO.

Figure 3.11.2 system logs



3.12 System tools

3.12.1 Firmware update

Upgrading of the firmware is possible through the Administrator Web interface using a TFTP Server or an Upload
Enter your TFTP Server IP address and firmware file location, then click start to update the firmware

Notes:
1. If enabled "Reset configuration to Factory Defaults", System will restore to factory default settings.
2. When update the firmware, please don't turn off the power. Or the system will get damaged.

Figure 3.12.1

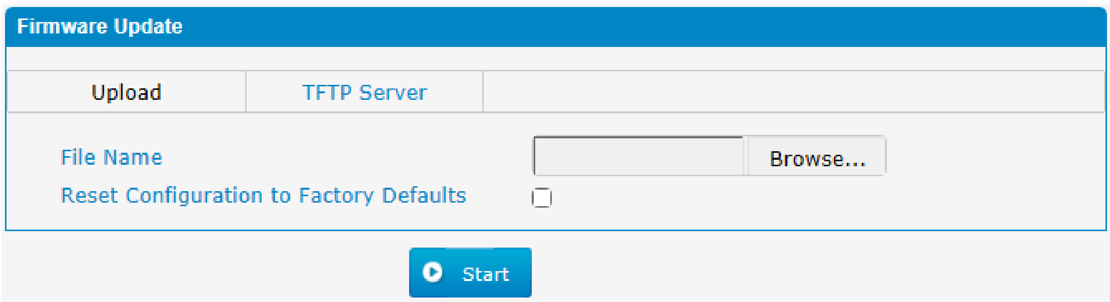


Table 3.12.1

Parameters	**Description**
Firmware update	Send package file from your computer to the device
File name	firmware

Factory reset	Reset Configuration to Factory Defaults
Browse	Choose File

3.12.2 SMTP Setting

To send the SMS or system alert to email address, please configure the Email settings first, and make sure SMTP test is successful.

Figure 3.12.2

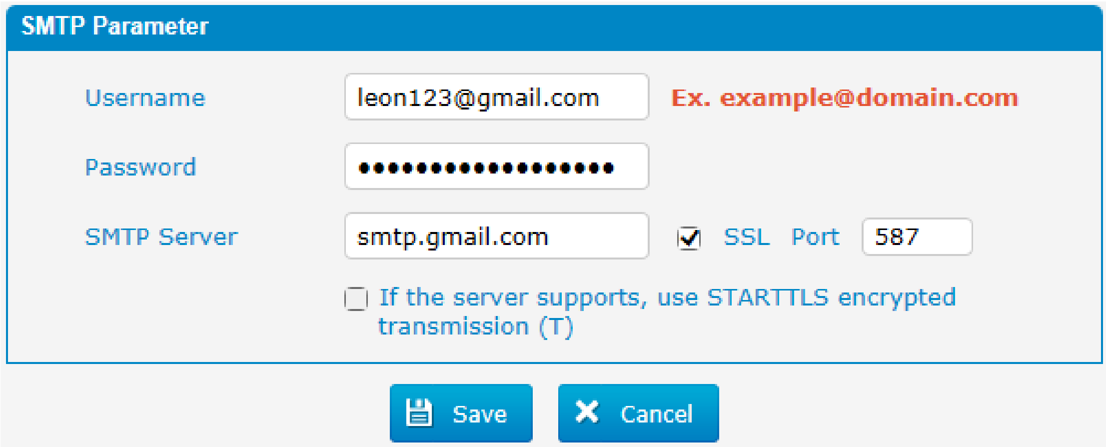


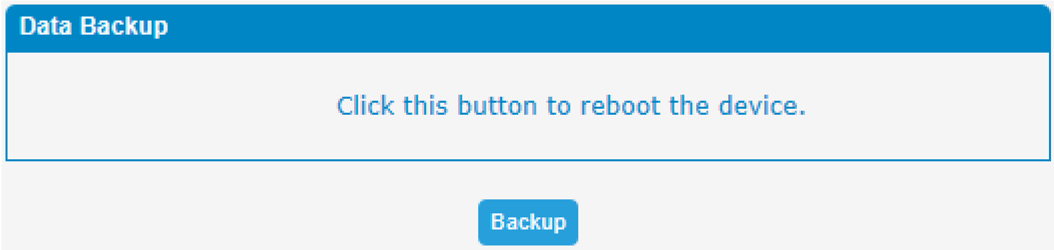
Table 3.12.2

Parameters	Description
E-mail Address	The E-mail Address that MWG1004 will use to send voice mail.
Password	The password for the email address used above
SMTP Server	The IP address or hostname of an SMTP server that the MWG1004 will connect to in order to send voice mail messages via email, i.e.mail.yourcompany.com.
SSL	If the server of sending email needs to authenticate the sender, you need to enable this. Note: Must be selected for Gmail or exchange server.
Port	SMTP Port: the default value is 25.
Use SSL/TLS to send secure message to server	If the server of sending email needs to authenticate the sender, you need to enable this. Note: Must be selected for Gmail or exchange server.

3.12.3 Data backup

We can backup up the configurations before reset MWG1004 to factory defaults

Figure 3.12.3



Click 'Backup' to download configuration file to your computer.

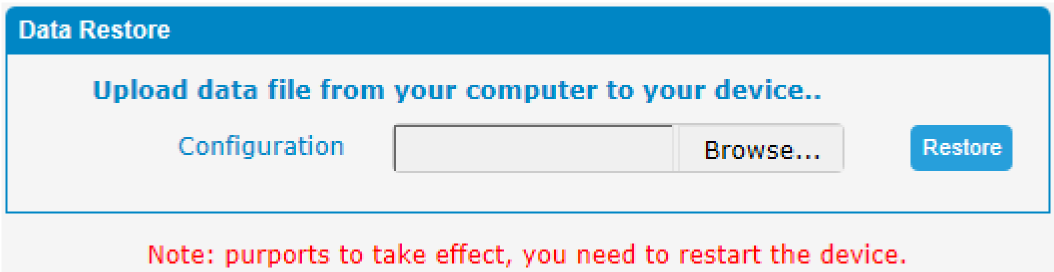
Notes:
1. Only configurations, custom prompts will be backed up.
2. When you have updated the firmware version, it’s not recommended to restore using old package.

3.12.4 Data restore

You can restore this configuration in case the unit loses it for any reason or to clone a unit with the configuration of another unit. The configuration backup configurations are in txt format. Please note that you can use a backup file from an older firmware version and use it in a unit with a more recent firmware version. However, a backup file from a newer firmware version than the one actually in the unit cannot be used for a restore operation on the unit.

Notes:
1. The upload process will last about 30s.
2. When you have updated the firmware version, it’s not recommended to restore using old package.

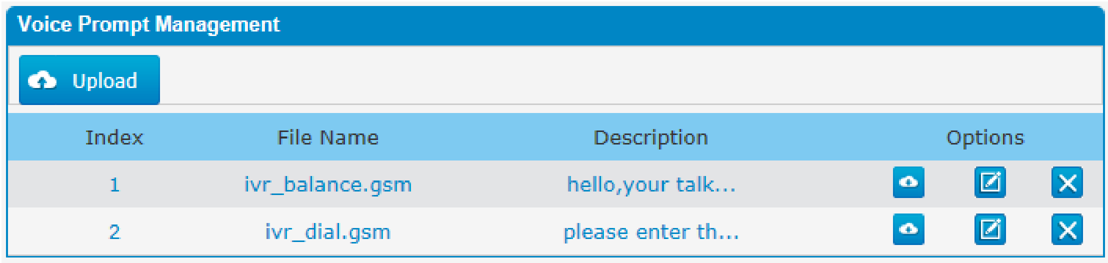
Figure 3.12.4



3.12.5 Voice Prompt Management

By default, when PSTN call incoming, the system will play the default IVR, and also the user can load custom IVR.

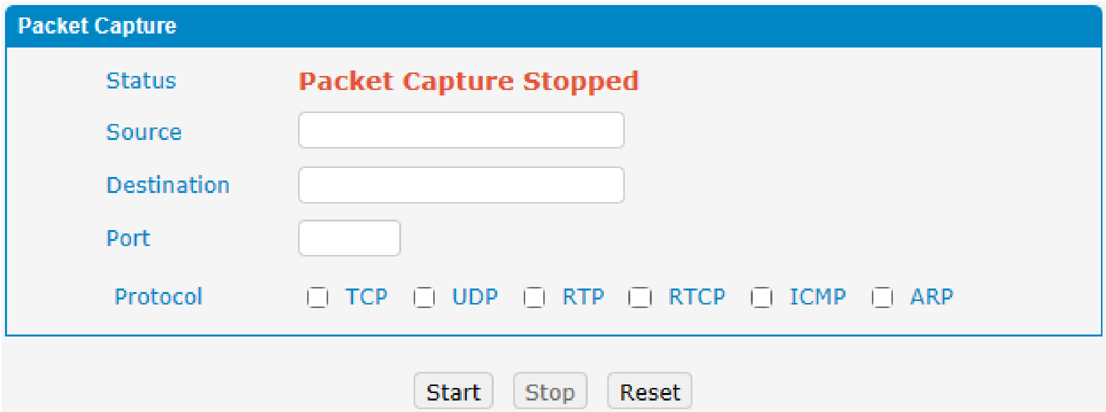
Figure 3.12.5 Voice Prompt Management



Note: the customize voice files can be recorded using Windows recording programs, the sound format is 8000Hz, 16 bit sampling in mono, with wav/gsm format, size of files cannot be exceed 190KB.

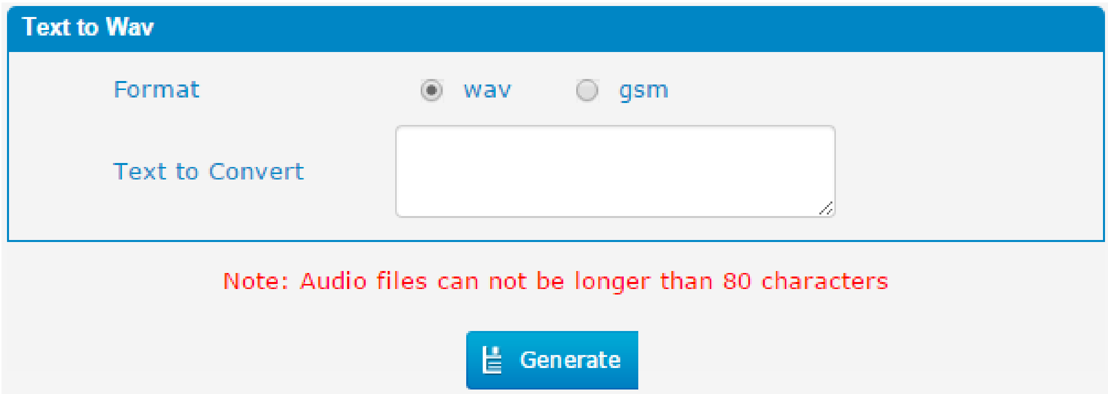
3.12.6 Packet Capture

Figure 3.12.6 Packet Capture



3.12.7 Text to wav

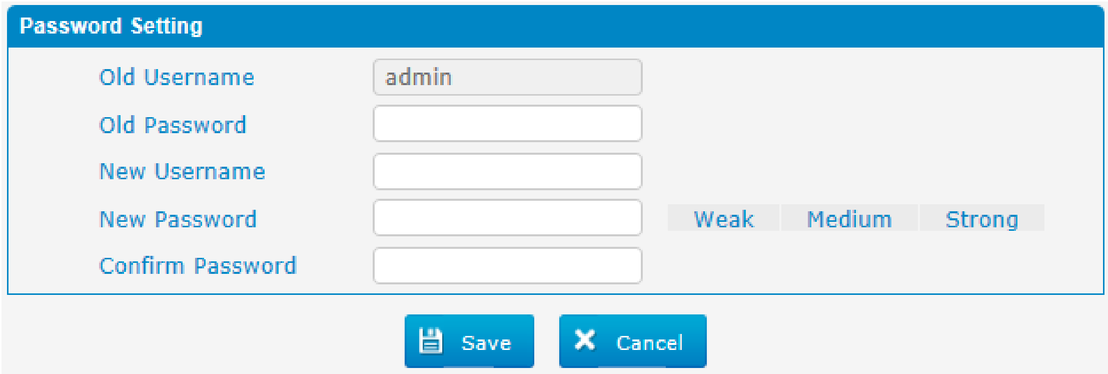
Figure 3.12.7 Text to wav



3.12.8 Password Setting

When using web Configuration, please enter default user name and password. User can modify the login name and password.

Figure 3.12.8 Password Setting



3.12.9 Time & Date

The Network Time Protocol (NTP) is a protocol and software implementation for synchronizing the clocks of computer systems over packet-switched, variable-latency data networks.
User need to fill the NTP Server Address and select Time Zone.

Figure 3.12.9 Time & Date parameter

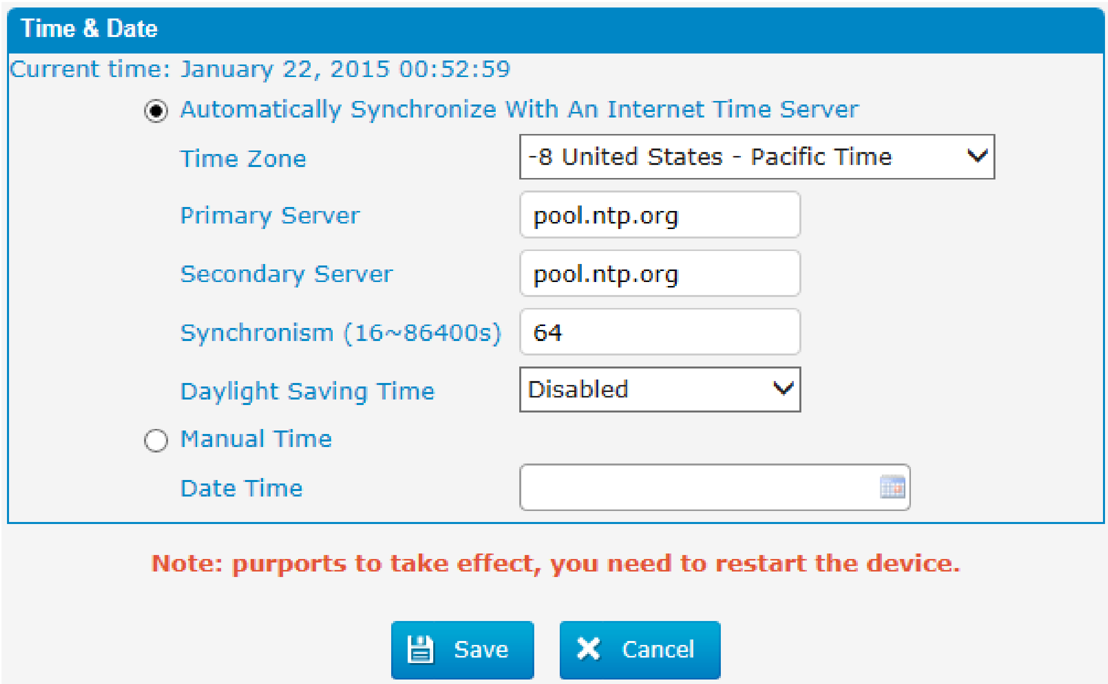


Table 3.12.9 Time & Date parameter

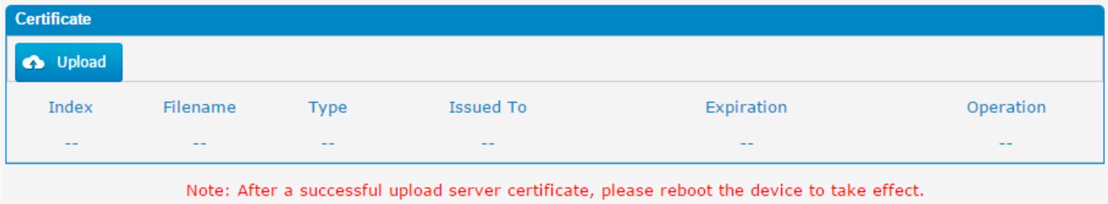
Parameters	**Description**
Time zone	You can choose your time zone here.
Primary server	Primary NTP Server Address
Secondary server	Secondary NTP Server Address
Synchronism	Set the time interval for checking local appliance’s time with the server

Daylight Saving Time	Set the mode to Automatic or disabled

3.12.10 Certificates

MWG1004 can support TLS trunk. Before you register TLS trunk to MWG1004, you should upload certificates first.

Figure 3.12.10 Certificates



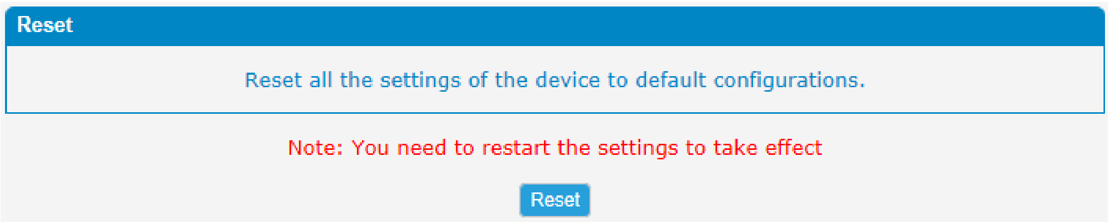
Trusted Certificate
This certificate is a CA certificate. When selecting "TLS Verify Client" as "Yes", you should upload a CA. The relevant IPPBX should also have this certificate.

Gateway Certificate
This certificate is server certificate. No matter selecting "TLS Verify Client" as "Yes" or "NO",you should upload this certificate to MWG1004. If IPPBX enables "TLS Verify server", you should also upload this certificate on IPPBX.

3.12.11 Factory reset

Be careful do this operation, after restore factory setting, all the parameters will be changed to the factory default.

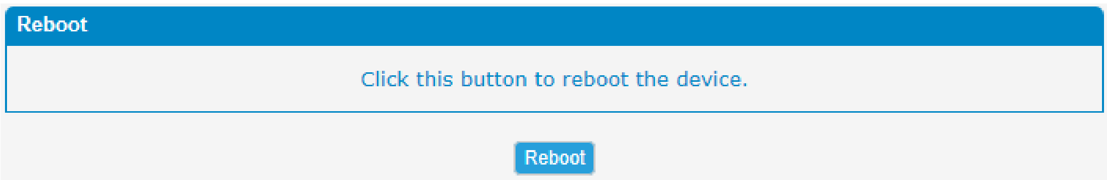
Figure 3.12.11 factory reset



Reset to Factory Defaults
Click this button to reset Factory Default settings

3.12.12 Reboot

Figure 3.12.12 Reboot



Warning: Rebooting the system will terminate all active calls!