MUC1004

IP PBX

Administrator guide V1.0

Version 1.0.0.15

Xiamen Maxincom Technologies Co., Ltd.

Table of Contents

1. Introduction

1.1 Overview

MUC1004—IP PBX for Small Business/Home Office

MUC1004 is a standalone embedded hybrid PBX for small businesses and remote branch offices of larger organizations. It is designed to bring enterprise-grade Unified Communications and Security Protection in an easy-to-manage fashion.

1.2 Product Features

● Alert	● Firewalls
● Blacklist	● HTTPS
● Call Back	● Integrated built-in packet capture tools
● Call Detail Records(CDR)	● Interactive Voice Response (IVR)
● Call Forward,Call Parking	● Intercom/Zone Prompt
● Call Pickup	● Music On Hold
● Call Recording	● Open VPN
● Call Routing	● Paging/Intercom
● Call transfer	● Phone Provisioning
● Call Waiting	● PIN Users
● Caller ID	● QoS
● Conference	● Queue
● DDNS	● Ring Group
● Define Office Time	● Speed Dial
● Direct Inward System Access (DISA)	● Spy functions
● Distinctive Ringtone	● Static Route
● Do Not Disturb(DND)	● VLAN
● External Storage	● Voicemail

1.3 Product Appearance

The appearance of MUC1004 shows as follow

Figure 1-3-1 Front view of MUC1004

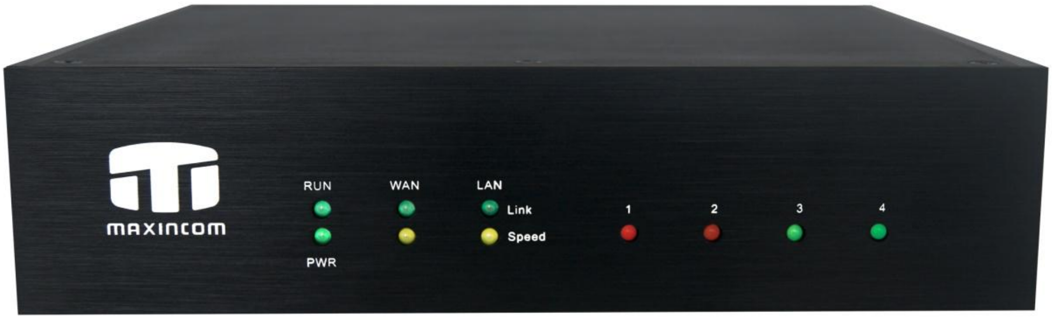


Table 1-3-1 Description of Front view

Index	Indicators	Description
1	RUN	On: Starting Off: Abnormal Blinking every 0.5s: Normal status
2	POWER	On: Power on Off: Power off
3	WAN,LAN	Green LED: indicates the Internet interface is in Link . Yellow LED: ON is indicates 100MBps Ethernet port.
4	1,2,3,4	Red LED stands for FXO port Orange LED indicates presence of a BRI port. Green LED stands for FXS port Red LED blinks: FXO port isn't connected to PSTN line. Alternately blinks Red and Green: FXO port has an incoming call. Alternately blinks Red and Green fast: FXO port is in a call. Alternately blinks Green and Red: FXS port is ringing. Alternately blinks Green and Red fast: FXS port is in a call.

Figure 1-3-2 Rear view of MUC1004

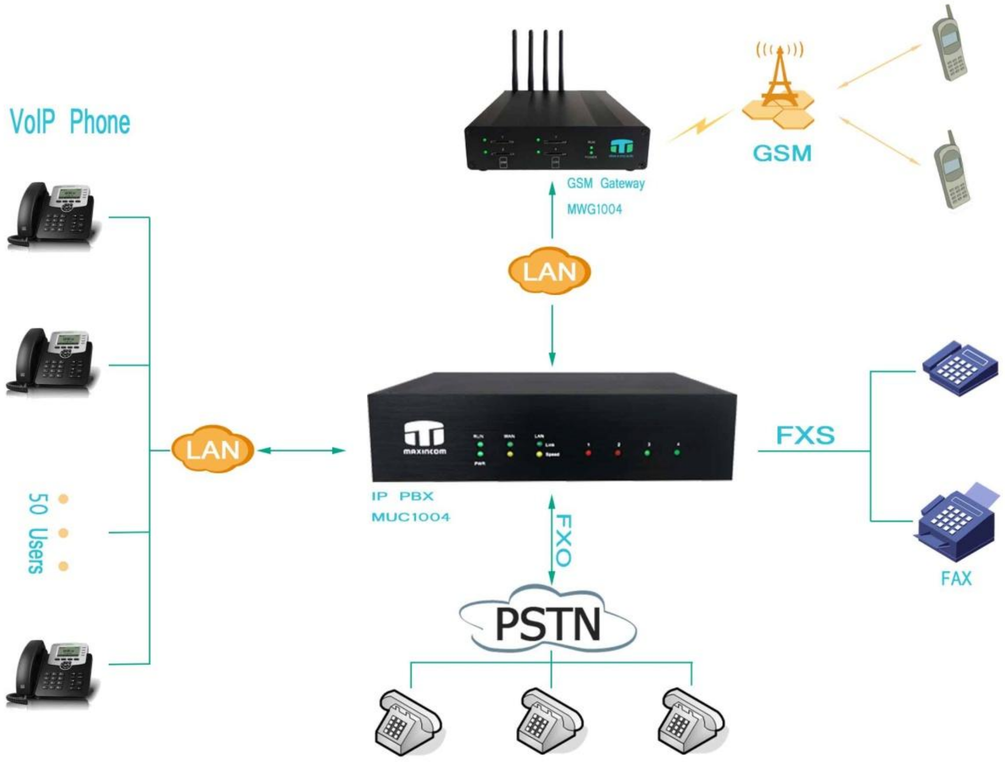
Table 1-3-2 Description of Rear view

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 1A

1.4 Scenario of Application

Application 1

Figure 1-4-1



Application 2

Figure 1-4-2

2. Installation Guide

2.1 Installation Notice

MUC1004 adapts 12VDC Power adapter , make sure AC power supply grounded well to ensure the reliability and stability;

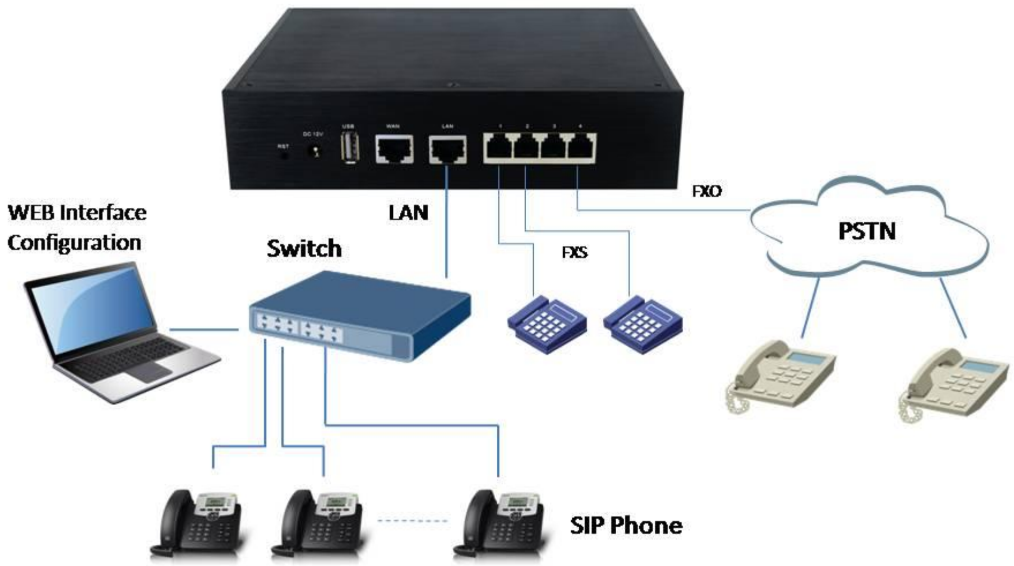
Notes: incorrect power connection may damage power adapter and device.

MUC1004 provides standard RJ45 with 10Mbps or 100Mbps interfaces.

2.2 Installation Procedure

2.2.1 Connect Drawing

Figure 2-2-1 Connect Drawing



3. WEB Interface Configuration

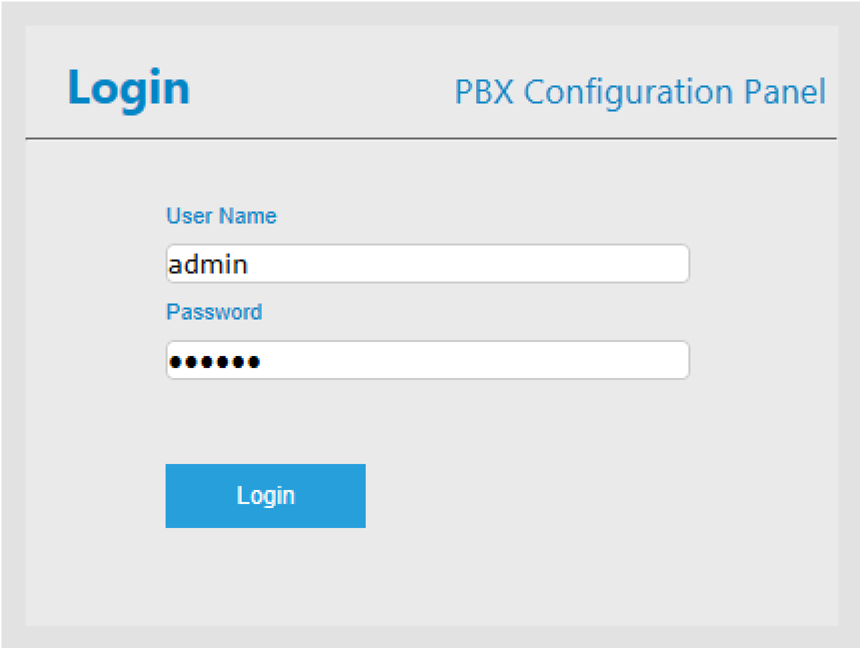
MUC1004 IP PBX has the same web interface. This charpter describes web configuration of MUC1004. The MUC1004 contains an embedded web server to set parameters by using the HTTP protocol. We are strongly recommend to access device with Google Chrome or Firefox Browser.

3.1 Access MUC1004 unit

Enter IP address of MUC1004 in IE/Google Chrome/Firefox Browser. The default IP of LAN port is 192.168.6.200. and the GUI shows as below:

In this example, the IP address is 192.168.6.91

Figure 3-1-1 WEB login interface

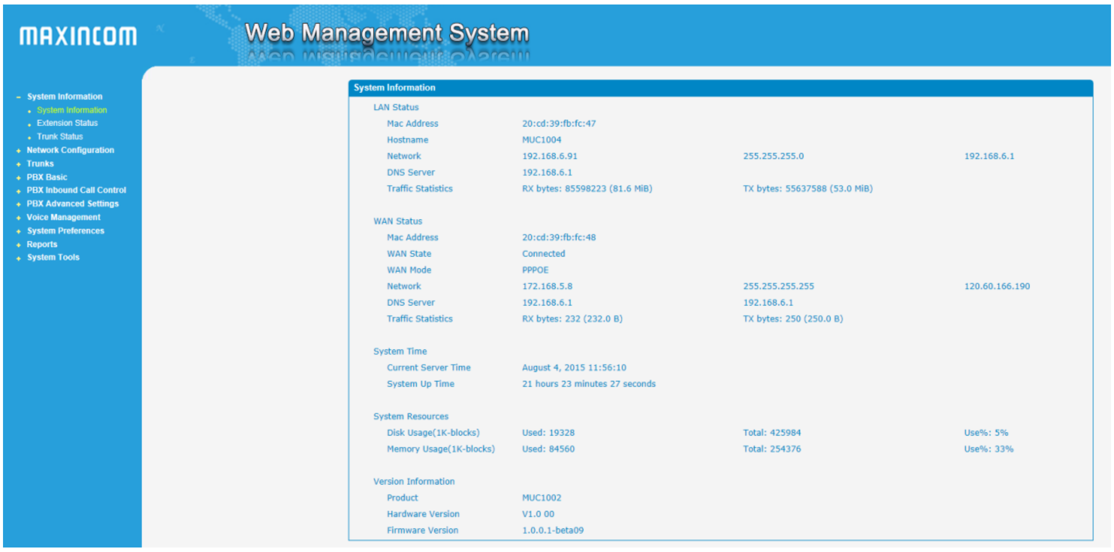


Enter username and password and then click "Login" in configuration interface. The default username and password are "admin/admin". It is strongly recommended, change the default password to a new password for system security .

3.2 Parameters Configuration

MUC1004 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

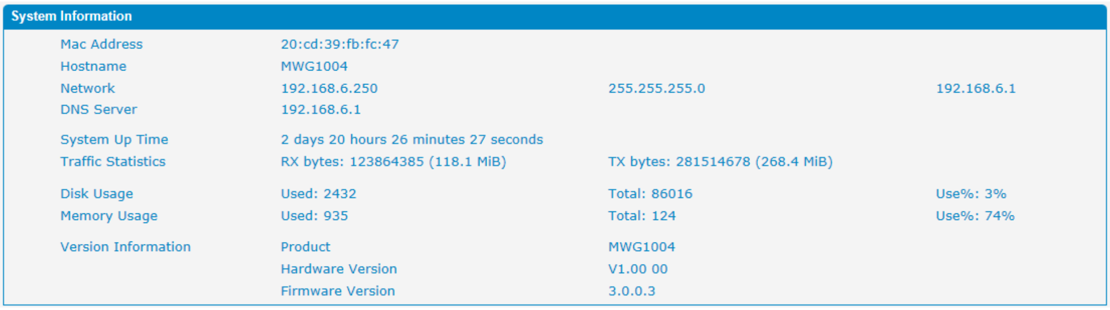


Table 3.3-1 System Information

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 Extensions Status

Figure 3.3-2 Extensions Status

Extension Status

Show Filter | Page 1 of 1(11 Records)

Status	Extension	Type	Name	Caller ID	DND	IP Address
Not Registered	100	SIP	100	100	OFF	
Registered (Idle)	101	SIP	101	101	OFF	192.168.6.105
Registered (Idle)	102	SIP	102	102	OFF	192.168.6.99
Not Registered	103	SIP	103	103	OFF	
Registered (Idle)	104	SIP	104	104	OFF	192.168.6.212
Registered (Idle)	105	SIP	105	105	OFF	192.168.6.105
Not Registered	106	SIP	106	106	OFF	
Registered (Idle)	601	FXS	601	601	OFF	
Registered (Idle)	602	FXS	602	602	OFF	
Registered (Idle)	603	FXS	603	603	OFF	
Registered (Idle)	604	FXS	604	604	OFF	

Extensions Status Description:

Not Registered — Extensions is not registered

Registered (Idle) — Extensions is idle

Registered (Ringing) — Extensions is ringing

Registered (Busy) — Extensions is busy

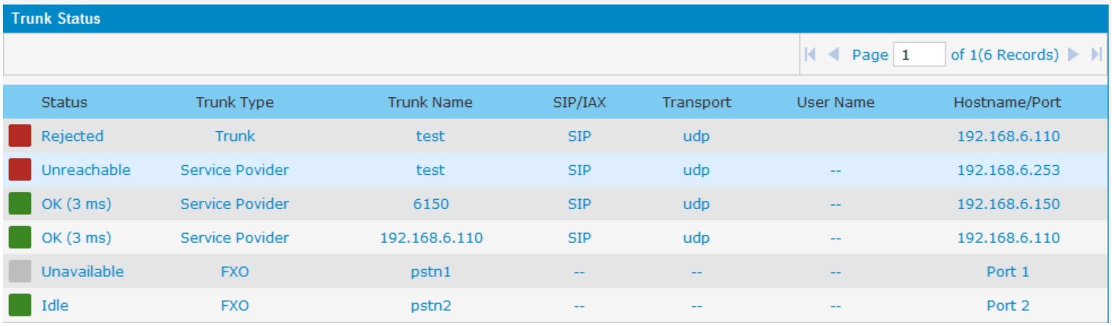
Registered (Hold) — Extensions is hold

Table 3.3-2 Extensions Status

Parameters	**Description**
Status	Indicates the status of Extensions
DND	Do Not disturb

3.3.3 Trunk Status

Figure 3-3-3 Trunk Stratus



Trunk Status

Page 1 of 1(6 Records)

Status	Trunk Type	Trunk Name	SIP/IAX	Transport	User Name	Hostname/Port
Rejected	Trunk	test	SIP	udp		192.168.6.110
Unreachable	Service Povider	test	SIP	udp	--	192.168.6.253
OK (3 ms)	Service Povider	6150	SIP	udp	--	192.168.6.150
OK (3 ms)	Service Povider	192.168.6.110	SIP	udp	--	192.168.6.110
Unavailable	FXO	pstn1	--	--	--	Port 1
Idle	FXO	pstn2	--	--	--	Port 2

Trunk Status Description:

VoIP Trunk:

Status

Rejected: Trunk registration failed.

Registered: Successful registration, trunk is ready for use.

Request Send: Registering.

Waiting: Waiting for authentication. ???

Service Provider:

Status

OK: Successful registration, trunk is ready for use.

Unreachable: The trunk is unreachable.

Failed: Trunk registration failed.

FXO Trunk:

Status

Idle: The port is idle.

Busy: The port is in use.

Unavailable: The port hasn't connected to the PSTN line.

More detail message, please refer to the LED indication of front panel.

Table 3-3-3 Trunk Status

Parameters	**Description**
Status	Shows the registration status of Trunk 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 LAN Configuration

Figure 3-4-1 LAN Configuration

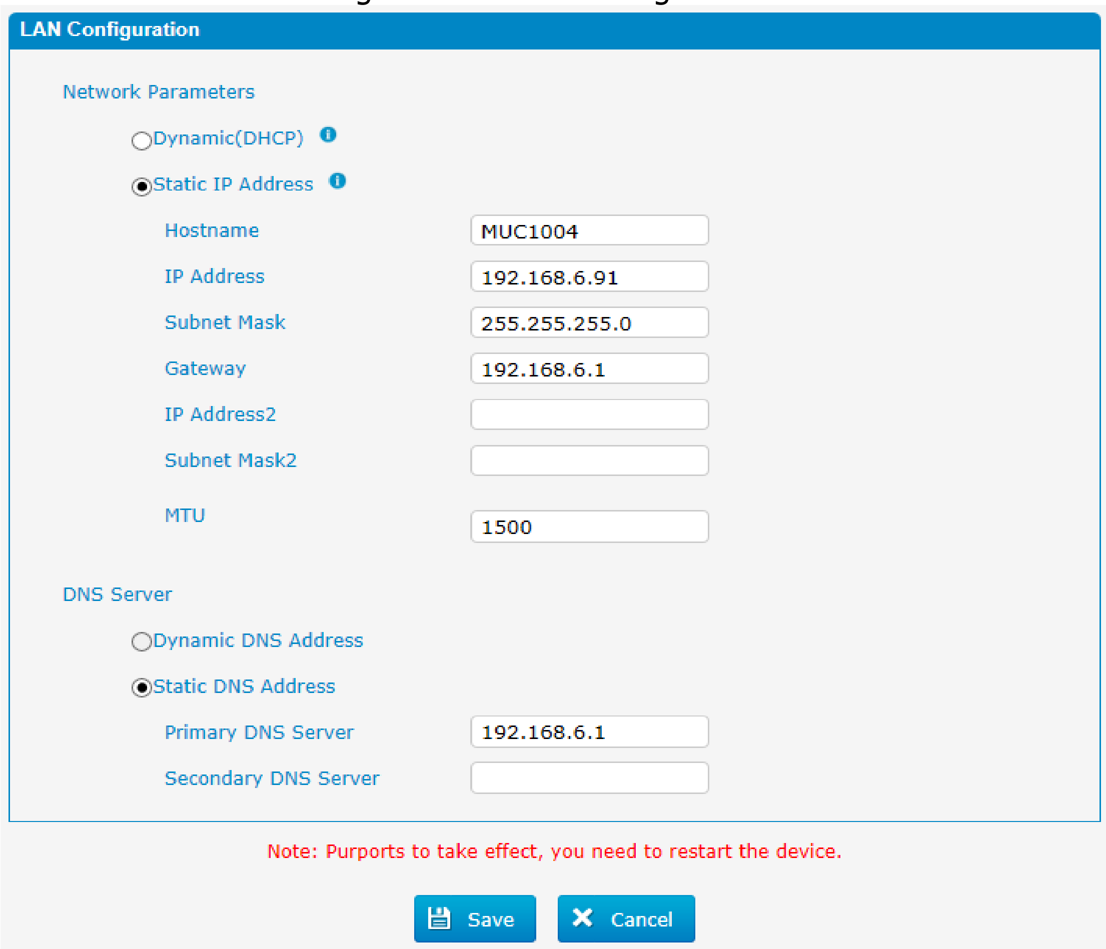
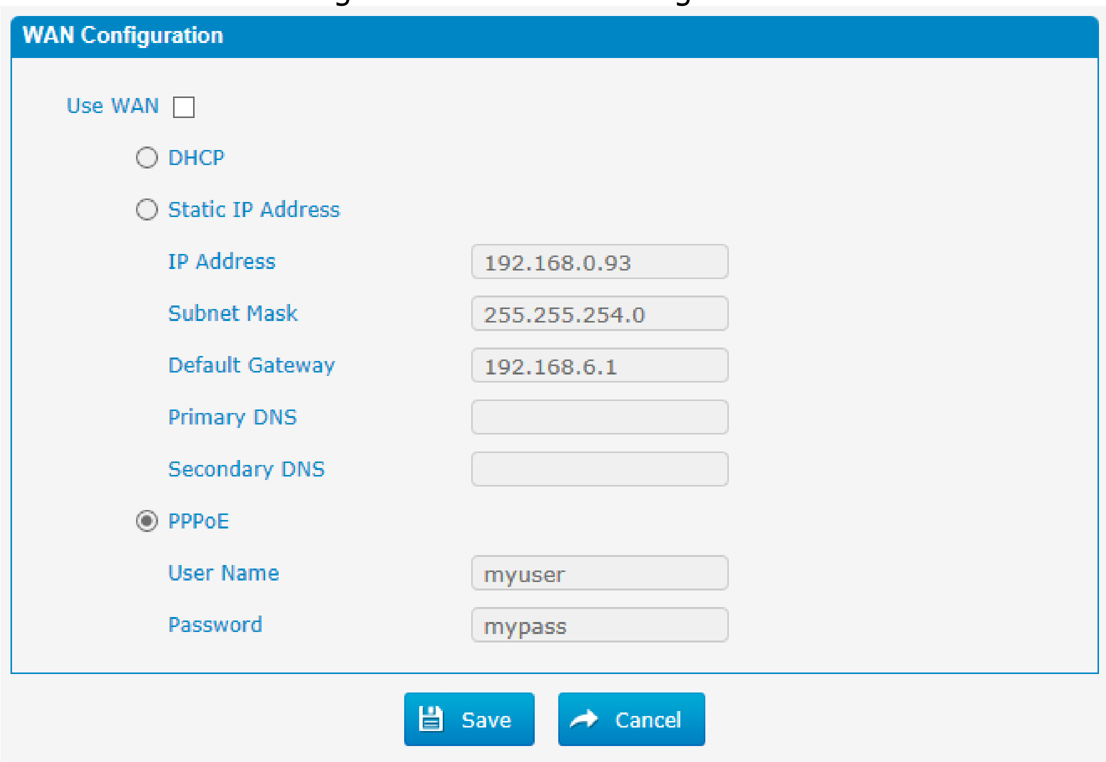


Table 3-4-1 Description of Local network

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 MUC1004
IP Address	Set the IP Address for MUC1004, It is recommended to configure a static IP address for MUC1004
Subnet Mask	Set the subnet mask for MUC1004
Gateway	Set the gateway for MUC1004
IP Address 2	Set the second IP Address for MUC1004
Subnet Mask2	Set the second subnet mask for MUC1004
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 MUC1004.
Secondary DNS Server	Set the Secondary DNS Server for MUC1004.

Figure 3-4-1-2 WAN Configuration



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: MUC1004 is not the VLAN server, a 3-layer switch is still needed, please configure the VLAN information there first, then input the details in MUC1004, so that the packages via MUC1004 will be added the VLAN label before sending to that switch.

Figure 3-4-2 VLAN Configuration

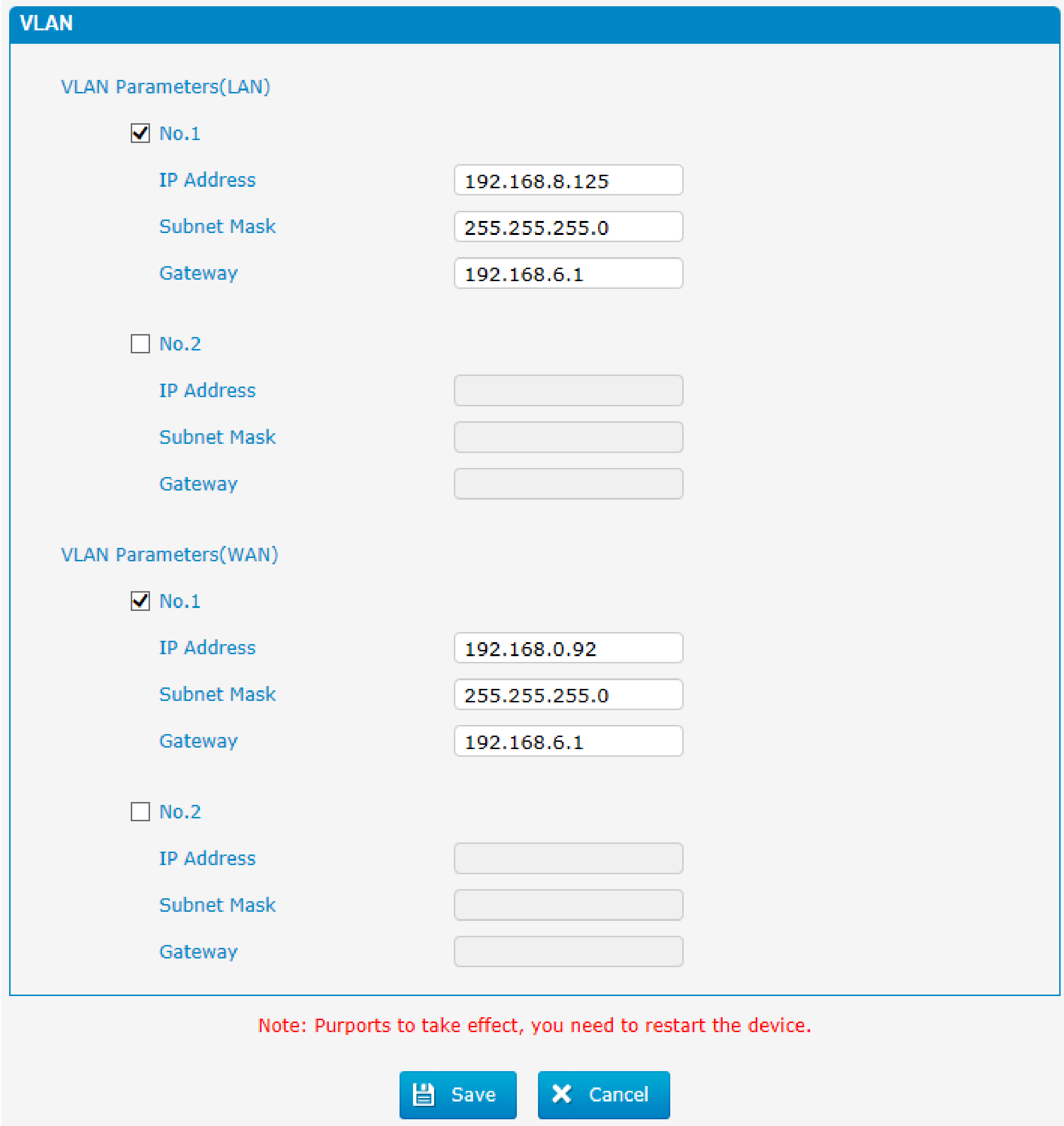


Table 3-4-2 Description of VLAN Configuration

Parameter	Description
NO.1	Click the NO.1 you can edit the first VLAN over LAN
IP Address	Set the IP Address for MUC1004 VLAN over LAN.
Subnet Mask	Set the Subnet Mask for MUC1004 VLAN over LAN.
Gateway	Set the Default Gateway for MUC1004 VLAN over LAN

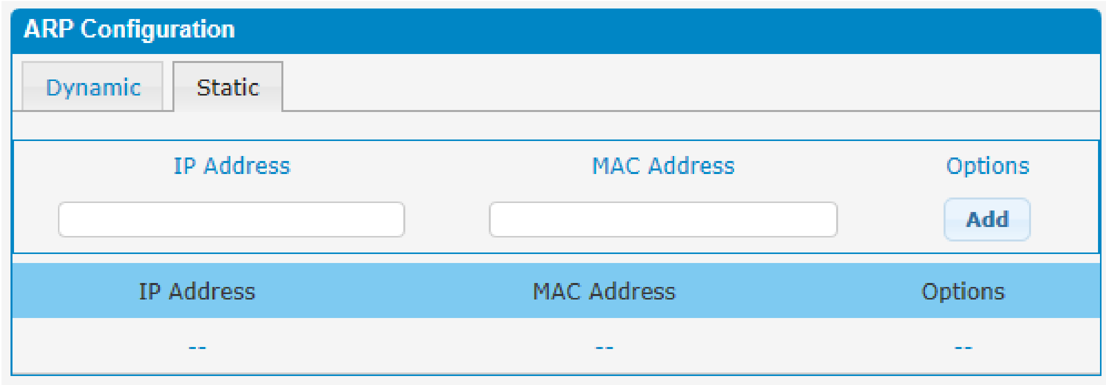
3.4.3 ARP

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:

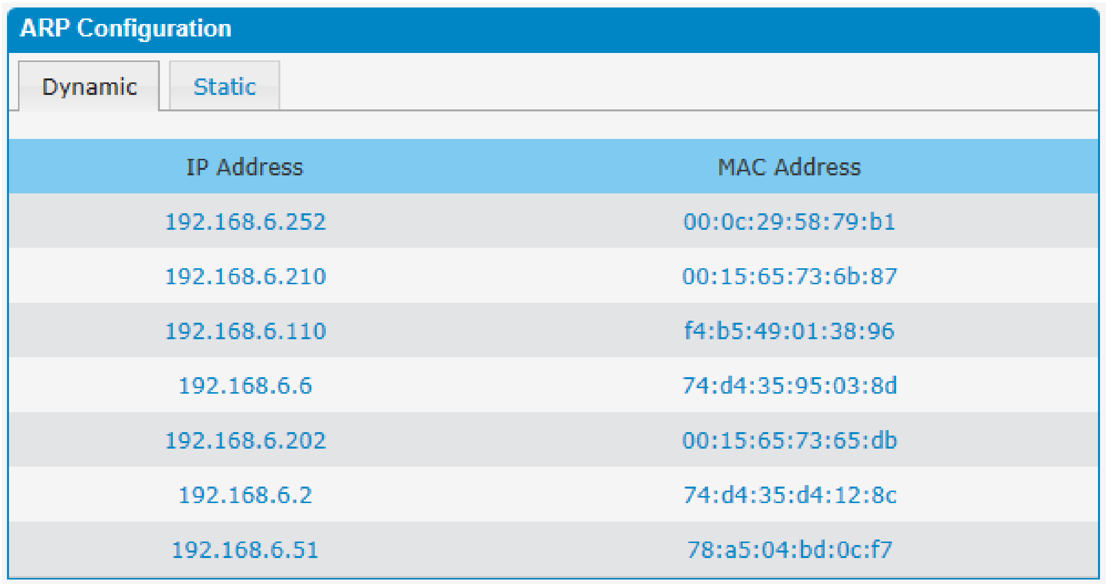
- Protect your network against ARP spoofing
- Prevent network confusion as a result of misconfigured network device

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. MUC1004 supports OpenVPN.

Figure 3-4-4 VPN Configuration

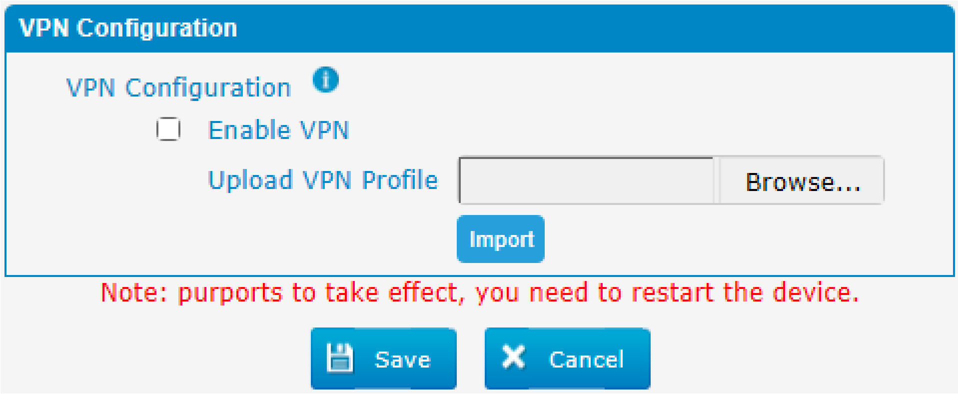


Table 3-4-4 Description of VPN Parameter

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. MUC1004 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

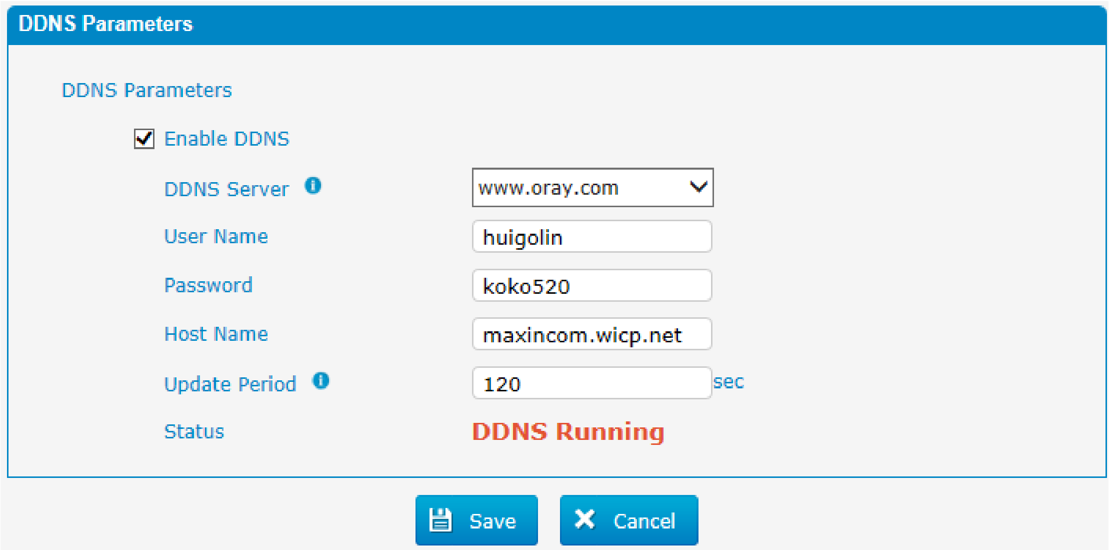


Table 3-4-5 Description of DDNS Server

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.
Host 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

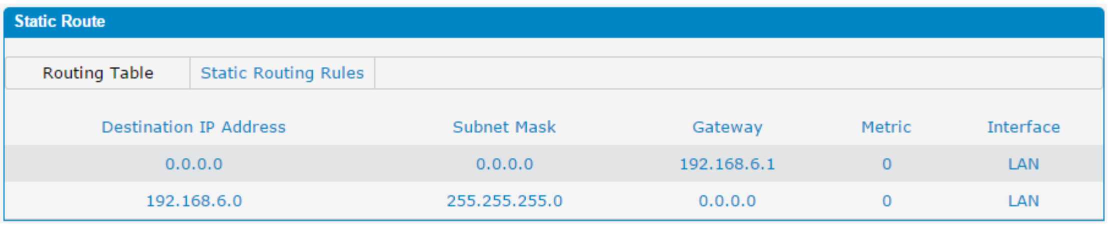
MUC1004 will have more than one internet connection in some situations but it has only one default gateway. You will need to set some Static Route for MUC1004 to force it to go out through different gateway when access to different internet.

The default gateway priority of MUC1004 from high to low is VPN/VLAN-> LAN port.

1) Route Table

The current route rules of MUC1004.

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

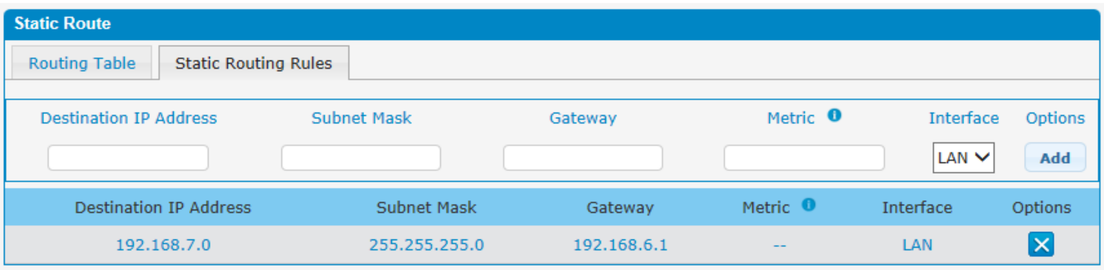
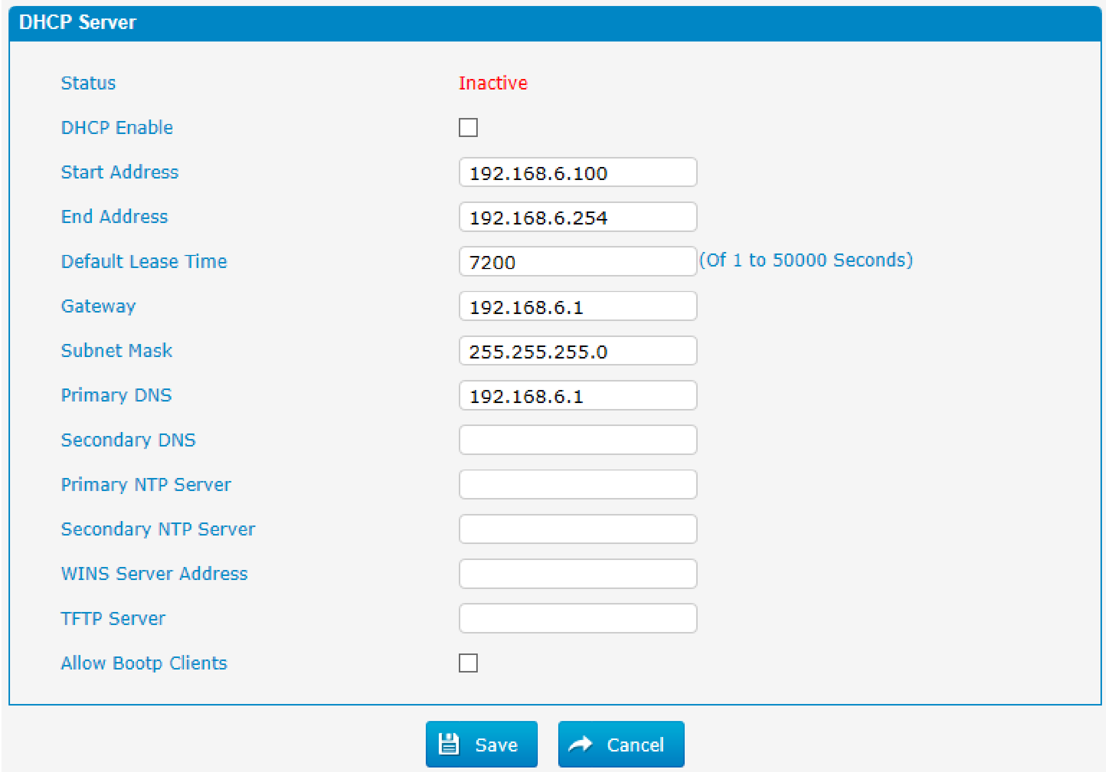


Table 3-4-6 Description of Static Routing

Parameters	Description
Destination IP Address	The destination network to be accessed to by MUC1004.
Subnet Mask	Specify the destination network portion.
Gateway	Define which gateway MUC1004 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.

3.4.7 DHCP Server

Figure 3-4-7 DHCP Server



3.5 Trunks

3.5.1 Analog Trunks(PSTN Trunks)

The public switched telephone network (PSTN) is the network of the world's public circuit-switched telephone networks.

Figure 3-5-1 Analog Trunks

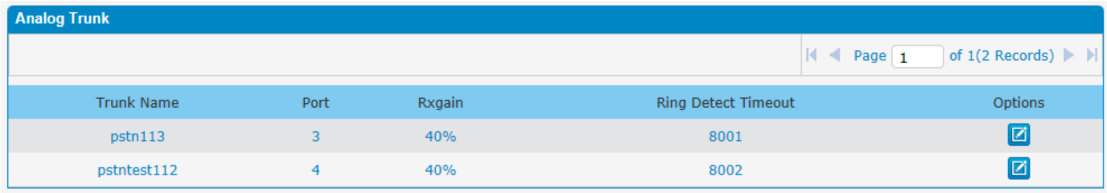


Figure 3-5-1a Analog Trunks Edit

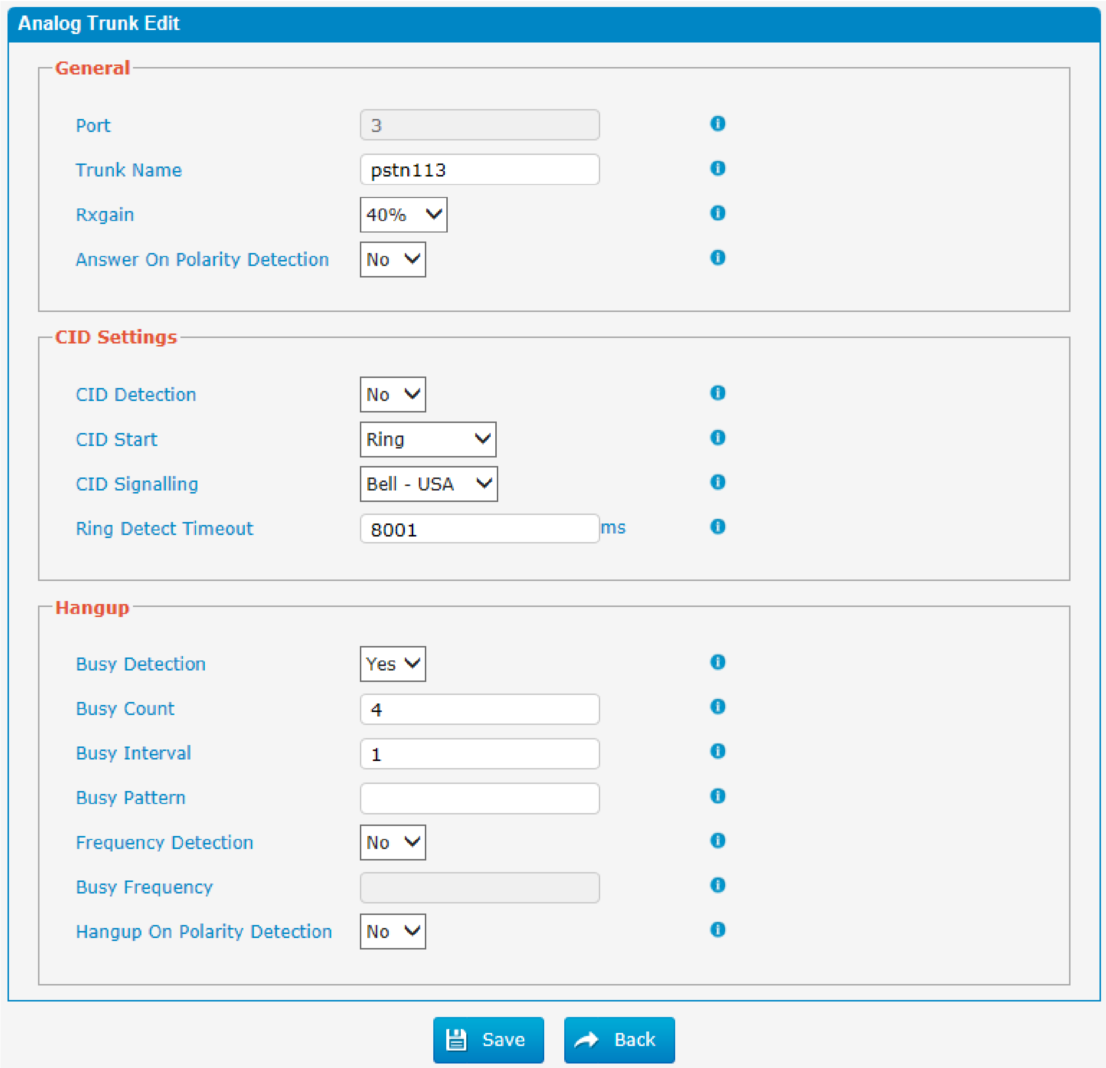


Table 3-5-1 Description of Analog Trunk

Parameters	Description
Trunk Name	A unique label used to identify this trunk when listed in outbound rules, incoming rules, etc.E.g. "pstn113".
Rxgain	Used to modify the volume level of this trunk. Normally, this setting does not need to be changed.
Answer on Polarity Detection	Use a polarity reversal to mark when a outgoing call is answered by the remote party
CID Detection	For FXO trunks, this option forces PBX to look for Caller ID on incoming calls.
CID Start	This option allows you to define the start of a Caller ID signal: Ring: Start when a ring is received (Caller ID Signaling: Bell_USA, DTMF). Polarity: Start when a polarity reversal is started (Caller ID Signaling: V23_UK, V23_JP,DTMF). Before Ring: Start before a ring is received (Caller ID Signaling: DTMF).
CID Signalling	This option defines the type of Caller ID signaling to use. It can be set to one of the following: Bell_USA: bell202 as used in the United States v23_UK: suitable in the UK v23_Japan: suitable in Japan v23-Japan pure: suitable in Japan DTMF: suitable in Denmark, Sweden, and Holland
Busy Detection	Busy Detection is used to detect far end hang-up or for detecting a busy signal. Select "Yes" to turn this feature on.
Budy Count	If Busy Detection is enabled, it is also possible to specify how many busy tones to wait for before disconnecting the call. The default is 4, but better results can be achieved if set to 6 or even 8. Remember, the higher the number, the more time will be required to release a channel. A higher setting lowers the probability that you will encounter random hang-ups.
Busy Interval	The busy detection interval
Busy Pattern	If Busy Detection is enabled, it is also possible to specify the cadence of your busy signal.In many Countries, it is 500msec on, 500msec off. If a Busy Pattern is not specified,The system will accept any regular sound-silence pattern that repeats <Busy Count> times as a busy signal. If you specify Busy Pattern, then the system will further check the length of the tone and silence, which will further reduce the chance of a false positive disconnection.

Frequency Detection	Used for Frequency Detection (Enable detecting the busy signal frequency or not).
Busy Frequency	If the Frequency Detection is enabled, you must specify the local frequency.
Hangup Polarity Reversal Detection	The call will be considered as "hang up" on a polarity reversal.

3.5.2 IP Trunk (peer to peer mode)

Figure 3-5-2 IP Trunk

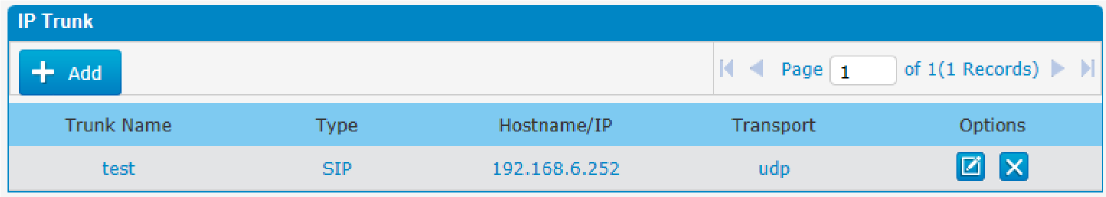


Figure 3-5-2a Add IP Trunk

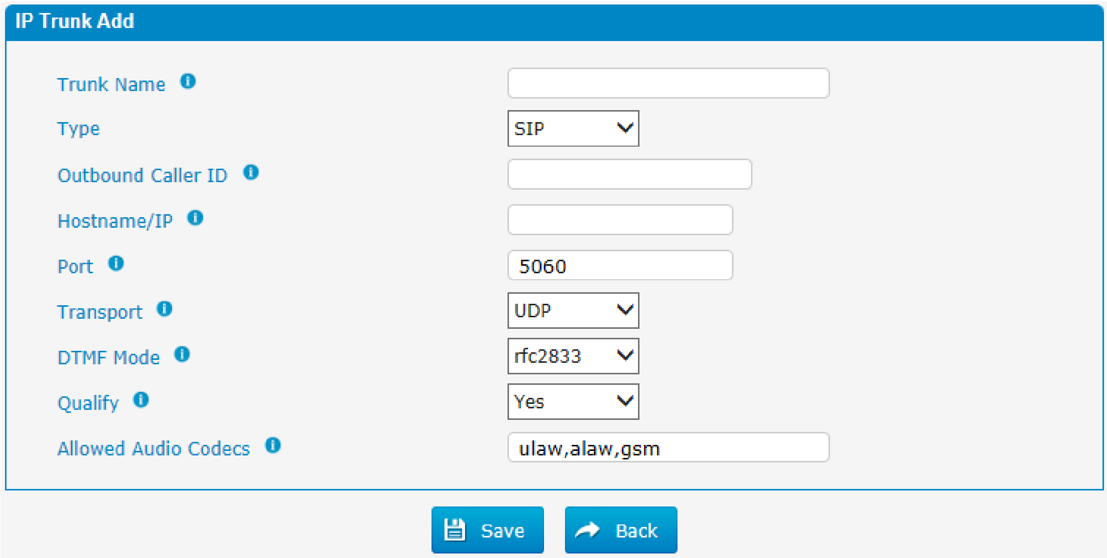


Table 3-8-2 Description of IP Trunk

Parameters	**Description**
IP Trunk	Add remote IP of Softswitch, SIP server which will send call traffics to gateway.
Trunk Name	It describes the trunk for the ease of identification.
Type	Choose the type of this trunk, SIP or IAX
Outbound Caller ID	Caller ID for calls placed on out this trunk
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, MUC1004 will ignore the reachability and the status of this account will be unmonitored.
Allow codecs	ulaw,alaw,gsm

3.5.3 VoIP Trunk

In this page, we can configure VoIP trunk (SIP/ IAX) you have got from provider with the authorization name and password.

Figure 3-5-3 VoIP Trunk

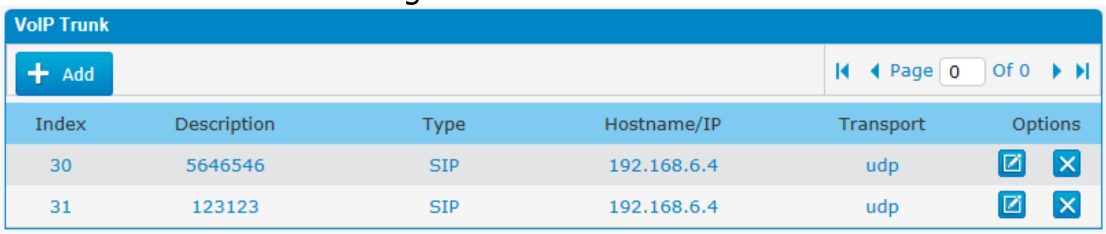


Figure 3-5-3a Add VoIP Trunk

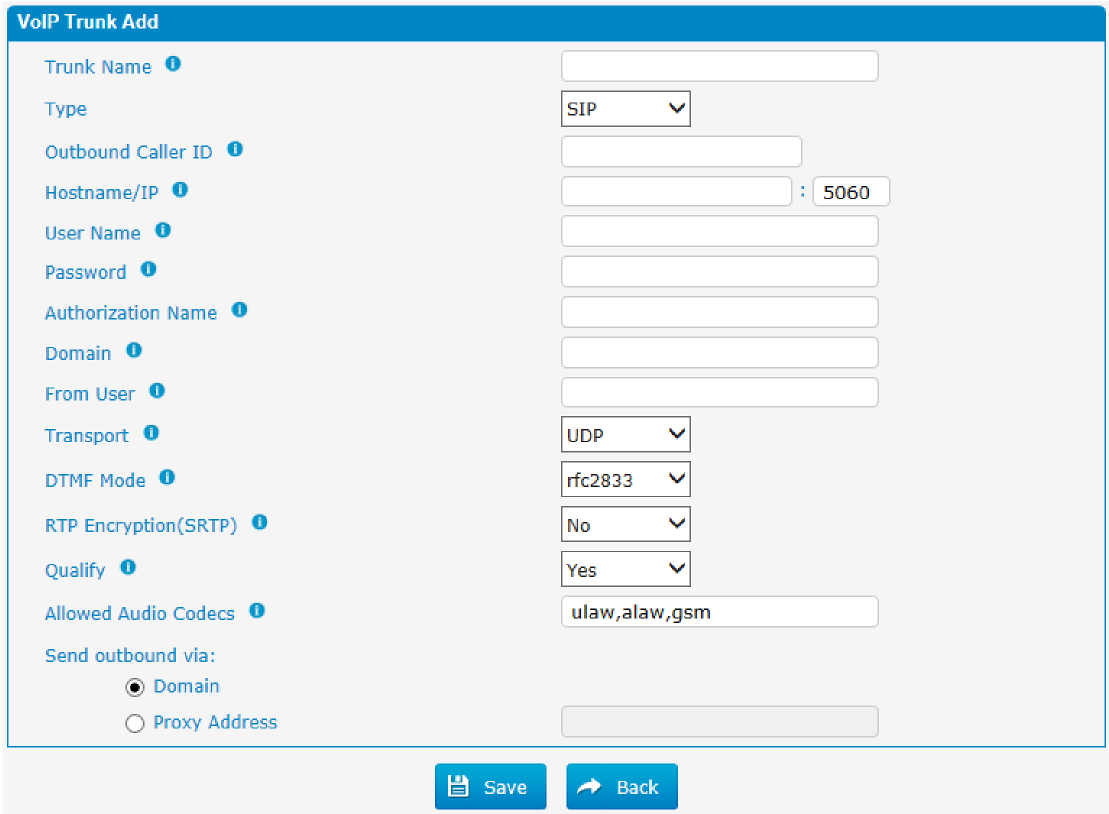


Table 3-5-3 Description of VoIP Trunk

Parameters	**Description**
Trunk Name	It describes the trunk for the ease of identification.
Type	Choose the type of this trunk, SIP or IAX
Outbound Caller ID	Caller ID for calls placed on out this trunk
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, MUC1004 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.

3.6 PBX Basic

3.6.1 Extensions

There are three types of extensions supported in MUC1004: SIP, IAX and analog extension(FXS).

Figure 3-6-1 Extensions

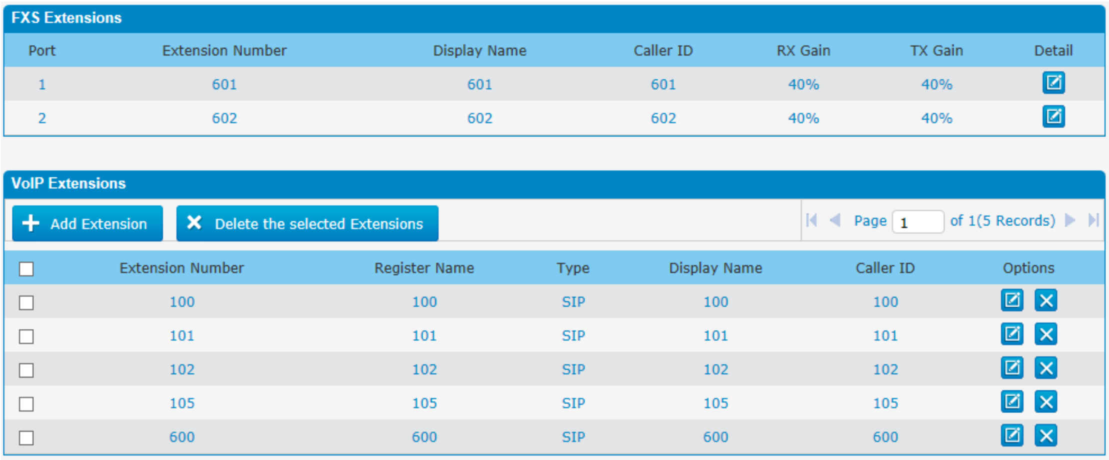


Figure 3-6-1a Fxs Extensions Edit

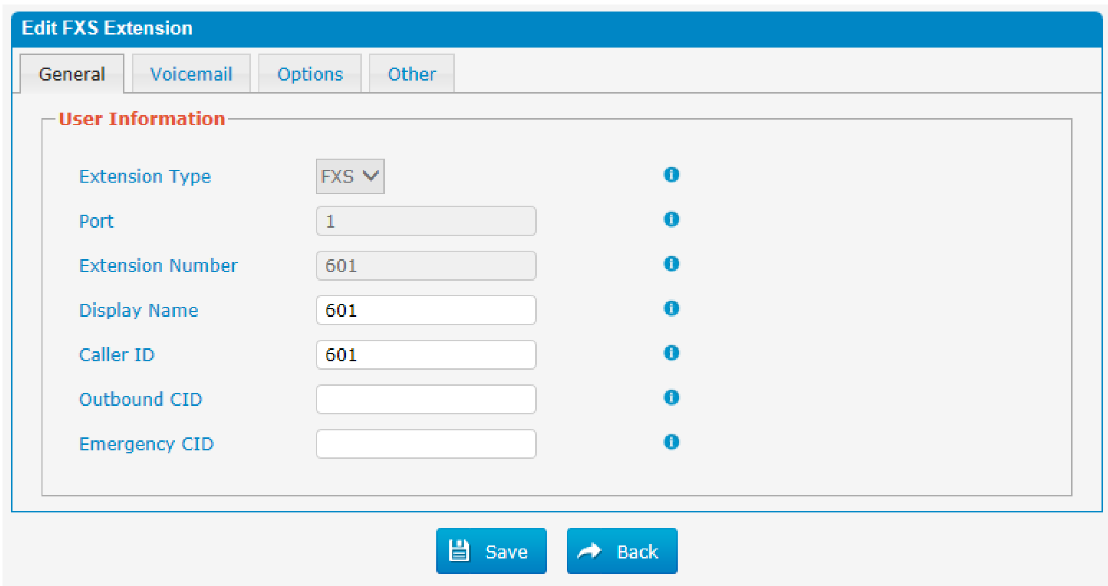
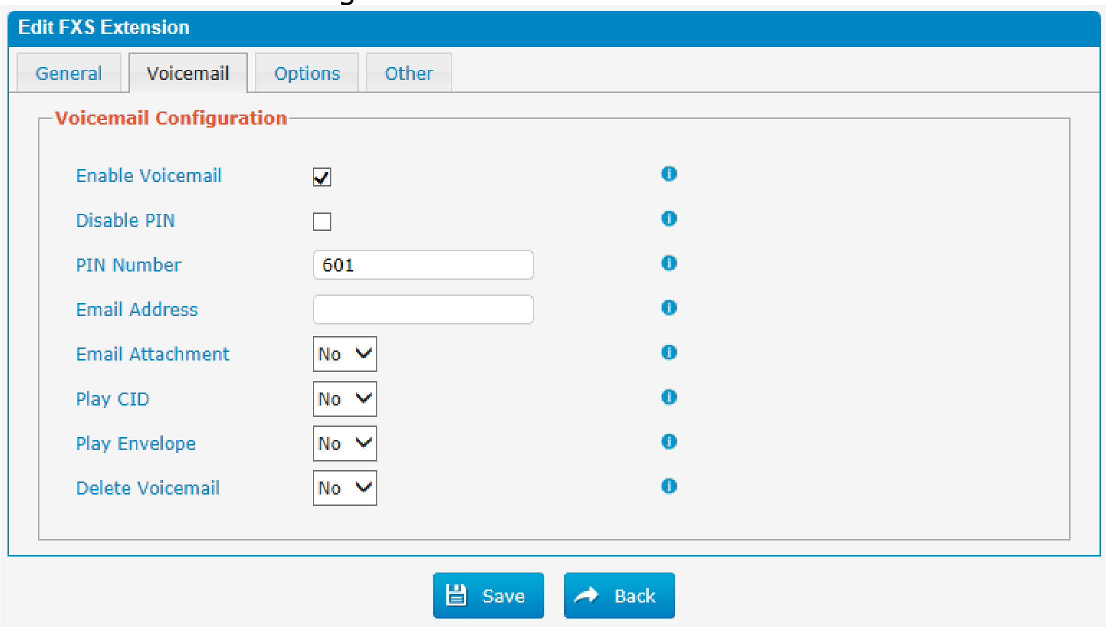


Table 3-6-1a FXS Extensions

Parameters	**Description**
Port	The extension correspond port.
Extensions Number	The numbered extension, e.g. 601, that will be associated with this particular User/Phone.
Display Name	A character-based name for this user, e.g. “Han Jones”.
Call ID	The Caller ID (CID) string will be used when this user calls another internal user.

Outbound CID	Overrides the caller ID when dialing out a trunk.Any setting here will override the common outbound caller ID set in the trunks admin . Format: "caller name" <######> Leave this field blank to disable the outbound caller ID feature for this user
Emergency CID	This Caller ID will always be set when dialing out an outbound route flagged as emergency.The emergency CID overrides all other callerID Settings.

Figure 3-6-1b Fxs Extensions Edit



·Enable Voicemail

Check this box if the user should have a voicemail account.

·Voicemail Access PIN #

Voicemail Password for this extension, e.g. "601".

·Email Address

This option defines whether or not voicemails/Fax is sent to the Email address as an
attachment.

Note: Please ensure that all voicemail settings are properly configured on the System

·Play Envelope

Envelope controls whether or nor the Voicemail system will play the message envelope (date/time) before playing the voicemail message.

Figure 3-6-1c FXS Extensions Edit

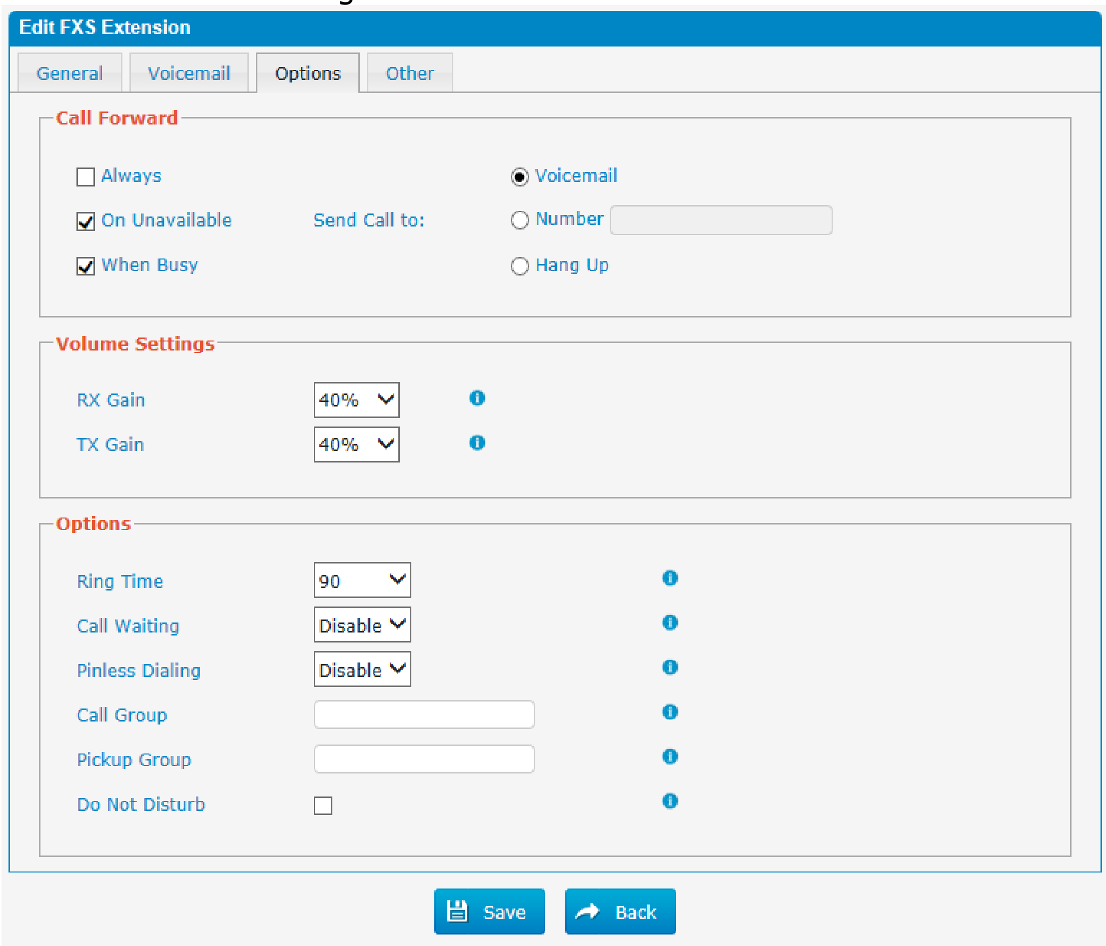
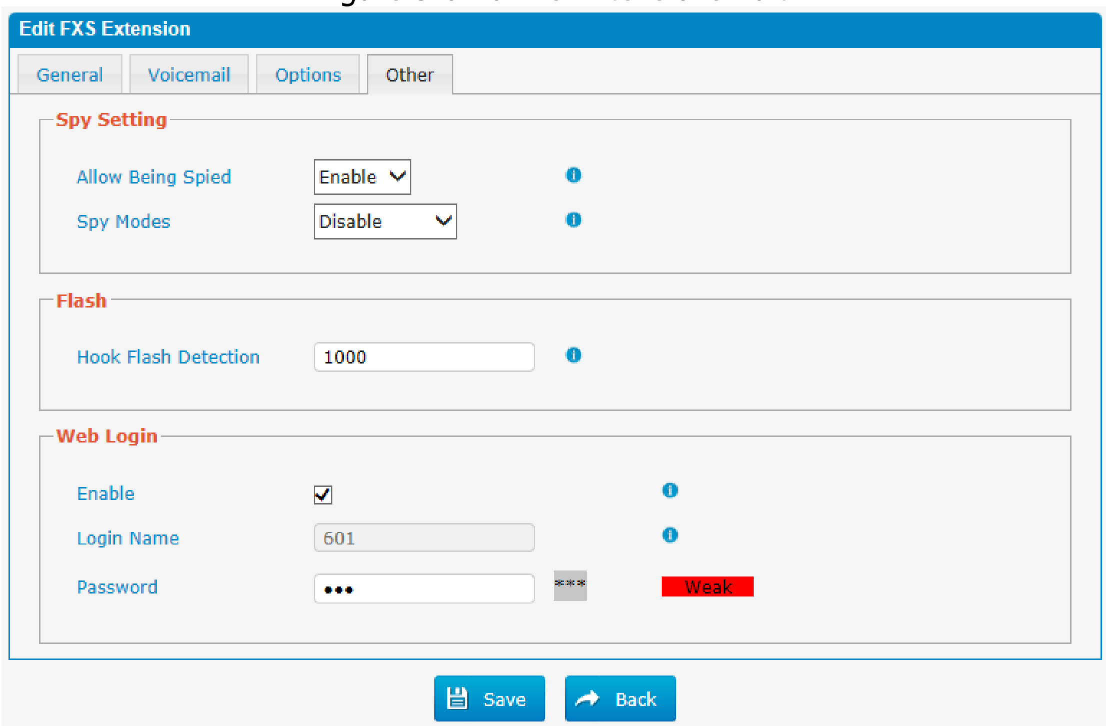


Figure 3-6-1d Fxs Extensions Edit



·Call Forwarding(Follow Me)

This function sets inbound call forwarding on an extension. An administrator can configure Call Forward for this extension.

· Volume Settings

Rxgain: The Volume sent to FXS extension.
Txgain: The Volume sent out by the FXS extension

·Call Waiting

Check this option if the extension should have Call Waiting capability. If this option is checked, the "When busy" follow me options will not be available.

·Pickup Group

If this extension belongs to a pickup group, any calls that ring this extension can be picked up by other extensions in the same pickup group by dialing the Call Pickup feature code(the default is *8).

Note: *8 is the default setting, it can be changed under Feature Codes -> General -> Call Pickup.

·Spy Settings

MUC1004 allows extension to monitor/barge in other conversation. Once this feature is enabled, the extension has the ability to monitor/barge in other calls using the feature codes for each spy mode. Refer to "Feature Codes" section for more information.

·spy modes

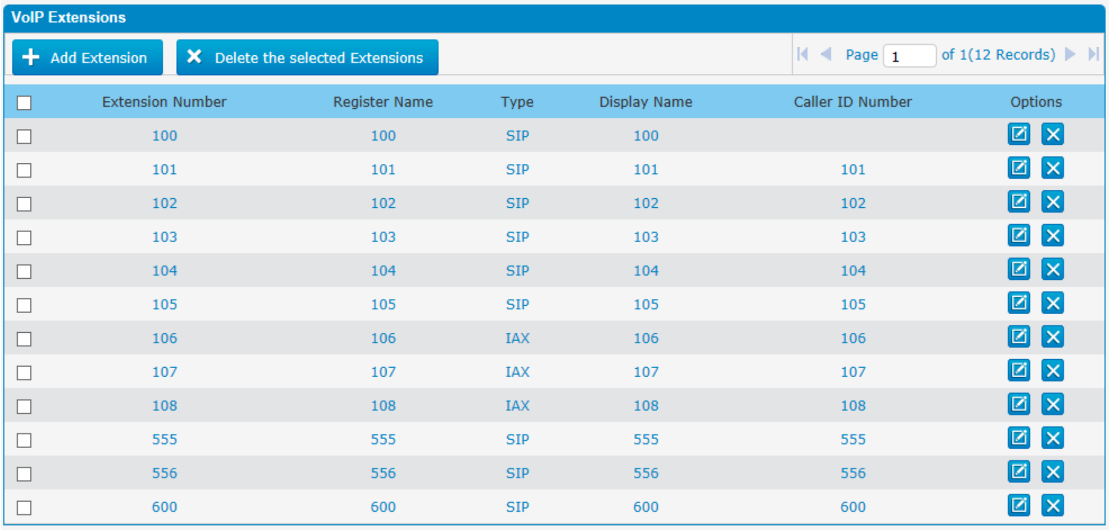
There are 4 spy modes available:
General spy: you have the permission to use the following 3 modes.
Quiet spy: you can only hear the call, but can't talk.
Whisper spy: you can hear the call, and can talk with the monitored extension.
Barge spy: you can hear the call and talk with them both.

·Hook Flash Detection

Sets the amount of time, in milliseconds, that must pass since the last hook-flash event received by MUC1004 before it will recognize a second event. If a second event occurs in less time than defined by Hook Flash Detection, then MUC1004 will ignore the event. The default value of Flash is 1000ms, and it can be configured in 1ms increments.

VoIP Extensions

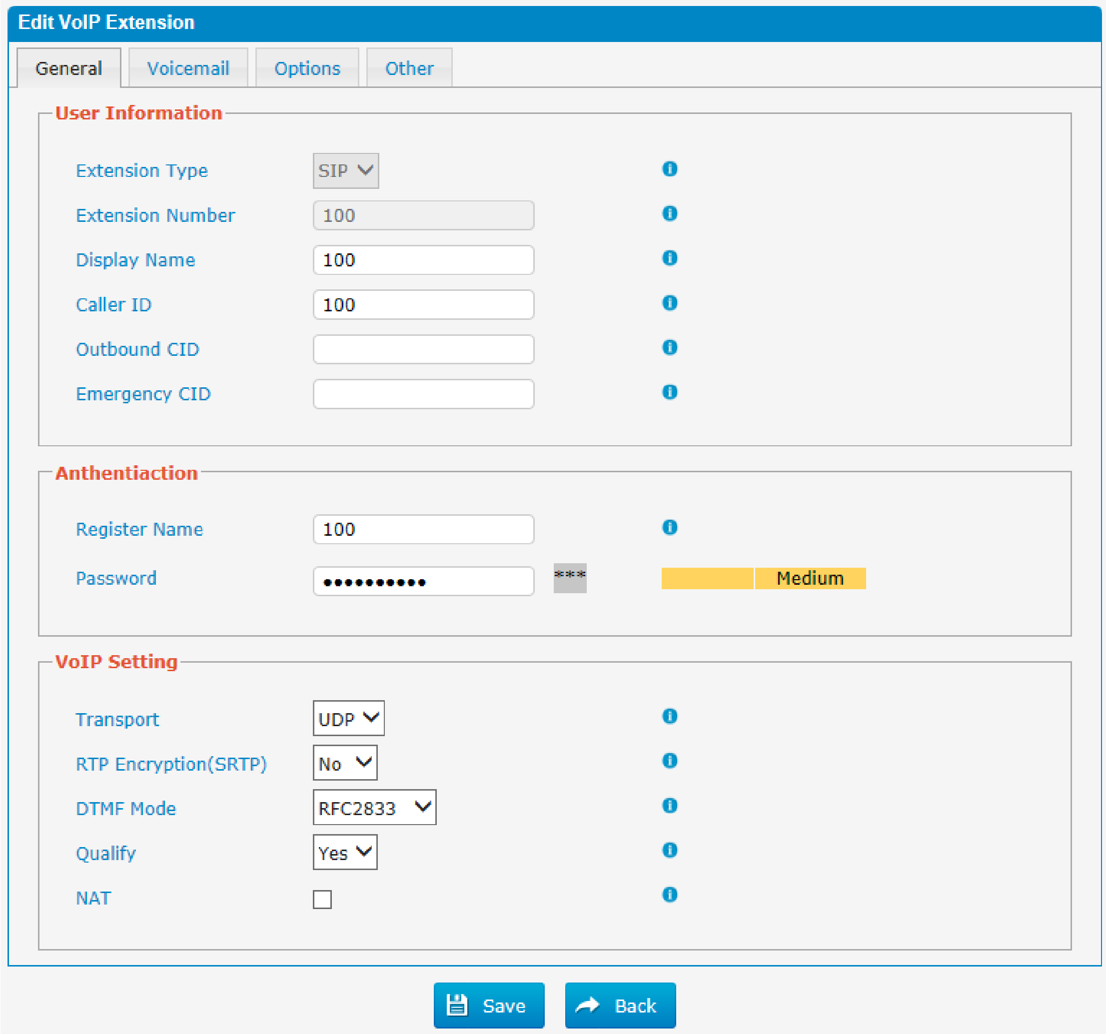
A VoIP extension is a SIP/IAX Account that allows an IP Phone or an IP soft phone client to register on MUC1004.



Extension Number	Register Name	Type	Display Name	Caller ID Number	Options
100	100	SIP	100		
101	101	SIP	101	101	
102	102	SIP	102	102	
103	103	SIP	103	103	
104	104	SIP	104	104	
105	105	SIP	105	105	
106	106	IAX	106	106	
107	107	IAX	107	107	
108	108	IAX	108	108	
555	555	SIP	555	555	
556	556	SIP	556	556	
600	600	SIP	600	600	

We can click "Add extension" to start.

Figure 3-6-1e VoIP Extensions Edit/Add



·Extension Type

Extension type: SIP, IAX or SIP/IAX.

SIP—The extension sends and receives calls using the VoIP protocol SIP.

IAX—The extension sends and receives calls using the VoIP protocol IAX.

·Extension Number

The numbered extension, e.g. 100, that will be associated with this particular User/Phone.

·Display Name

A character-based name for this user, e.g. "Han Jones".

·Caller ID

The Caller ID will be used when this user calls another internal extension.

·Register Name

It is for extension registration validation. Users will not be able register the extension if the authorization name is incorrect even though the username and password are correct.

·Password

The password for this extension, but it is not a fixed one. When you add new extension,a random and robust password will be generated like "0e3lx9Iz".

·Transport

This will be the transport method used by the extension. The options are UDP (default) or TCP or TLS.

·Enable SRTP

Enable extension for SRTP (RTP Encryption).

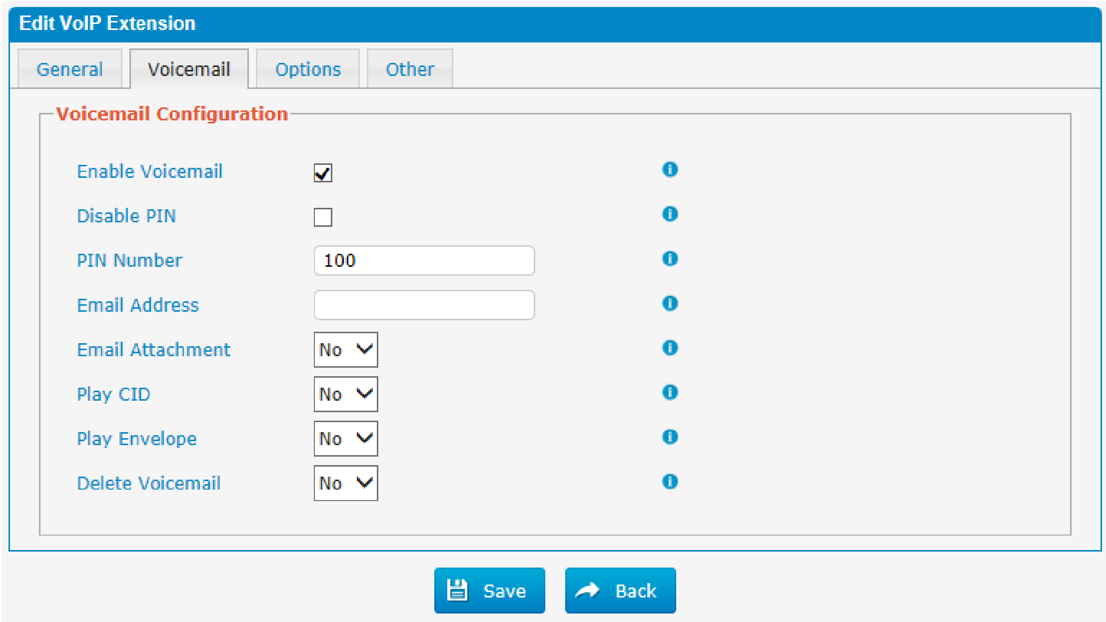
·DTMF Mode—RFC2833, Info, Short Info,Inband, Auto.

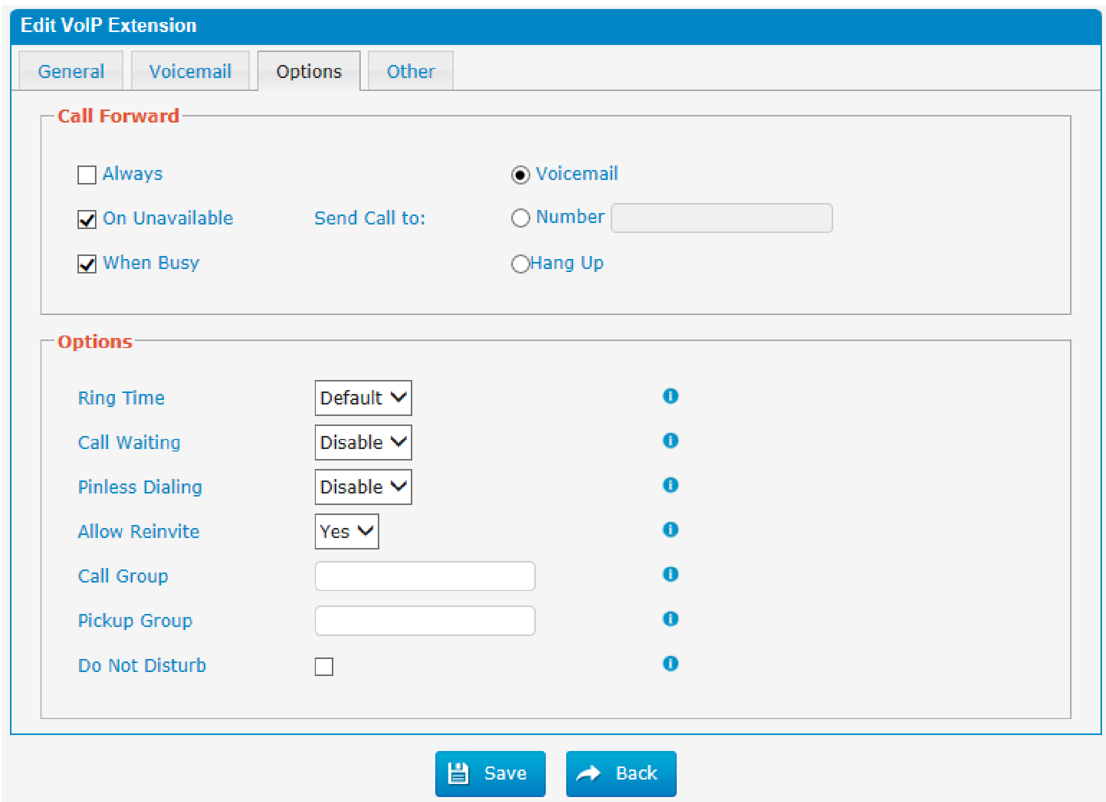
·Qualify

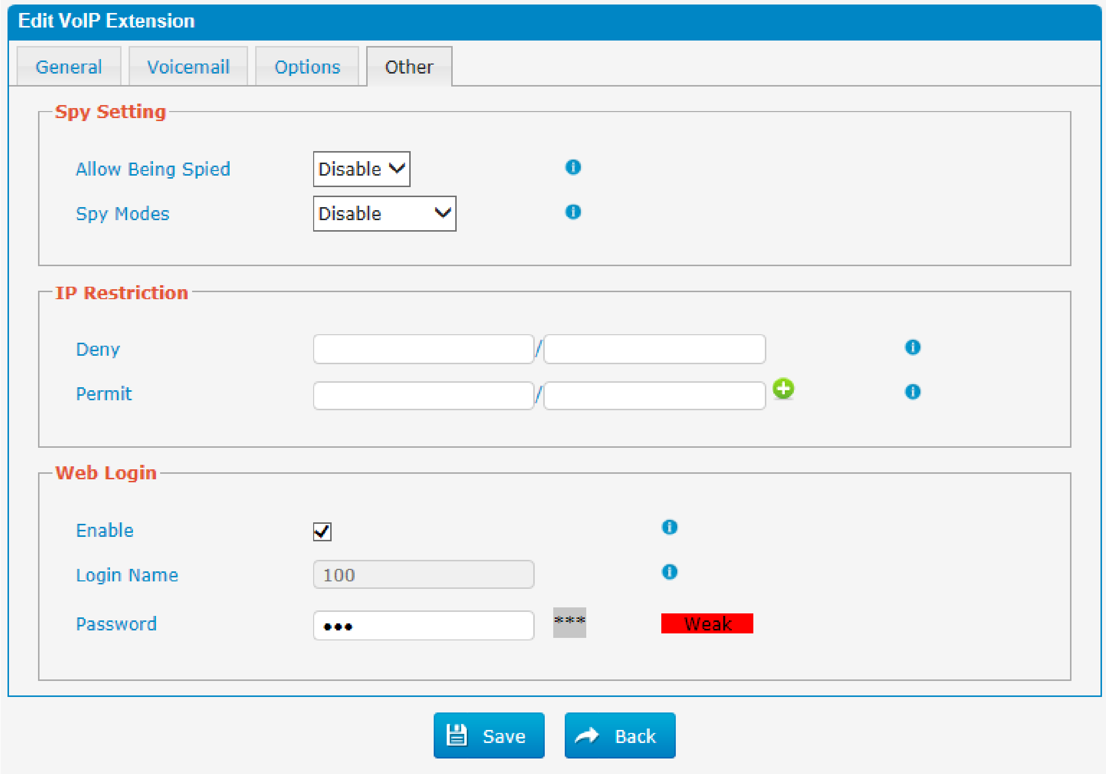
Send check alive packets to IP phones.

·NAT

This setting should be used when the system is using a public IP address to communicate with devices hidden behind a NAT device (such as a broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP and/or RTP ports.


Edit VoIP Extension
General
Voicemail
Options
Other
Voicemail Configuration
Enable Voicemail
Disable PIN
PIN Number
100
Email Address
Email Attachment
No
Play CID
No
Play Envelope
No
Delete Voicemail
No
Save
Back


Edit VoIP Extension
General
Voicemail
Options
Other
Call Forward
Always
On Unavailable
When Busy
Send Call to:
Voicemail
Number
Hang Up
Options
Ring Time
Default
Call Waiting
Disable
Pinless Dialing
Disable
Allow Reinvite
Yes
Call Group
Pickup Group
Do Not Disturb
Save
Back



3.6.2 Feature Codes

There are many feature codes available in MUC1004, which allow users to dial from extension side to realize the exact feature.

Figure 3-6-2 Feature Codes

Feature Codes

		Use Default?	Feature Status
General			
Call Pickup	*8	☑	Enable
Call Trace	*69	☑	Enable
Directed Call Pickup	*08	☑	Enable
Attended Transfer	*2	☑	Enable
Blind Transfer	##	☑	Enable
One Touch Record	*1	☑	Enable
Call Forward			
Call Forward All Activate	*72	☑	Enable
Call Forward All Deactivate	*73	☑	Enable
Call Forward Busy Activate	*90	☑	Enable
Call Forward Busy Deactivate	*91	☑	Enable
Call Forward No Answer Activate	*52	☑	Enable
Call Forward No Answer Deactivate	*53	☑	Enable
Call Forward to Voicemail	*900	☑	Enable
Call Forward to Number	*901	☑	Enable
Call Forward Hang Up	*902	☑	Enable
Call Waiting			
Call Waiting - Activate	*70	☑	Enable
Call Waiting - Deactivate	*71	☑	Enable
Do-Not-Disturb (DND)			
DND Activate	*78	☑	Enable
DND Deactivate	*79	☑	Enable
DND Toggle	*76	☑	Enable
Speed Dial			
Speed Dial Prefix	*0	☑	Enable
Voicemail			
Voicemail Main Menu	*97	☑	Enable
Dial Voicemail	*98	☑	Enable
Direct Dial Prefix	#	☑	Enable
Parking Lot			
Call Parking	*85	☑	Enable
ChanSpy			
Quiet Mode	*93	☑	Enable
Whisper Mode	*94	☑	Enable
Barge Mode	*95	☑	Enable
Paging and Intercom			
Intercom Prefix	*80	☑	Enable
User Intercom Allow	*54	☑	Enable
User Intercom Disallow	*55	☑	Enable

Save

3.6.3 Speed dial

Figure 3-6-3 Speed Dial

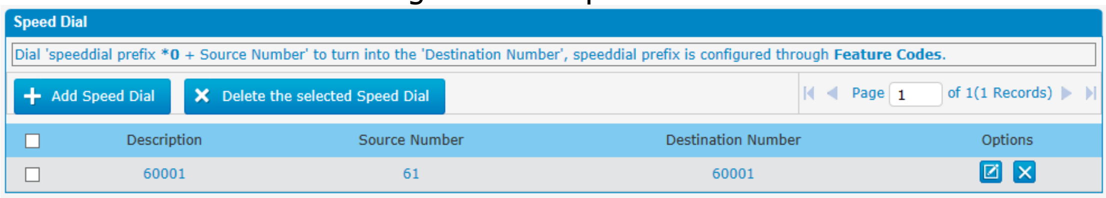


Figure 3-6-3a Speed Dial Add

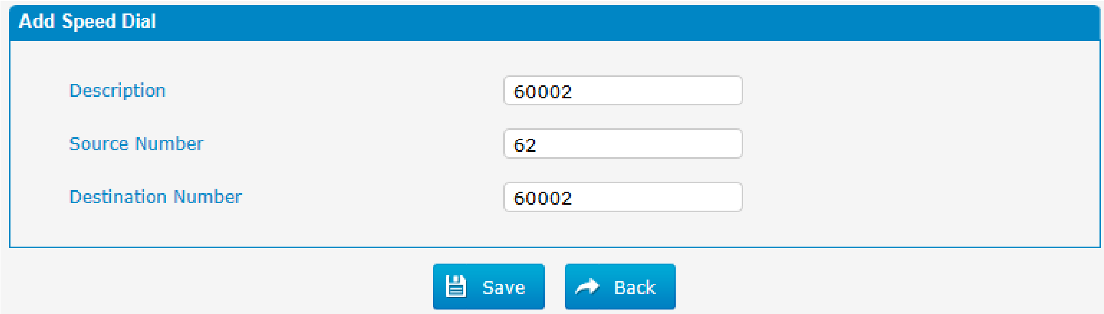


Table 3-6-3 Description of Speed Dial

Parameters	**Description**
Source Number	The speed dial number.
Destination Number	The number you want to call. E.g. the source number is "33". The destination number is 5528369. The prefix number is *90. You can use an extension with any type to dial *9033, then it will call the number 5528369. The predix of Speed dial is setting on "feathur codes" **Note:** Don't forget to add the outbound dial prefix if you would like to dial the speed dial number through trunk.

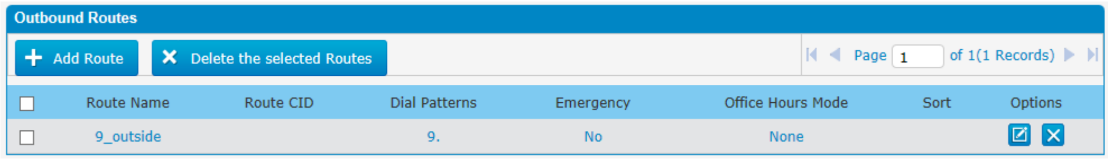
3.6.4 Outbound Routes

In this page, we can configure the outbound rules to control the outgoing calls.

Notes:

1. The max number of outbound route is 32.
2. If the dial patterns are the same in several routes, MUC1004 will choose the available routes from top to the last one.
3. When you have created a new extension, please edit the outbound route so that it can dial out too.

Figure 3-6-4 Outbound Routes



We can create outbound route or use the default route "9_outside" (dial 9+numbers to dial out). Also you can delete multiple outbound routes at once as required.

Figure 3-6-4a Outbound Routes Edit

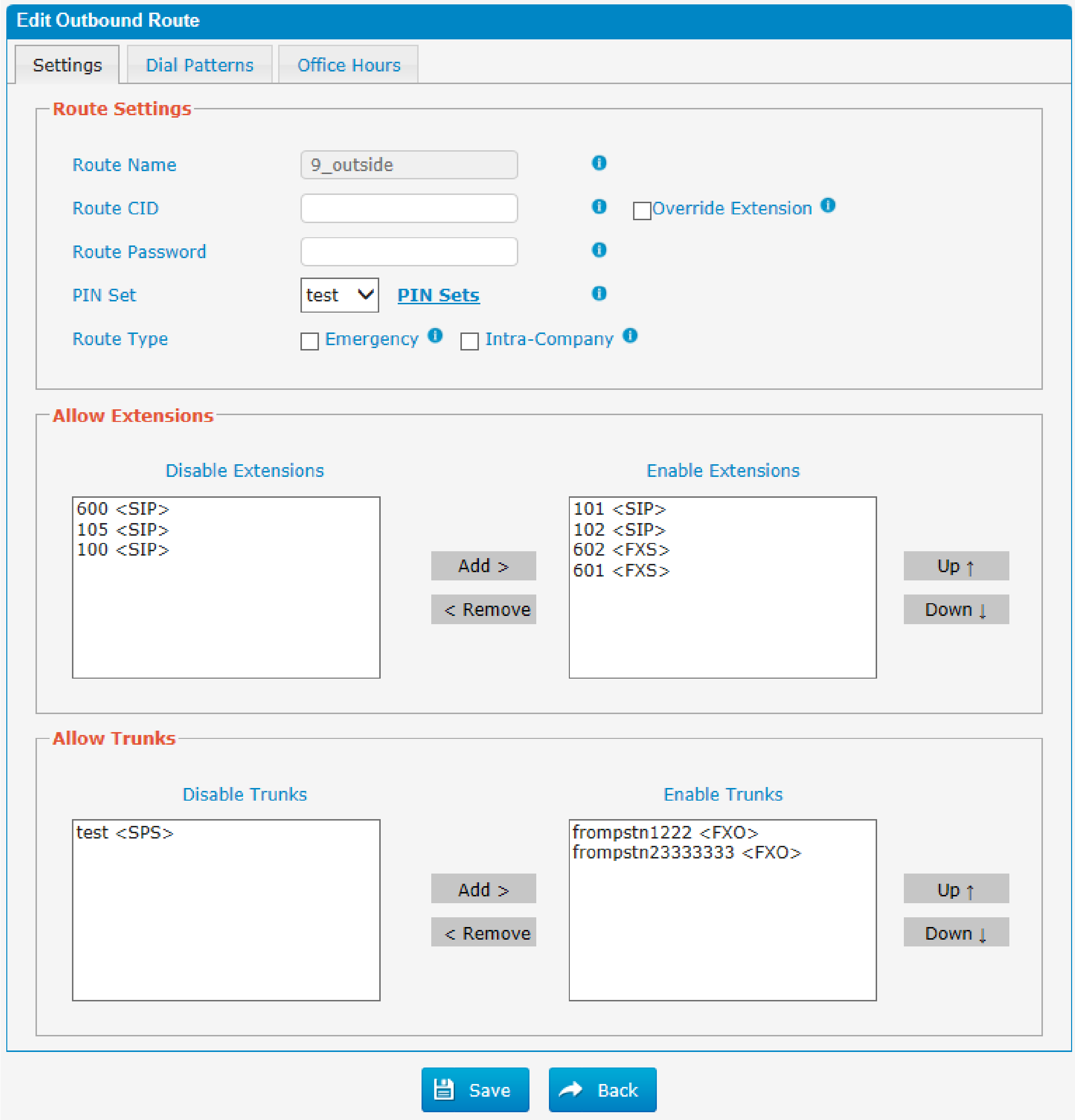


Figure 3-6-4b Outbound Routes Edit

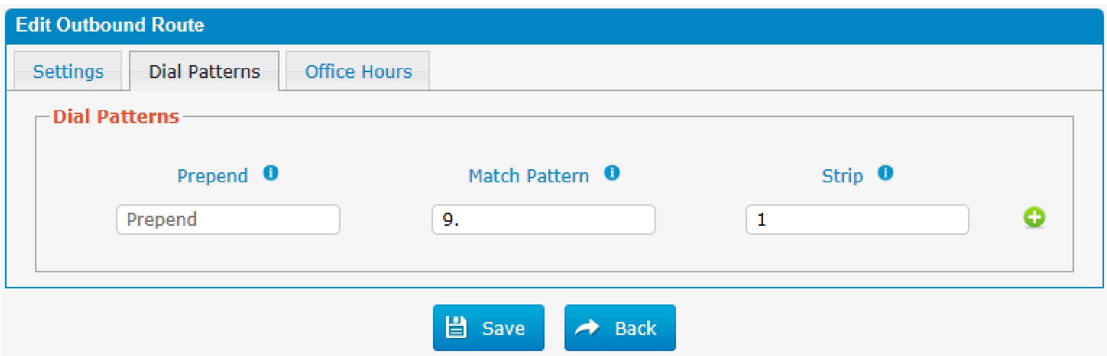
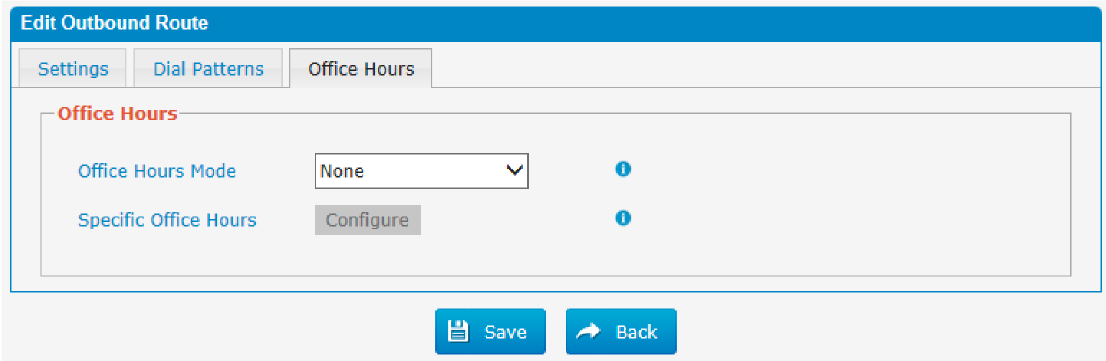


Figure 3-6-4c Outbound Routes Edit



·Route Name

Name of this Outbound Route. E.g. "Local" or "Long Distance".

·Password

The route password can be used to protect this route from being accessed without a password. You can choose one of the passwords in the PIN list that you can click the "Pin Settings" to edit it in "Pin Settings" page.

·PIN set

Optional: Select a PIN Set to use.If using this option,Leave the route password field blank.

·Office Hours

When a specific office hour is selected, this outbound route can only be used during this office hour, and can't be used in non-office hours.

·Dial Pattern

Outbound calls that match this dial pattern will use this outbound route.
There are a number of dial pattern characters that have special meanings:
X: Any Digit from 0-9
Z: Any Digit from 1-9
N: Any Digit from 2-9
[12345-9] : Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)
The "**.**" Character will match any remaining digits. For example, "9011." will match any phone number that starts with "9011", excluding "9011" itself.
The "**!**" will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.
Example 1: **1[5-8]6** will match 156,166,176,186.
Example 2: **1NXXNXXXXX** will match a phone number starting with a 1, followed by a 3-digit area code, and then 6-digit number.

·Strip

Allows the user to specify the number of digits that will be stripped from the front of the phone number before the call is placed. For example, if users must press 0 before dialing a phone number, one digit should be stripped from the dial string before the call is placed.

·Prepend

These digits will be prepended to the phone number before the call is placed. For example,if a trunk requires 10-digit dialing, but users are more comfortable with 7-digit dialing, this field could be used to prepend a 3-digit area code to all 7-digit phone numbers before calls are placed.

·Add

Add multiple dial patterns in this outbound route.

·Allow Extensions

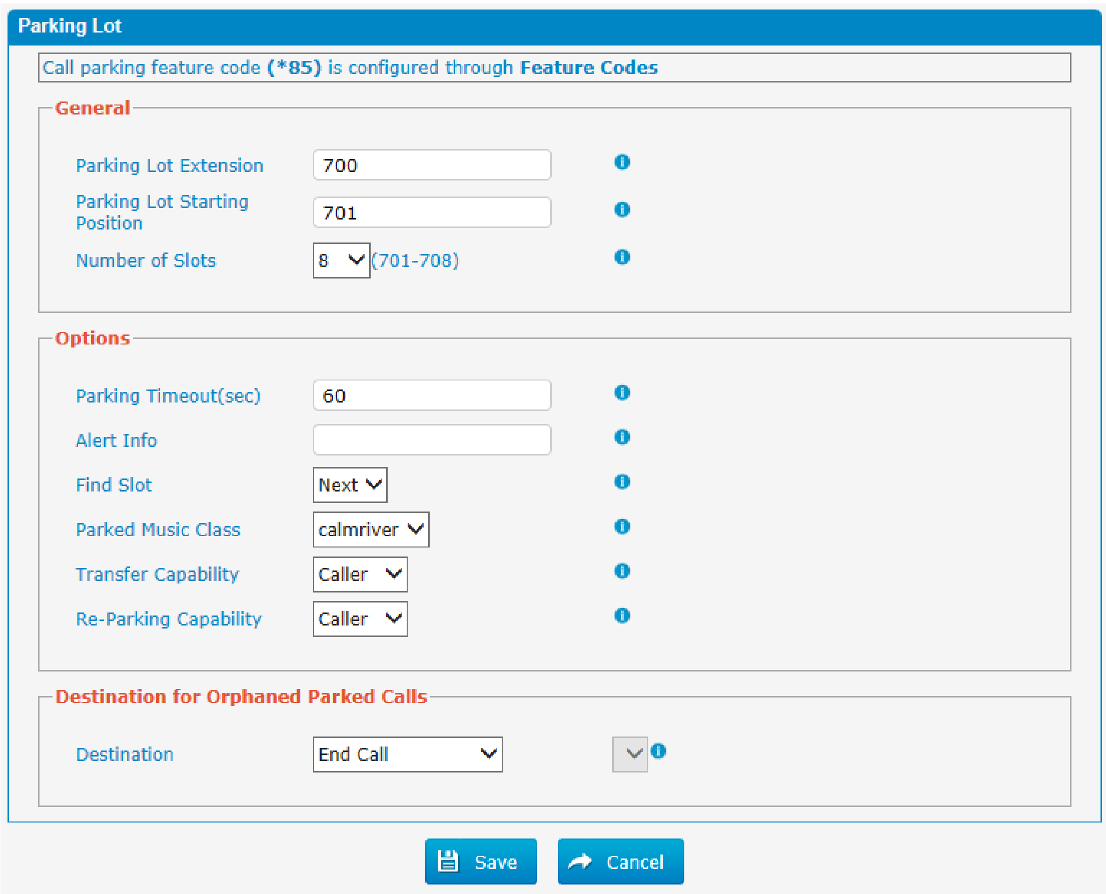
Define the extensions that will be permitted to use this outbound route.

·Allow Trunks

Define the trunks that can be used for this outbound route.

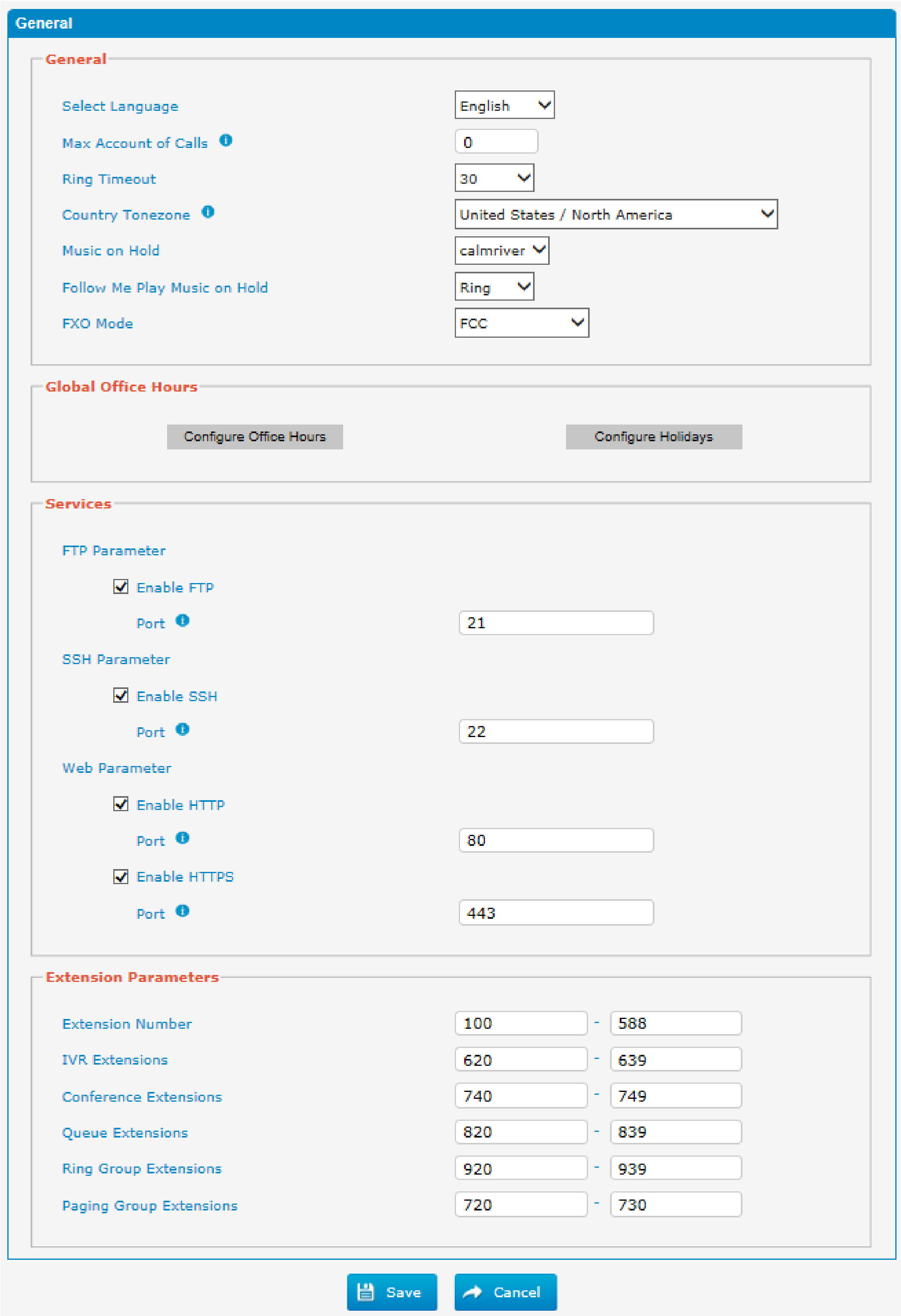
3.6.5 Parking Lot

Figure 3-6-5 Parking Lot



3.6.6 General Preferences

Figure 3-6-6 General Preferences

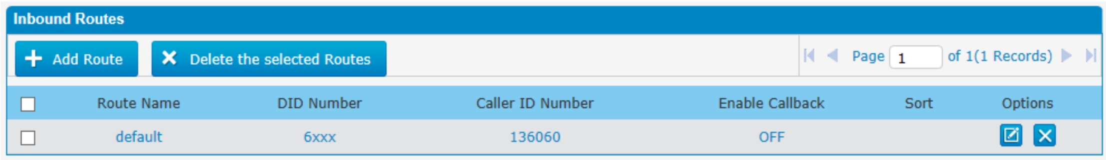


3.7 PBX Inbound Call Control

3.7.1 Inbound Routes

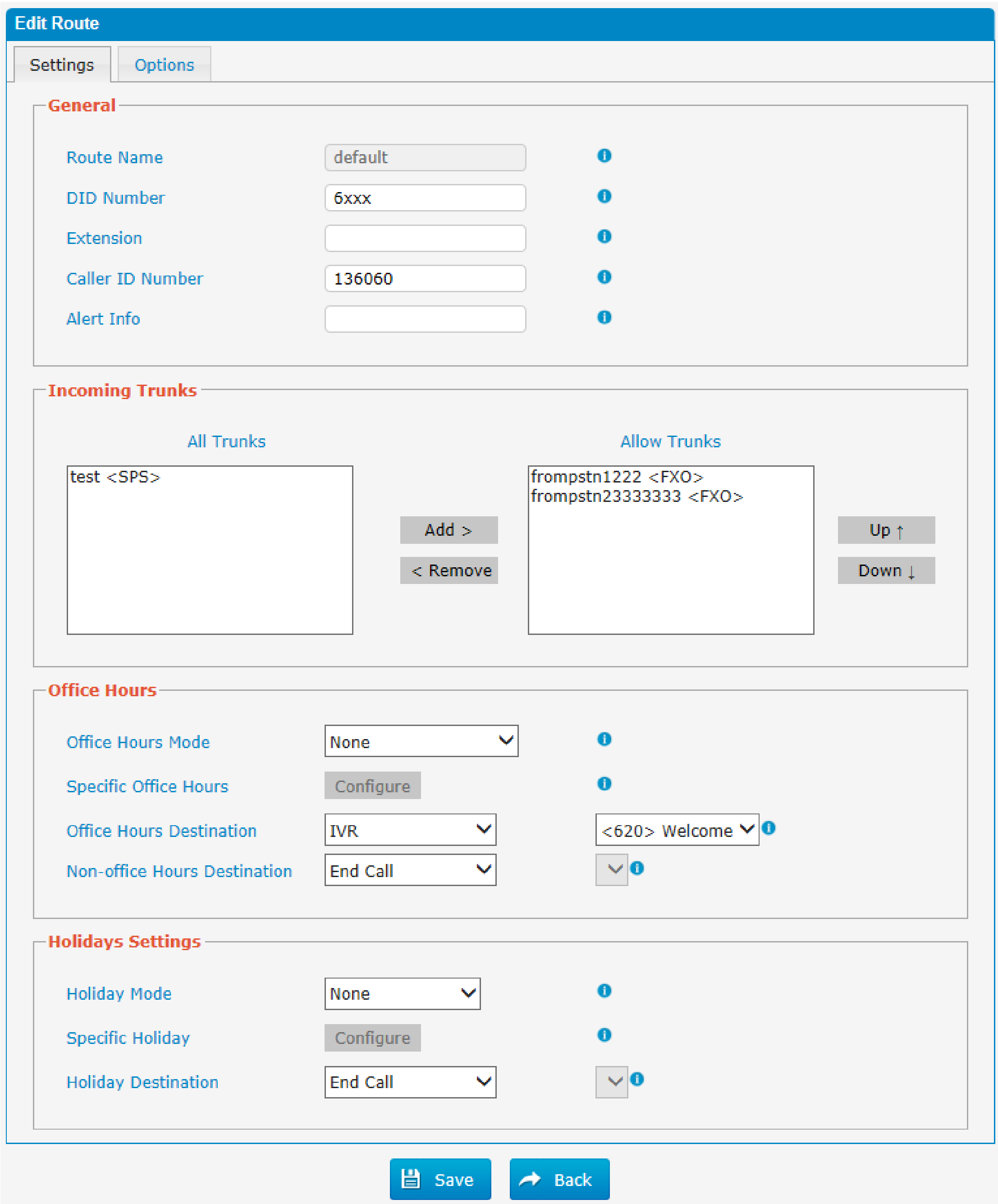
Inbound routing processes incoming call traffic to destination extensions during office hours or outside office hours

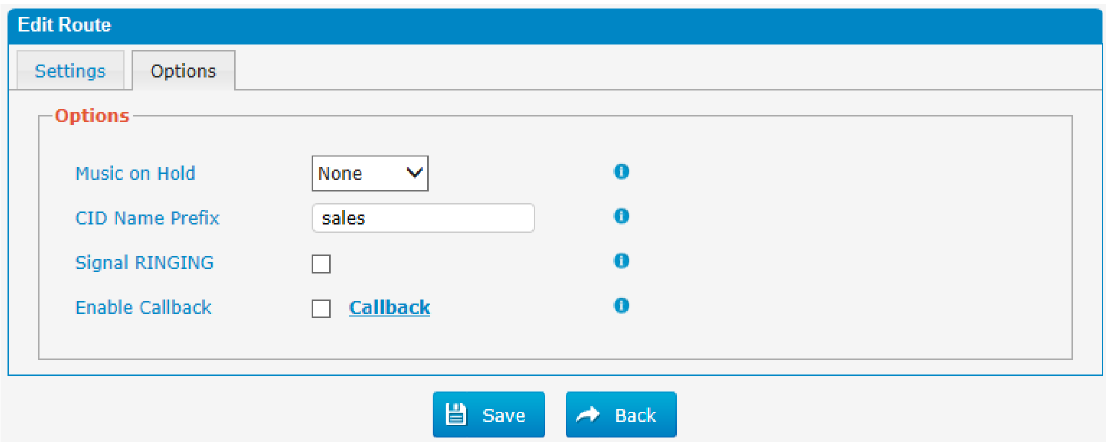
Figure 3-7-1 Inbound Routes



There is a default inbound route for all the trunks and set IVR as the destination, you can edit it or create a new one for your demands or you can delete multiple outbound routes at once as required.When an incoming call arrives, the system will first check “Holidays”.

Figure 3-7-1a Inbound Routes Edit


Edit Route
Settings
Options
General
Route Name
default
DID Number
6xxx
Extension
Caller ID Number
136060
Alert Info
Incoming Trunks
All Trunks
Allow Trunks
test <SPS>
Add >
< Remove
frompstn1222 <FXO>
frompstn23333333 <FXO>
Up ↑
Down ↓
Office Hours
Office Hours Mode
None
Specific Office Hours
Configure
Office Hours Destination
IVR
<620> Welcome
Non-office Hours Destination
End Call
Holidays Settings
Holiday Mode
None
Specific Holiday
Configure
Holiday Destination
End Call
Save
Back


Edit Route
Settings
Options
Options
Music on Hold
None
CID Name Prefix
sales
Signal RINGING
Enable Callback
Callback
Save
Back

1) General

·Route Name

A name for this inbound route. E.g. "default".

·DID Number

Define the expected DID Number if this trunk passes DID on incoming calls. Leave this field blank to match calls with any or no DID info. You can also use pattern matching to match a range of numbers. The following patterns may be used:

X: Any Digit from 0-9

Z: Any Digit from 1-9

N: Any Digit from 2-9

[12345-9]: Any digit in the brackets (in this example, 1, 2, 3, 4, 5, 6, 7, 8, 9)

The "**.**" Character will match any remaining digits. For example, "9011." will match any phone number that starts with "9011", excluding "9011" itself.

The "**!**" will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX** will match any 7-digit phone number.

Example 2: **1NXXNXXXXX** will match a phone number starting with a 1, followed by a 3-digit area code, and then 6-digit number.

·Extension

Define the extension for DID number. This field is only valid when you use BRI, SIP, SPS or SPX trunk for this inbound router. You can only input number and "-" in this field and the format can be xxx or xxx-xxx. The count of the number must be only one or equal to the count of the DID number.

·Caller ID Number

Define the Caller ID Number to be matched on incoming calls. Leave this field blank to match any or no DID info.

You can also use a pattern match (e.g. 2[345]X) to match a range of numbers.

The following patterns may be used:

X: Any Digit from 0-9

Z: Any Digit from 1-9

N: Any Digit from 2-9

[12345-9]: Any digit in the brackets (in this example, 1, 2, 3, 4, 5, 6, 7, 8, 9)

The "**.**" Character will match any remaining digits. For example, "9011." will match any phone number that starts with "9011", excluding "9011" itself.

The "**!**" will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX** will match any 7 digits phone number.

Example 2: **1NXXNXXXXX** will match a phone number starting with a 1, followed by a 3-digit area code, and then 6-digit number.

·Alert info

Alert info can be used for distinctive ring with SIP devices.

2) Incoming Trunks

This area allows you to select which trunks will be member trunks for this route. To make a trunk a member of this route, please move it to the "Selected" box.

3) Office hours

Select one defined business days office hours.

·Office Hours Mode

Optional office hours mode to be used for this route.
None: Disable office hours for this route.
Gloal office hours: It is configured through general preferences.
Specific office hours: Use the specific office hours settings.

·Office Hours Destination

Configure where to route the incoming calls during office hours.

·End Calls

Route the incoming calls to end calls, the system will auto hang up the call.

·Extension

Route the incoming calls to a specific extension.

·Voicemail

Route the incoming calls to extension's voicemail.

·IVR

Route the incoming calls to a specific IVR.

·Ring Group

Route the incoming calls to a specific Ring Group.

·Conference Room

Route the incoming calls to a specific Conference Room.

·DISA

Route the incoming calls to a specific DISA.

·Queues

Route the incoming calls to a specific Queue.

·Outbound Routes

Route the incoming calls to a specific outbound route.

This function is mainly used for the connection of two branches.
For example: Company A locates headquarters in the USA with a branch B in China. A and B both have a MUC1004 phone system.
Now if staff of A would like to make a call to a telephone or mobile phone in China from the extension of A but via the FXS line of B, that can be done by this configuration.

·Non-office Hours Destination

Configure where to route the incoming calls during non-office hours.

4) Holidays Settings

Define where the calls will be routed during Holidays.

·Holiday Mode

Select which defined Holiday to use.
None: Disable holiday for this route.
Gloal holiday: It is configured through general preferences.
Specific holiday: Use the holiday settings.

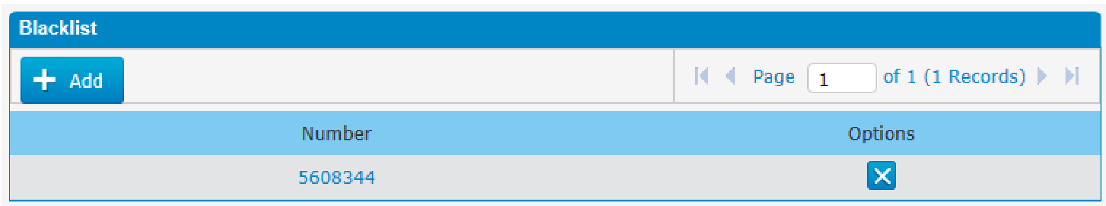
·Holiday Destination

Configure where to route the incoming calls during holidays.

3.7.2 Blacklist

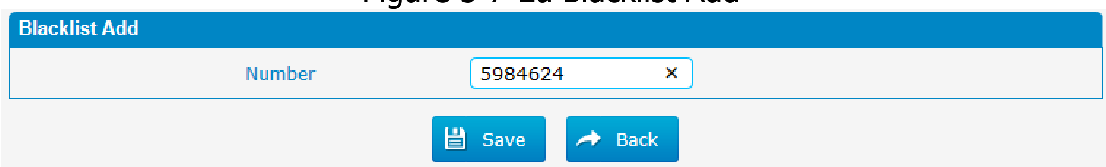
Blacklist is used to block an incoming/outgoing call. If the number of incoming/outgoing call is listed in the number blacklist, the caller will hear the following prompt: "The number you have dialed is not in service. Please check the number and try again". The system will then disconnect the call.

Figure 3-7-2 Blacklist



We can add a number to blacklist

Figure 3-7-2a Blacklist Add



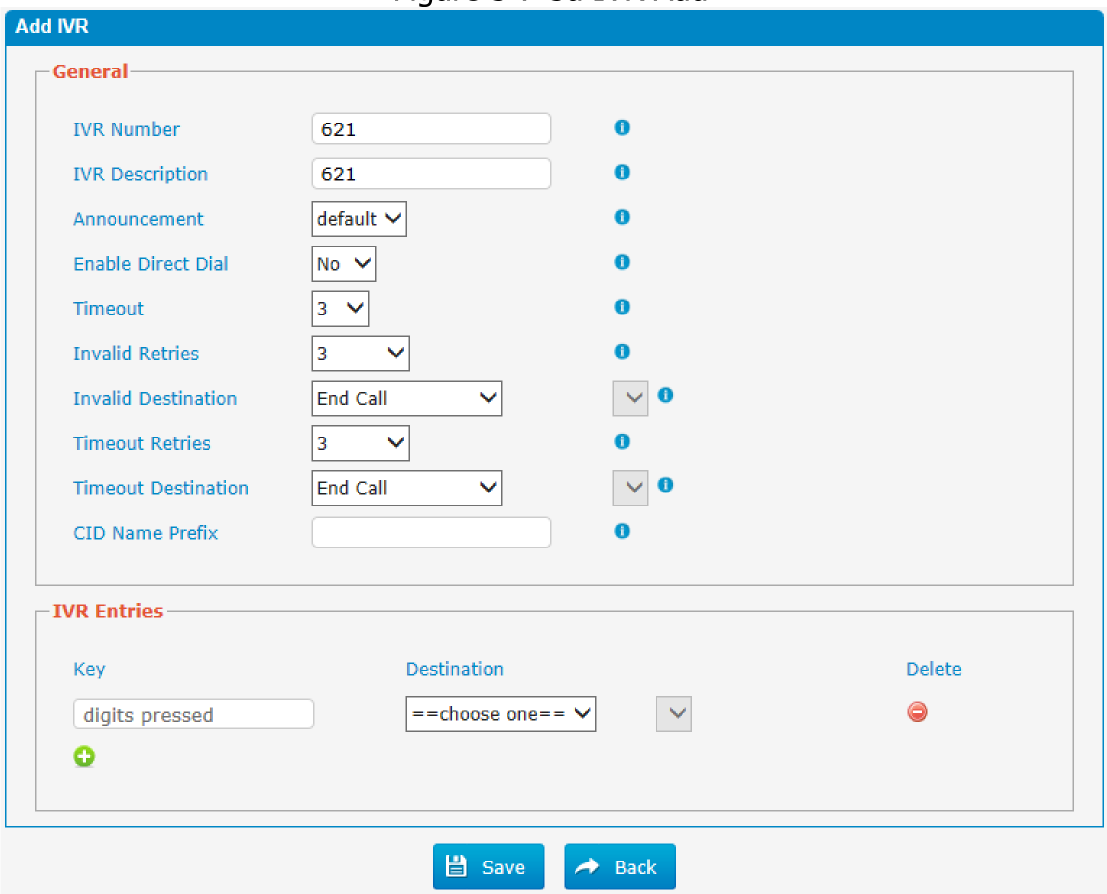
3.7.3 IVR

When there's an inbound call aims at Auto Attendant, MUC1004 will play an IVR recording and route the caller to the requested destination (for example, "Welcome to XX company,for sales press 1, for technical support press 2, for operator press 0", etc.). The system will transfer the call to corresponding extension according to DTMF digits input by the user.

Figure 3-7-3 IVR

There is a default IVR here, we can edit it directly or add IVR by yourself.

Figure 3-7-3a IVR Add



·Number

MUC1004 treats IVR as an extension; you can dial this extension number to reach the IVR from internal extensions.

·Description

Description of this IVR.

·Announcement

Greeting to be playd on entry to the IVR.

·Enable Direct Dial

Allow the caller to dial other extensions number directly.

·Timeout

The number of times that the selected IVR prompt will be played.

·Invalid

Define the invalid action. The invalid action is triggered if the user enters a DTMF digit that is not defined for this IVR.

·Key

The Key pressed when the callers hear the IVR prompt.

·Destination

Where will MUC1004 route the call when the action occurs.

3.7.4 Queue

Call Queues give users (e.g. call centers) an efficient means to have their calls answeredin the order they were received to deliver top tier customer service.

Figure 3-7-4 Queue

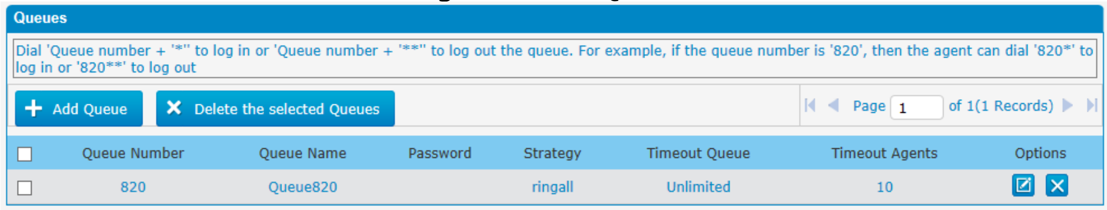
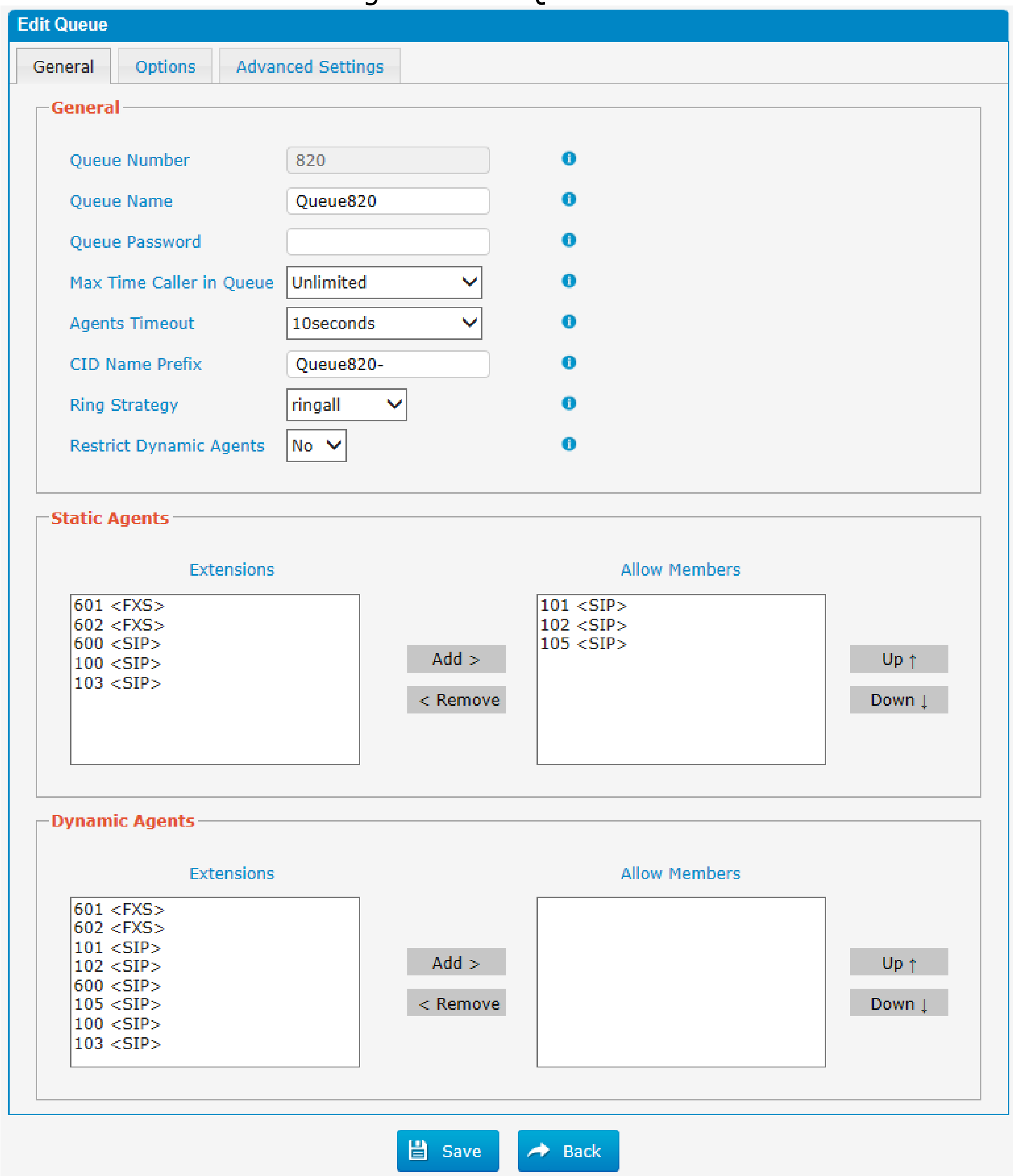
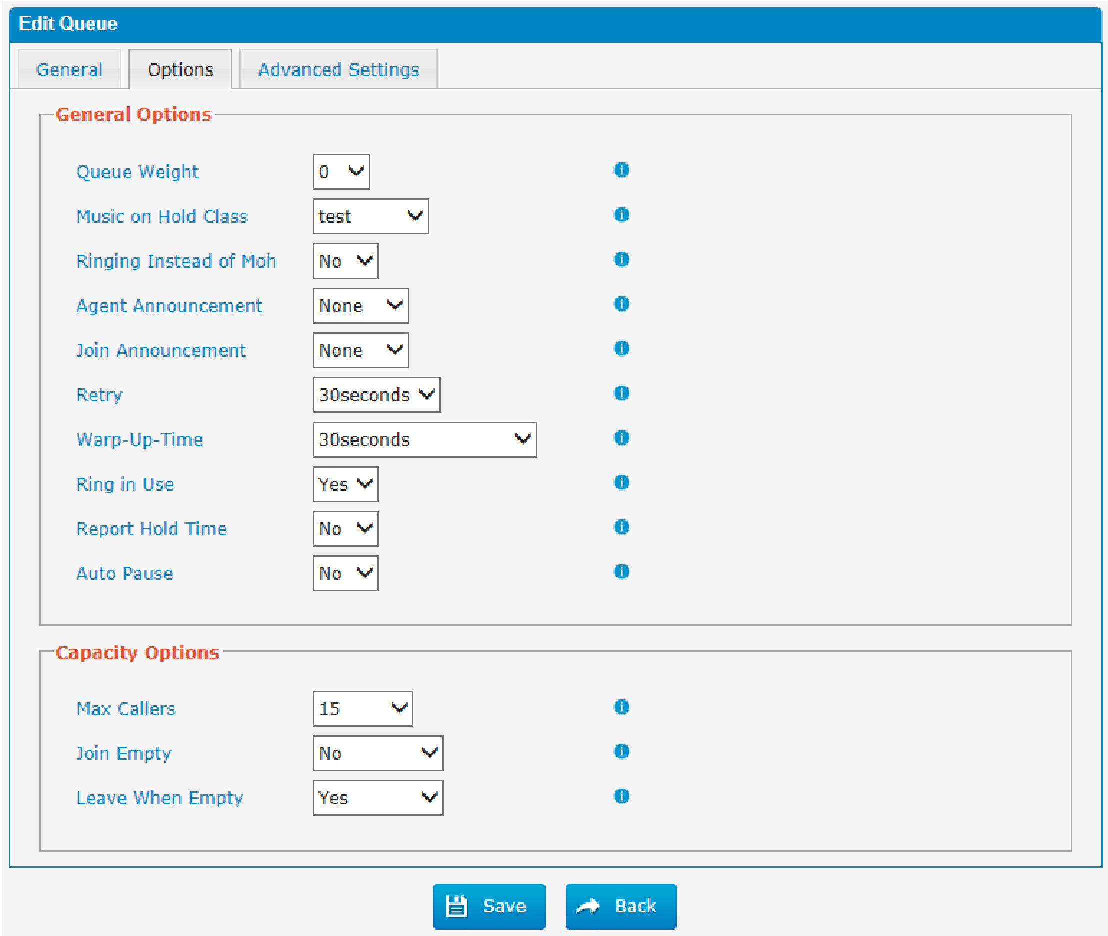
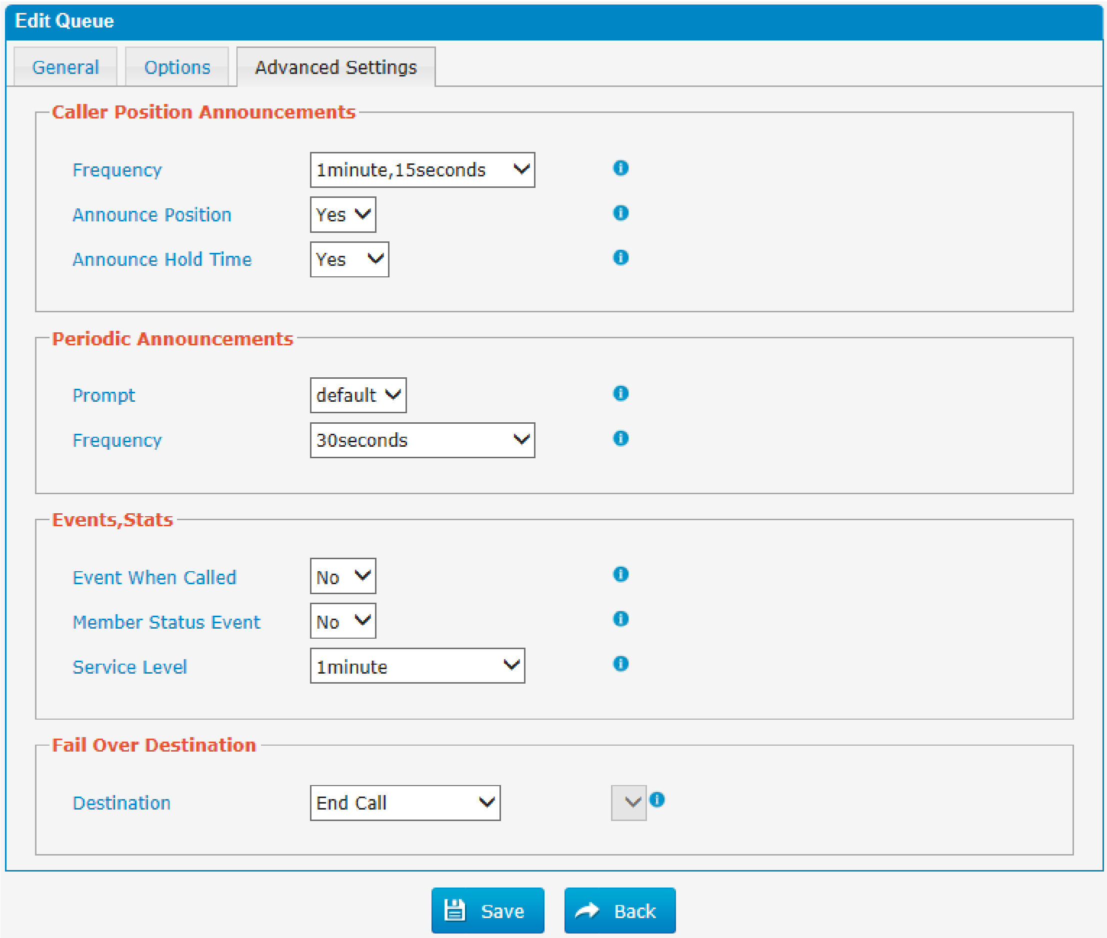


Figure 3-7-4a Queue Edit




Edit Queue
General
Options
Advanced Settings
General Options
Queue Weight
0
Music on Hold Class
test
Ringing Instead of Moh
No
Agent Announcement
None
Join Announcement
None
Retry
30seconds
Warp-Up-Time
30seconds
Ring in Use
Yes
Report Hold Time
No
Auto Pause
No
Capacity Options
Max Callers
15
Join Empty
No
Leave When Empty
Yes
Save
Back



·Queue Number

Use this number to dial into the queue, or transfer callers to this number to put them into the queue.

·Queue Name

A name for the Queue.

·Queue Password

You can require agents to enter a password before they can log in to this queue.

·Queue Max Caller Time

The maximum number of seconds a caller can wait in a queue before being pulled out (0 for unlimited).

·Queue Agent Timeout

The number of seconds an agent's phone can ring before we consider it a timeout.

·Queue Ring Strategy

This option sets the Ringing Strategy for this Queue. The options are

RingAll: Ring all available Agents simultaneously until one answers.

LeastRecent: Ring the Agent which was least recently called.

FewestCalls: Ring the Agent with the fewest completed calls.

Random: Ring a Random Agent.
RRmemory: Round Robin with Memory, Remembers where it left off in the last ring pass.
Linear: Rings agents in the other specified,for dynamic agents in the other they logged in.

·Static Agents

This selection shows all users. Selecting a user here makes them an agent of the current queue.

·Music on Hold

Select the "Music on Hold" Class for this Queue.

·Agent Announcement

Announcement played to the Agent prior to bridging in the caller.

·Join Announcement

Announcement played to callers once prior to joining the queue.

·Retry

The number of seconds we wait before trying all the phones again.

·Wrap-up time

How many seconds after the completion of a call an Agent will have before the Queue can ring them with a new call. The default is 30.

·Join Empty

This option controls whether callers can join a call queue that has no agents. There are two options,
Yes: Callers can join a call queue without agents or only unavailable agents
No: Callers cannot join a queue when there are no agents in the queue.
The default option is No.

·Leave When Empty

This option controls whether callers already on hold are forced out of a queue that has no agents. There are two options.
Yes: Callers are forced out of a queue when no agents are logged in.
No: Callers will remain in a queue with no agents.

.Caller Position Announcements

·Frequency

How often to announce queue position and estimated hold time.
Note: "0 seconds" means disabling the announcement.

·Announce Position

Announce position of caller in the queue

·Announce Hold Time

Enabling this option causes MUC1004 to announce the hold time to the caller periodically based on the frequency timer. Either yes or no; hold time will not be announced if <1 minute.

.Periodic Announcements

·Prompt

Select a prompt file to play periodically.

·Frequency

How often to announce a prompt to the caller.

·Events

If a caller presses the key while waiting in the queue, this setting selects which action should process the key press.

·Fail over destination

Define the failover action. A failover occurs after the user reach the Queue max wait time.

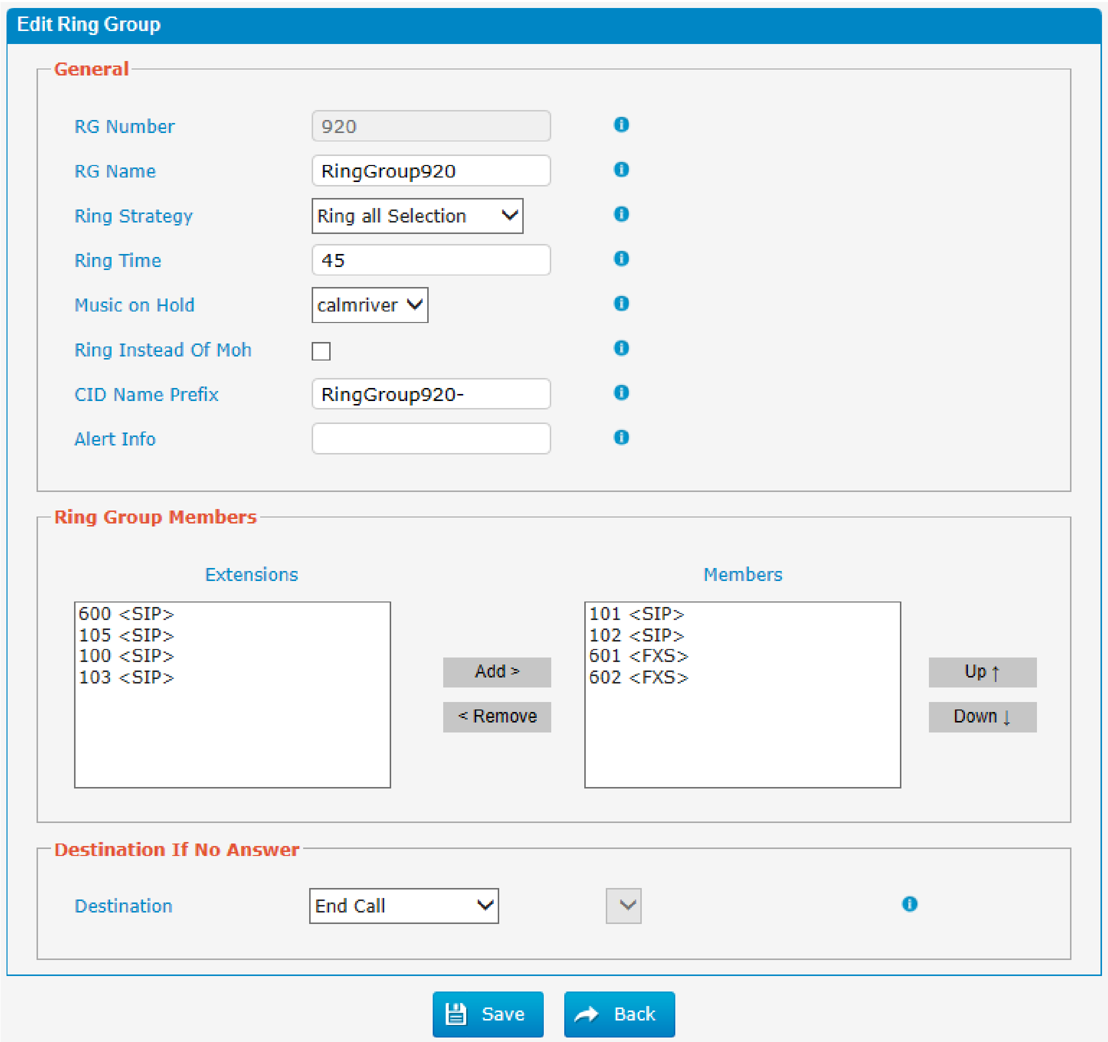
3.7.5 Ring Groups

Ring groups can be configured to balance the call traffic for multiple users and give callers a higher level of availability for incoming calls. Multiple ring methods and voicemail are supported.

Note: Call forward(follow me) feature in extension page will not take effect when it's ringing as an agent.

Figure 3-7-5 Ring Groupns

Figure 3-7-5a Ring Groupns Edit



·Ring Group Number

This option defines the numbered extension that can be dialed to reach this group.

·Ring Group Name

This option defines a name for this group, e.g. "Sales". "Ring Group Name" is a label to help you identify this group in the group list.

·Ring Strategy

This option sets the Ringing Strategy for this Group. The options are as follows:

1. Ring All Simultaneously: Ring all available Extensions simultaneously.
2. Ring Sequentially: Ring each extension in the group one at a time.

·Ring time

1. If the strategy is "Ring All Simultaneously", it means the number of seconds to ring this group before routing the call according to the

"Destination if No Answer" settings.

2. If the strategy is "Ring Sequentially", it means the number of seconds to ring a single extension before moving onto the next one.

·Music on hold

If you select a music on hold class to play,instead of "ring",they will hear that instead ringing while they are waiting for someone to pick up

·Ring Group Members

An extension can be made a member of this ring group by moving it into the "Selected" box.

·Destination If No Answer

When all members on this group fail to answer the call, system will handle the call according to the selected destination.

·CID name prefix

You can optionally prefix the caller ID name when ringing extensions in this group, ie: if you prefix with "Sales:",a call from John doe would display as "Sales:John doe" on the extensions that ring.

3.7.6 Conferences

Conference Calls increase employee efficiency and productivity, and provide a more cost-effective way to hold meetings. Conference agents can dial * to access the settings options and the admin can kick the last user out and can lock the conference room.

Figure 3-7-6 Conferences

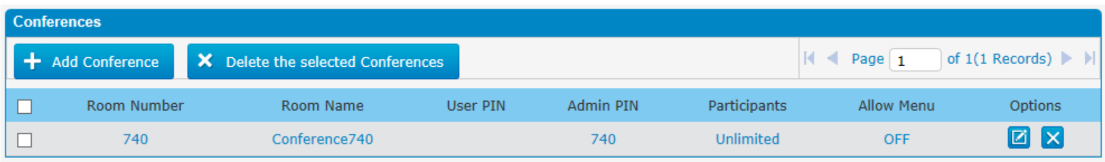
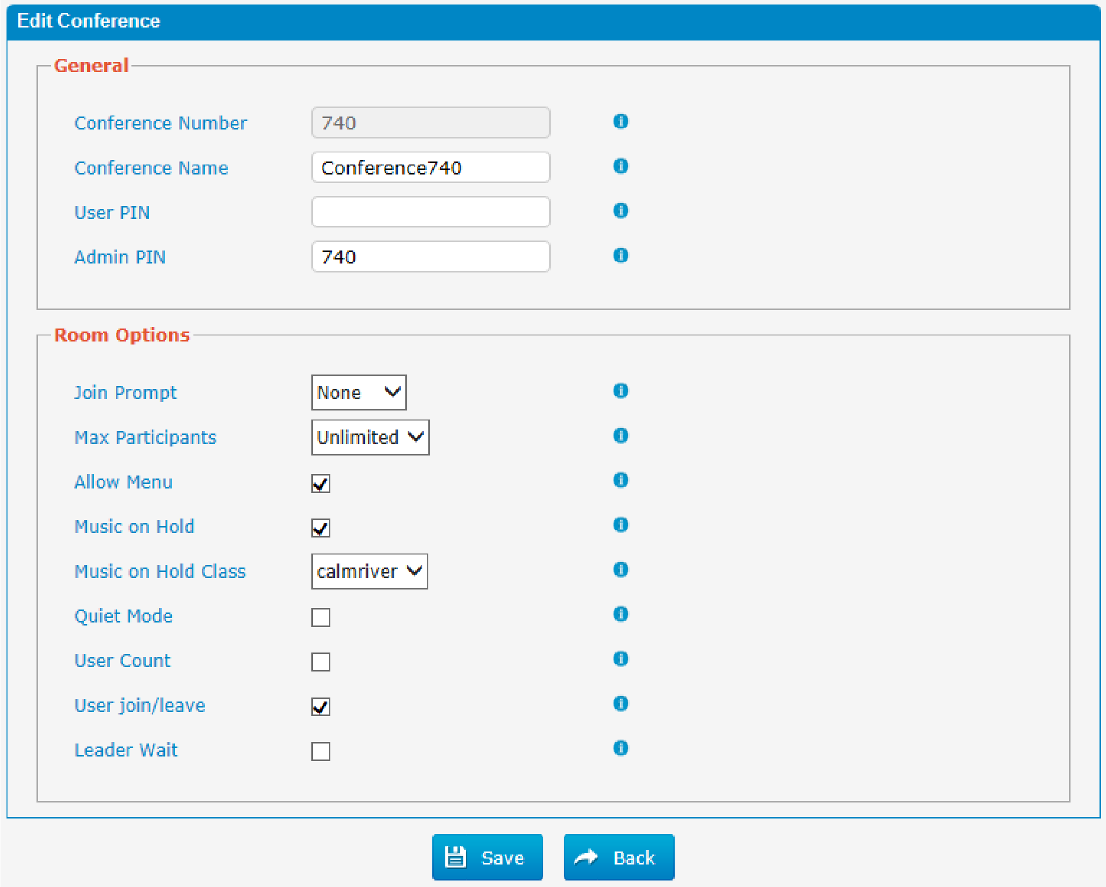


Figure 3-7-6a Conferences Edit/Add



·Conference Number

This is the number dialed to reach this Conference Room.

·User PIN

Set a PIN that must be entered in order to access this conference room (e.g. 1234).

·Admin PIN

Enter a PIN number for the admin user

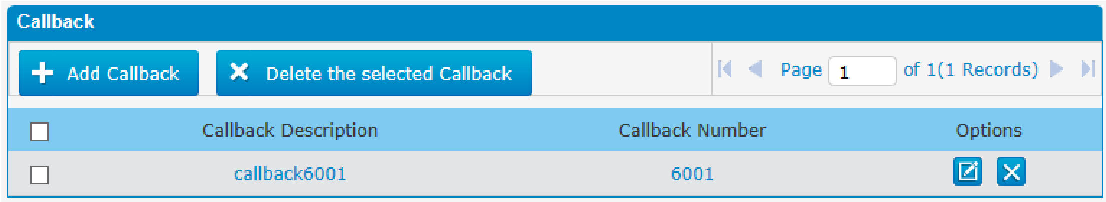
3.7.7 Callback

MUC1004 allows caller A to dial an inbound route number, and after hearing the ring, A can hang up the call or wait for MUC1004 to cut off the call, then MUC1004 will call A with this number. When A picks up the call, A can dial the number he wants to call; MUC1004 will call the number with its outbound route.

Notes:

1. If you'd like to use callback feature, please make sure it's enabled on the inbound route setting panel.
2. No callback rules needed to be set if the trunk supports call back with the caller ID directly.

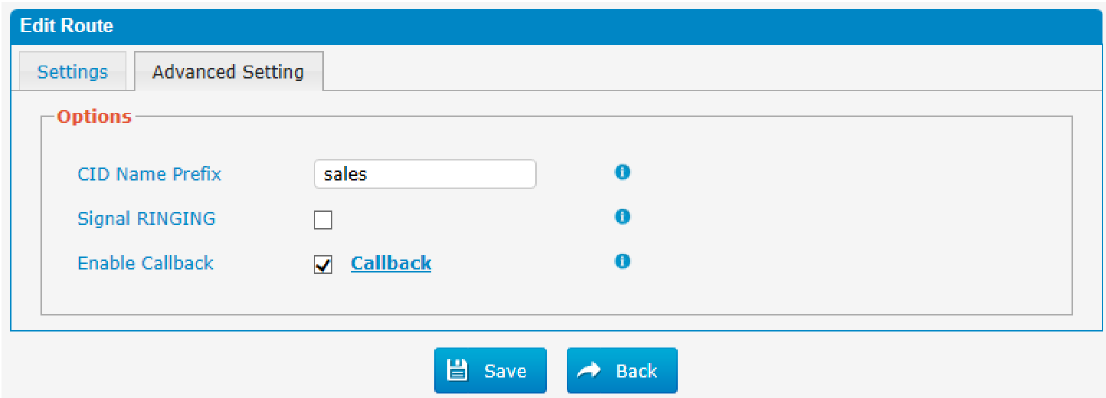
Figure 3-7-7 Callback



Follow the steps below to use this function.

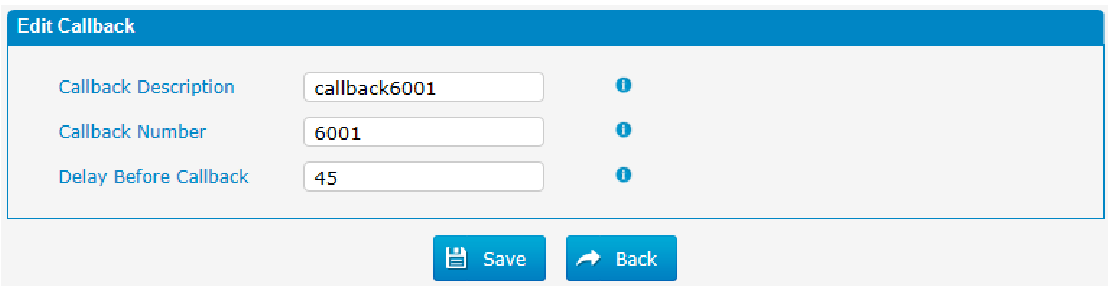
Step 1: Enable Callback.

Inbound Routes—Choose "Yes" on" Enable Callback" to enable this function.



Step 2: Create Callback number.

Figure 3-7-7 Callback Edit/Add



3.8 PBX Advanced Settings

3.8.1 SIP settings

This is the SIP settings in MUC1004, including General settings, NAT, Codecs, Qos, Response code and Advanced settings.
This section describes how to configure SIP server and SIP parameters

3.8.1.1 SIP General setting

Figure 3.8.1.1 SIP General Setting

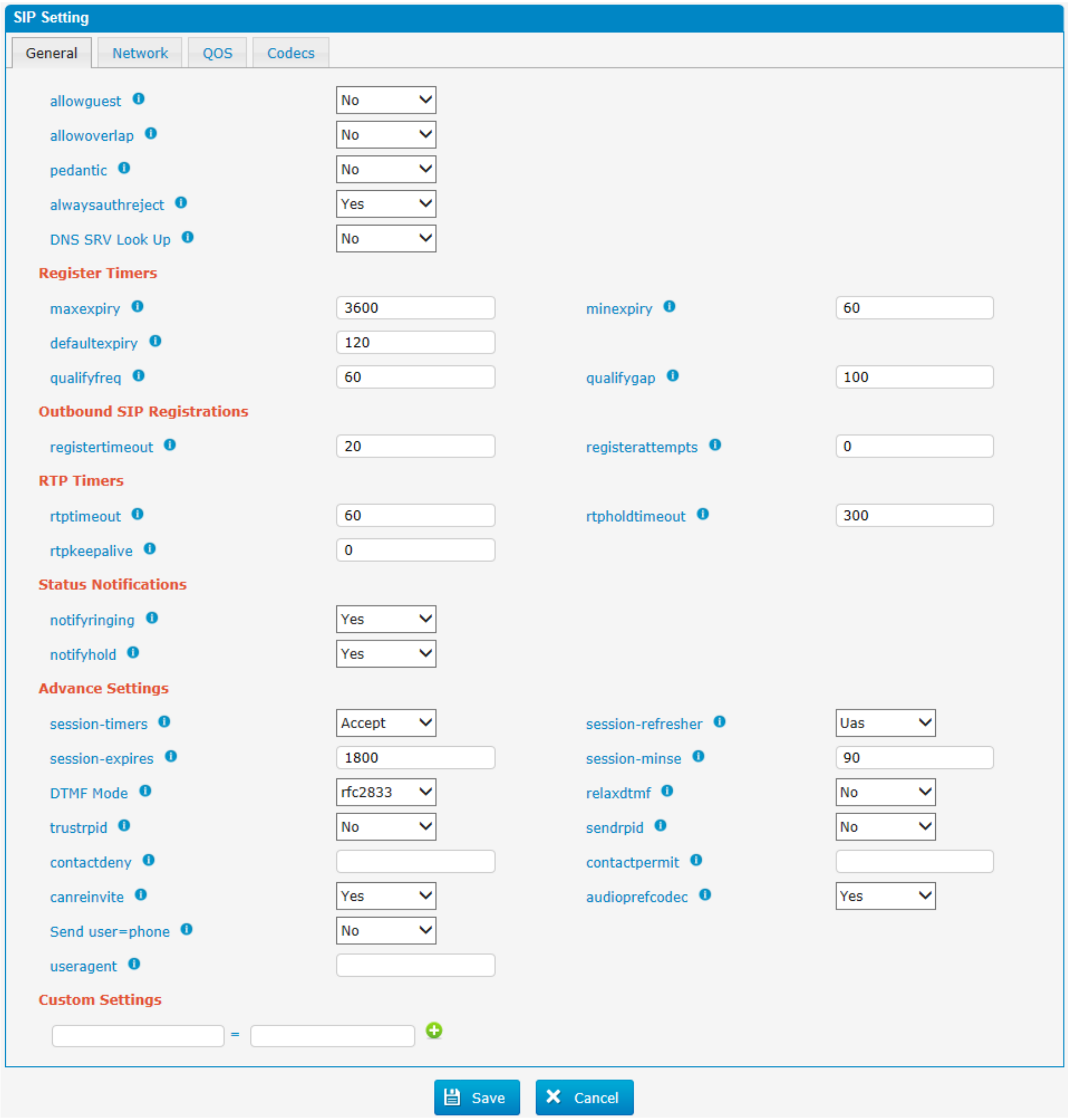


Table 3.8.1.1

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 MUC1004 rejects "Register"or "Invite" packets, MUC1004 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 “MUC1004”, you can change it if needed.

3.8.1.2 SIP Network Configuration

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

Figure 3.8.1.2 SIP Network Configuration

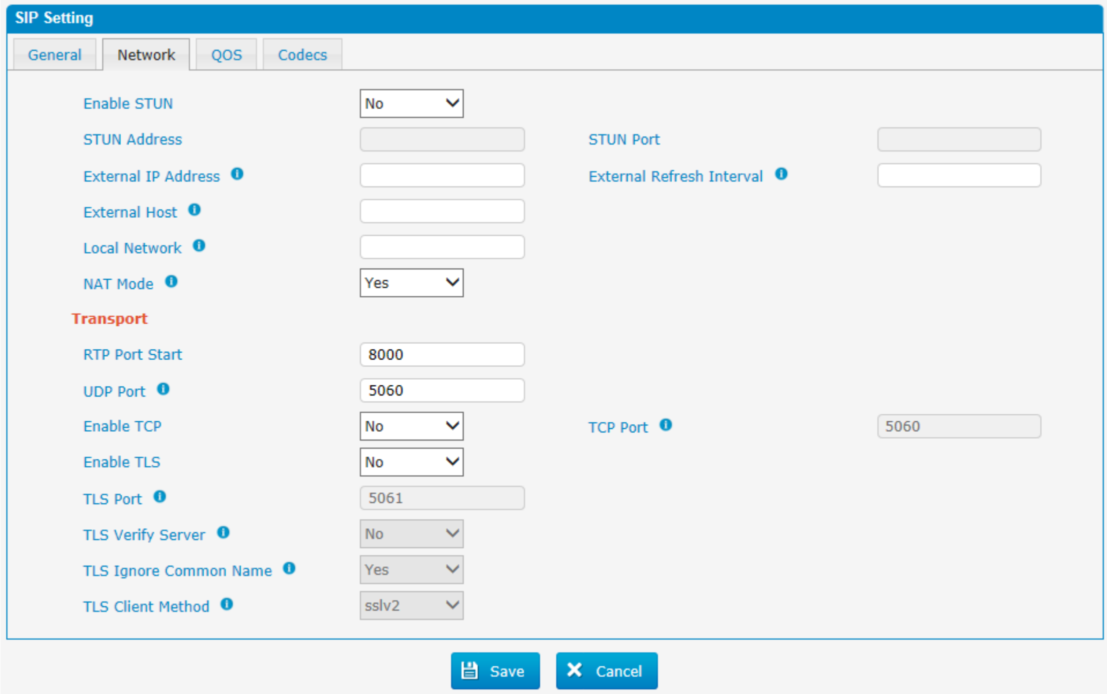


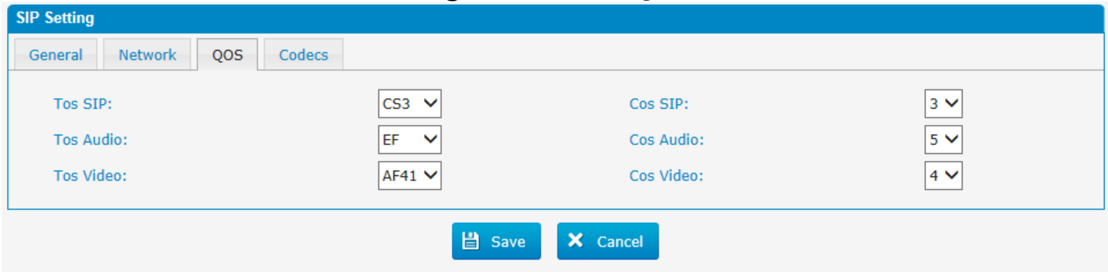
Table 3.8.1.2

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 MUC1004 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 MUC1004 as a TLS server, whether or not to verify client's certificate. It is "No" by default.
TLS Client Method	When using MUC1004 as TLS client, specify the protocol for outbound TLS connections. You can select it as tlsv1, sslv2 or sslv3.

3.8.1.3 Qos

Figure 3.8.1.3 Qos



3.8.1.4 codecs

We can choose the allowed codec in MUC1004, 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 3.8.1.4 codecs

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

3.8.2 IAX setting

IAX is the Internal Asterisk Exchange protocol, you can connect to MUC1004 or register IAX trunk to another IAX server. It's supported by the asterisk-based IPPBX.

Figure 3.8.2 IAX setting

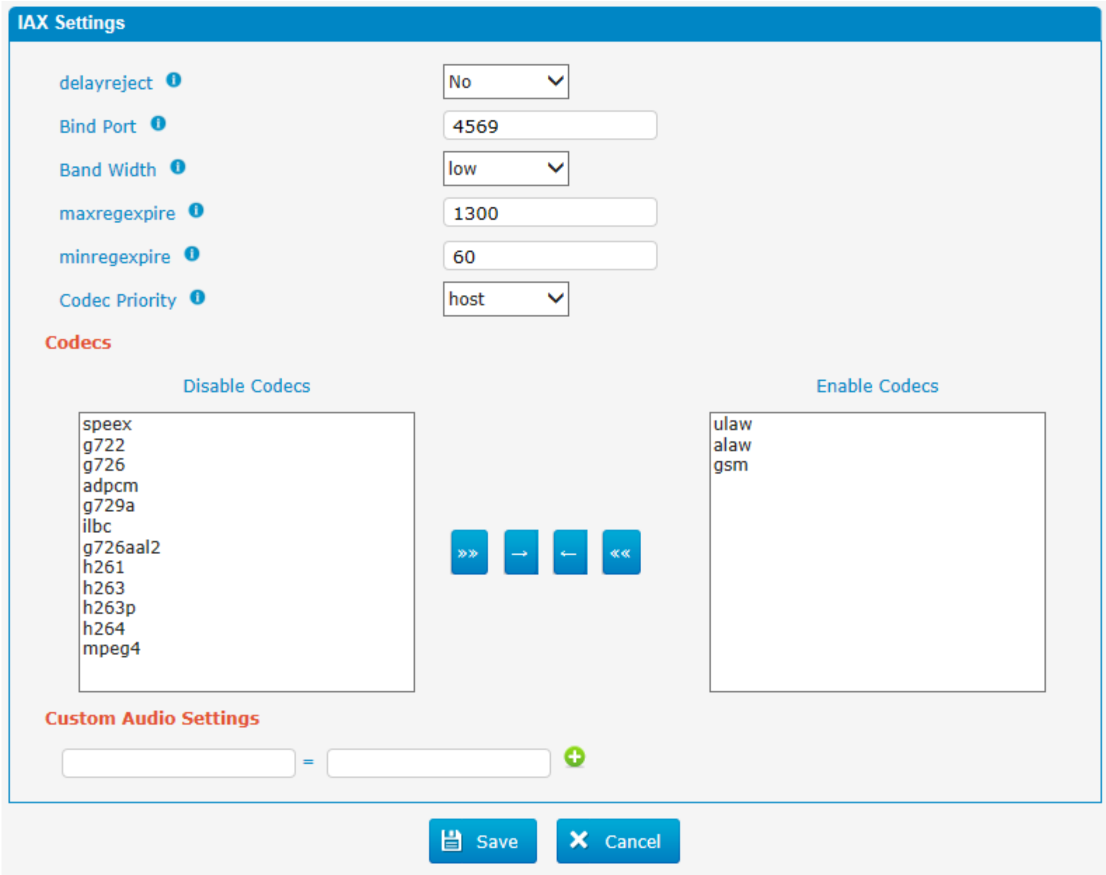


Table 3.8.2

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.

3.8.3 PIN Sets

In this page users can manage all the passwords of outbound routes, PIN User, and DISA.

Figure 3.8.3 PIN sets

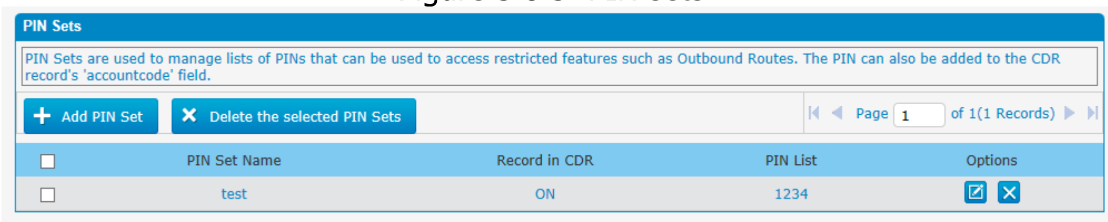
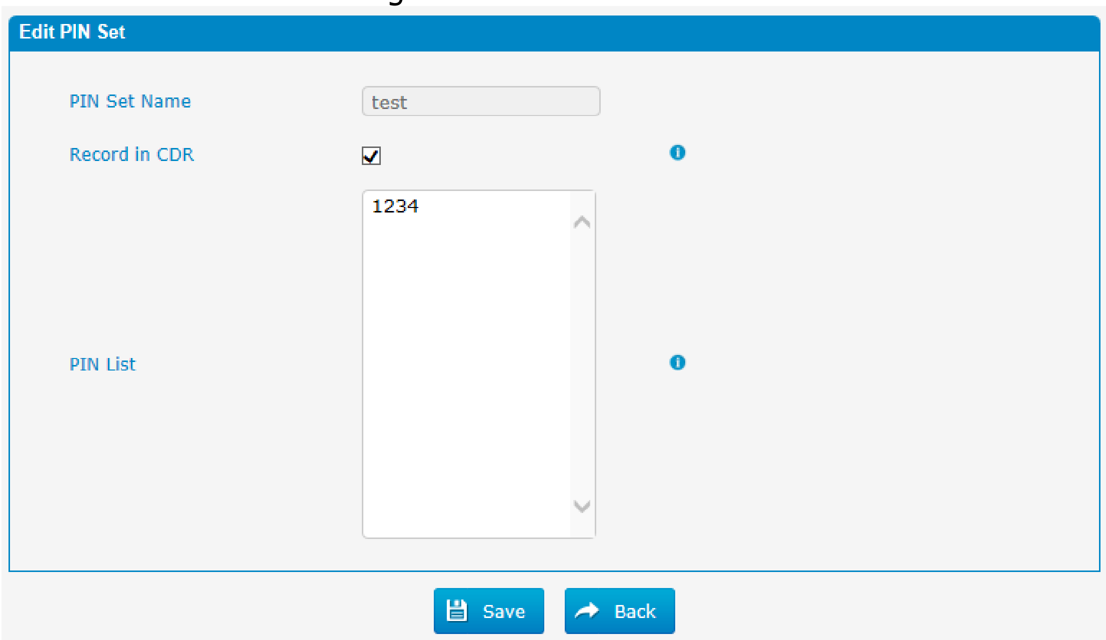


Figure 3.8.3a PIN Set Edit



·PIN Set Name

A character-based name for this PIN list, e.g. "testPIN"

·Record in CDR

If set yes, the PIN code will be displayed in call log.

·PIN list

PIN list is a numeric field. Letters and punctuation are not allowed in this field. Fill in one PIN and if you end with enter for each PIN, you could create multiple PINs.

3.8.4 DISA

DISA (Direct Inward System Access) allows someone calling in from outside the telephone switch (PBX) to obtain an "internal" system dial tone and make calls as if they were using one of the extensions attached to the telephone switch. To use DISA, a user calls a DISA number, which invokes the DISA application. The DISA application in turn requires the user to enter a PIN number, followed by the pound sign (#). If the PIN number is correct, the user will hear dial tone on which a call may be placed. Obviously, this type of access has serious security implications, and great care must be taken not to compromise your security.

Figure 3.8.4 DISA

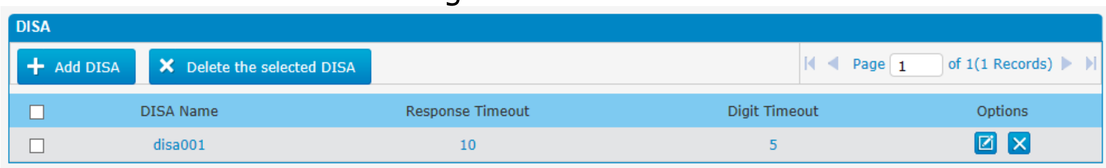
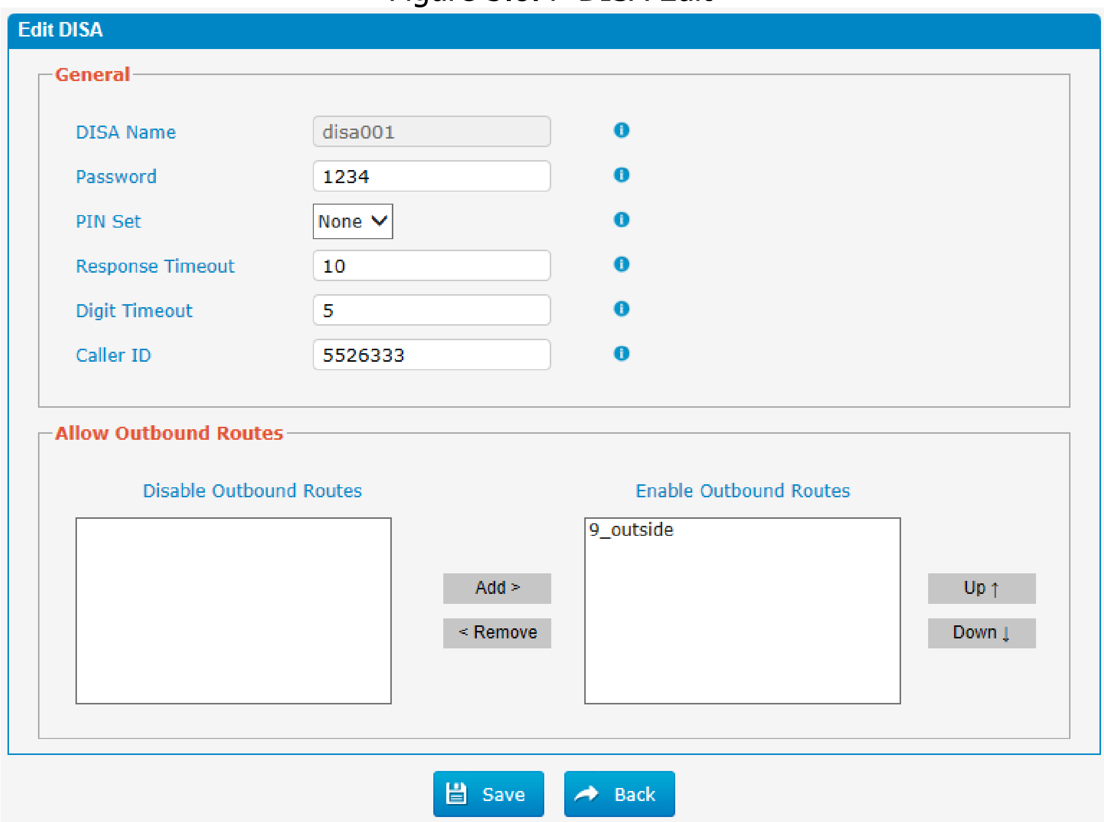


Figure 3.8.4 DISA Edit



·DISA Name

Give this DISA application a name to help you identify it.

·Password

The password for this DISA.

·PIN Set

Optional: select a PIN set to use.If using this option,leave the password field blank

·Response Timeout

The maximum amount of time the system will wait before hanging up the call if the user has dialed an incomplete or invalid number. The default is 10 seconds.

·Digit Timeout

The maximum amount of time permitted between each digit when the user is dialing an extension number. The default is 5 seconds.

·Allow Outbound Routes

Used to set the outbound routes that can be accessed from this DISA.

3.8.5 Paging and Intercom

Paging is used to make an announcement over the speakerphone to a phone or group of phones. Targeted phones will not ring, but instead answer immediately into speakerphone mode. Please note that this section is for configuring paging groups. If you would like to configure Intercom settings, please open the PBX Basic -> Feature Codes screen.

Note: A paging group can have a maximum of 20 members.

Figure 3.8.5 Paging and Intercom

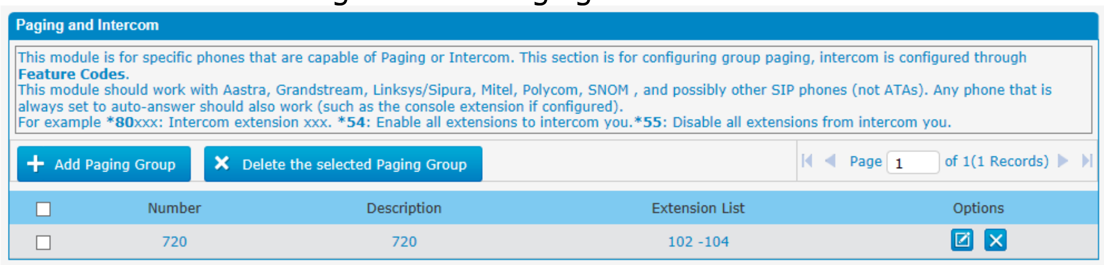
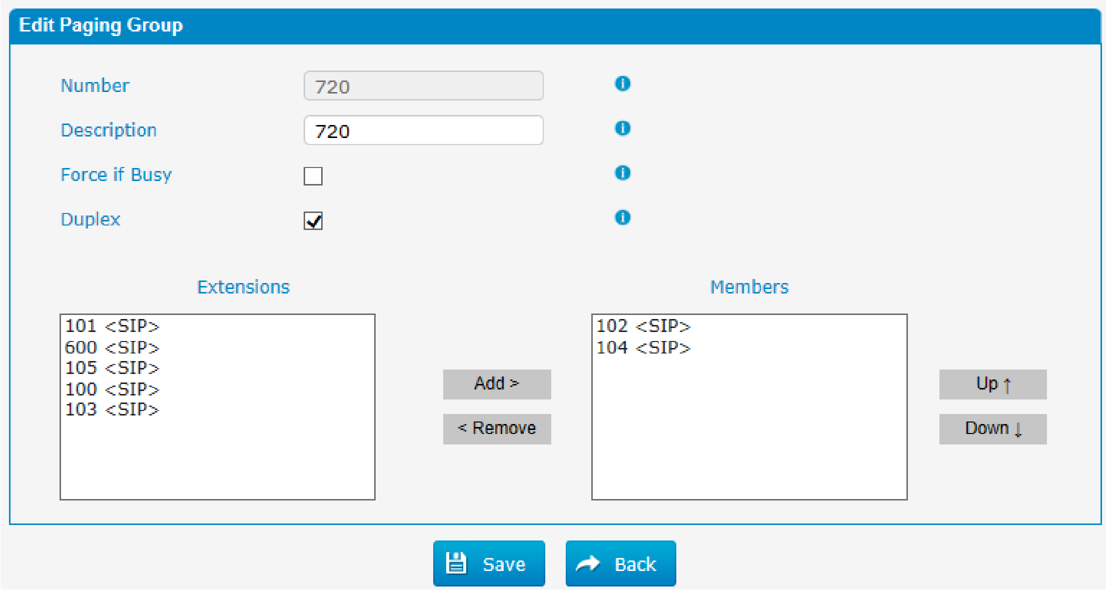


Figure 3.8.5a Paging and Intercom Edit/Add



·Paging Group Number

Define the numbered extension that may be dialed to reach this group.

·Duplex

Paging is typically one way for announcements only. Checking this will make paging duplex, allowing all users in the group to talk and be heard by all.

·Force if busy

If selected,will not check if the device is in use before paging it,This means conversations can be interrupted by a page(depending on how the device handles it).This is useful for "emergency" paging groups.

3.9 Voice Management

3.9.1 Voice prompt Recordings

We can record or upload the prompts in this page; you can also play it directly to confirm if it's a valid one, you can also download it and save it as a backup.

Figure 3-9-1 Voice prompt Recording

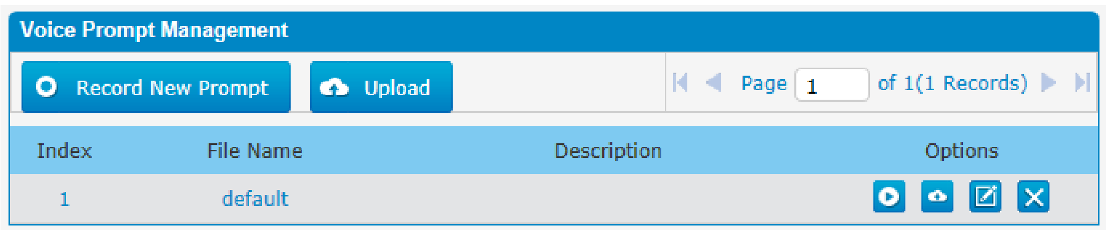
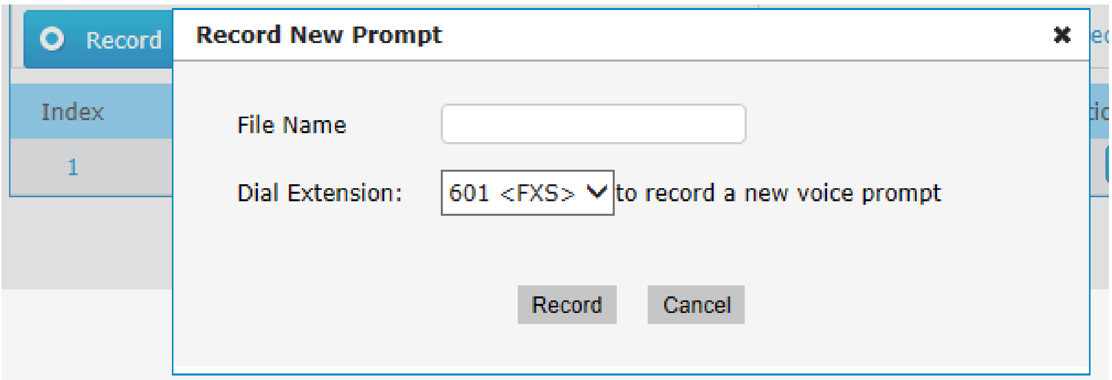


Figure 3-9-1a Record New Prompt



The administrator can record custom prompts by doing the following:

1) Click "Record New Custom Prompt".

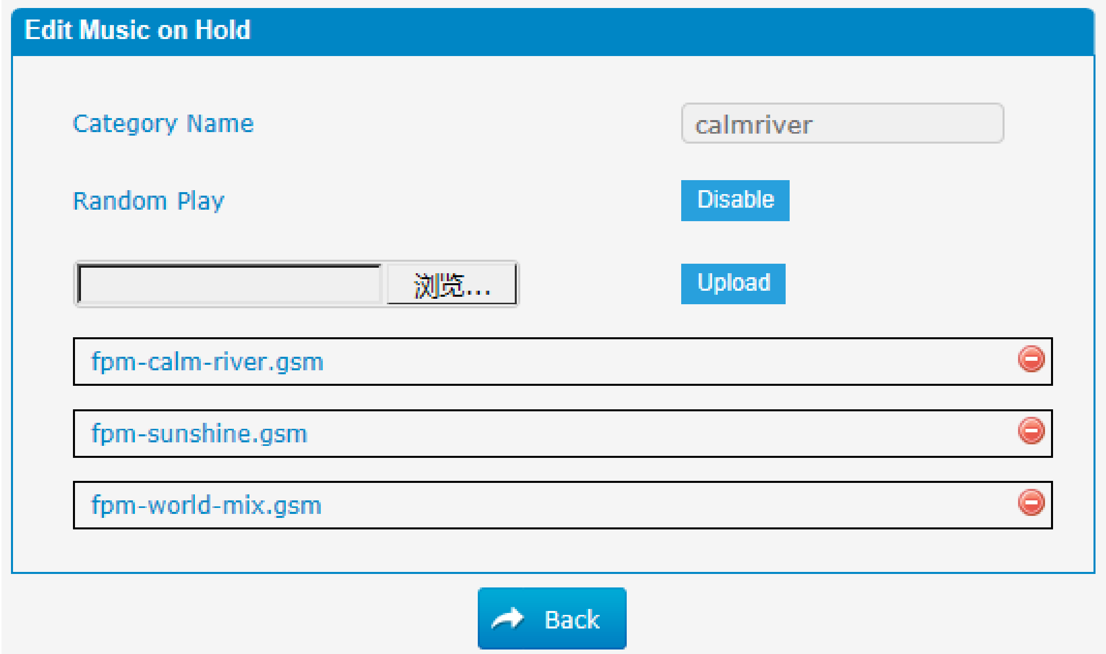
2) Input the desired file name on the popup window and choose an extension to call for recording (such as vp500).

3) Click "Record". The selected extension will ring and you can pick up the phone to start recording.

3.9.2 Music on Hold

Figure 3-9-2 Music on Hold

Figure 3-9-2a Music on Hold Edit



The administrator can upload on hold music as follows:

1) Click "Browse" to choose the desired audio file.
2) Click "Upload" to upload the selected file.

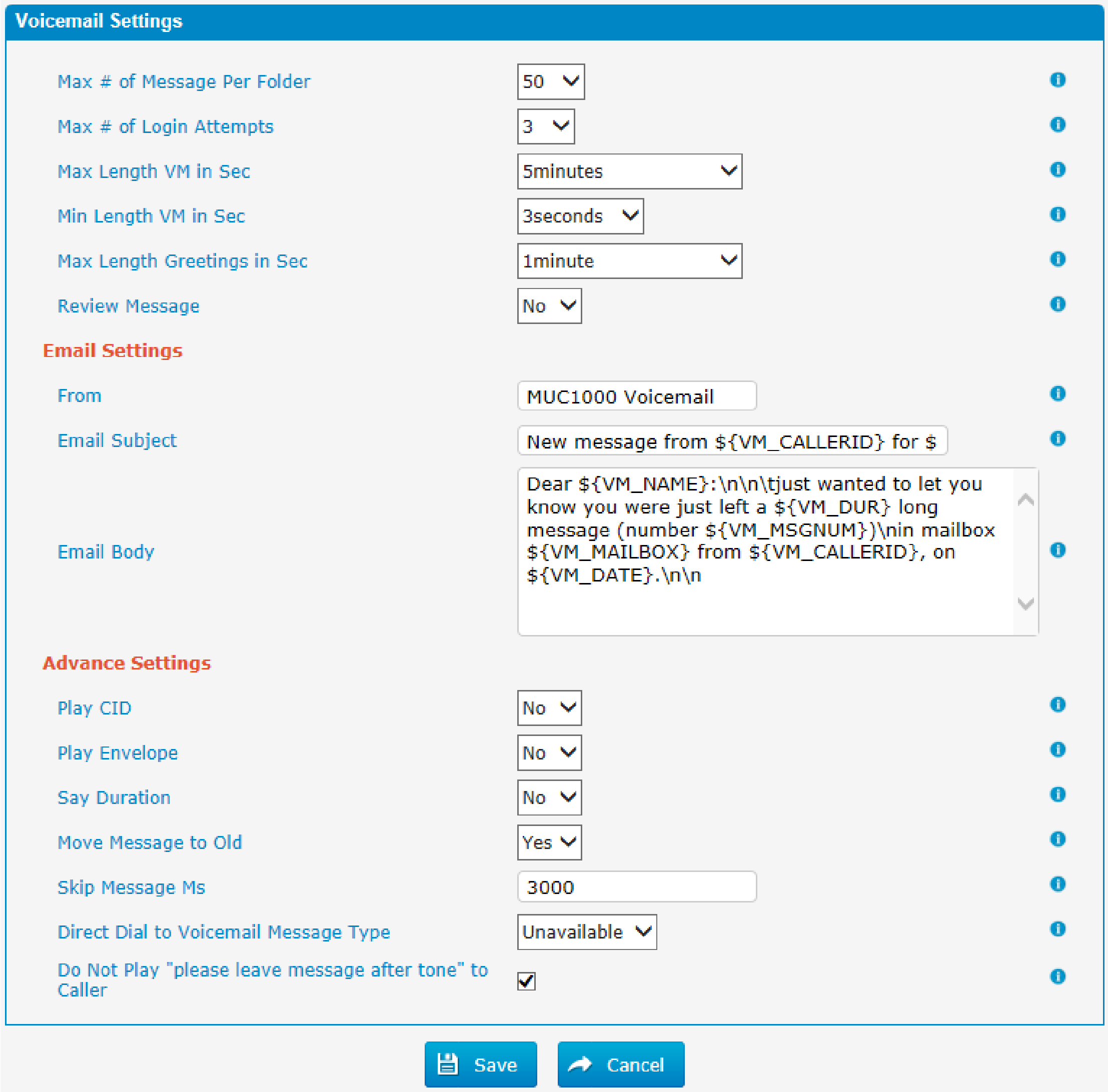
Note: The file size must not be larger than 1.8 MB, and the file must be WAV format:

GSM 6.10 8 kHz, Mono, 1 Kb/s;
Alaw/Ulaw 8 kHz, Mono, 1 Kb/s;
PCM 8 kHz, Mono, 16 Kb/s.

3.9.3 Voicemail Settings

In this page, we can configure some settings for voicemail feature, including general voicemail settings and SMTP settings, which is used for "voicemail to email".

Figure 3-9-3 Voicemail Setting



·Max # of Messages per Folder

Set the maximum number of messages that can be stored in a single voicemail box.

·Max # of Login Attempts

Max number of failed login attempts

·Max Length VM in Sec

Set the maximum length of a single voicemail message.

·Min Length VM in Sec

Set the minimum length of a single voicemail message. Messages below this threshold will be automatically deleted.

·Max Length Greetings in Sec

Max length of greeting in seconds.

·Review Message

Allow sender to review/record their message before save it(No by default)

·Play CID

Say the called ID information before the message

·Play envelope
Turn on/off envelope playback before message playback.
·Say Duration
Turn on/off the duration information before the message.
·Move message to old
Move heard messages to the "old" folder automatically
·Direct dial to voicemail message type
Default message type to use when dialing direct to an extensions voicemail

3.10 System Preferences

3.10.1 Firewall Rules

Figure 3.10.1 Firewall Rules

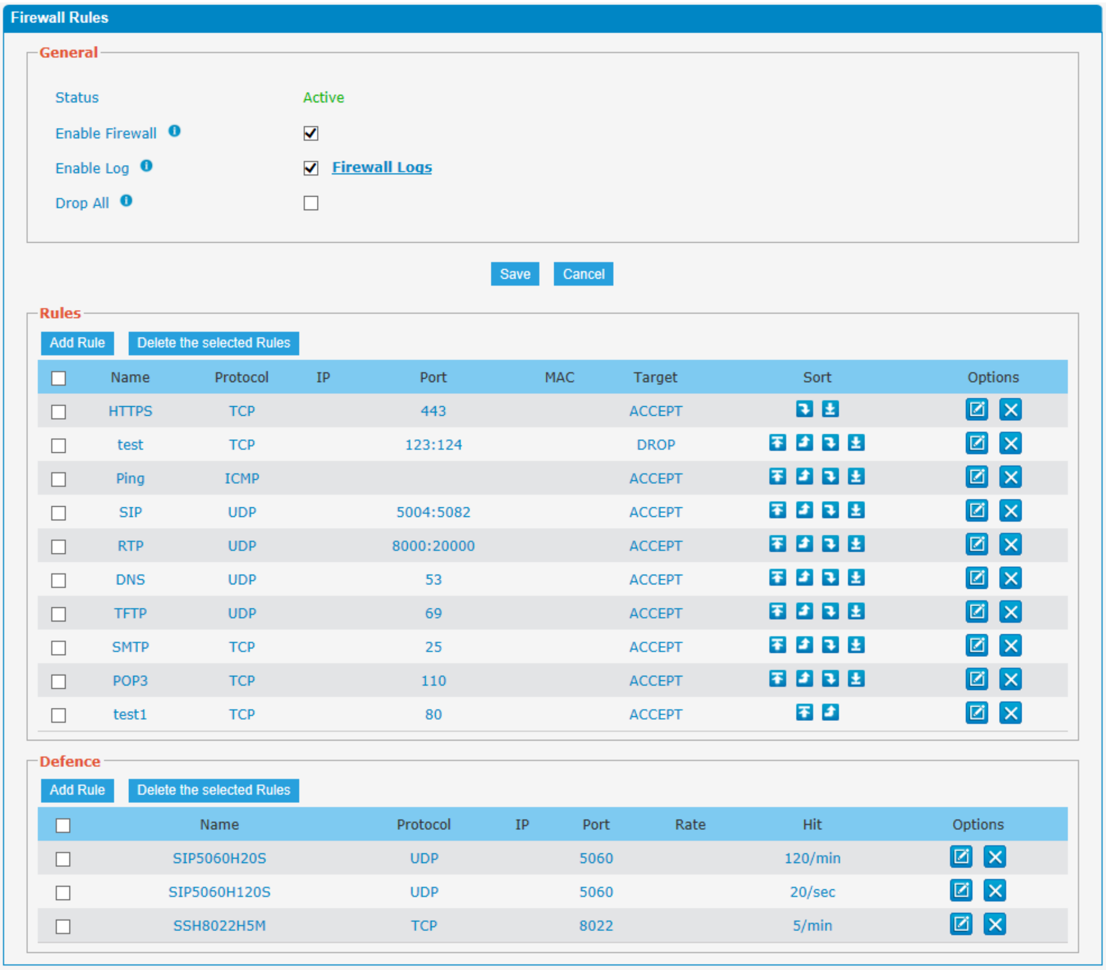


Figure 3.10.1a Firewall Rules Edit/Add

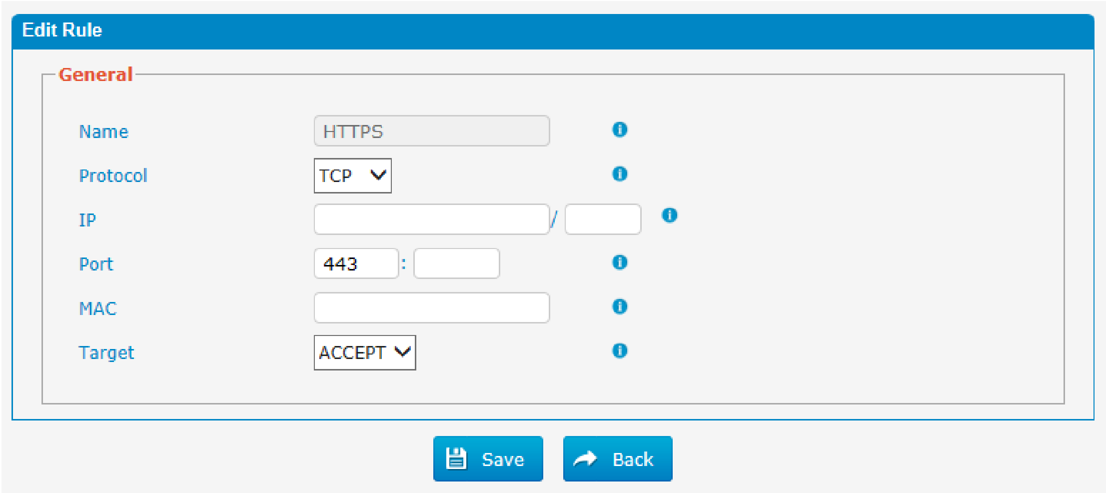
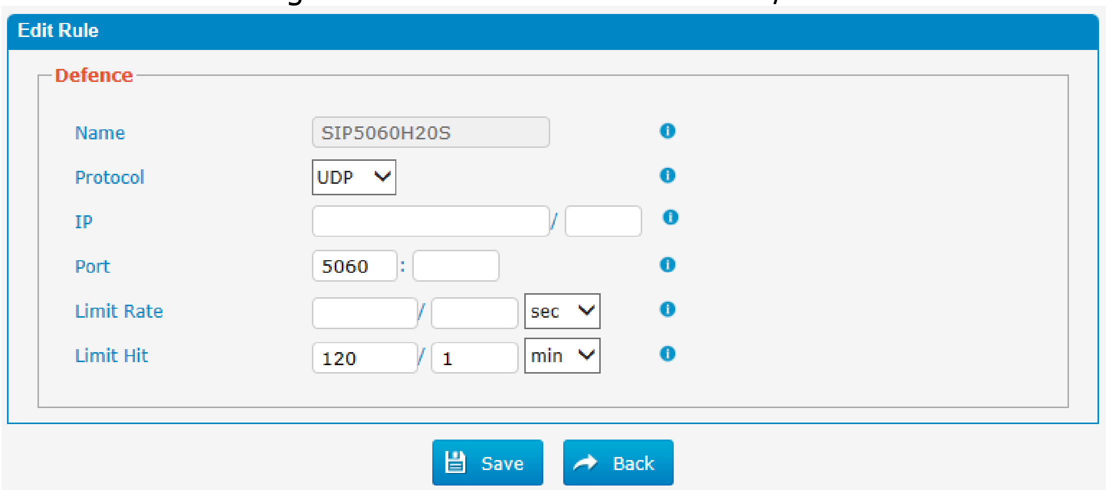


Figure 3.10.1b Firewall Defence Edit/Add



3.10.2 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 3.10.2

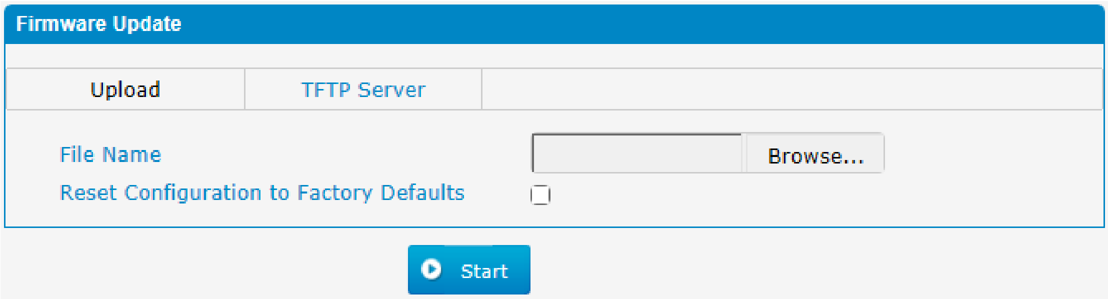


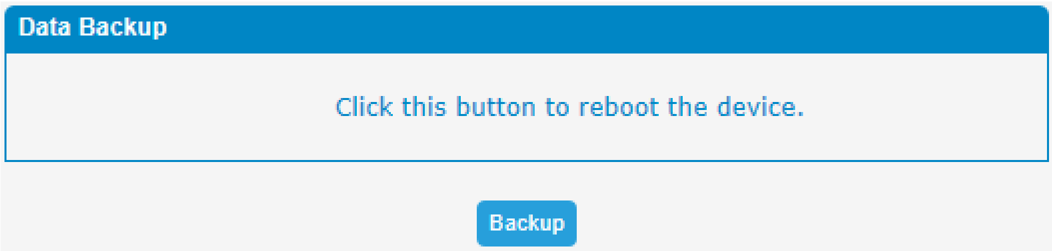
Table 3.10.2

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

3.10.3 Data backup

We can backup up the configurations before reset MUC1004 to factory defaults

Figure 3.10.3



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.

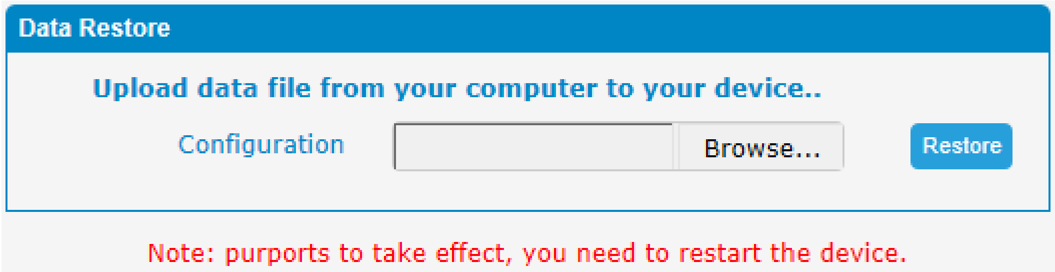
3.10.4 Data restore

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.

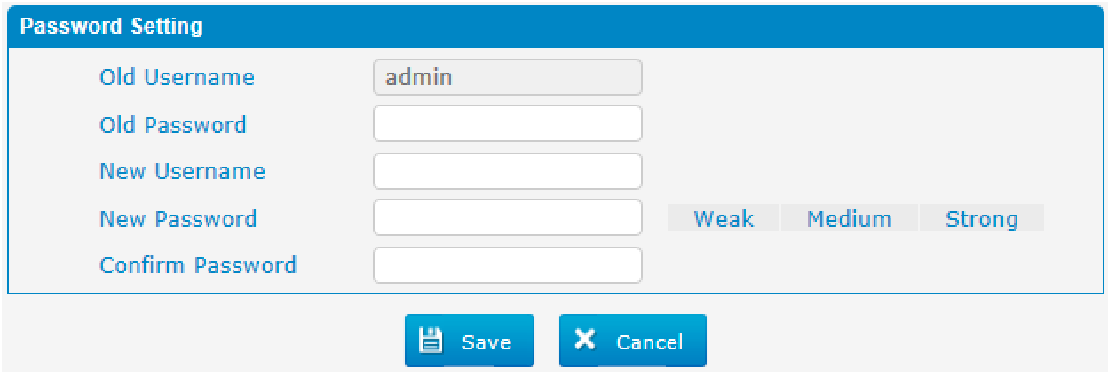
Figure 3.10.4



3.10.5 Password Setting

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

Figure 3.10.5 Password Setting



3.10.6 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 3.10.6 Time & Date parameter

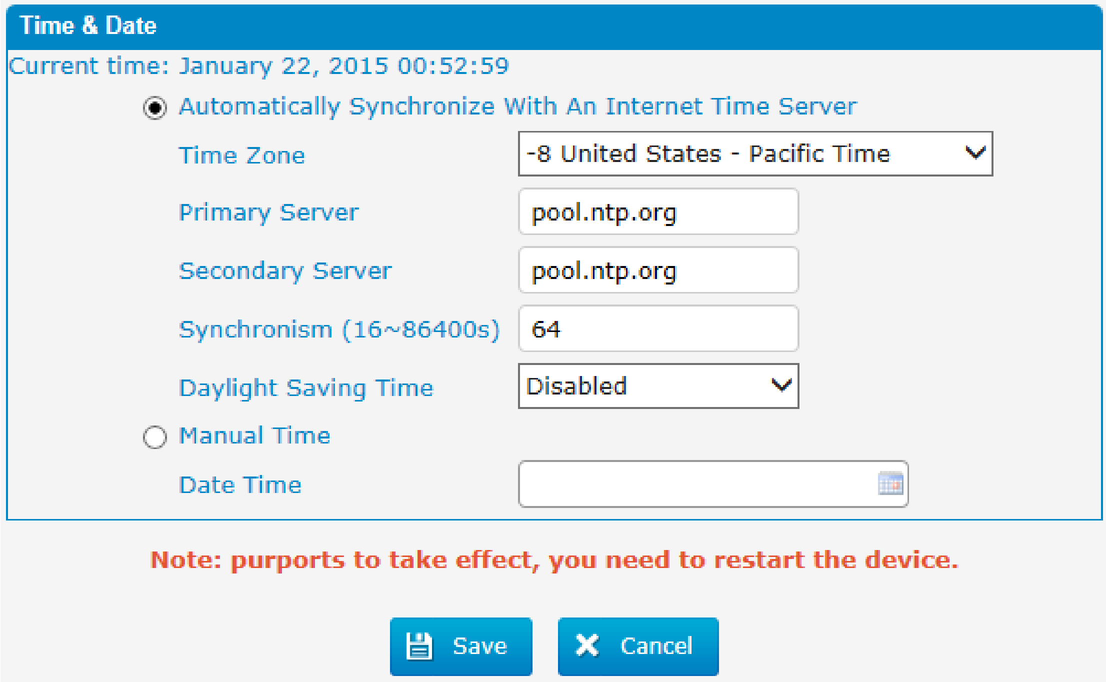


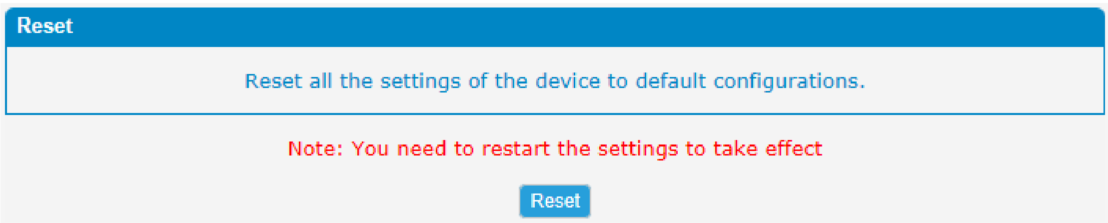
Table 3.10.6 Time & Date parameter

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

3.10.7 Factory reset

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

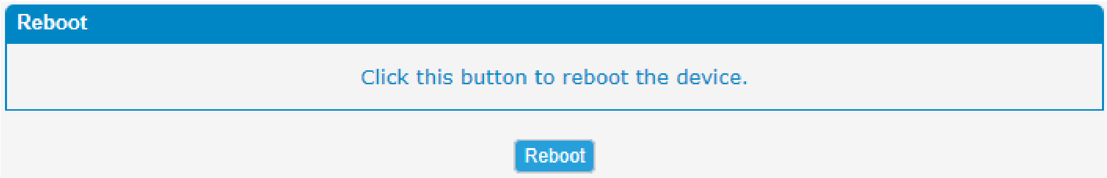
Figure 3.10.7 factory reset



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

3.10.8 Reboot

Figure 3.10.8 Reboot



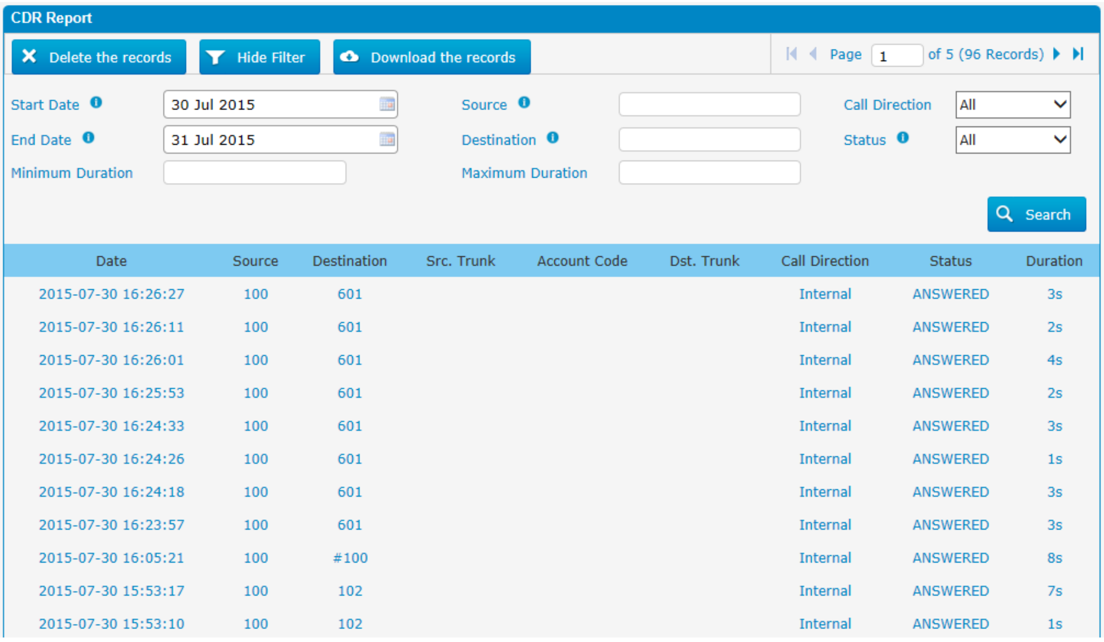
Warning: Rebooting the system will terminate all active calls!

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



Date	Source	Destination	Src. Trunk	Account Code	Dst. Trunk	Call Direction	Status	Duration
2015-07-30 16:26:27	100	601				Internal	ANSWERED	3s
2015-07-30 16:26:11	100	601				Internal	ANSWERED	2s
2015-07-30 16:26:01	100	601				Internal	ANSWERED	4s
2015-07-30 16:25:53	100	601				Internal	ANSWERED	2s
2015-07-30 16:24:33	100	601				Internal	ANSWERED	3s
2015-07-30 16:24:26	100	601				Internal	ANSWERED	1s
2015-07-30 16:24:18	100	601				Internal	ANSWERED	3s
2015-07-30 16:23:57	100	601				Internal	ANSWERED	3s
2015-07-30 16:05:21	100	#100				Internal	ANSWERED	8s
2015-07-30 15:53:17	100	102				Internal	ANSWERED	7s
2015-07-30 15:53:10	100	102				Internal	ANSWERED	1s

Table 3.11.1 CDR Report

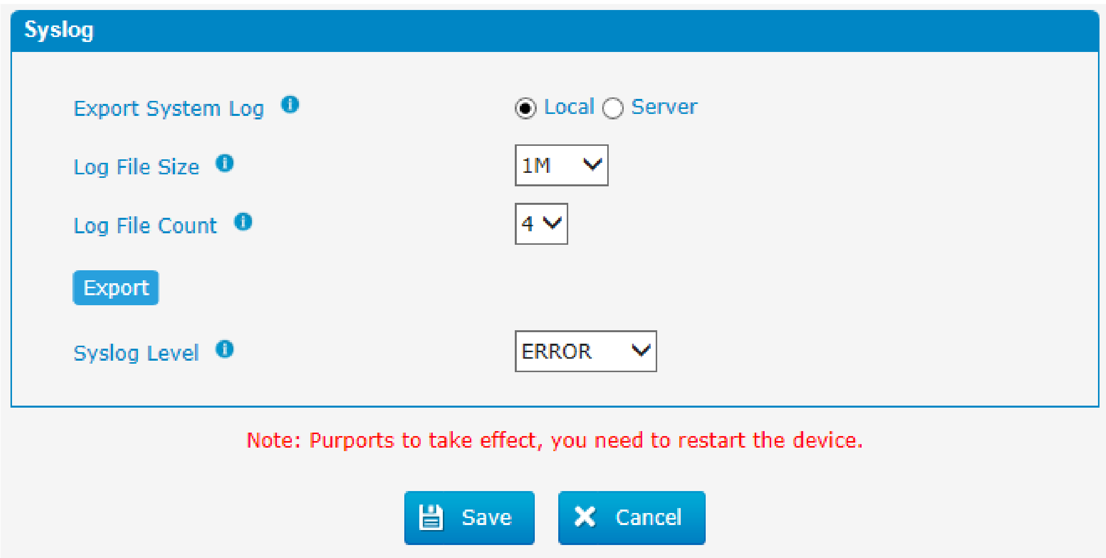
Parameters	Description
Date	start and end time of calls
Source	Call number
Destination	Called number
Src channel	Source channel
Dst channel	Destination channel

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

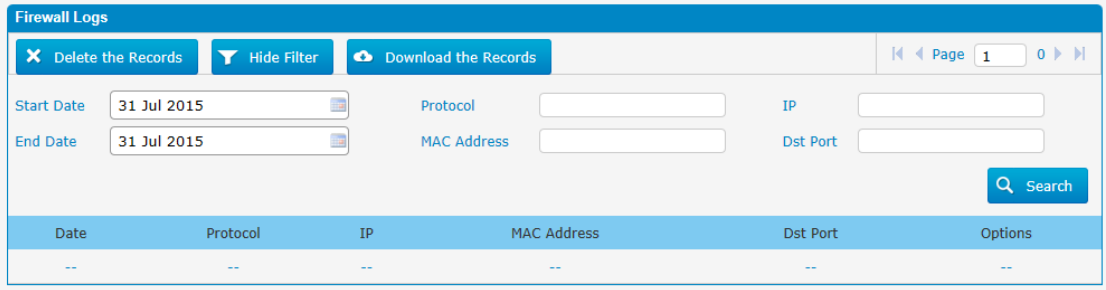
Syslog is a standard for network device data logging. It allows separation of the software that generates messages from the system that stores them and the software that reports and analyzes them. It also provides devices which would otherwise be unable to communicate a means to notify administrators 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



3.11.3 Firewall logs

Figure 3.11.3 Firewall logs



3.12 System tools

3.12.1 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 3.12.1

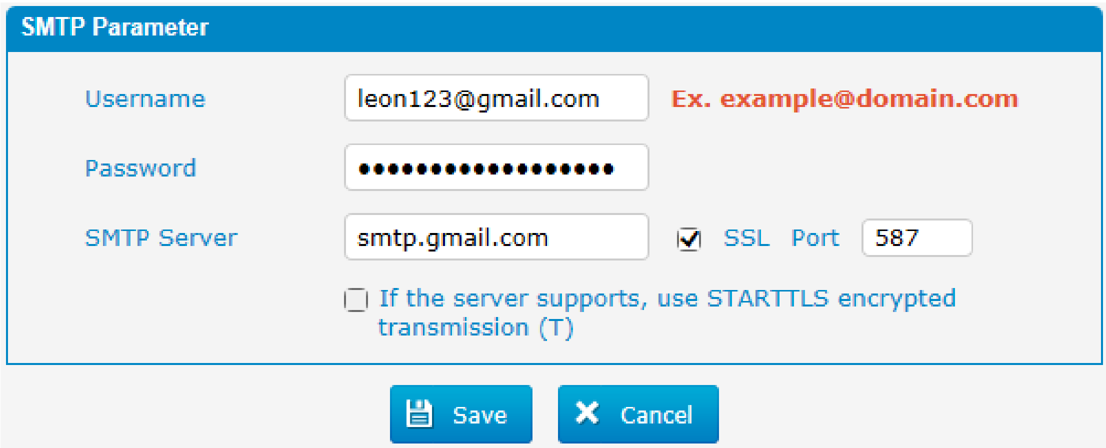
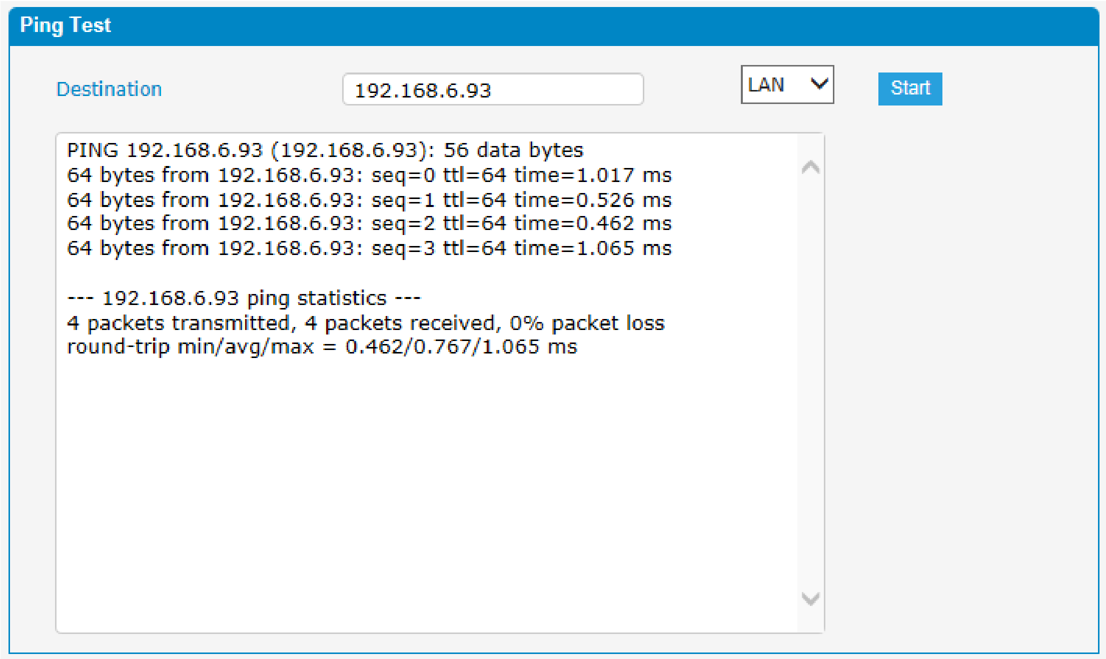


Table 3.12.1

Parameters	**Description**
Username	The E-mail Address that MUC1004 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 MUC1004 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.

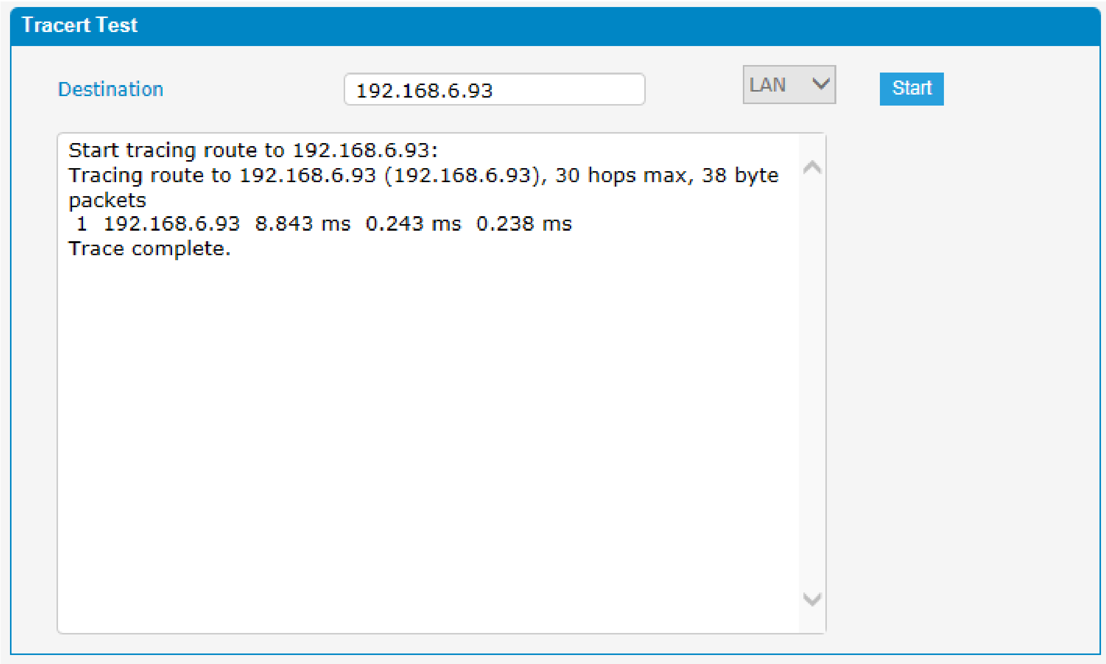
3.12.2 Ping

Figure 3.12.2 Ping



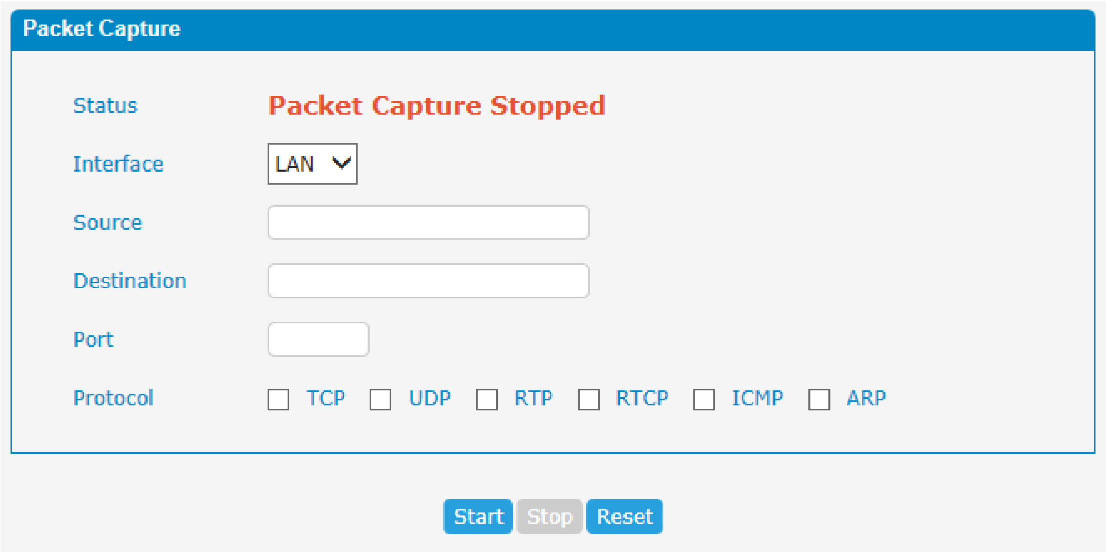
3.12.3 Tracert

Figure 3.12.3 Tracert



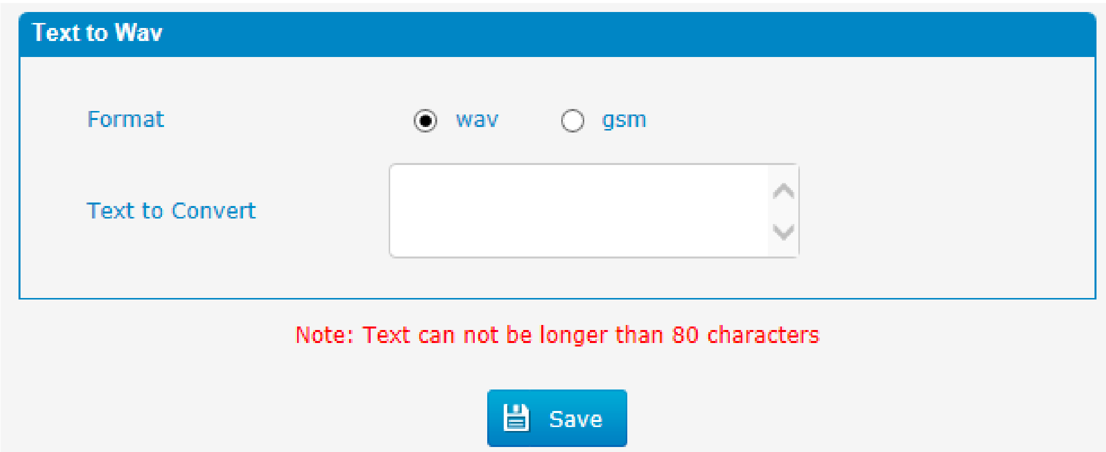
3.12.4 Packet Capture

Figure 3.12.4 Packet Capture



3.12.5 Text to wav

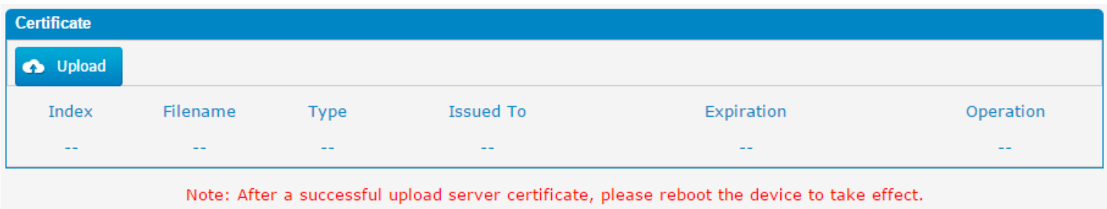
Figure 3.12.5 Text to wav



3.12.6 Certificates

MUC1004 can support TLS trunk. Before you register TLS trunk to MUC1004, you should upload certificates first.

Figure 3.12.6 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 MUC1004. If IPPBX enables "TLS Verify server", you should also upload this certificate on IPPBX.