NEXO 450/8100/16300

IP PBX

IP PBX Administrator Manual

Version 1.2

Table of Contents

1. Introduction

1.1 Overview

NEXO Series PBX—IP PBX for Small Business/Home Office

NEXO 450/8100/16300 IP PBX 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.

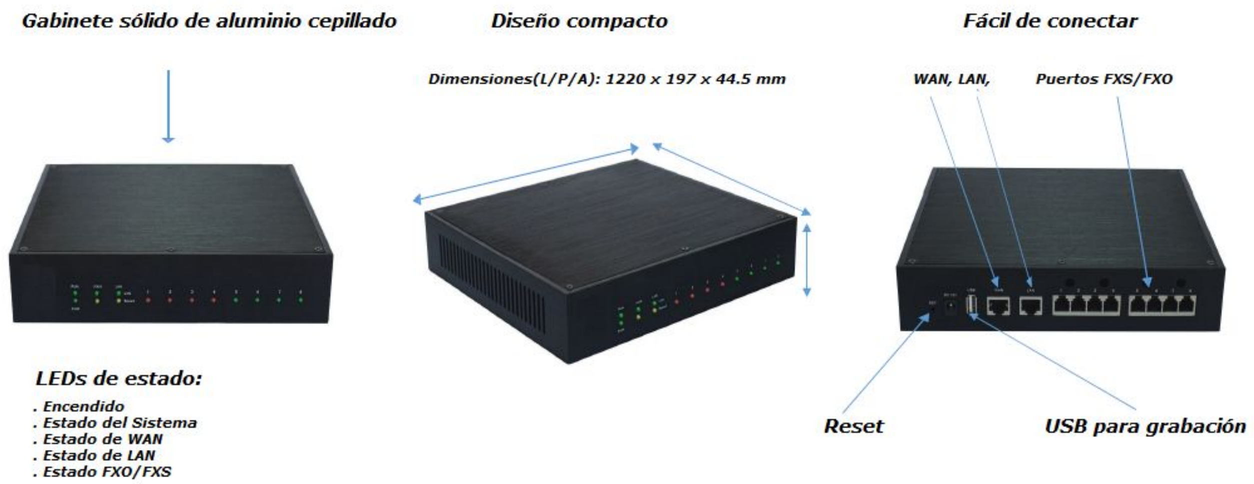
1.2 Product Features

● 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
●T.30,T.38 Faxes	●Alert Settings
●IP Blacklist	●AMI Settings
●Extension CDR	

1.3 Product Appearance

IP PBX NEXO 450

IP PBX NEXO 8100



IP PBX NEXO 16300

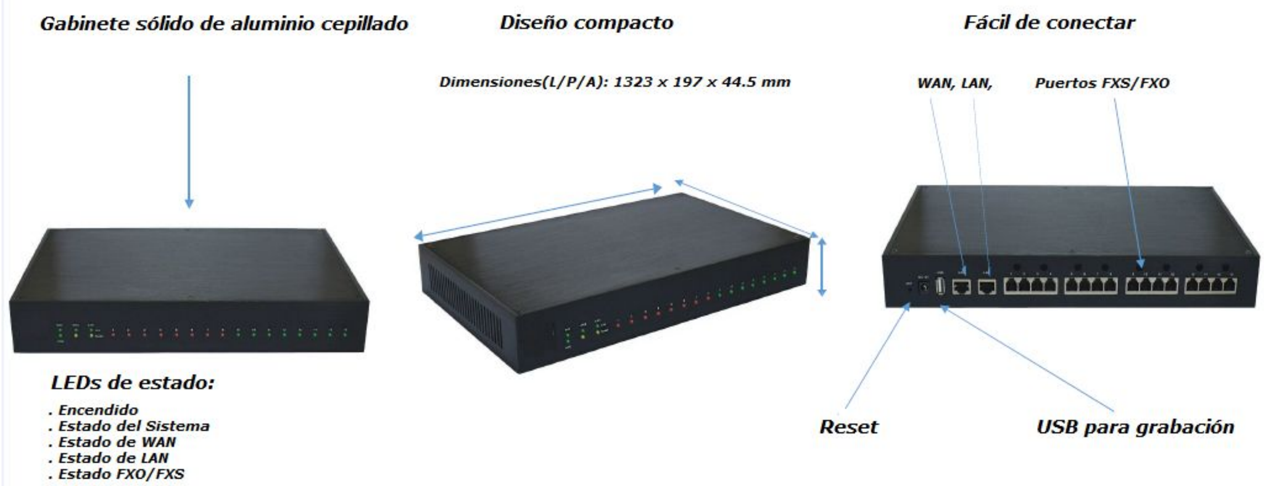


Table 1-3-1 Description of Front view

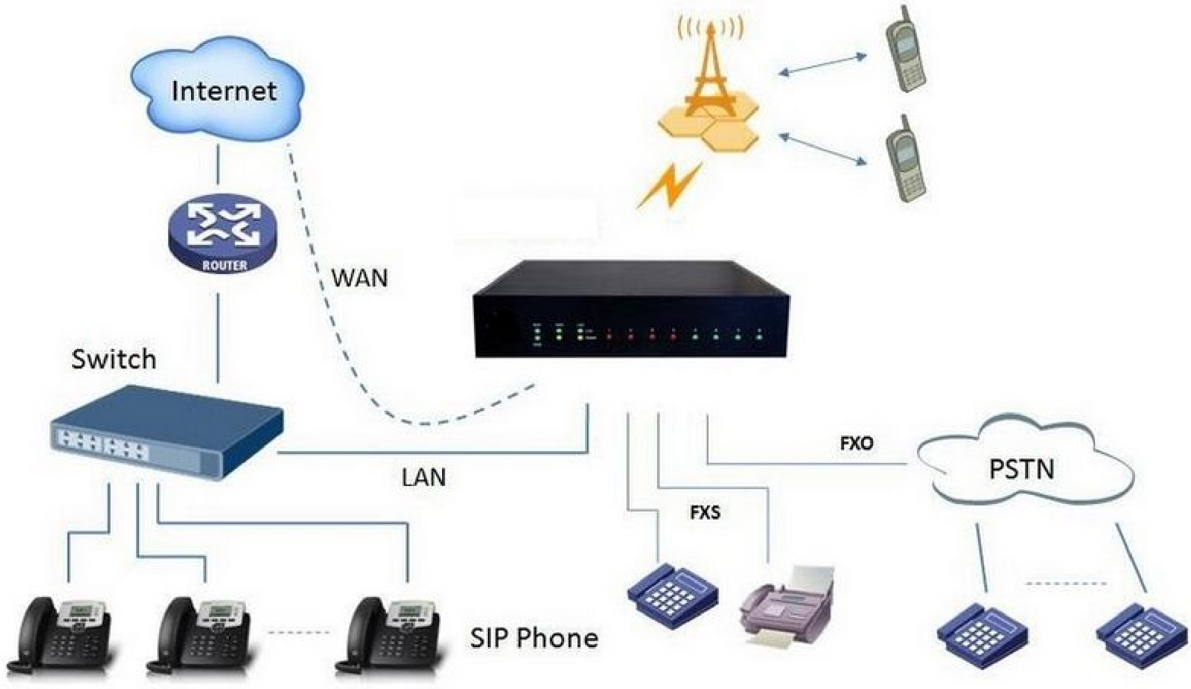
Index	Indicators	Description
1	RUN	On: Starting Off: Abnormal Blinking every 0.5s: Normal status
2	PWR	On: Power on Off: Power off
3	WAN,LAN	Green LED: indicates the Internet interface is in Link . Yellow LED: ON is indicates 100MBps Ethernet port.
4	1~4,(5~8),(9~16)	Red LED stands for FXO port Orange 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.

Table 1-3-2 Description of Rear view

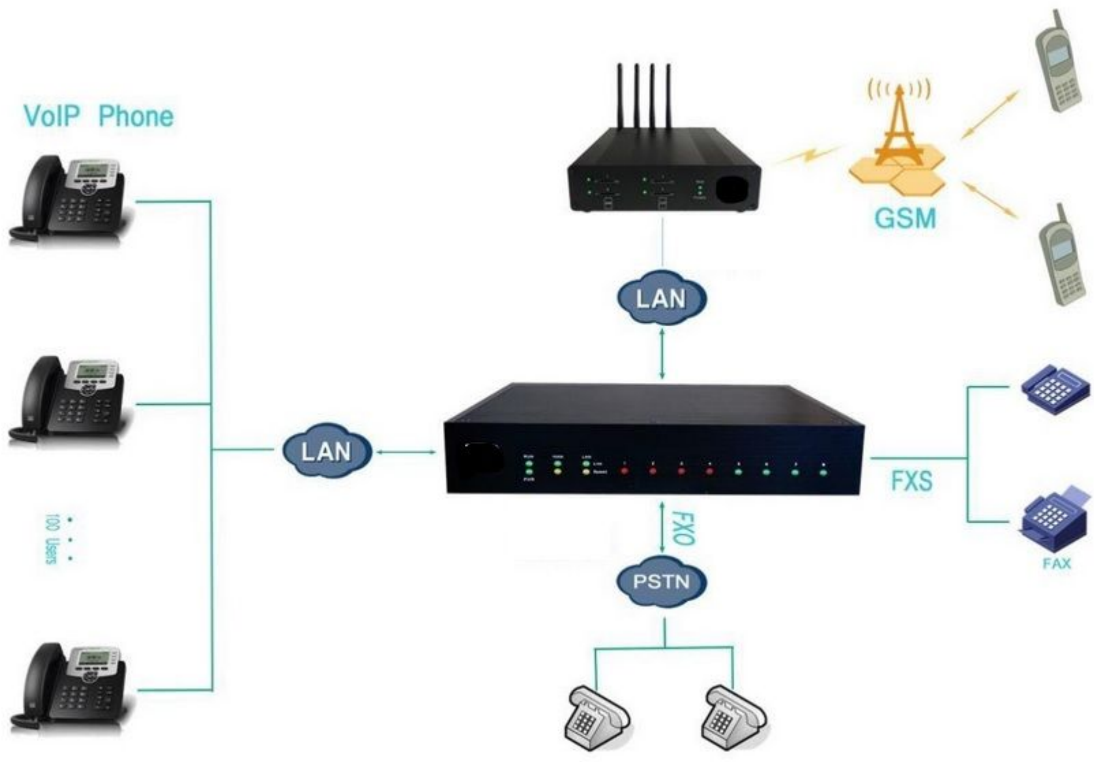
Index	Interface	Description
1	RST	Reset button to restore default IP and password or restore factory setting. Hold RST button 8 seconds, RUN LED being ON during this time
2	DC 12V	Power connector of DC power. Input: DC12V 3A/DC12
		1A (NEXO 450 only)
3	USB	For the storage of call recording files
4	WAN,LAN	NEXO 8100 provides two 10/100 adaptive RJ45 Ethernet ports, marked as LAN and WAN. **-LAN port** :LAN port is for the connection to Local Area Network **-WAN port:**WAN port is the netword port for the connection to internet. It supports "DHCP server","PPPoE/dynamic DNS",and"static IP"for IP address assignment.
5	1~4,(5~8),(9~16)	**FXO port** (red light): For the connection of PSTN lines or FXS port of traditional PBX. NEXO 8100 users could make or receive calls via FXO port. FXS port (green light): For the connection of analog phones. BRI port(orange port): For the connection of ISDN BRI lines. NEXO 8100 users could make or receive calls via BRI port. Note:The sequence number of the port corresponds to that of the indicator lights in the front panel.

1.4 Scenario of Application

Application 1 - Figure 1.4.1



Application 2 - Figure 1.4.2



2. Installation Guide

2.1 Installation Notice

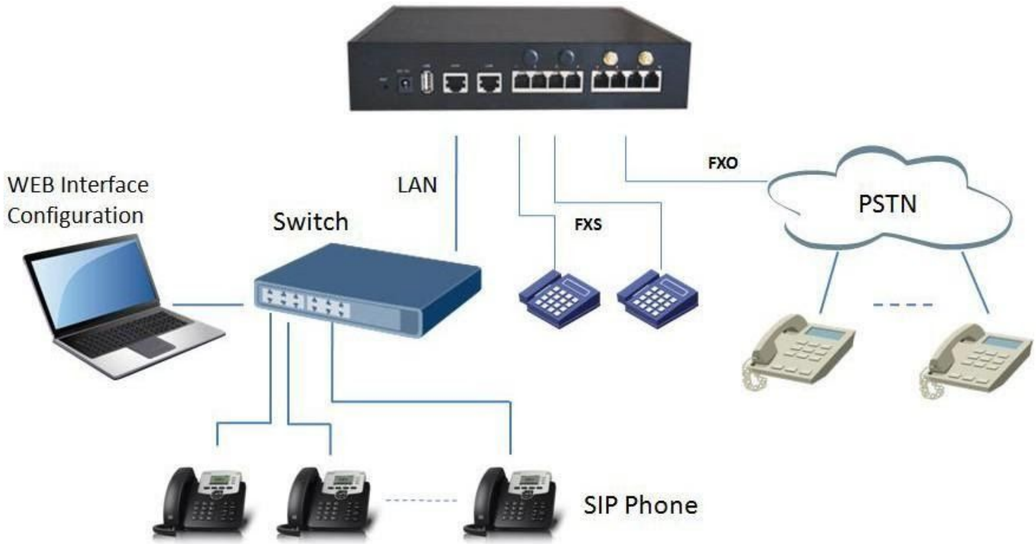
We use the NEXO 8100 device as an installation case as follows:

NEXO 8100 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.
NEXO 8100 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

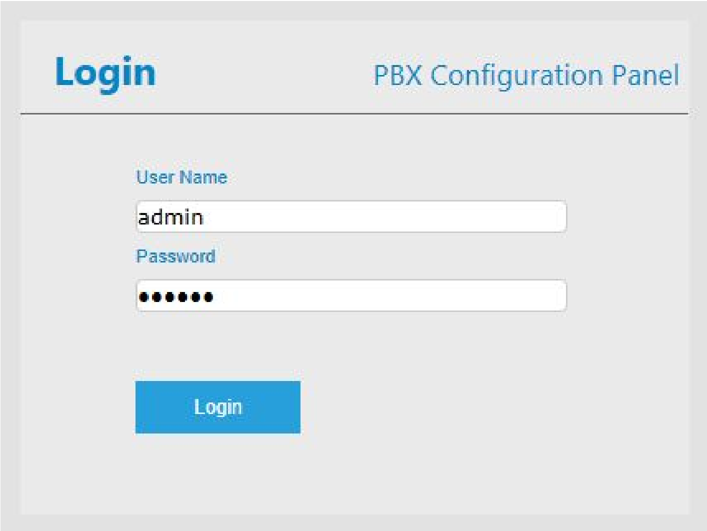
PBX IP PBX has the same web interface. This chapter describes web configuration of PBX. The PBX contains an embedded web server to set parameters by using the HTTP protocol. We are strongly recommend to access device with Google Chrome or Firefox Browser.

We use the NEXO 8100 device as a configuration case as follows:

3.1 Access NEXO 8100 unit

Enter IP address of NEXO 8100 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

PBX 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

Free Busy Hold Unavailable Ringing

Extension Status

Show Filter

Extension	Extension	Extension	Extension	Extension
100(SIP 192.168.6.33)	101(SIP 192.168.6.33)	102(SIP 192.168.6.33)	103(SIP 192.168.6.33)	104(SIP 192.168.6.33)
105(SIP)	106(SIP 192.168.6.33)	107(SIP)	108(SIP 192.168.6.33)	109(SIP)
110(SIP)	111(SIP)	112(IAX)	113(SIP)	114(SIP)
115(SIP)	116(SIP)	117(SIP)	118(SIP)	119(SIP)
120(SIP)	121(SIP)	122(SIP)	123(SIP)	124(SIP)
125(SIP)	126(SIP)	127(SIP)	128(SIP)	129(SIP)
130(SIP)	131(SIP)	132(SIP)	133(SIP)	134(SIP)
135(SIP)	136(SIP)	137(SIP)	138(SIP)	139(SIP)
140(SIP)	141(SIP)	142(SIP)	143(SIP)	144(SIP)
145(SIP)	146(SIP)	147(SIP)	148(SIP)	149(SIP)
603(FXS)	604(FXS)	--	--	--

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

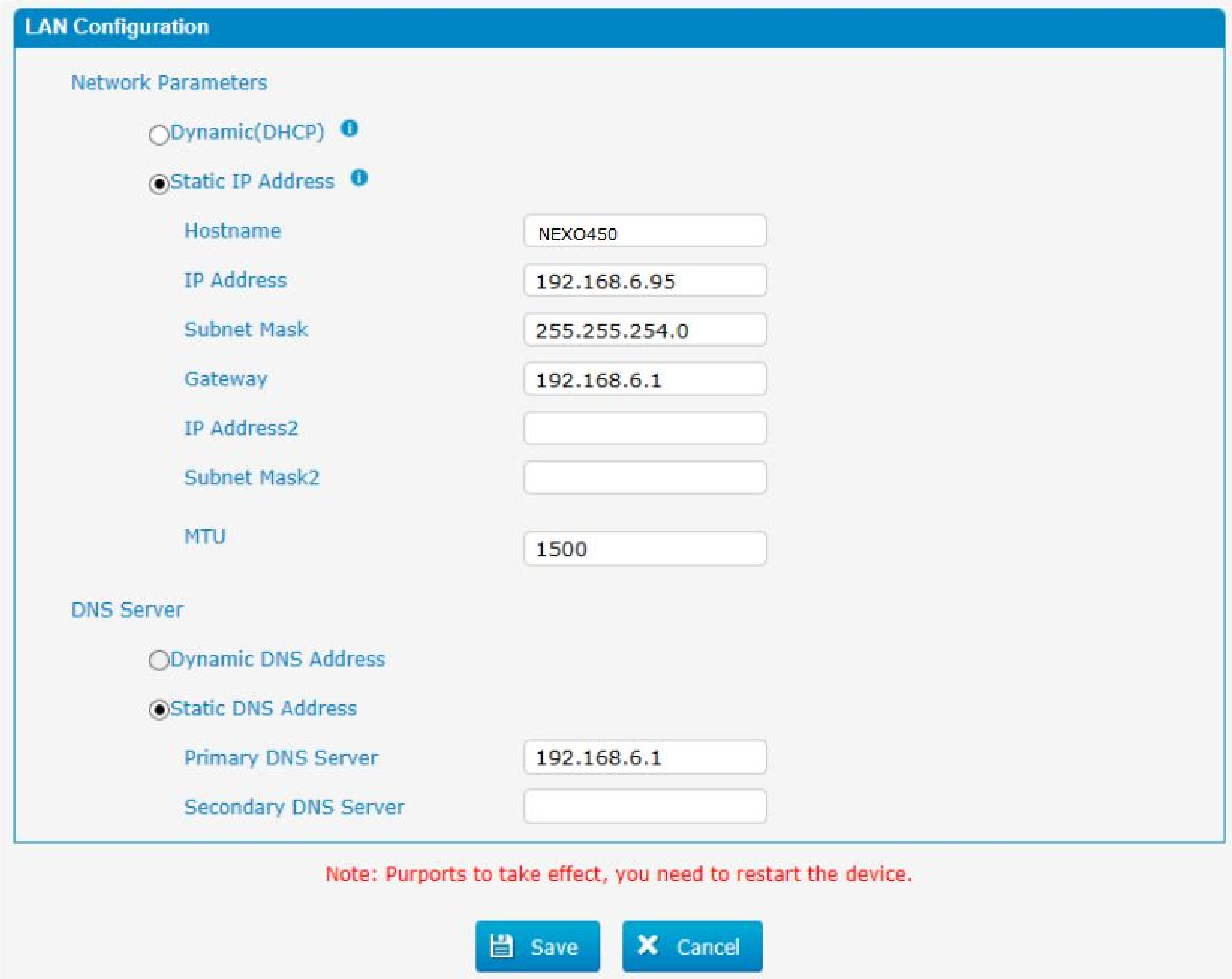


Table 3.4.1 Description of Local network

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 PBX
IP Address	Set the IP Address for PBX, It is recommended to configure a static IP address for PBX
Subnet Mask	Set the subnet mask for PBX
Gateway	Set the gateway for PBX
IP Address 2	Set the second IP Address for PBX

Subnet Mask2	Set the second subnet mask for PBX
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 PBX.
Secondary DNS Server	Set the Secondary DNS Server for PBX.

Figure 3.4.1.2 WAN Configuration

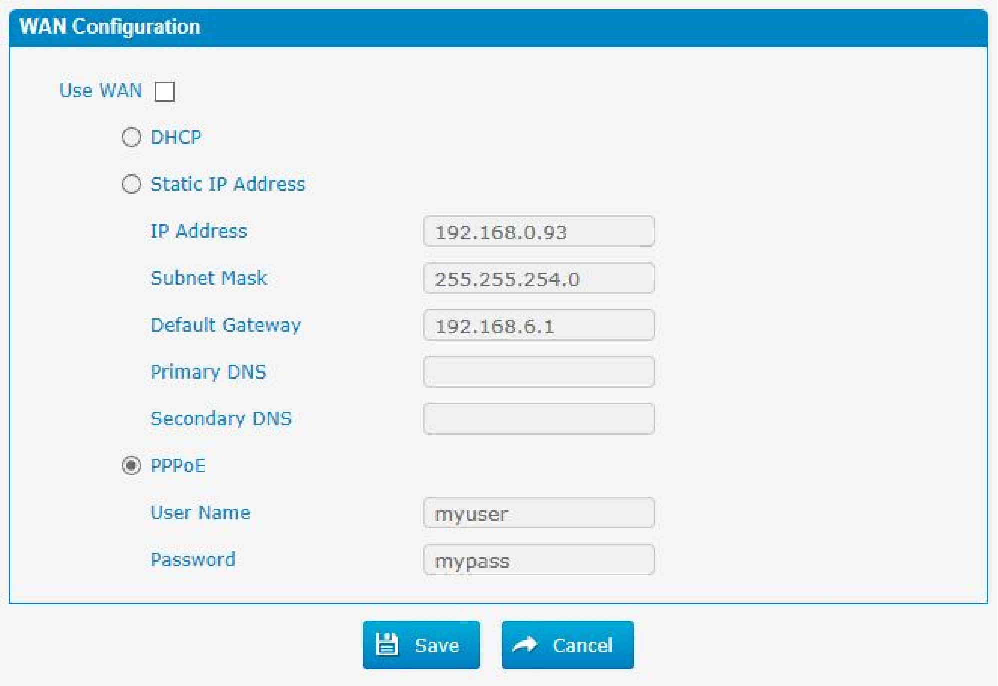


Table 3.4.1.2 Description of WAN Configuration

Parameter	**Description**
Use WAN	Enalbe use wan
Dynamic (DHCP)	Enable the device obtain IP Address automatically
Static IP Address	Configure the "IP Address", "Subnet Mask" and "Default Gateway" by manual
IP Address	Set the IP Address for PBX, It is recommended to configure a static IP address for PBX
Subnet Mask	Set the subnet mask for PBX
Default Gateway	Set the default gateway for PBX
Primary DNS	Set the primary DNS Server for PBX.
Secondary DNS	Set the Secondary DNS Server for PBX.
PPPoE	Use PPPoE to achieve IP address
User Name	PPPoE user name
Password	PPPoE password

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

Figure 3.4.2 VLAN Configuration

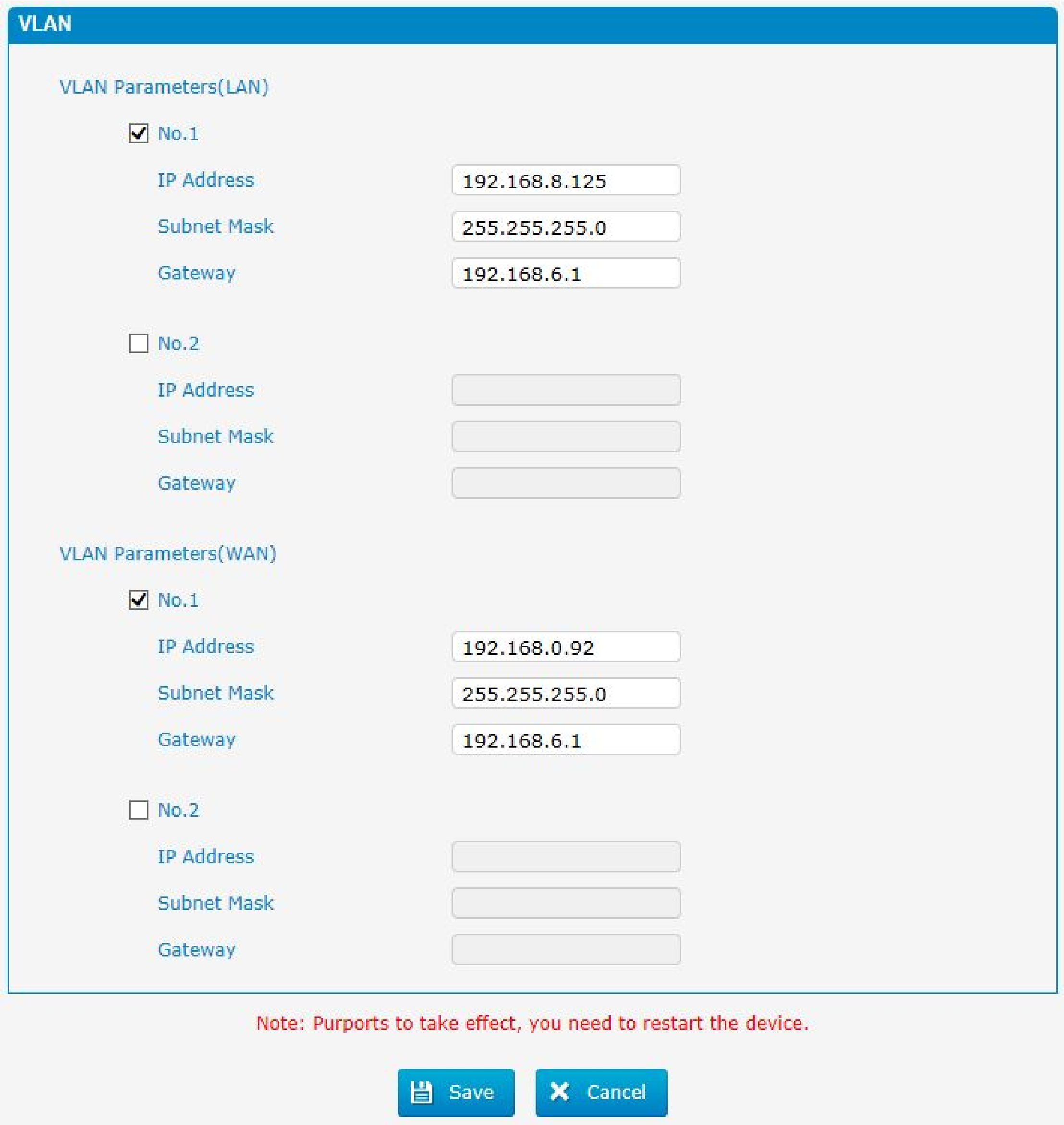


Table 3.4.2 Description of VLAN Configuration

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

3.4.3 ARP Configuration

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 Click "Dynamic ARP" to check ARP buffer

Figure 3.4.3a Dynamic ARP

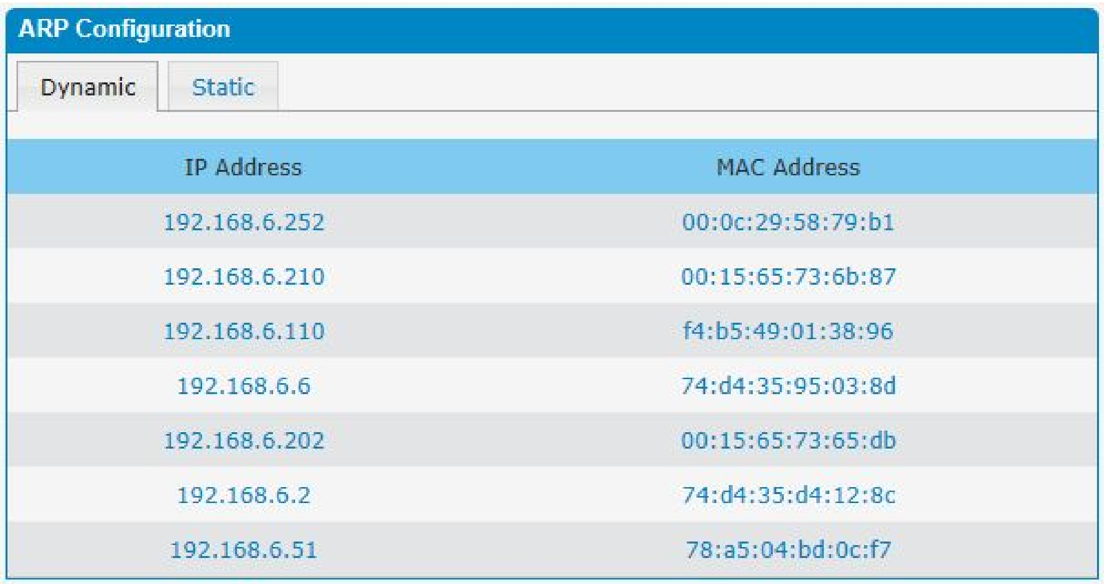
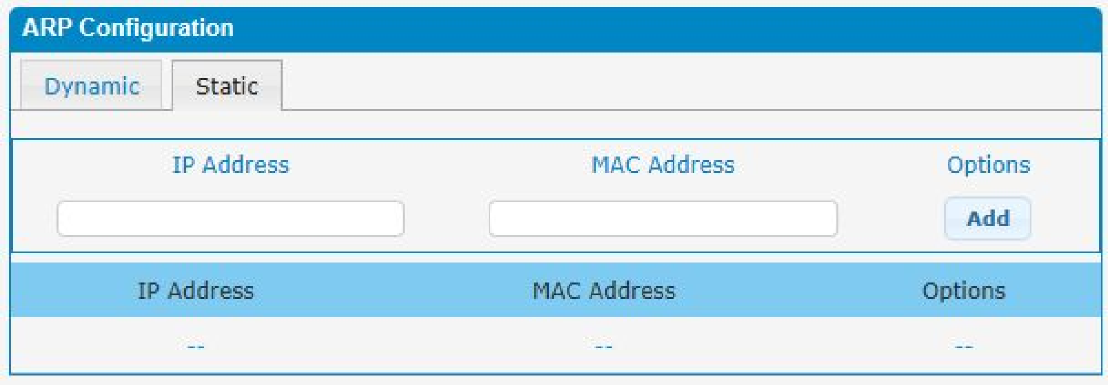


Figure 3.4.3 Add ARP



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. PBX supports OpenVPN.

Figure 3.4.4 VPN Configuration

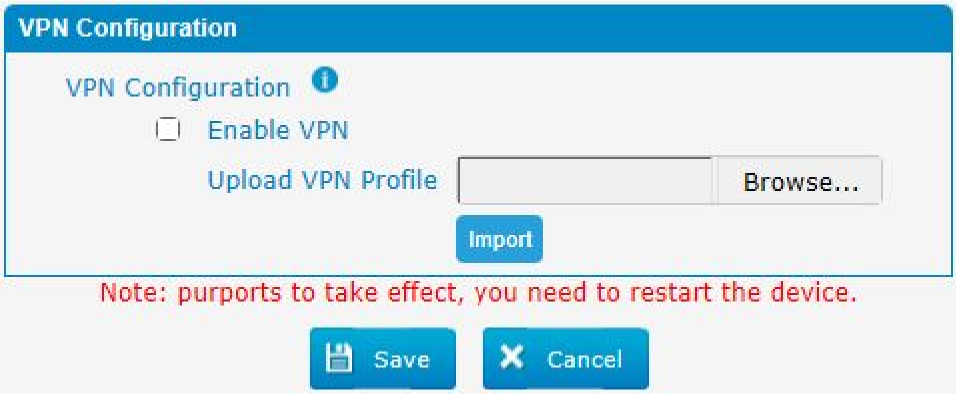


Table 3.4.4 Description of VPN Parameter

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. PBX works as VPN client mode only.
3. Upload file *.tar with *.conf in it.

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

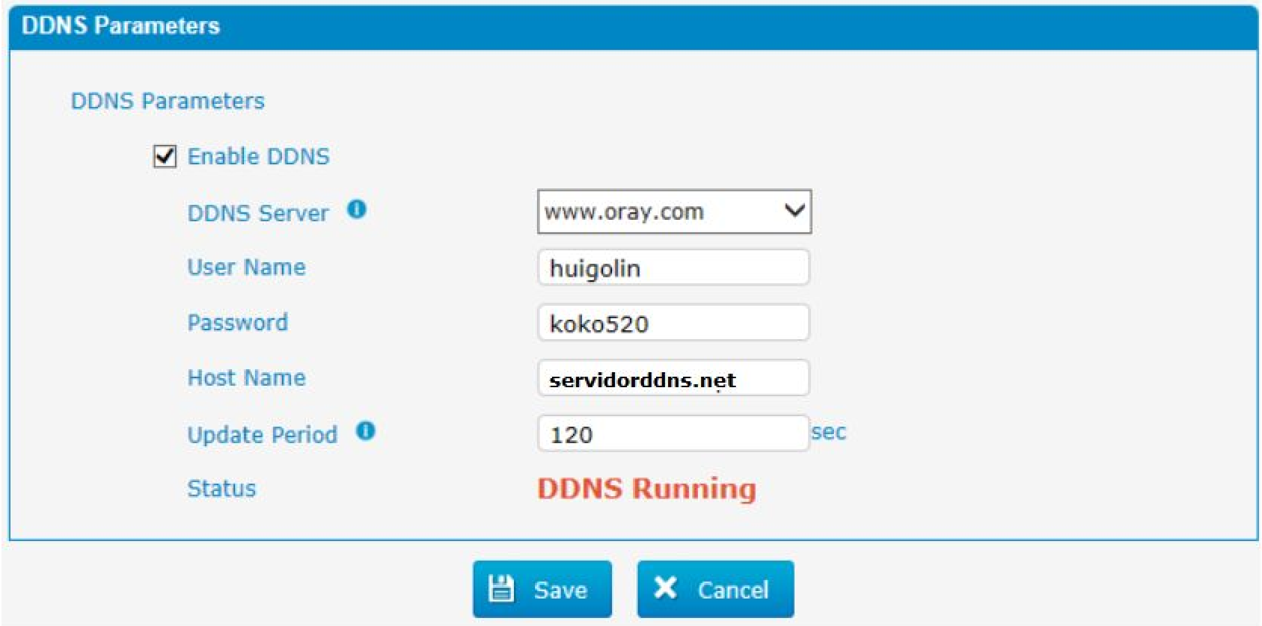


Table 3.4.5 Description of DDNS Server

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

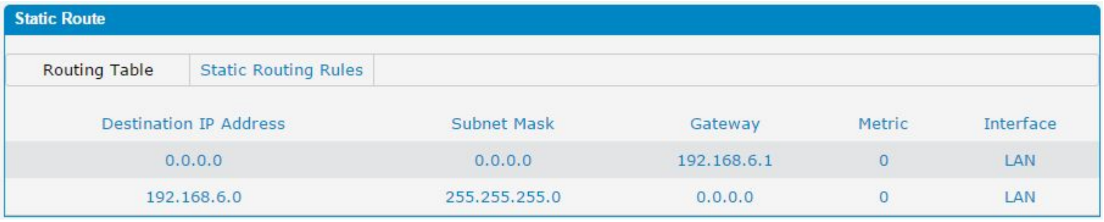
3.4.6 Static Route

PBX 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 PBX to force it to go out through different gateway when access to different internet. The default gateway priority of PBX from high to low is VPN/VLAN-> LAN port.

1) Route Table

The current route rules of PBX.

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

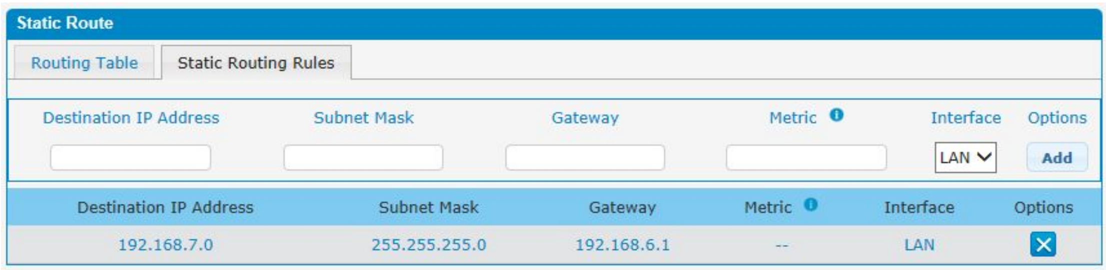


Table 3.4.6 Description of Static Routing

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

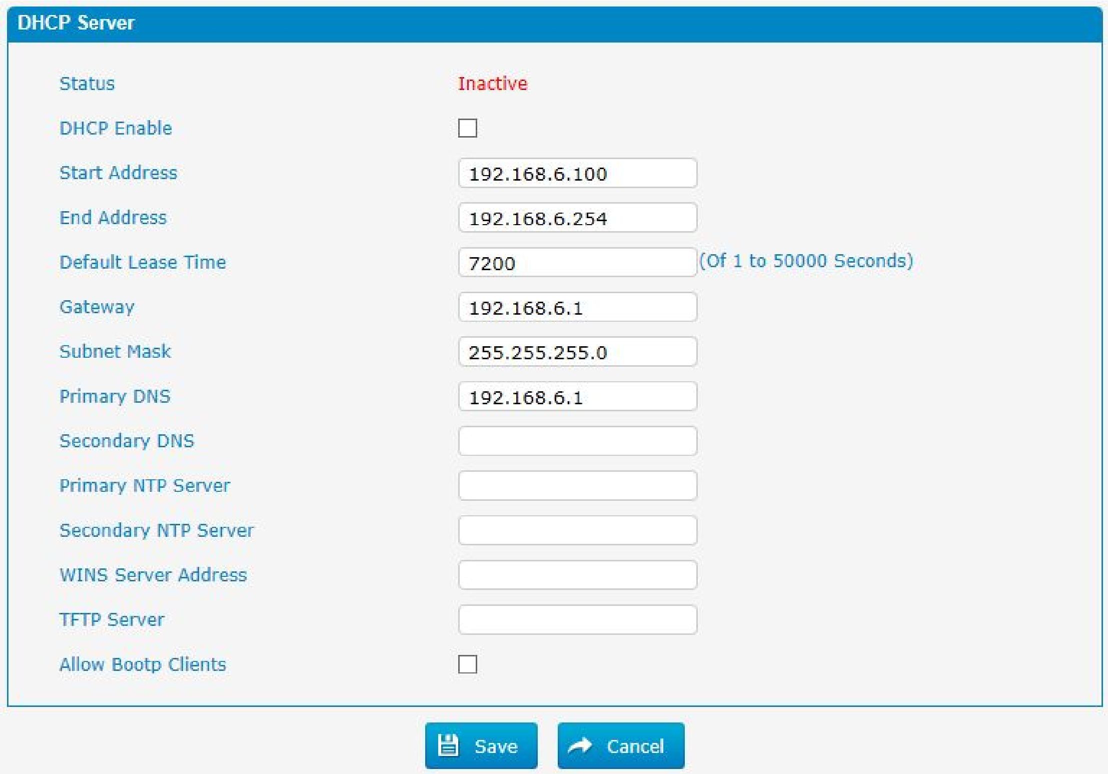


Table 3.4.7 Description of DHCP Server

Parameters	Description
Status	DHCP service status
DHCP Enable	Enable DHCP service
Start Address	Start IP of DHCP IP pool
End Address	End IP of DHCP IP pool
Default Lease Time	Default lease time
Gateway	Gateway address
Subnet Mask Address	Specify the destination network portion.
Primary DNS	Set the primary DNS Server for PBX.

Secondary DNS	Set the Secondary DNS Server for PBX.
Primary NTP Server	Set the primary NTP Server
Secondary NTP Server	Set the Secondary NTP Server
WINS Server Address	Set the WINS Server Address
TFTP Server Server	Set the TFTP Server
Allow Bootp Clients	Allow bootp clients

3.5 Trunks

3.5.1 Physical Trunks(PSTN and GSM 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

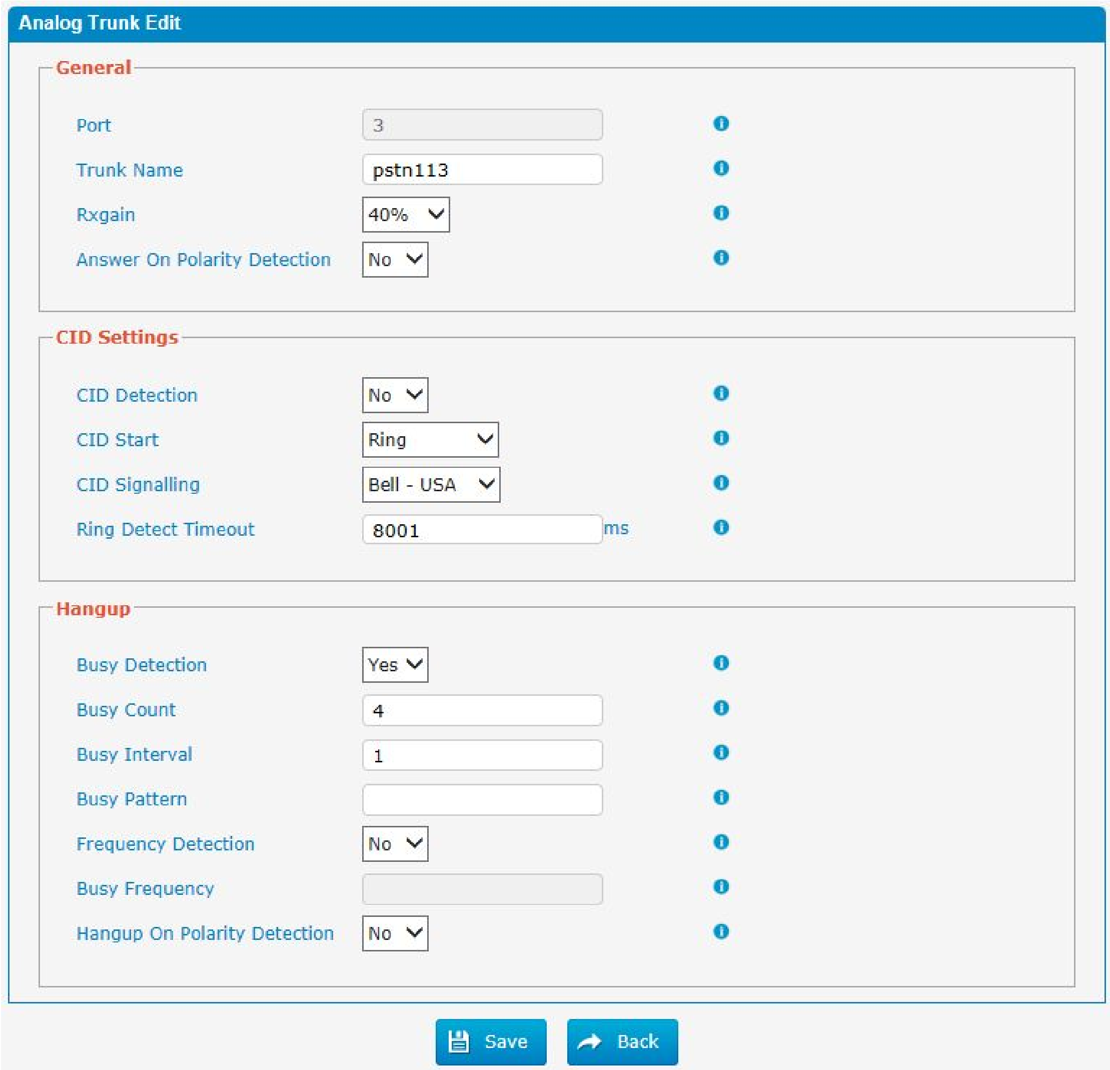


Table 3.5.1 Description of Analog Trunk

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.

Figure 3.5.1b GSM Trunks

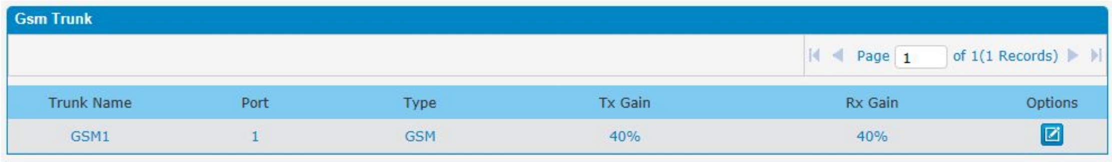


Figure 3.5.1c GSM Trunks Edit

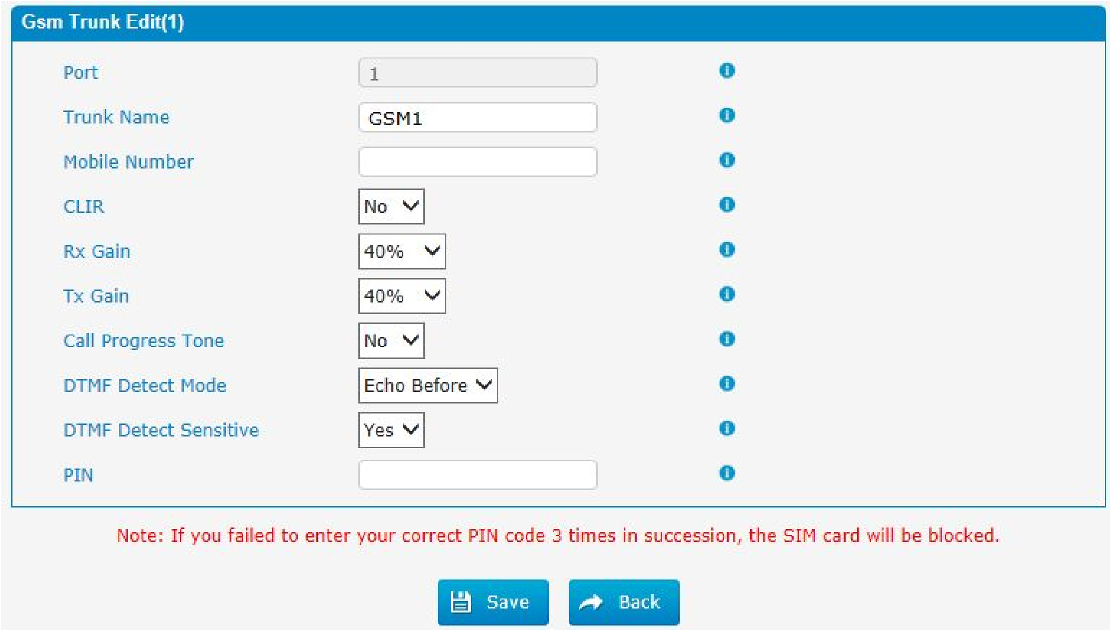


Table 3.5.1c Description of GSM Trunk

Parameters	**Description**
Port	A port for this trunk.
Trunk Name	A name for this trunk.
Mobile Number	Mobile number for this trunk.
CLIR	Calling Line Identification Restriction.
Rx Gain	The receive volume.
Tx Gain	The transfer volume.
Call Progress Tone	A ringback for this trunk.
DTMF Detect Mode	Set default dtmfmode for detect DTMF. Default: Echo Before Echo Before: Detect DTMF before echocan. Echo After: Detect DTMF after echocan.
DTMF Detect Sensitive	DTMF detect sensitive.
PIN	The PIN is normally associated with the SIM card.

3.5.2 IP Trunk (Peer to Peer Mode)

Figure 3.5.2 IP Trunk

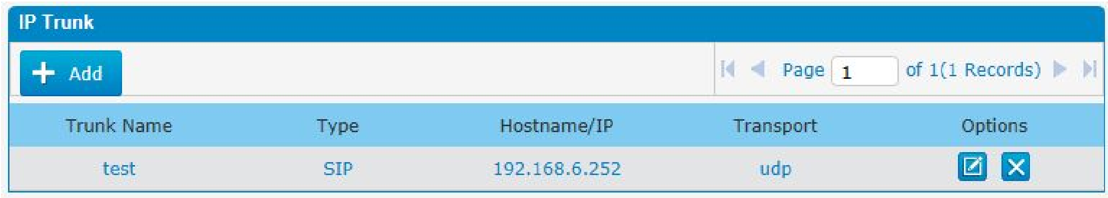


Figure 3.5.2a Add IP Trunk

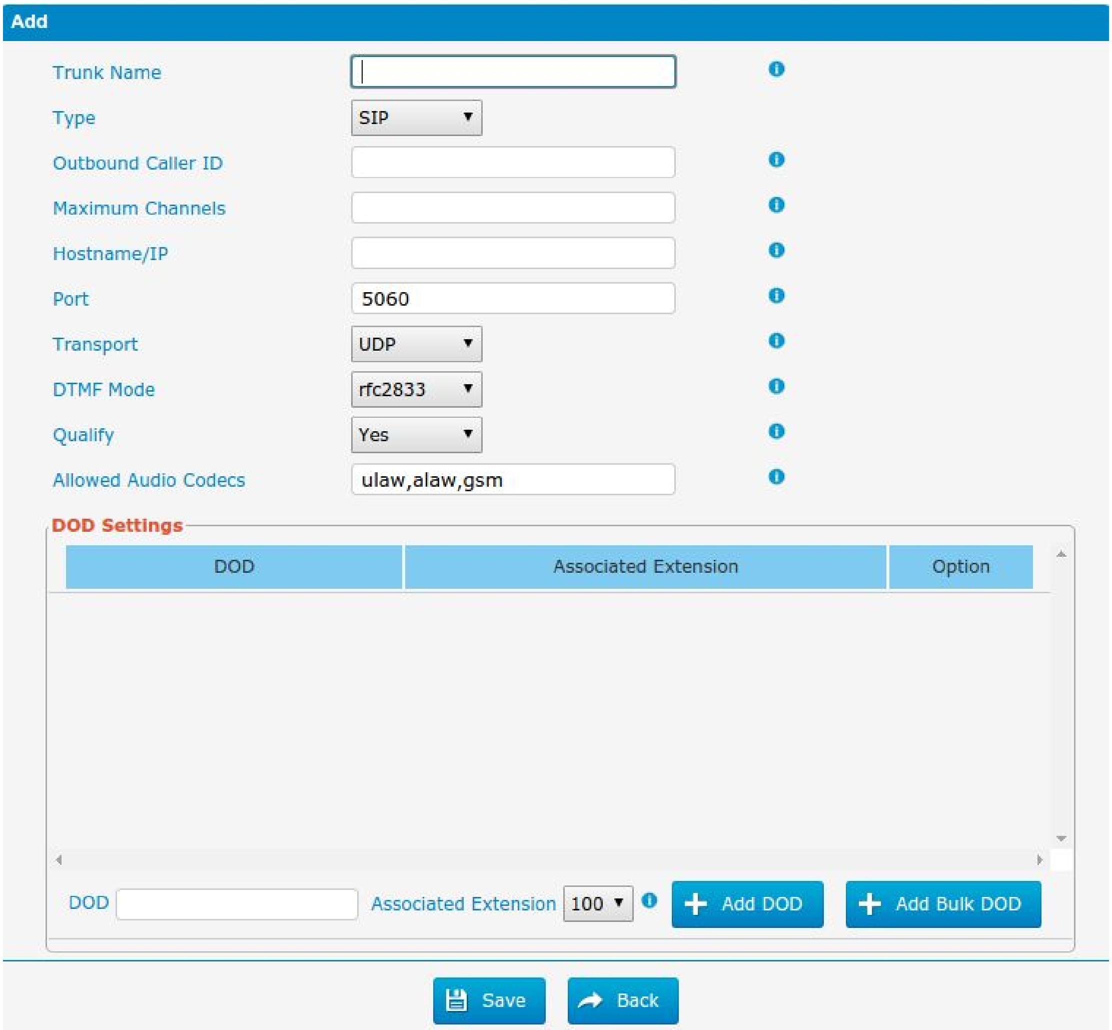


Figure 3.5.2b Add Bulk Dod

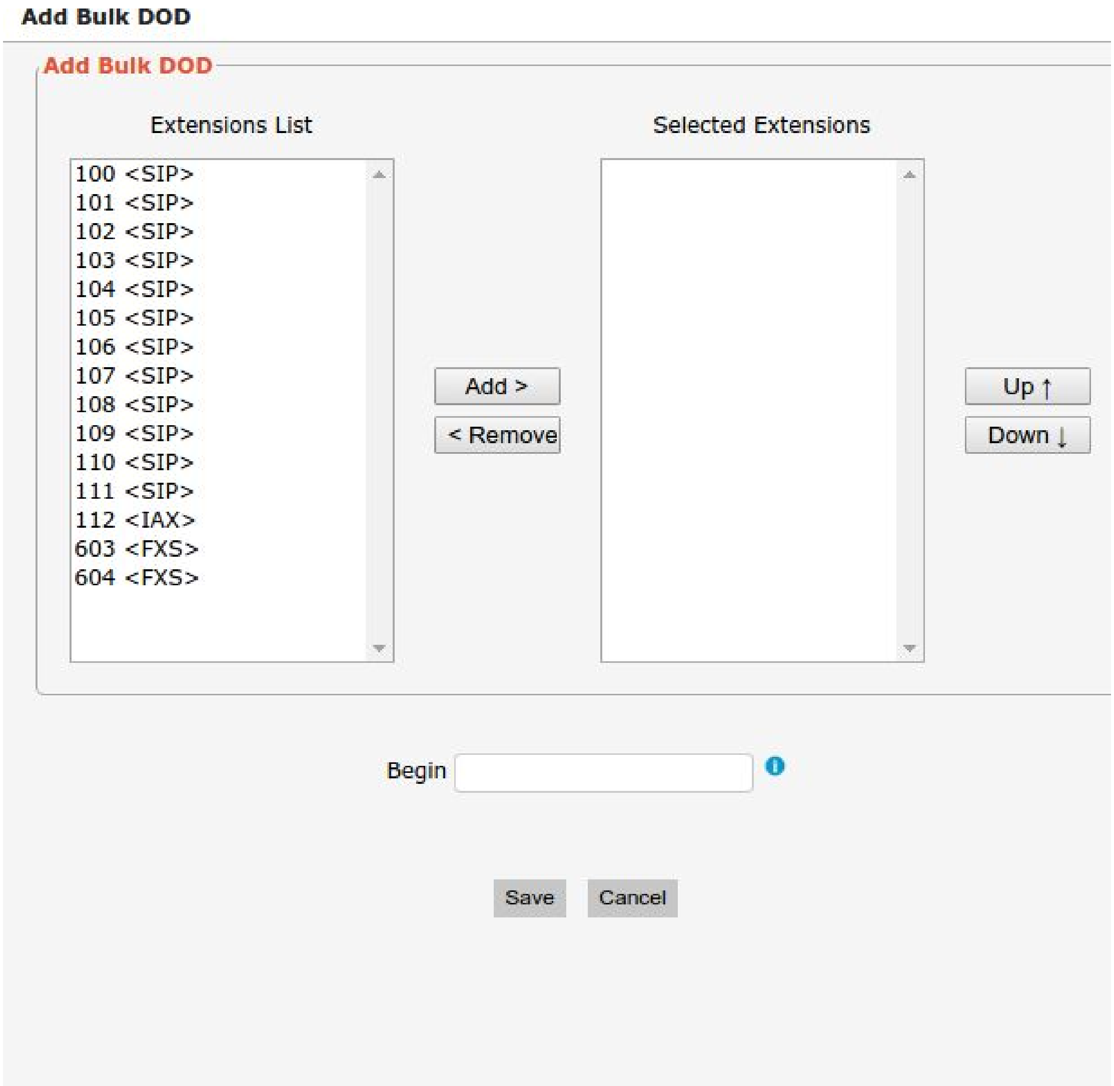


Table 3.5.2 Description of IP Trunk

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, PBX will ignore the reachability and the status of this account will be unmonitored.
Allow codecs	ulaw,alaw,gsm
DOD Settintings	Add dod number to associated extension.
Add Bulk DOD	Add bulk dod number to associated extensions which begin with Begin number

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

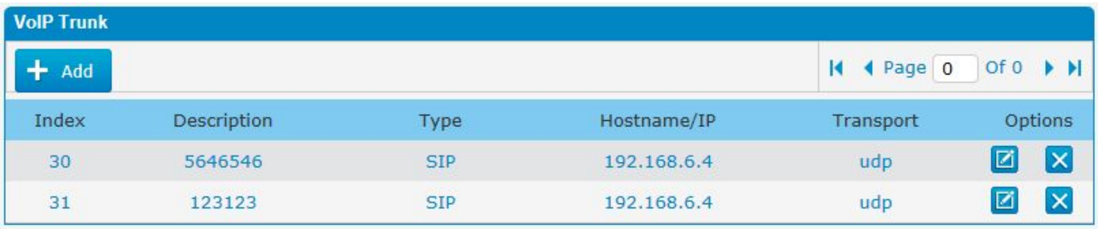


Figure 3.5.3a Add VoIP Trunk

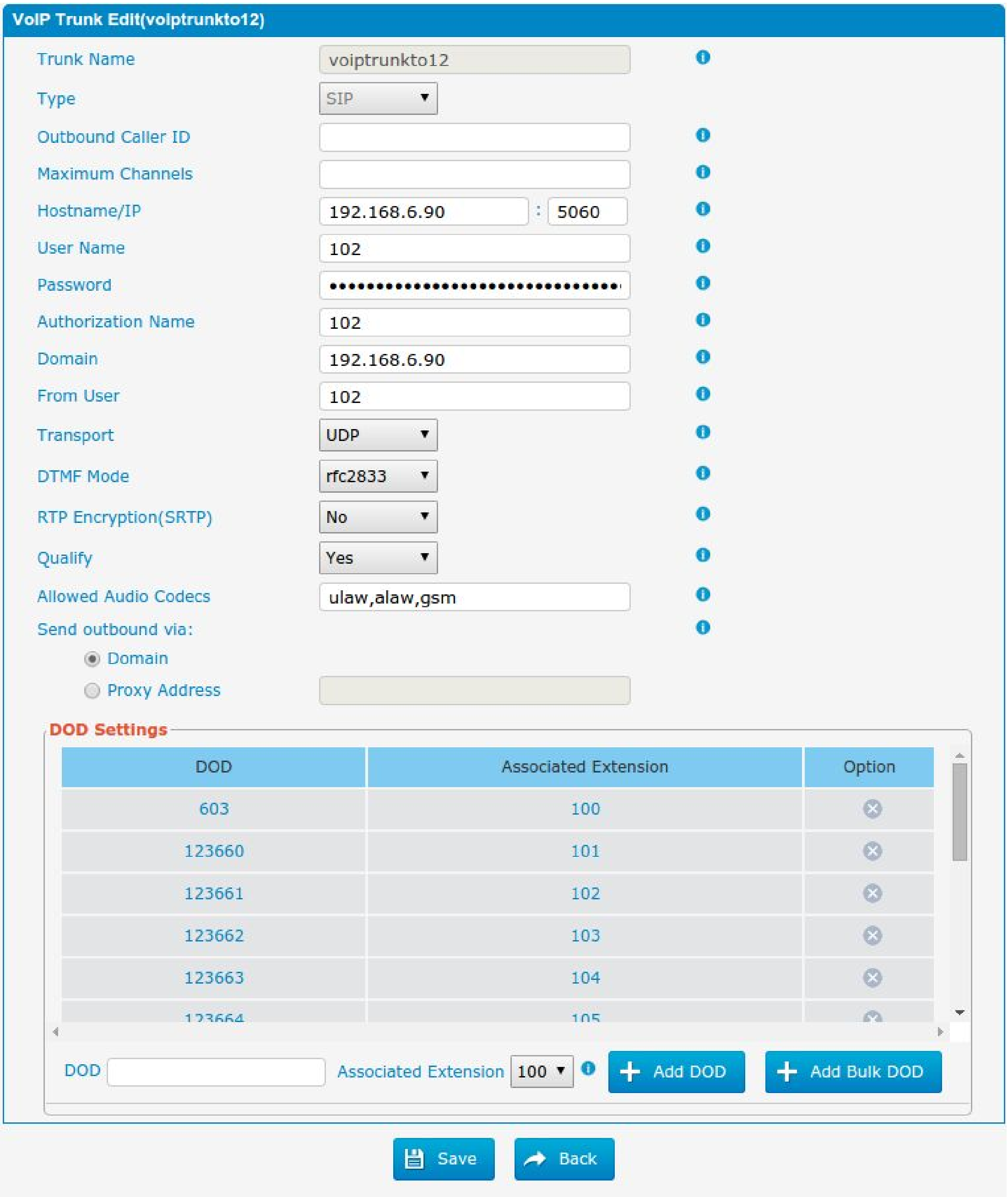


Figure 3.5.3b Add Bulk DOD

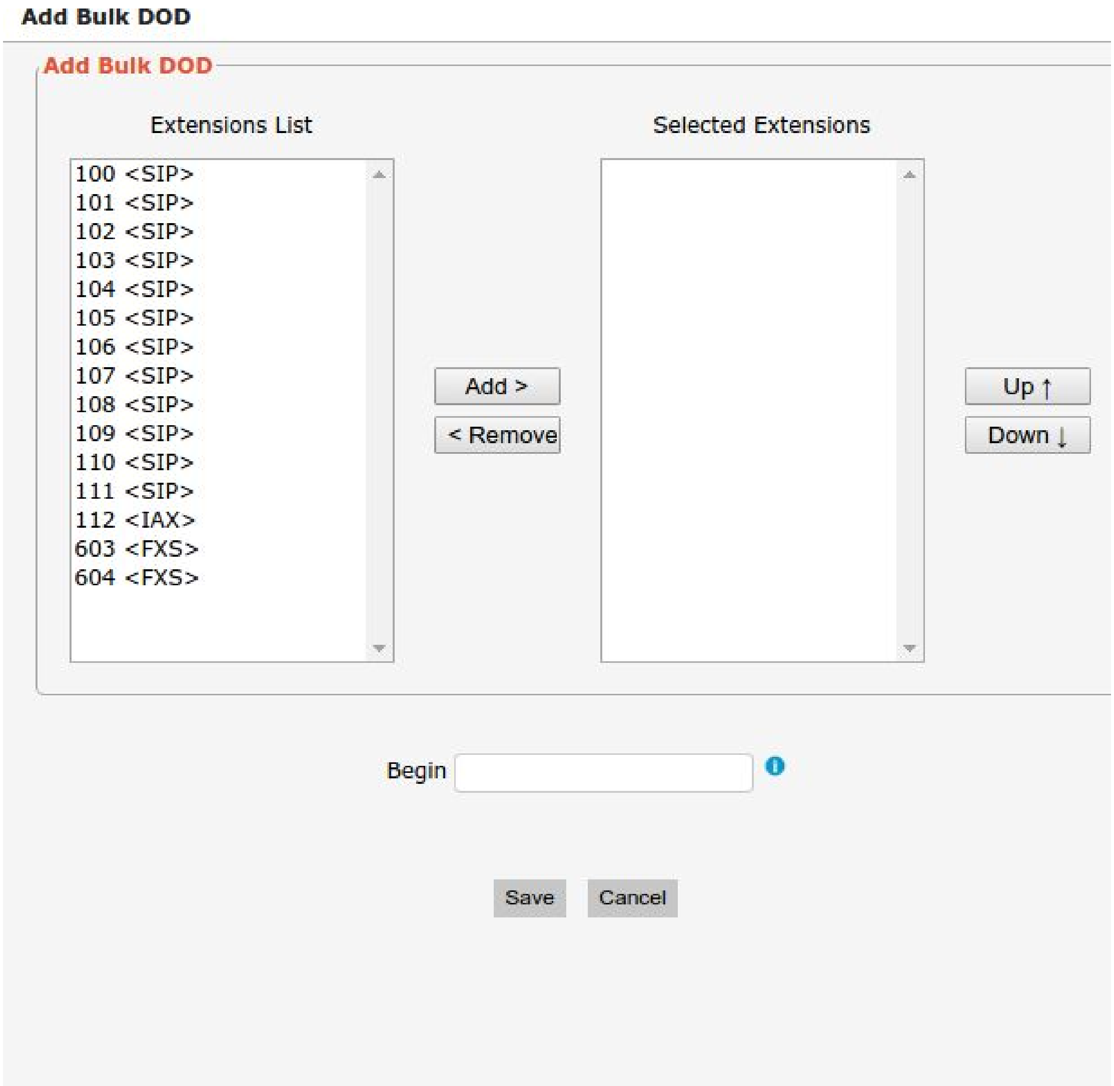


Table 3.5.3 Description of VoIP Trunk

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, PBX 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.
DOD Settintings	Add dod number to associated extension.
Add Bulk DOD	Add bulk dod number to associated extensions which begin with Begin number

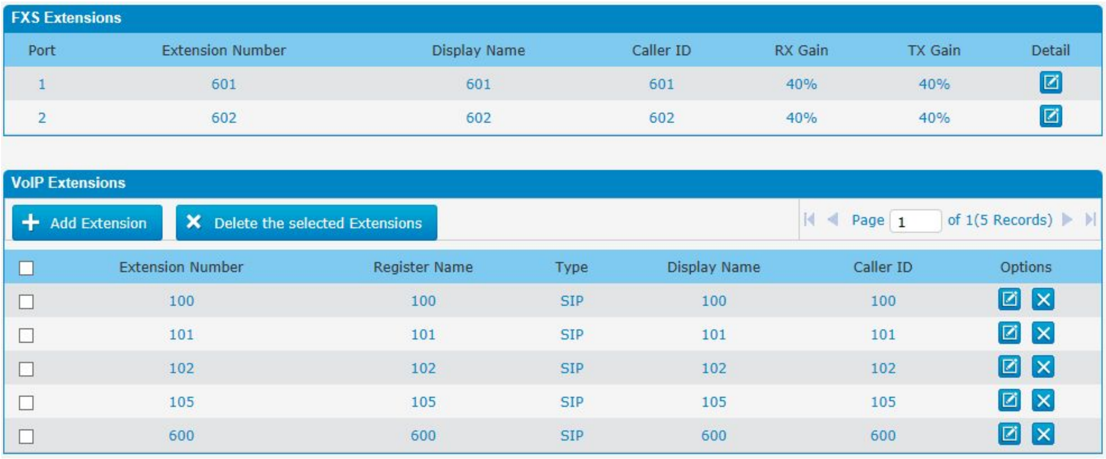
3.6 PBX Basic

3.6.1 Extensions

3.6.1.1 FXS Extensions

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

Figure 3.6.1.1 Extensions



FXS Extensions

Port	Extension Number	Display Name	Caller ID	RX Gain	TX Gain	Detail
1	601	601	601	40%	40%	
2	602	602	602	40%	40%	

VoIP Extensions

Add Extension | Delete the selected Extensions | Page 1 of 1(5 Records)

	Extension Number	Register Name	Type	Display Name	Caller ID	Options
	100	100	SIP	100	100	
	101	101	SIP	101	101	
	102	102	SIP	102	102	
	105	105	SIP	105	105	
	600	600	SIP	600	600	

Figure 3.6.1.1a Fxs Extensions Edit

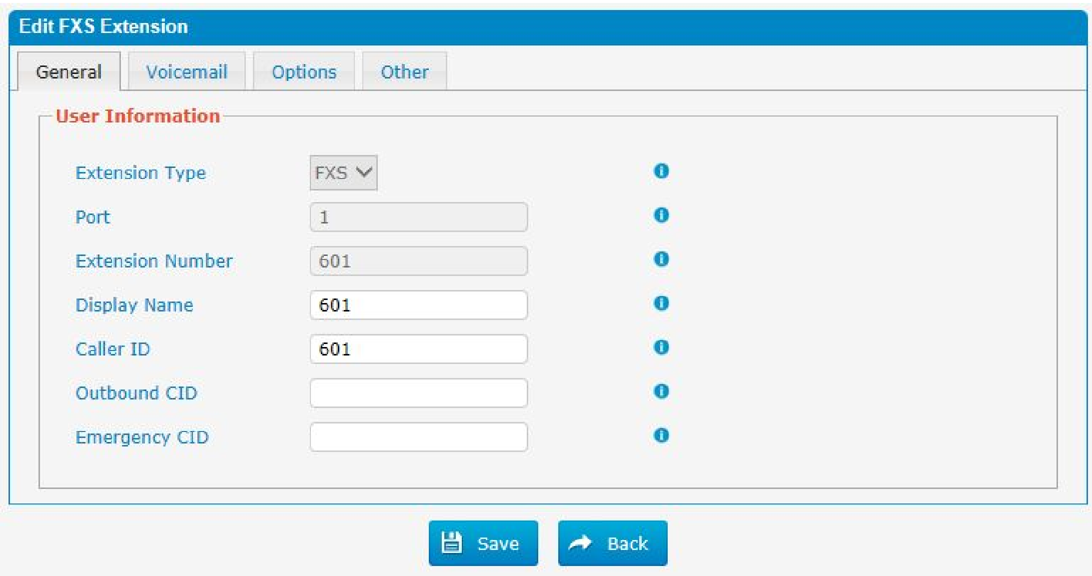


Table 3.6.1.1a FXS Extensions

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 3.6.1.1b Fxs Extensions Voicemail

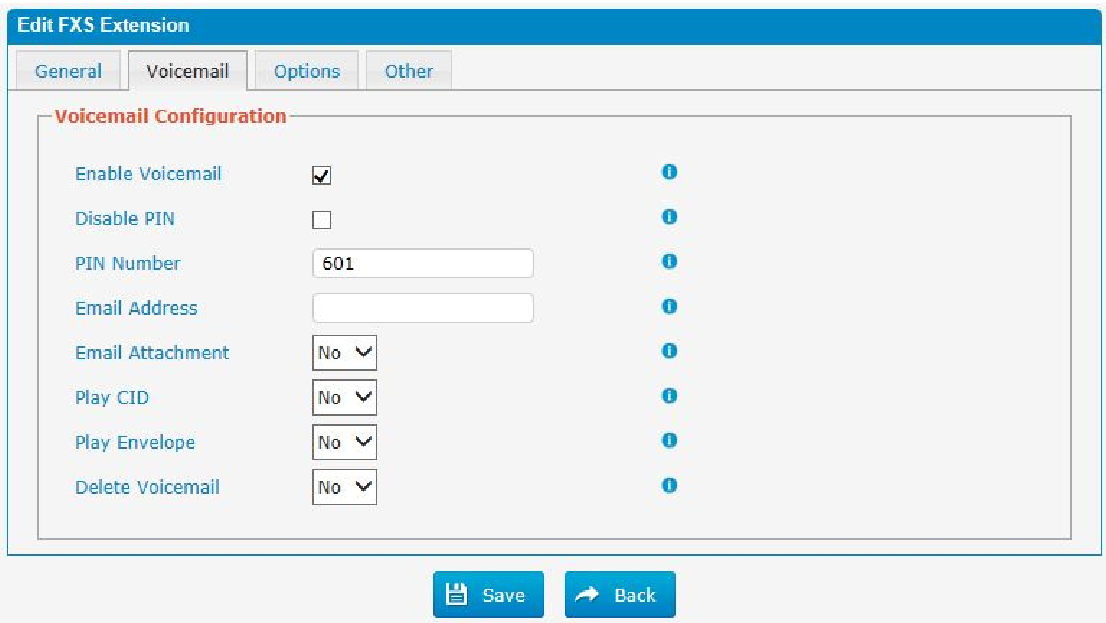


Table 3.6.1.1b Description of FXS Extensions Voicemail

Parameters	Description
Enable Voicemail	Check this box if the user should have a voicemail account.
Disable PIN	Disable voicemail PIN authentication.
PIN Number	Password used to access the Voicemail system.e.g. “601”.

Email Address Email Attachment	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 CID	Read back caller's telephone number prior to playing the incoming message.
Play Envelope	Envelope controls whether or nor the Voicemail system will play the message envelope (date/time) before playing the voicemail message.
Delete Voicemail	the message will be deleted from the Voicemailbox (after having been emailed).

Figure 3.6.1.1c FXS Extensions Options

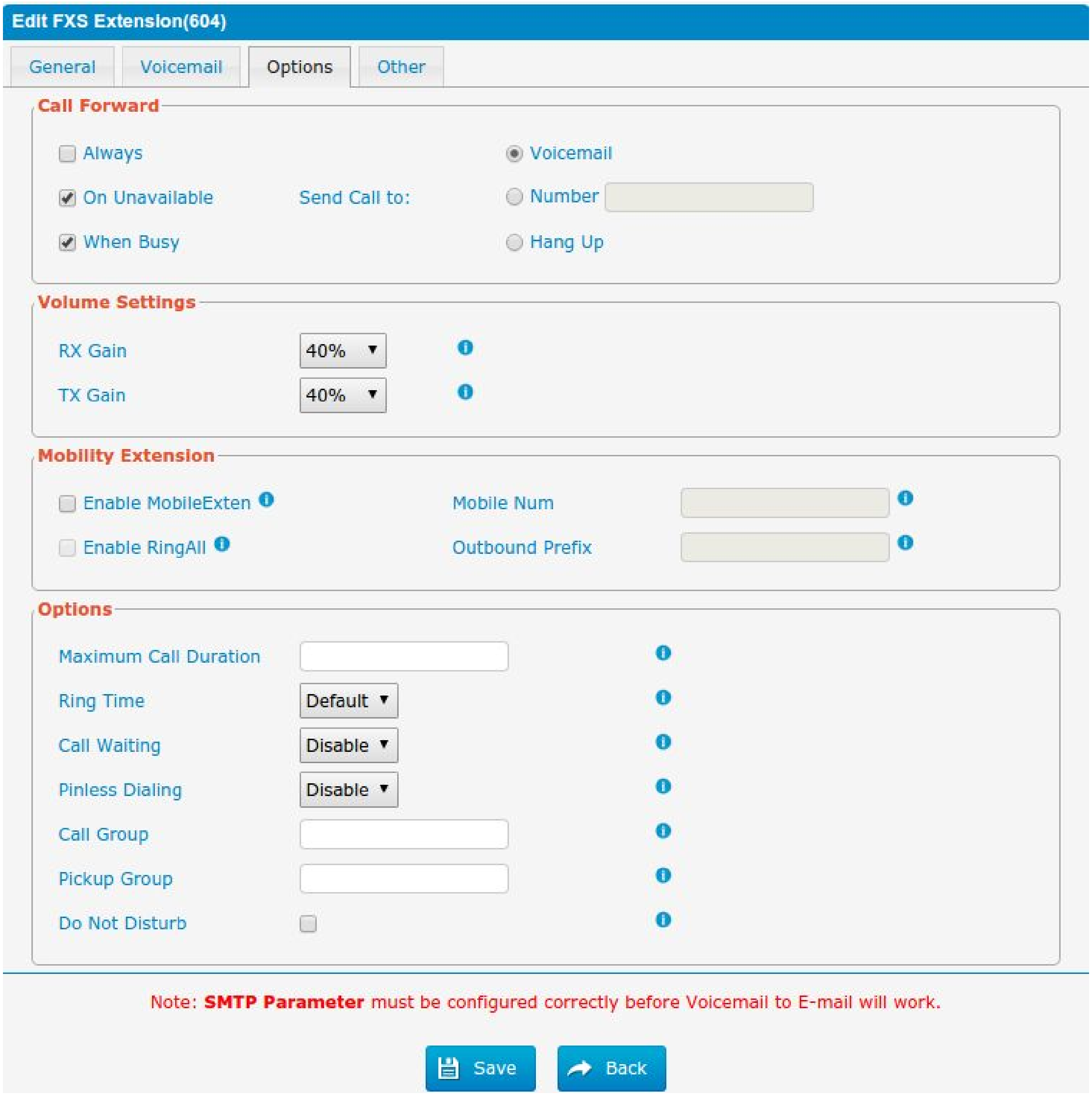


Table 3.6.1.1c Description of FXS Extensions Options

Parameters	**Description**
Call Forward (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
Mobility Extension	Mobile Num: if you set a mobile number as mobility extension,while you call in PBX with this mobile number, the mobile phone will get all permission of the associated extension.for example: dialing the extension, playing the voicemail. Enable RingAll: when someone calls the associated extension,your mobile phone will ring together,what you need is set outbound route and set Outbound Prefix number.
Maximum Call Duration	The absolute maximum amount of time permitted for a call,it only valid for outbound calls
Ring Time	Number of seconds to ring prior to going to voicemail.
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.
Pinless Dialing	Pinless Dialing will allow this extension to bypass any pin codes normally required on outbound calls
Call Group	Call group for peer/user
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.
Do Not Disturb	Do Not Disturb

Figure 3.6.1.1d Fxs Extensions Other

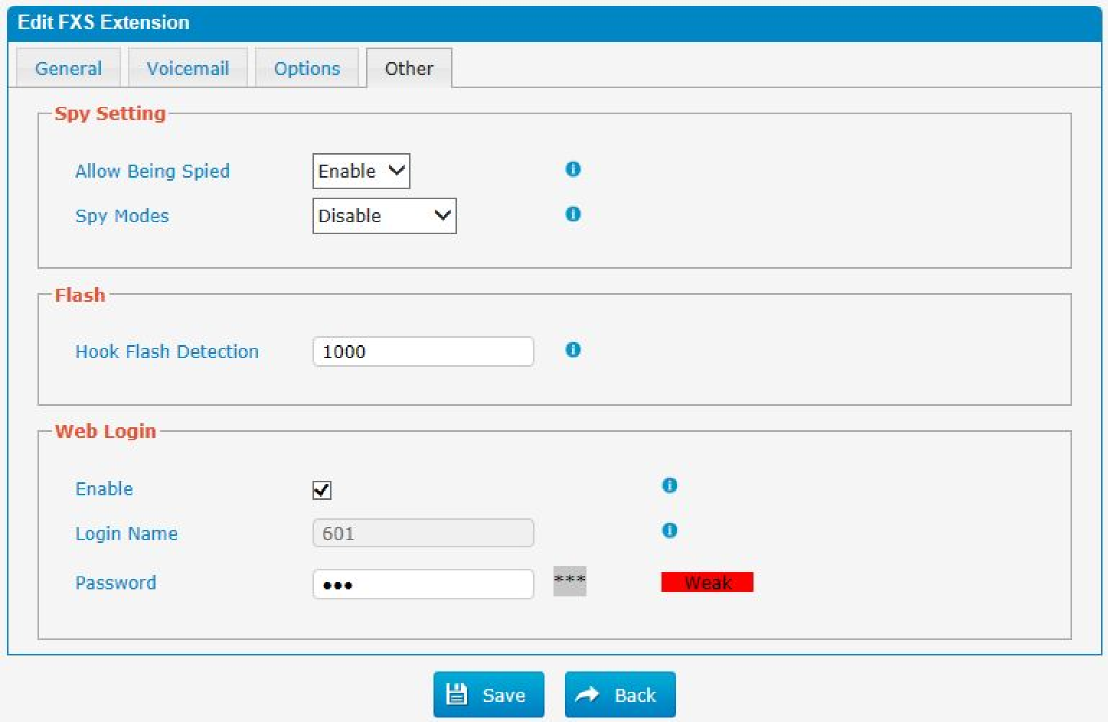


Table 3.6.1.1d Description of FXS Extensions Other

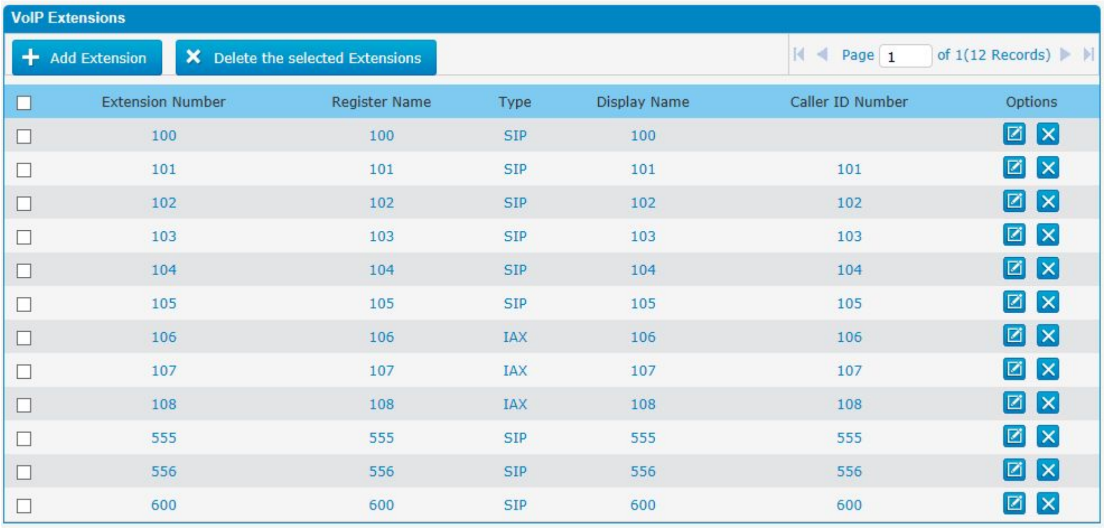
Parameters	Description
Spy Settings	PBX allows extension to monitor/barge in other conversation. Once this feature is enabled, the extension has the ability to monitor/barge in other calls using the feature codes for each spy mode. Refer to "Feature Codes" section for more information.
spy modes	There 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.
Flash	Sets the amount of time, in milliseconds, that must pass since the last hook-flash event received by PBX before it will recognize a second event. If a second event occurs in less time than defined by Hook Flash Detection, then PBX will ignore the event. The default value of Flash is 1000ms, and it

	can be configured in 1ms increments.
Web Login	Extension web login setttings.

3.6.1.2 VoIP Extensions

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

Figure 3.6.1.2 VoIP Extensions Edit/Add



VoIP Extensions

Add Extension | Delete the selected Extensions | Page 1 of 1(12 Records)

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	

Figure 3.6.1.2a VoIP Extensions Edit/Add

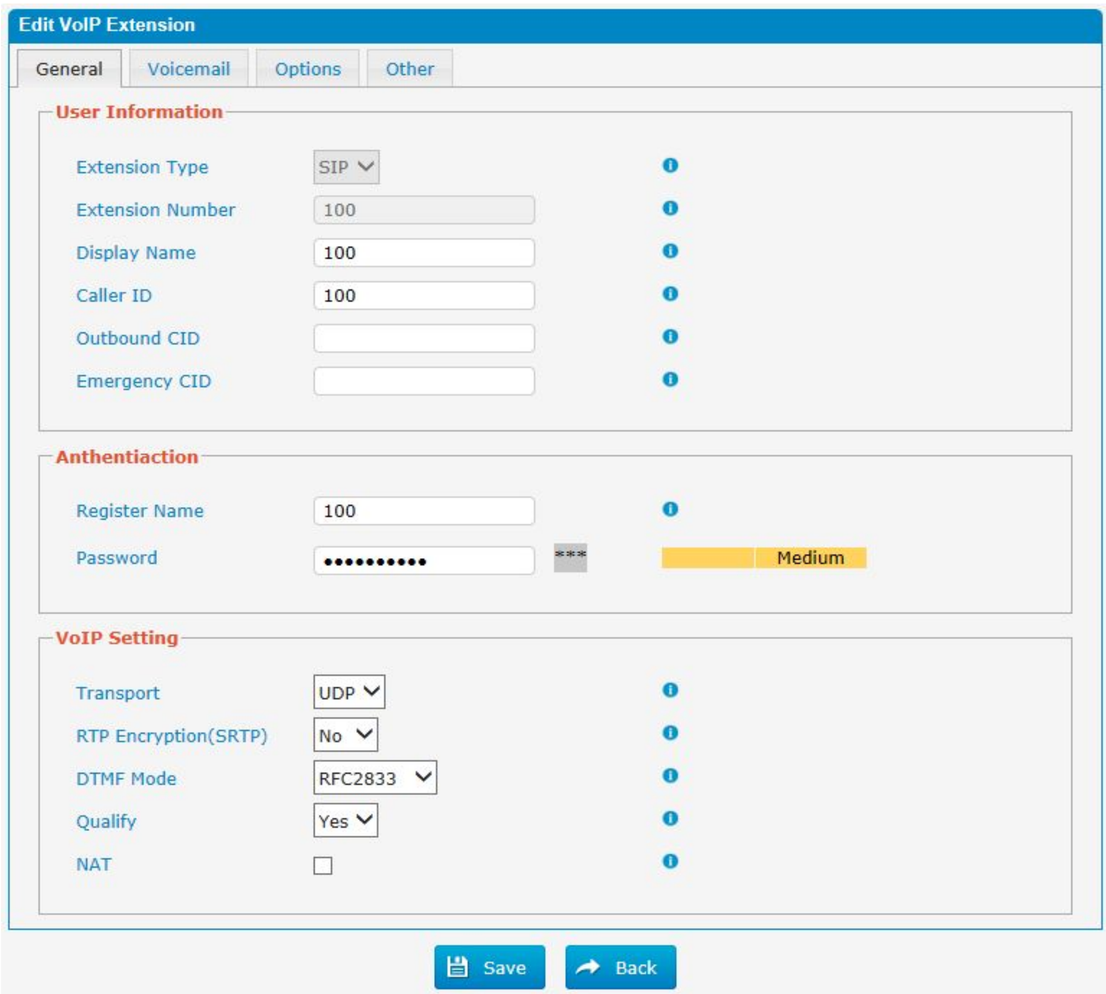


Table 3.6.1.2a Description of VoIP Extensions Edit/Add

Parameters	**Description**
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.
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.
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.
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.

Figure 3.6.1.2b VoIP Extensions Voicemail

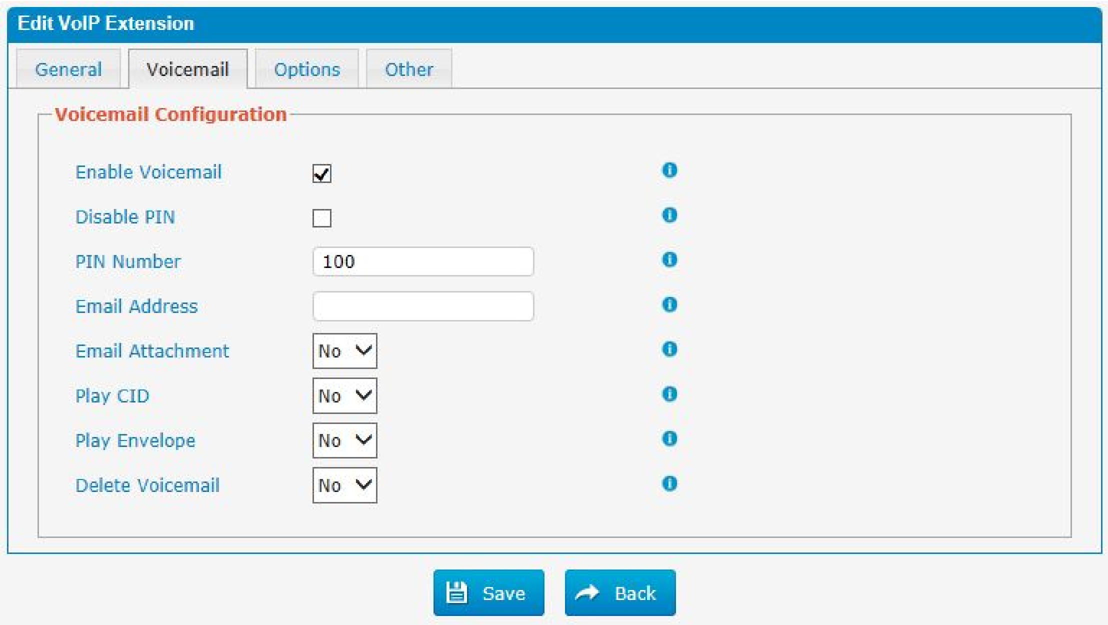


Table 3.6.1.2b Description of VoIP Extensions Voicemail

Parameters	**Description**
Enable Voicemail	Check this box if the user should have a voicemail account.
Disable PIN	Disable voicemail PIN authentication.
PIN Number	Password used to access the Voicemail system.e.g. "100".
Email Address Email Attachment	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 CID	Read back caller's telephone number prior to playing the incoming message.
Play Envelope	Envelope controls whether or nor the Voicemail system will play the message envelope (date/time) before playing the voicemail message.
Delete Voicemail	the message will be deleted from the Voicemailbox (after having been emailed).

Figure 3.6.1.2c VoIP Extensions Options

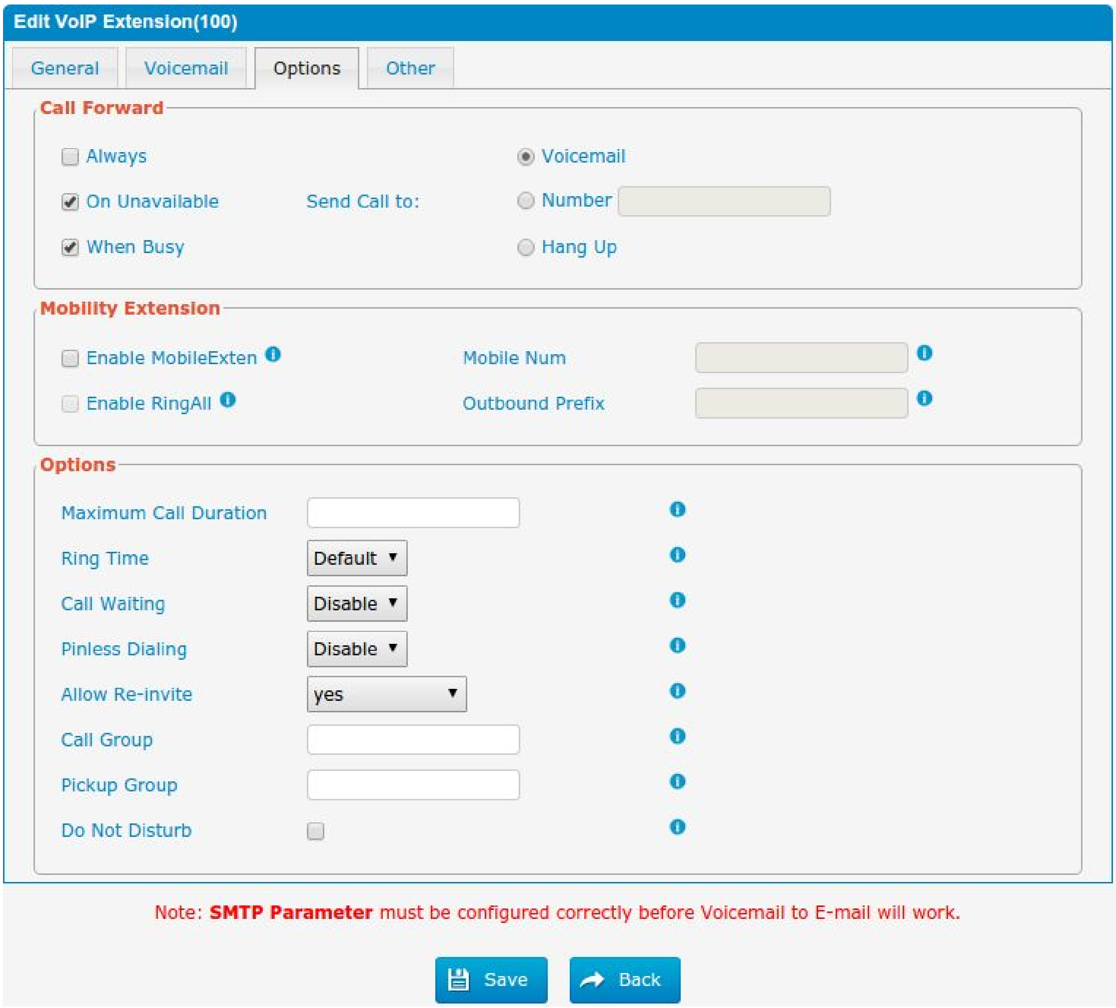


Table 3.6.1.2c Description of VoIP Extensions Options

Parameters	**Description**
Call Forward (Follow Me)	This function sets inbound call forwarding on an extension. An administrator can configure Call Forward for this extension.
Mobility Extension	Mobile Num: if you set a mobile number as mobility extension,while you call in PBX with this mobile number, the mobile phone will get all permission of the associated extension.for example: dialing the extension, playing the voicemail. Enable RingAll: when someone calls the associated extension,your mobile phone will ring together,what you need is set outbound route and set Outbound Prefix number.
Maximum Call	The absolute maximum amount of time permitted for a call,it only valid for outbound calls

Duration	
Ring Time	Number of seconds to ring prior to going to voicemail.
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.
Allow Re-invite	Re-Invite policy for this device. yes: Allow RTP media direct. no: Deny re-invites. nonat: Allow reinvite when local, deny reinvite when NAT. update: Use UPDATE instead of INVITE. update,nonat: Use UPDATE when local, deny when NAT.",
Pinless Dialing	Pinless Dialing will allow this extension to bypass any pin codes normally required on outbound calls
Call Group	Call group for peer/user
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.
Do Not Disturb	Do Not Disturb

Figure 3.6.1.2d VoIP Extensions Other

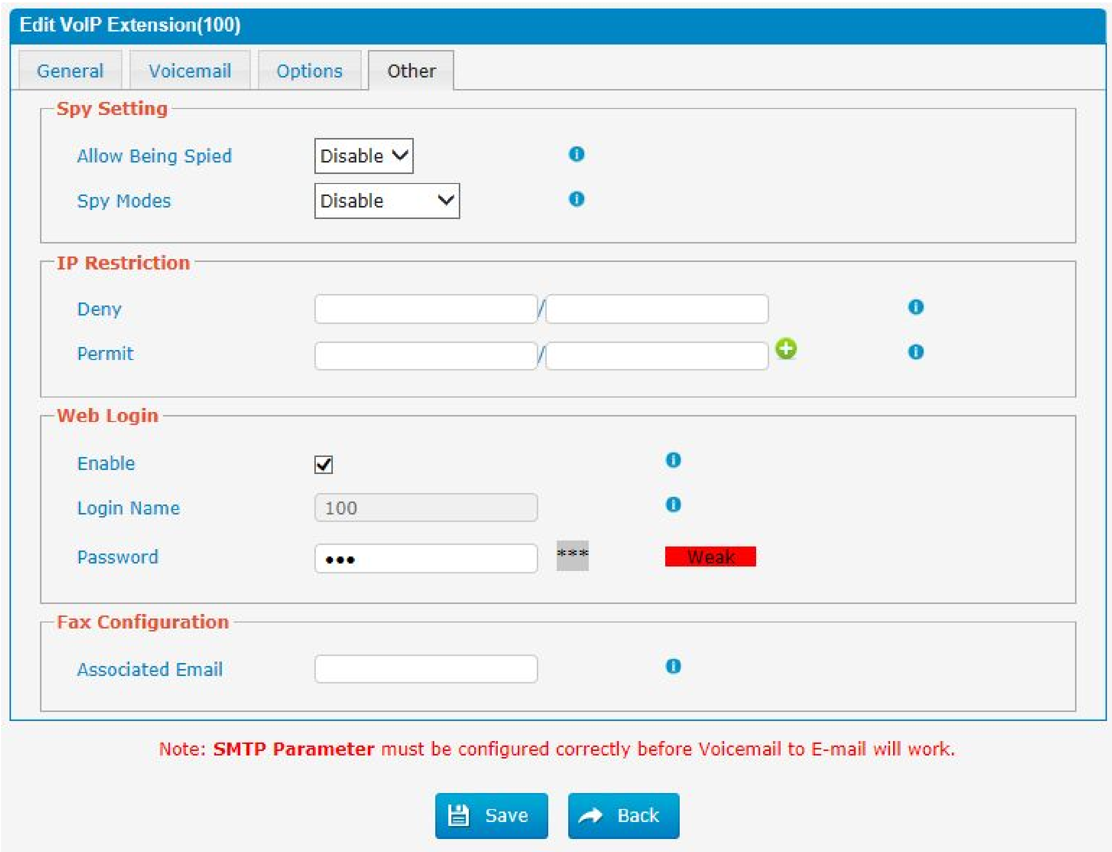


Table 3.6.1.2d Description of VoIP Extensions Other

Parameters	**Description**
Spy Settings	PBX 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.

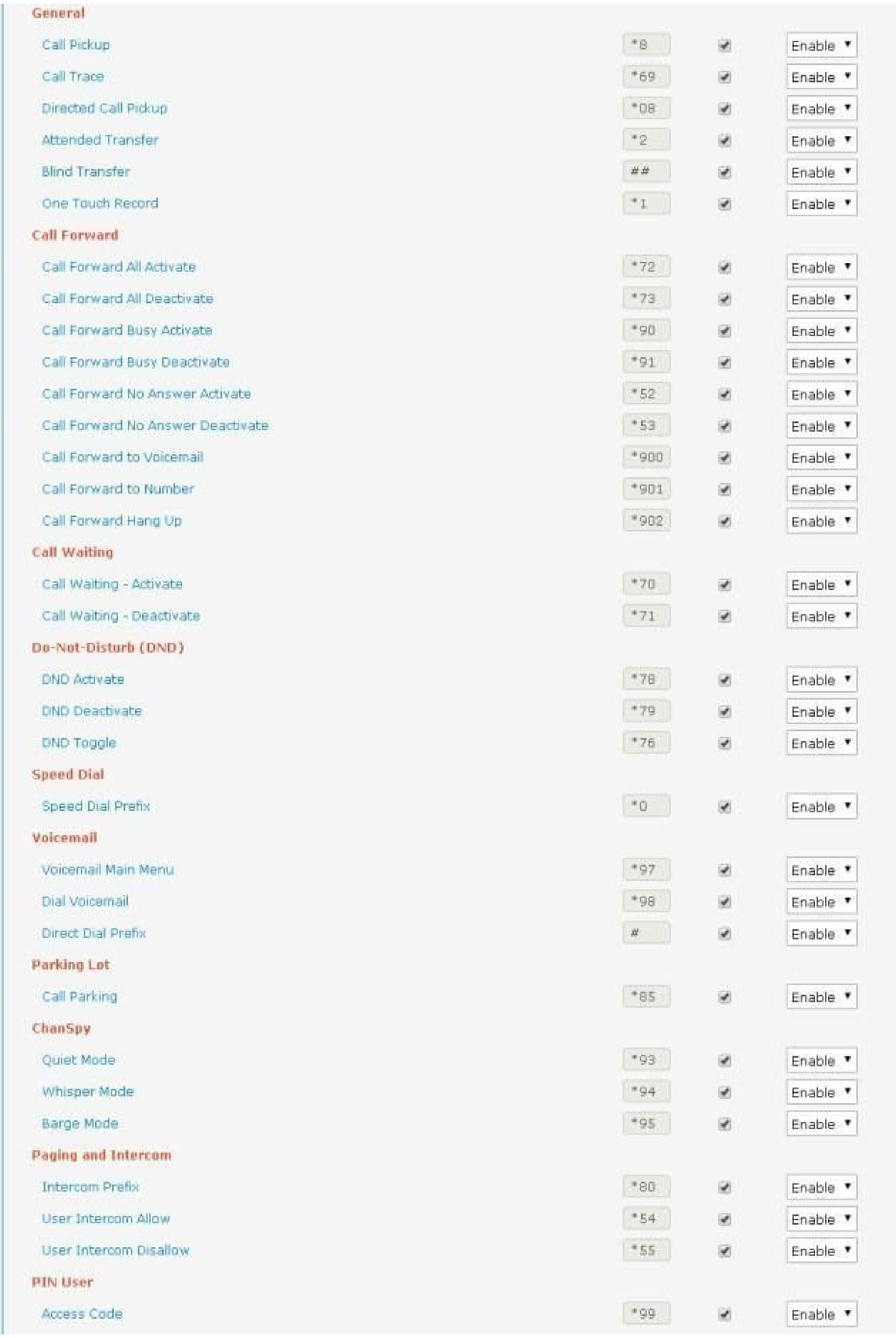
IP Restriction	IP Restriction Settings Default leave it blank on ”IP Restriction” configuration. it indicate that registration of remote extension is allowed(remote extension IP Address is not deny)

	Deny: IP Address range to deny access to,in the form of network/netmask, e.g.0.0.0.0/0.0.0.0 **Permit:** IP Address range to deny access to,in the form of network/netmask,this can be a very useful security option when dealing with remote extensions that are at a known location(such as a branch office) or within a known ISP range for some home office situations. e.g.192.168.6.1/255.255.255.0
Web Login	Extension web login setttings.
Fax Configuration	Associated Email: the email address that FAXs are send to. It is used for T.38 FAX

3.6.2 Feature Codes

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

Figure 3.6.2 Feature Codes



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
PIN User			
Access Code	*99	✔	Enable

Table 3.6.2 Description of Feature Codes

Label	**Feature Codes**	**Description**
Call Pickup	*8	Pickup extension
Call Trace	*69	Trace last call number,and press 1,dial this number out.
Directed Call Pickup	*08	[featurecode] + extension number Pickup specify extension
Attended Transfer	*2	[featurecode] + extension number Specify transfer to extension
Blind Transfer	##	[featurecode] + extension number After the success of the transfer to extension will automatically hang up
One Touch Record	*1	Start recording in call, stop recording when Enter again
Call Forward All Activate	*72	Call forward all activate
Call Forward All Deactivate	*73	Call forward all deactivate
Call Forward Busy Activate	*90	Call forward busy activate
Call Forward Busy Deactivate	*91	Call forward busy deactivate
Call Forward No Answer Activate	*52	Call forward no answer activate
Call Forward No Answer Deactivate	*53	Call forward no answer deactivate
Call Forward to Voicemail	*900	Call forward to voicemail
Call Forward to Number	*901	Call forward to number
Call Forward Hang Up	*902	Call forward to hang up
Call Waiting -	*70	Call waiting activate

Activate		
Call Waiting - Deactivate	*71	Call waiting deactivate
DND Activate	*78	DND activate
DND Deactivate	*79	DND deactivate
DND Toggle	*76	DND toggle
Speed Dial Prefix	*0	[featurecode] + Speed Dial Source Number = Speed Dial Destination Number
Voicemail Main Menu	*97	Into voicemail main menu
Dial Voicemail	*98	Check extension voicemail
Direct Dial Prefix	#	[featurecode] + Extension number Leave a message to Specify extension
Call Parking	*85	Eg. Park a call to extension 701
Quiet Mode	*93	[featurecode] + Extension number you can only hear the call, but can't talk.
Whisper Mode	*94	[featurecode] + Extension number you can hear the call, and can talk with the monitored extension.
Barge Mode	*95	[featurecode] + Extension number you can hear the call and talk with them both.
Intercom Prefix	*80	[featurecode] + Extension number
User Intercom Allow	*54	Allow user intercom
User Intercom Disallow	*55	Disable user intercom
Access Code	*99	[featurecode] + [password] Get into PIN Users function

3.6.3 Speed dial

Figure 3.6.3 Speed Dial

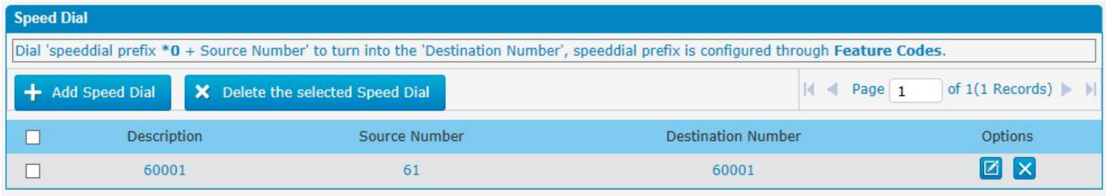


Figure 3.6.3a Speed Dial Add

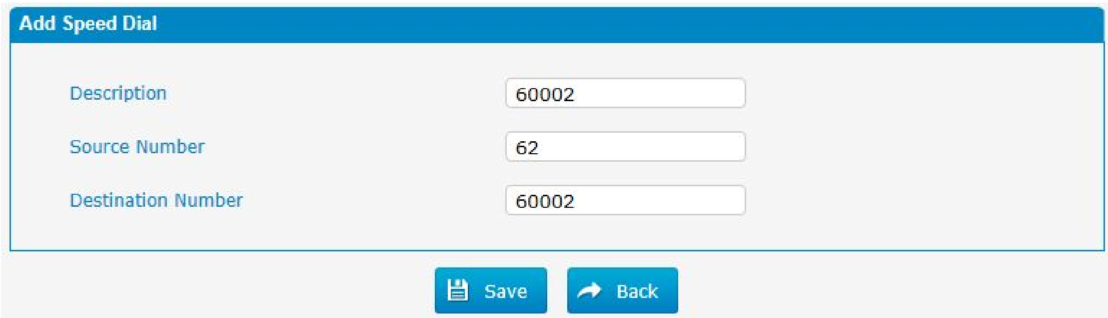


Table 3.6.3 Description of Speed Dial

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.

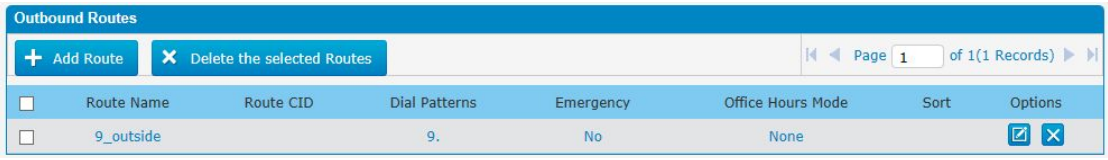
3.6.4 Outbound Routes

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, PBX 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

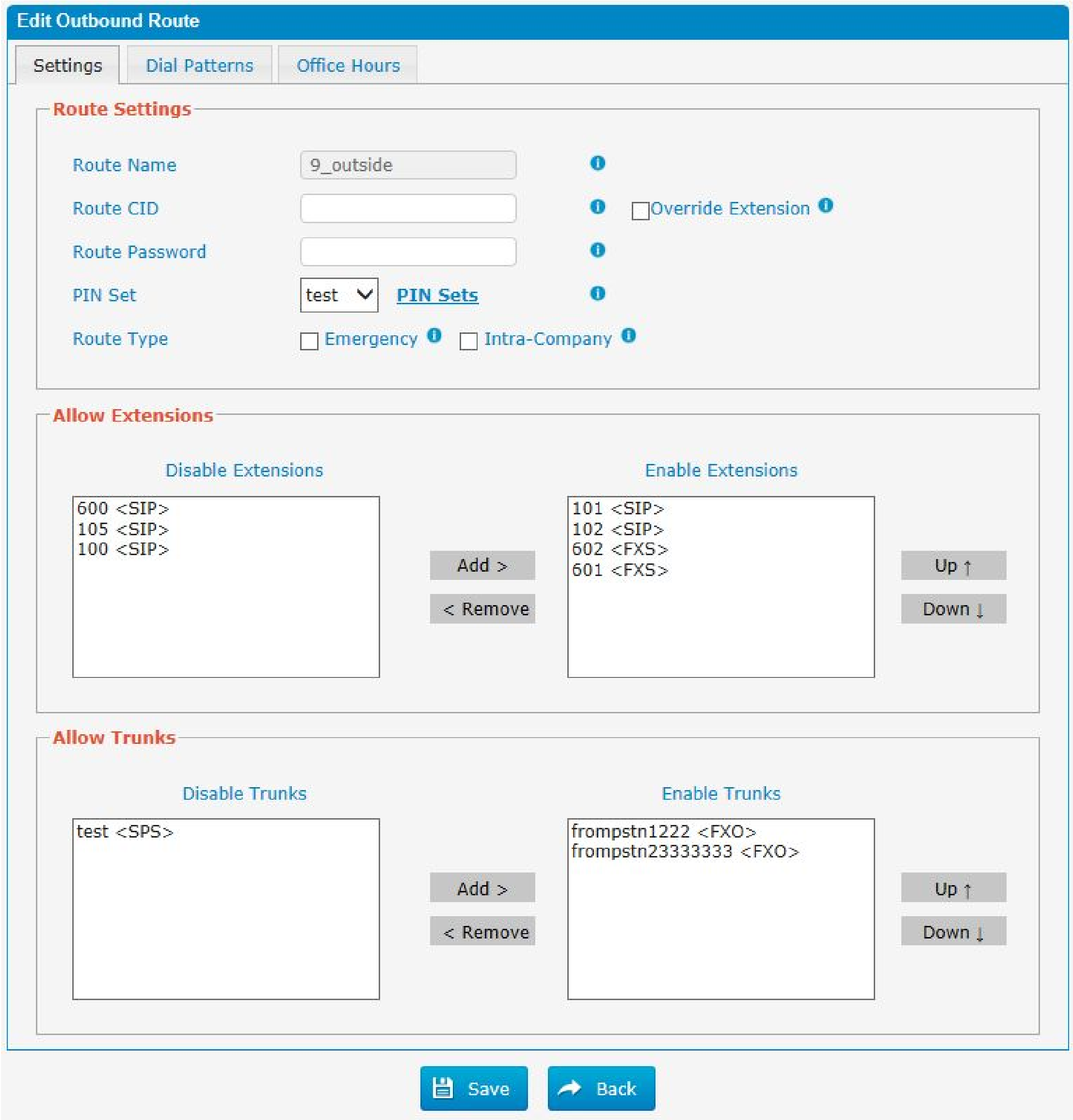


Table 3.6.4a Description of Outbound Routes Edit

Parameters	**Description**
Route Name	Name of this Outbound Route. E.g. "Local" or "Long Distance".
Route CID	CID of this route
Override Extension	Whether ovrride extension cid
Route Passwd	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.
Route Type	● Emergency ● Intra-Company
Disable Extensions	All disable extensions
Enable Extensions	Define the extensions that will be permitted to use this outbound route.
Disable Trunks	All disable trunks
Enable Trunks	Define the trunks that can be used for this outbound route.

Figure 3.6.4b Outbound Routes Edit

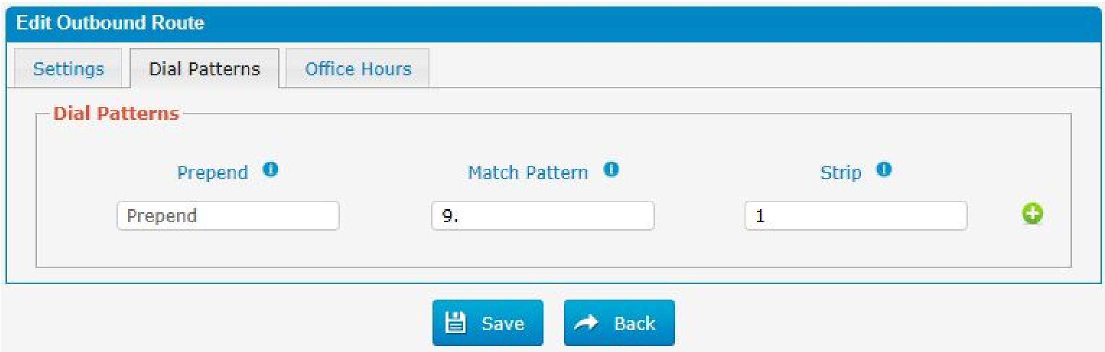


Table 3.6.4b Description of Outbound Routes Edit

Parameters	**Description**
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.
Match 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.
Add	Add multiple dial patterns in this outbound route.

Figure 3.6.4c Outbound Routes Edit

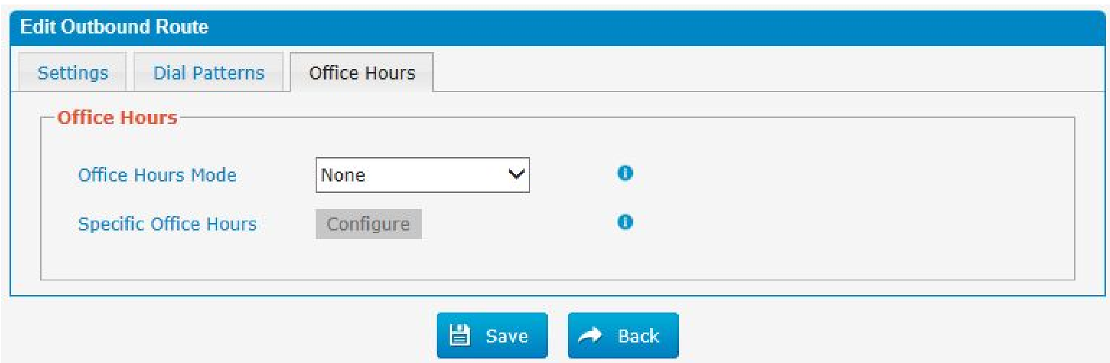


Table 3.6.4c Description of Outbound Routes Edit

Parameters	**Description**
Office Hours Mode	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.
Speciffic Office Hours	Configure specific office hour

3.6.5 Parking Lot

Figure 3.6.5 Parking Lot

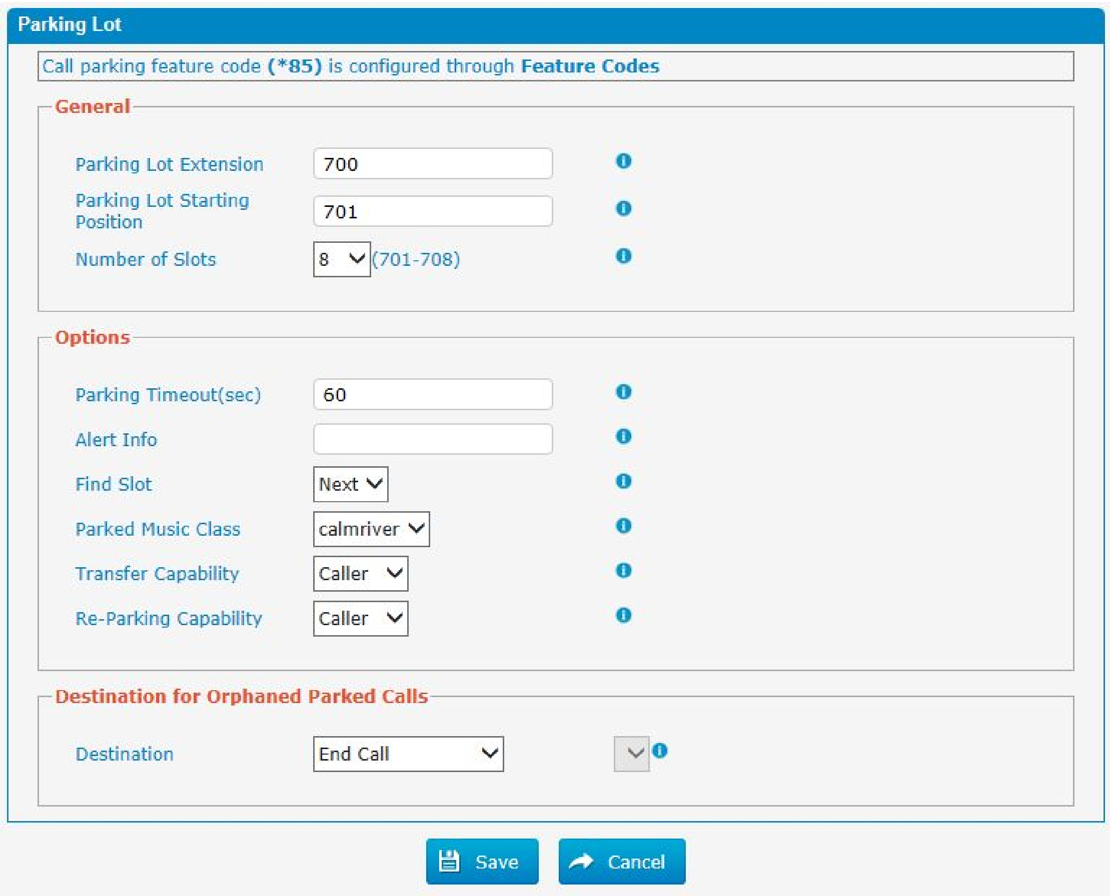


Table 3.6.5 Description of Parking Lot

Parameters	**Description**
Parking Lot Extension	This is the extension where you will transfer a call to park it.
Parking Lot Staring Postion	The starting postion of the parking lot
Number of Slots	The total number of parking lot spaces to configure. Example, if 700 is the extension and 8 slots are configured, the parking slots will be 701-708
Parking Timeout (sec)	The timeout period in seconds that a parked call will attempt to ring back the original parker if not answered(0 for 45s).
Alert Info	This can create distinct rings on some SIP phones and can serve to alert the recipients that the call is from an Orphaned parked call.

Parked Music Class	This is the music class that will be played to a parked call while in the parking lot UNLESS the call flow prior to parking the call explicitly set a different music class, such as if the call came in through a queue or ring group.
Transfer Capability	Enables or disables DTMF based transfers when picking up a parked call.
Re-Parking Capability	Enables or disables DTMF based parking when picking up a parked call.
Destination	Destination to send the call to after Timeout Recording is played.

3.6.6 Time Groups

Figure 3.6.6 Time groups configure

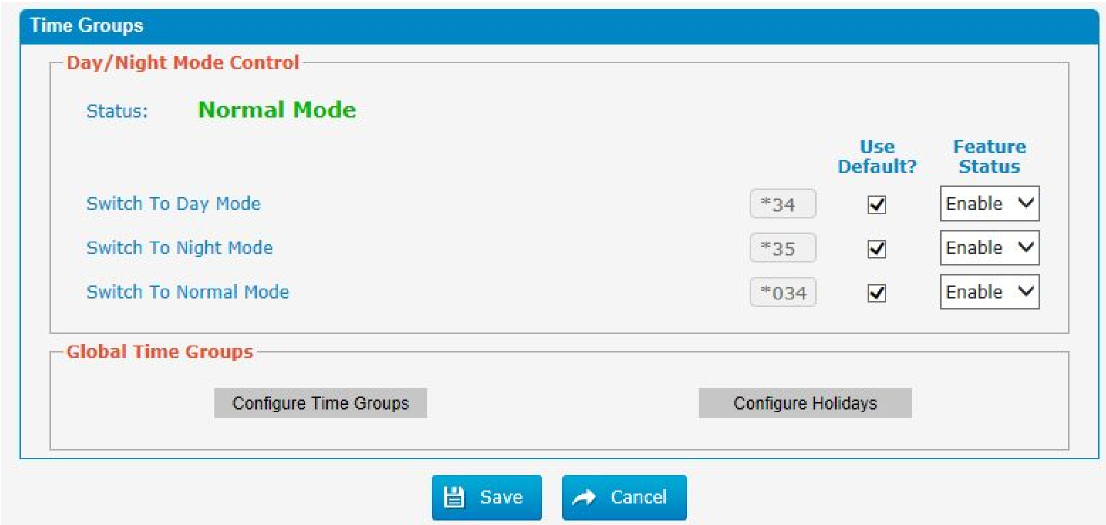
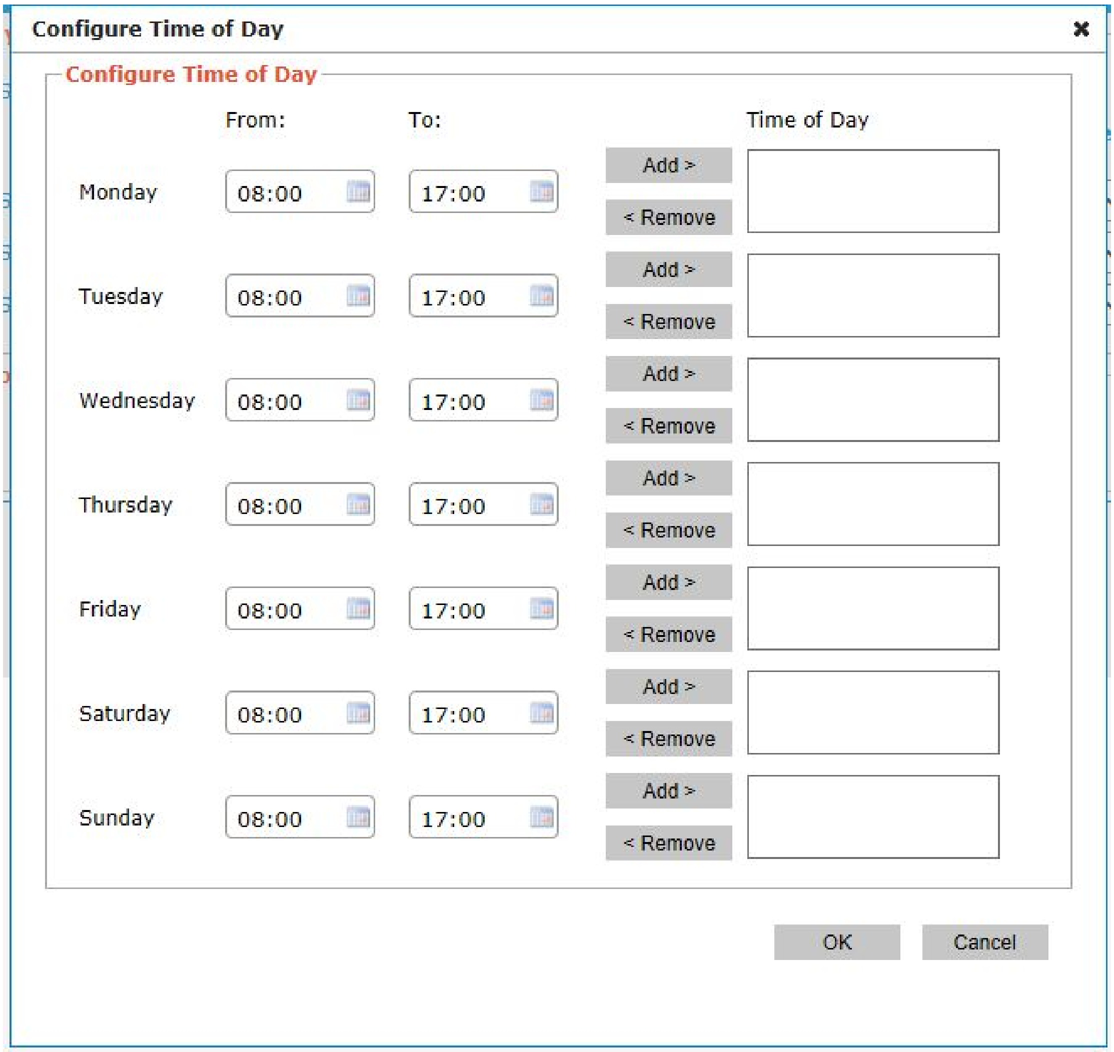


Figure 3.6.6a Time groups configure



3.6.7 General Preferences

Figure 3.6.7 General Preferences

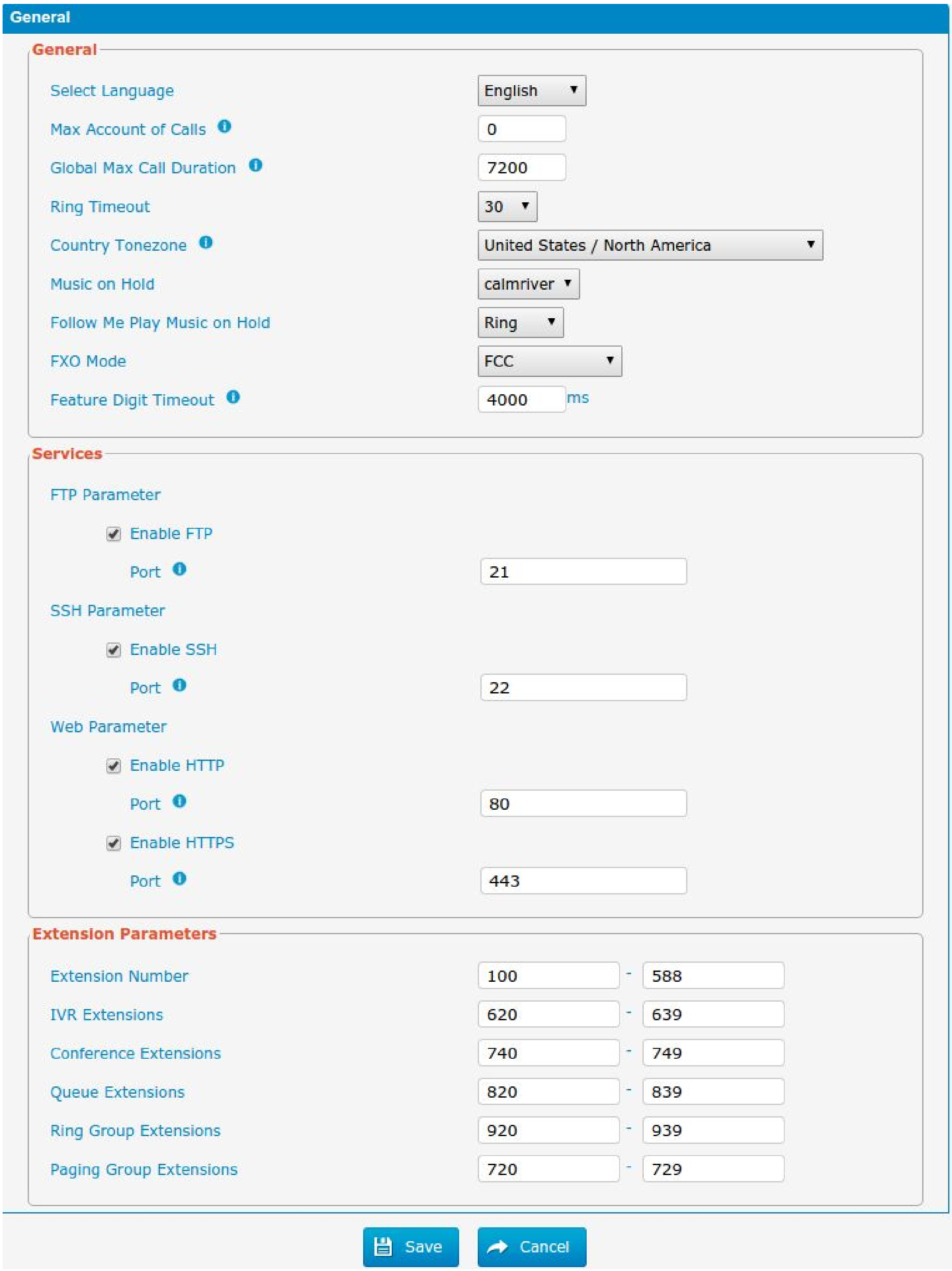


Table 3.6.7 Description of General Preferences

Parameters	**Description**
Select Language	Web label language selection English and Chinese-S
Max Account of Calls	Maximum concurrent calls limit(0 for unlimited)
Global Max Call Duration	The absolute maximum amount of time permitted for a call. A setting of 0 disables the timeout.
Ring Timeout	Global extension ring timeout.
Country Tonezone	Please select your country or nearest neighboring country to enable the default dial tone, busy tone, and ring tone for your region.
Muisc on Hold	Select MOH music
Follow Me Play Music on Hold	Mucic of follow me Ring: normal ring back tone Defaul: default MOH music None: silence
FXO Mode	FXO coutry mode
Feature Digigt Timeout	Max time (ms) between digits for feature activation.
Enable FTP	FTP services, Default Port 21
Enable SSH	SSH services, Default Port 8022
Enable HTTP	HTTP services, Default Port 80
Enable HTTPS	HTTPS services, Default Port 443
Extension Number	The scope of VoIP Extension
IVR Extensions	The scope of IVR
Conference Extensions	The scope of conference extension
Queue Extensions	The scope of queue extension
Ring Group Extensions	The scope of ring group

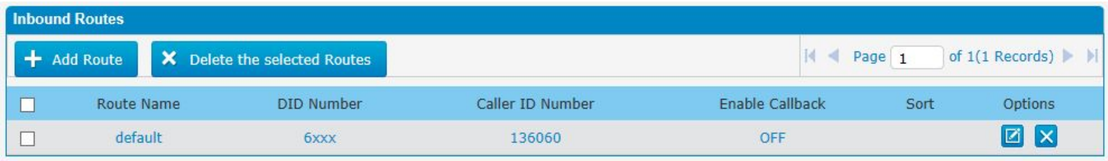
Paging Group Extensinos	The scope of paging group

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

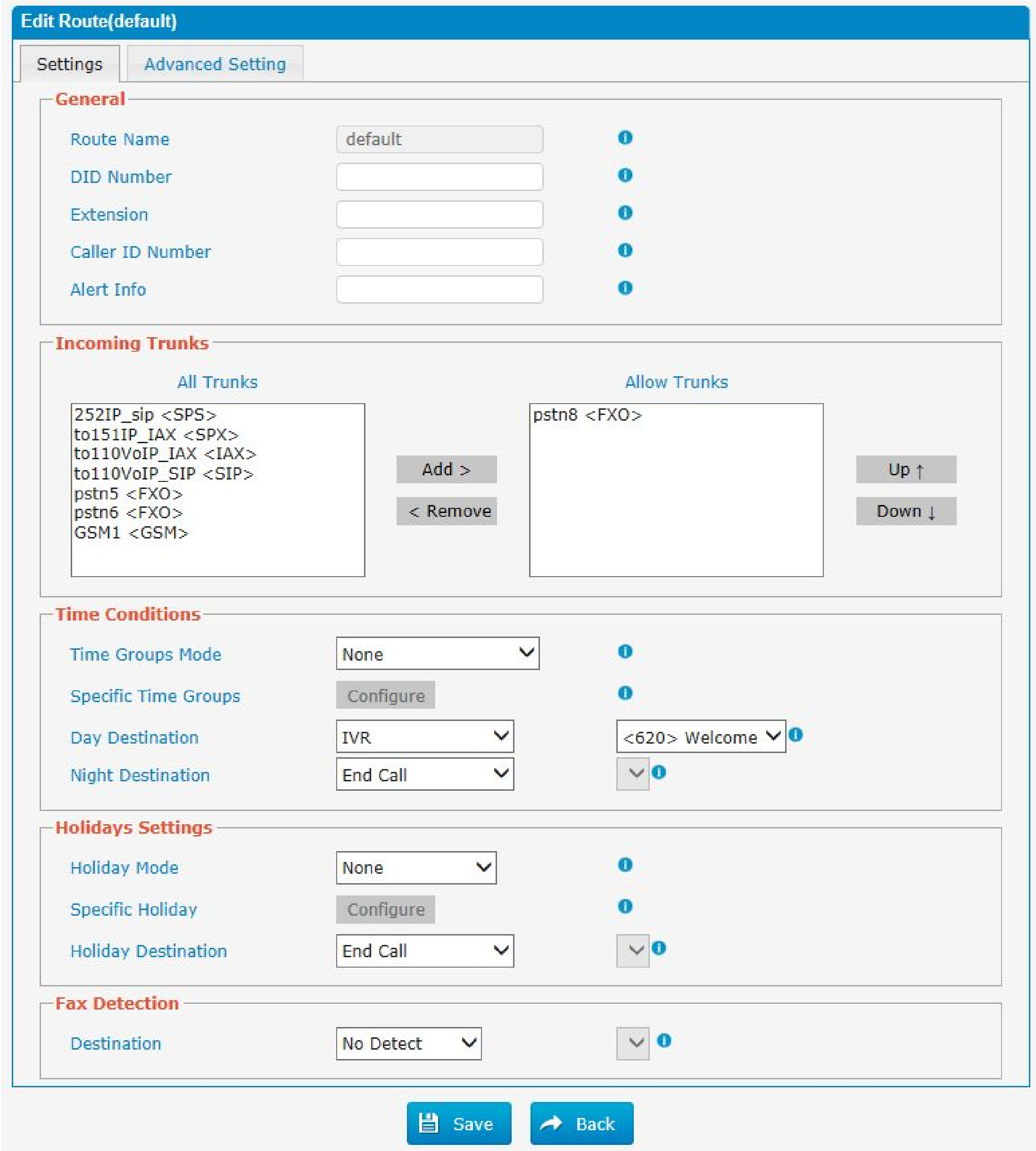


Table 3.7.1a Description of Inbound Routes Edit

Parameters	**Description**
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.
All Trunks	List all available trunks
Allow 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.
Time Groups Mode	Select time groups mode. ● 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.
Specific Time Groups	Set specific time groups
Day Destination NightDestination	● 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 PBX 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.
Holiday Mode	Define where the calls will be routed during Holidays. ● 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.
Specific Holiday	Specific holiday time groups
Holiday Destination	Configure where to route the incoming calls during holidays.
Destination	Fax detect destination

Figure 3.7.1b Inbound Routes Edit

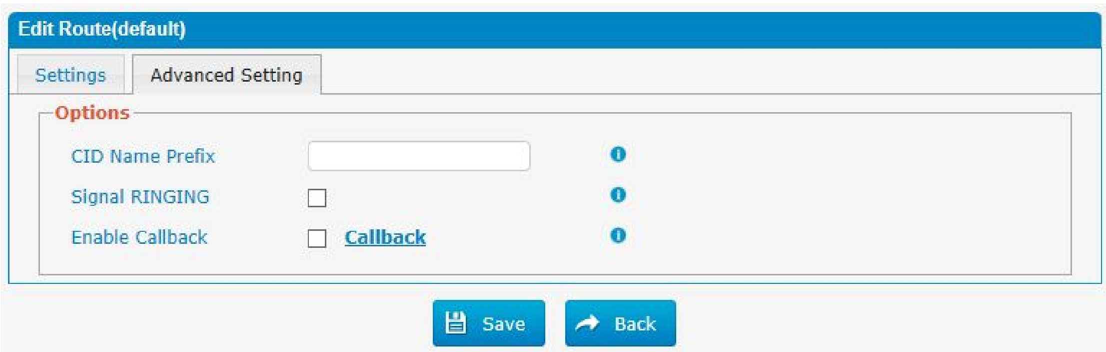


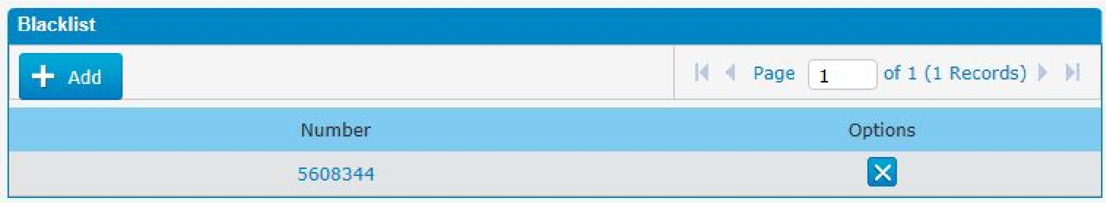
Table 3.7.1b Description of Inbound Routes Edit

Parameters	**Description**
CID Name Prefix	Set inbound CID prefix
Signal RINGING	Some devices or providers require RINGING to be sent before ANSWER. You'll notice this happening if you can send calls directly to a phone, but if you send it to an IVR, it won't connect the call.
Enable Callback	Enable callback

3.7.2 Blacklist

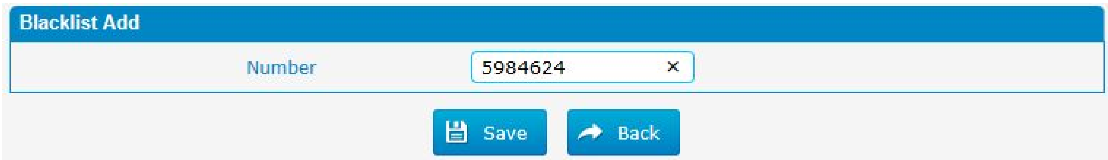
Blacklist is used to block an incoming/outgoing call. If the number of incoming/outgoing call is listed in the number blacklist, the caller will hear the following prompt: "The number you have dialed is not in service. Please check the number and try again". The system will then disconnect the call.

Figure 3.7.2 Blacklist



We can add a number to blacklist

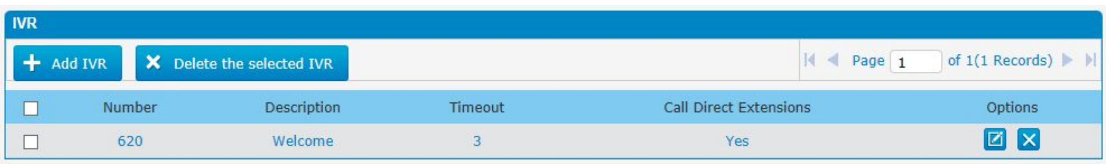
Figure 3.7.2a Blacklist Add



3.7.3 IVR

When there"s an inbound call aims at Auto Attendant, PBX 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

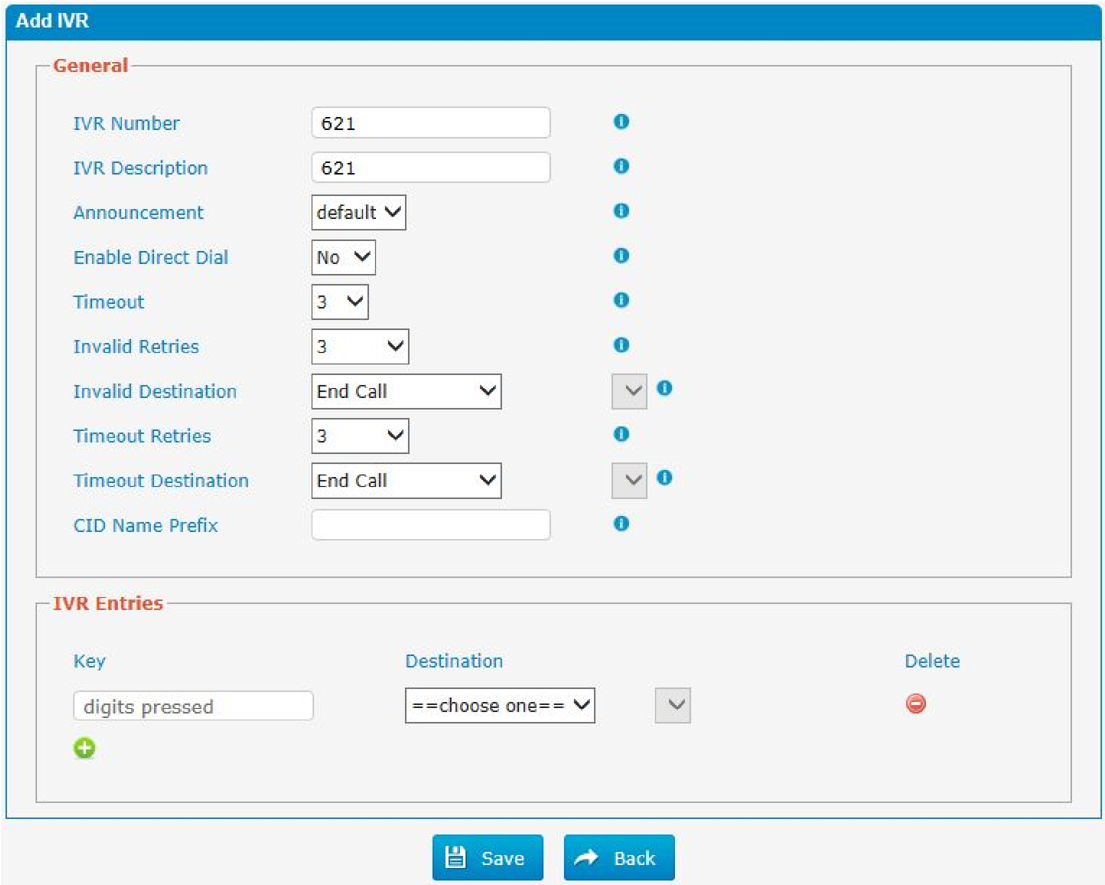


Table 3.7.3a Description of IVR Add/Edit

Parameters	**Description**
IVR Number	PBX treats IVR as an extension; you can dial this extension number to reach the IVR from internal extensions.
IVR 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 Retries	Invalid retries number of keys
Invalid Destination	Destination when Number of times more than the settings.

Timeout Retries	Retry timeout
Timeout Destination	Destination of timeout
CID Name Prefix	IVR CID preifx name
Key	The Key pressed when the callers hear the IVR prompt.
Destination	Where will PBX route the call when the action occurs.
Delete	Delete a key to the destination IVR record.
Add	Add a key to the destination IVR record.

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

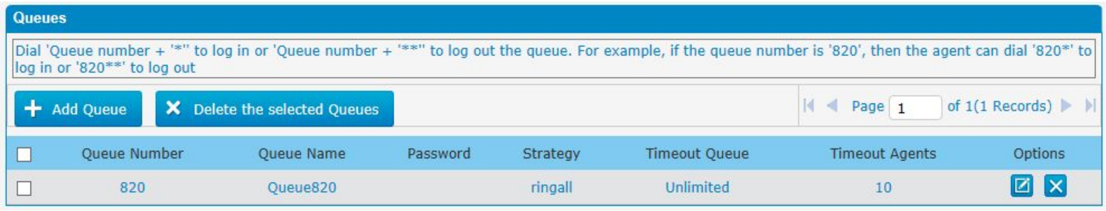


Figure 3.7.4a Queue General

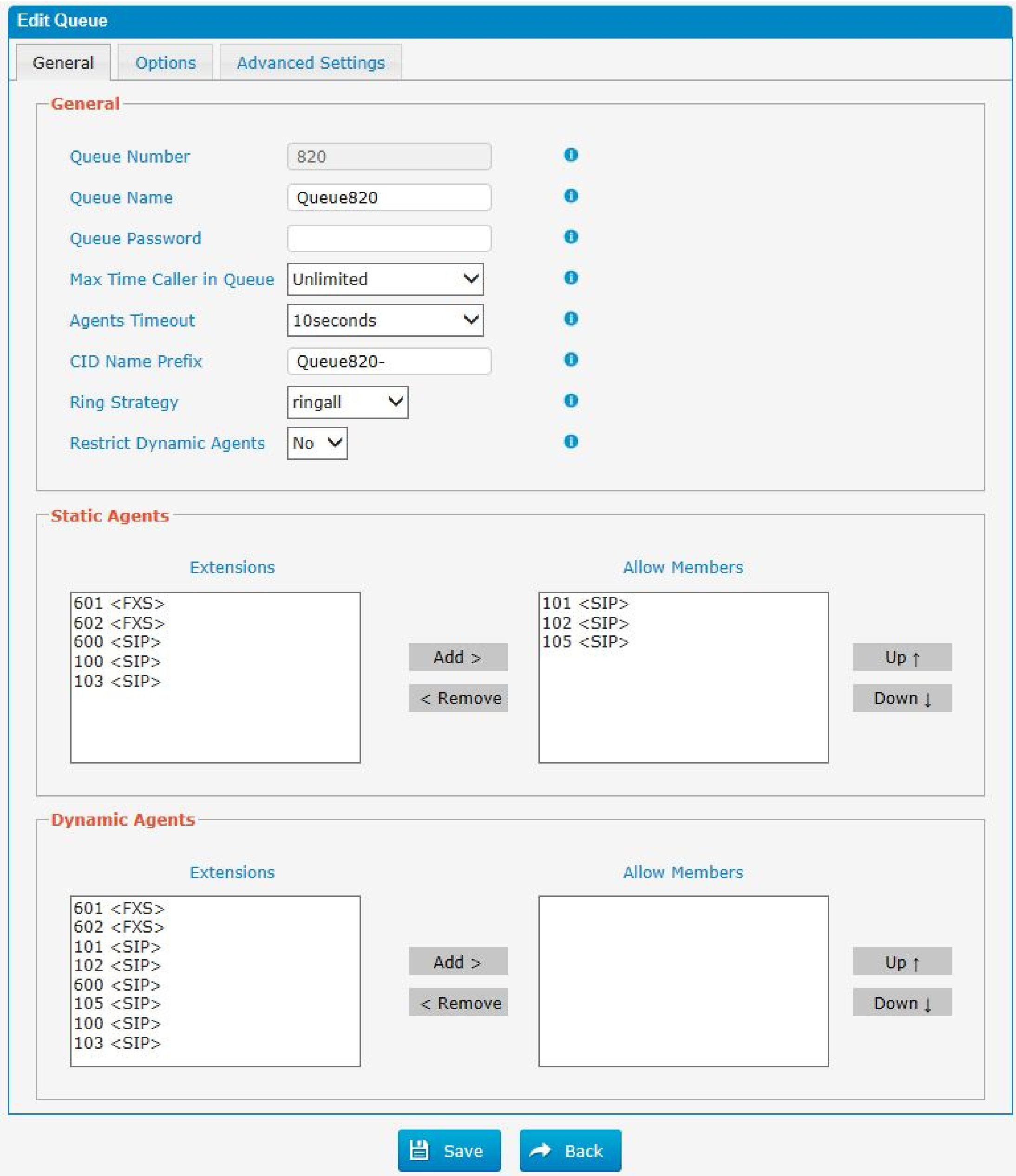


Table 3.7.4a Description of Queue General

Parameters	**Description**
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.
Max Time Caller in Queue	The maximum number of seconds a caller can wait in a queue before being pulled out (0 for unlimited).
Agents Timeout	The number of seconds an agent's phone can ring before we consider it a timeout.
CID Name Prefix	CID preifx name
Alert Info	Alert info can be used for distinctive ring with SIP devices.
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.
Restrict Dynamic Agents	Restrct dynamic agents
Static Agents	This selection shows all users. Selecting a user here makes them an agent of the current queue.
Dynamic Agents	Select dynamic agents

Figure 3.7.4b Queue Options

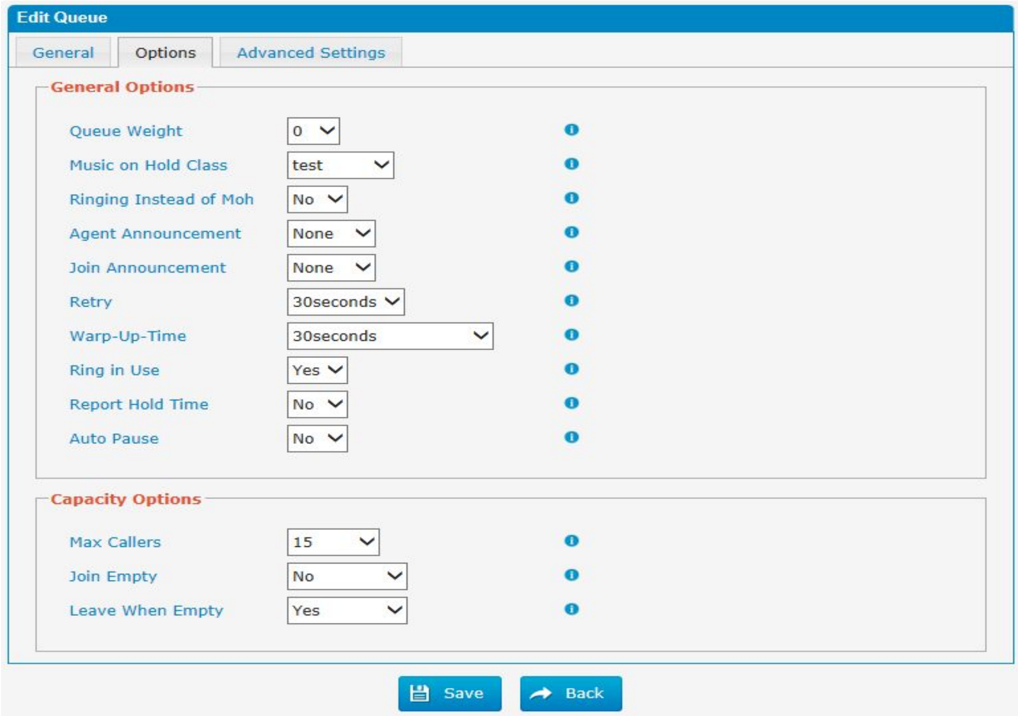


Table 3.7.4b Description of Queue Options

Parameters	**Description**
Queue Weight	Gives queues a 'weight' option, to ensure calls waiting in a higher priority queue will deliver its calls first if there are agents common to both queues.
Music on Hold Class	Music (MoH) played to the caller while they wait in line for an available agent.
Ringing Instead of Moh	Enabling this option make callers hear a ringing tone instead of Music on Hold.
Agent Announcement	Announcement played to the Agent prior to bridging in the caller
Join Announcement	Announcement played to callers prior to joining the queue.
Retry	The number of seconds we wait before trying all the phones again.
Warp-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.(0 for no delay).
Ring In Use	If set to no,the queue will avoid sending calls to members whose devices are known to be 'in use'.
Report Hold Time	If you wish to report the caller's hold time to the member before they are connected to the caller, set this to yes.

Max Callers	Maximum number of people waiting in the queue.
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.
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.

Figure 3.7.4c Queue Advanced Settings

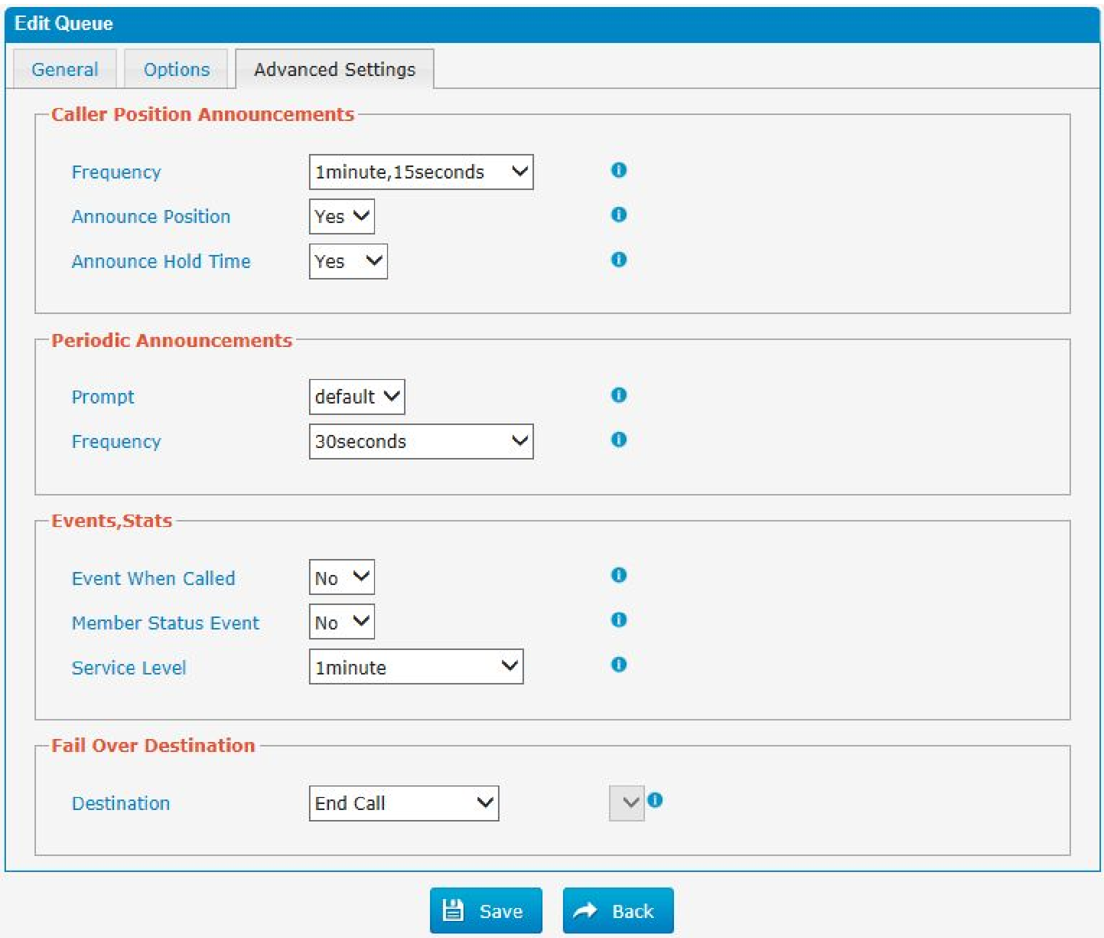


Table 3.7.4c Description of Queue Advanced Settings

Parameters	**Description**
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 PBX 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.
Prompt	Select a prompt file to play periodically.
Frequency	How often to announce a prompt to the caller.
Event When Called	If a caller presses the key while waiting in the queue, this setting selects which action should process the key press.
Member Status Event	
Service Level	
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 Grounps

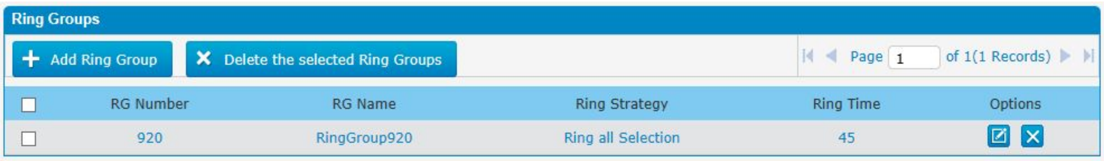


Figure 3.7.5a Ring Groups Edit

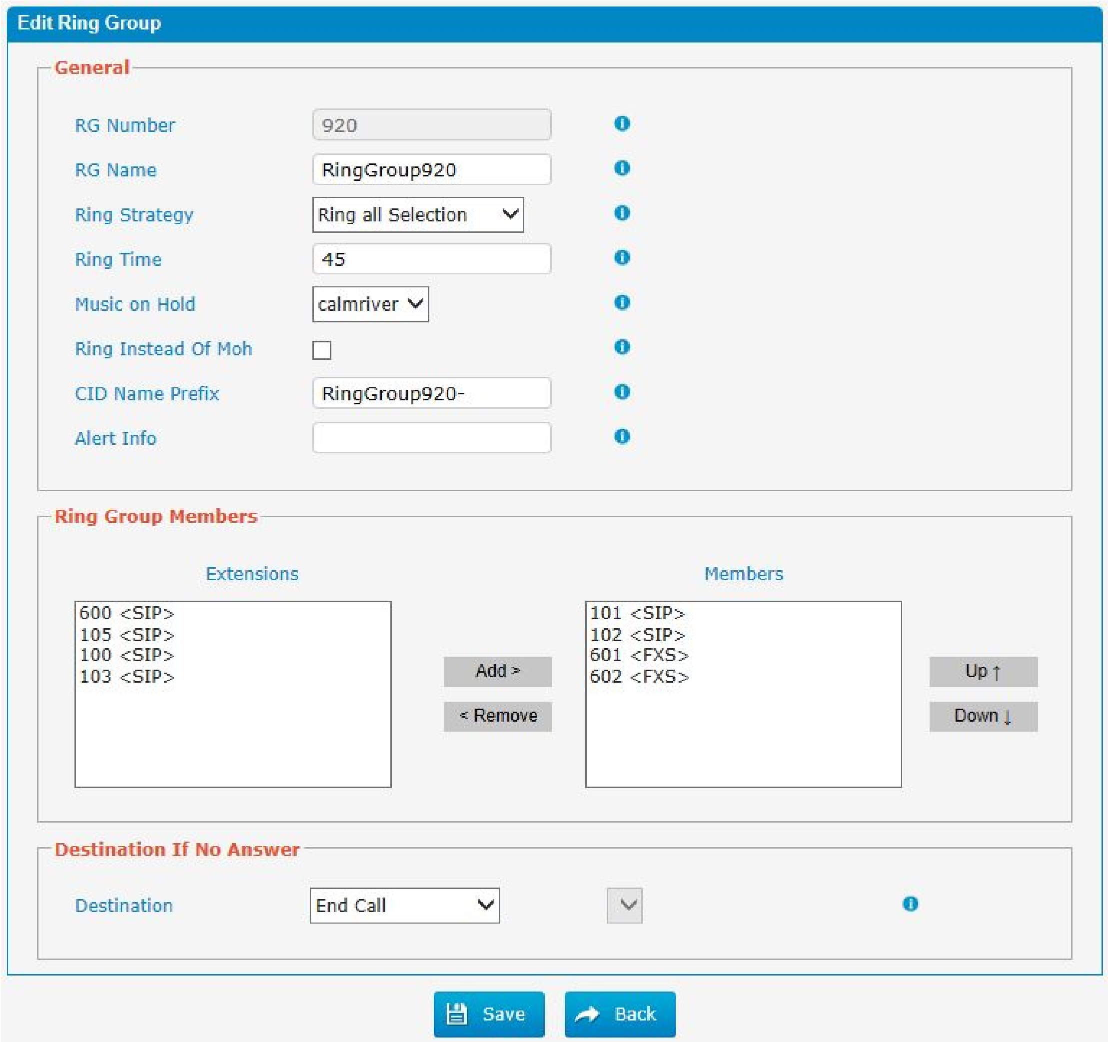


Table 3.7.5a Description of Ring Groups Edit

Parameters	**Description**
RG Number	This option defines the numbered extension that can be dialed to reach this group.
RG 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: ● Ring All Simultaneously: Ring all available Extensions simultaneously. ● 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 instead Of Moh	Enabling this option make callers hear a ringing tone instead of Music on Hold.
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.
Alert Info	Alert info can be used for distinctive ring with SIP devices.

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

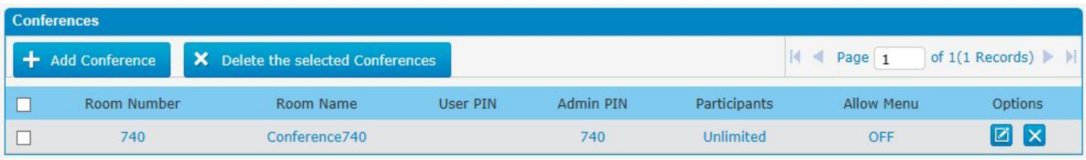


Figure 3.7.6a Conferences Edit/Add

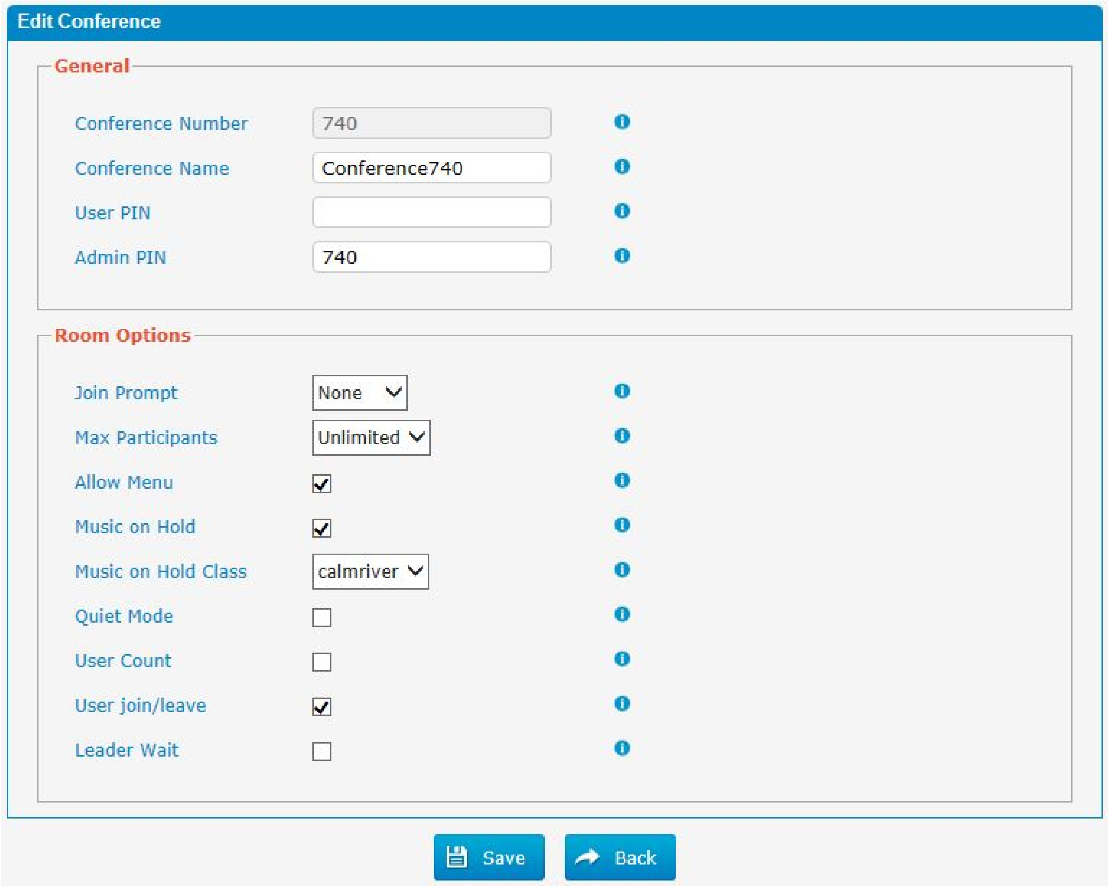


Table 3.7.6a Description of Conferences Edit/Add

Parameters	**Description**
Conference Number	This is the number dialed to reach this Conference Room.
Conference Name	This option defines a name for this conference, e.g. "Sales". "Conference Name" is a label to help you identify this conference in the conference list.
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
Join Prompt	Message to be played to the caller before joining the conference.
Max Paticipants	Maximum Number of users allowed to join this conference.
Allow Menu	Present Menu (user or admin) when '*' is received ('send' to menu)
Music on Hold	Enable Music On Hold when the conference has a single

	caller.
Muisc on Hold Class	Music (or Commercial) played to the caller while they wait in line for the conference to start.
Quiet Mode	Quiet mode (do not play enter/leave sounds)
User Count	Announce user(s) count on joining conference
User join/leave	Announce user join/leave
Leader Wait	Wait until the conference leader (admin user) arrives before starting the conference.

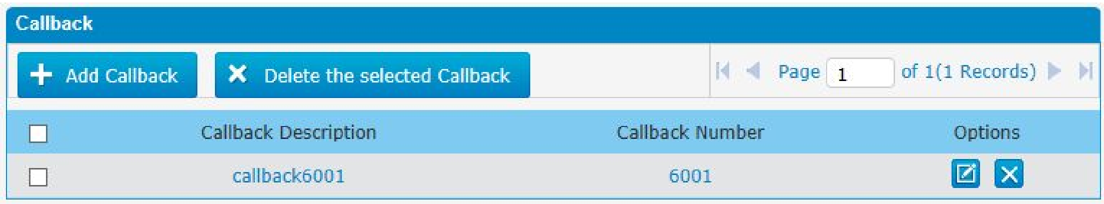
3.7.7 Callback

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

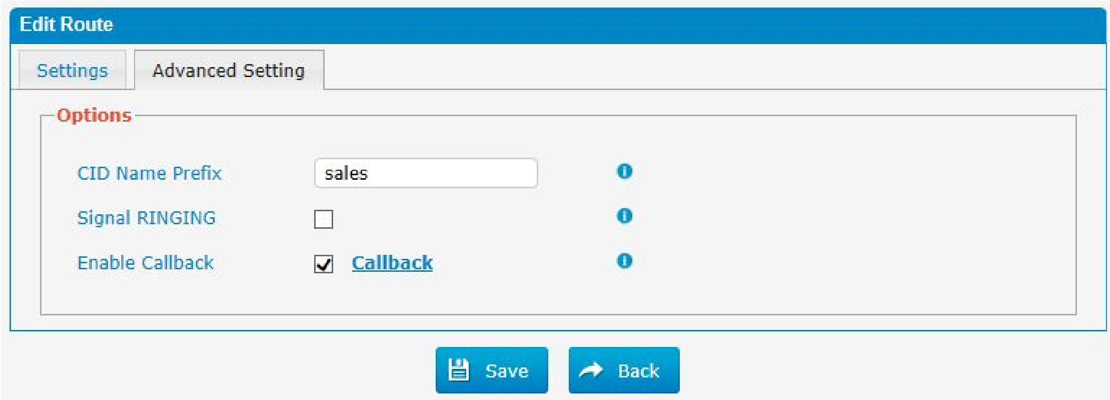


Follow the steps below to use this function.

Step 1: Enable Callback.

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

Figure 3.7.7a Inbound route Callback settings



Step 2: Create Callback number.

Figure 3.7.7b Callback Edit/Add

3.8 PBX Advanced Settings

3.8.1 SIP settings

This is the SIP settings in PBX, 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 General

Figure 3.8.1.1 SIP General Setting

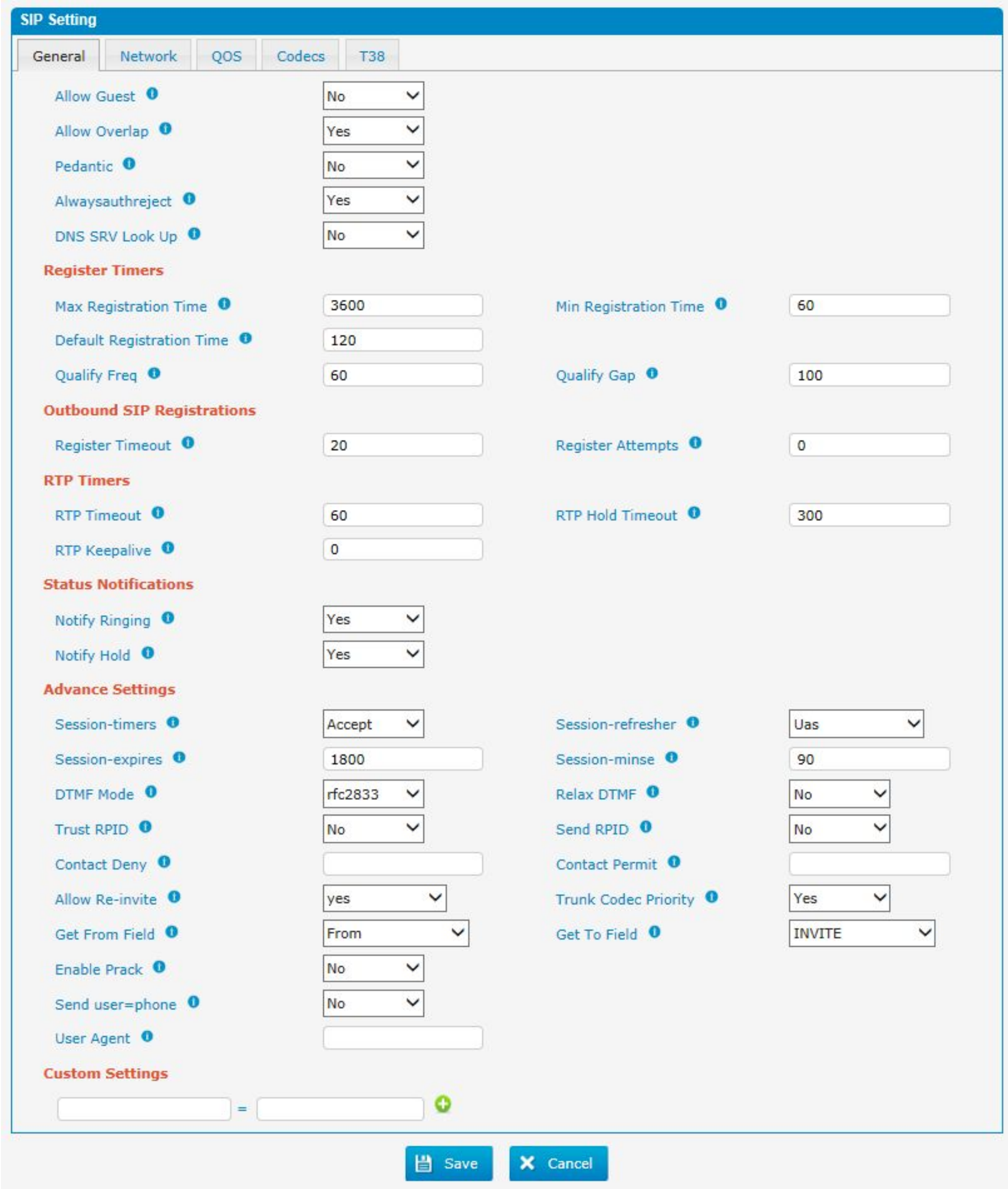


Table 3.8.1.1 Description of SIP General Setting

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

3.8.1.2 Network

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

Figure 3.8.1.2 SIP Network Configuration

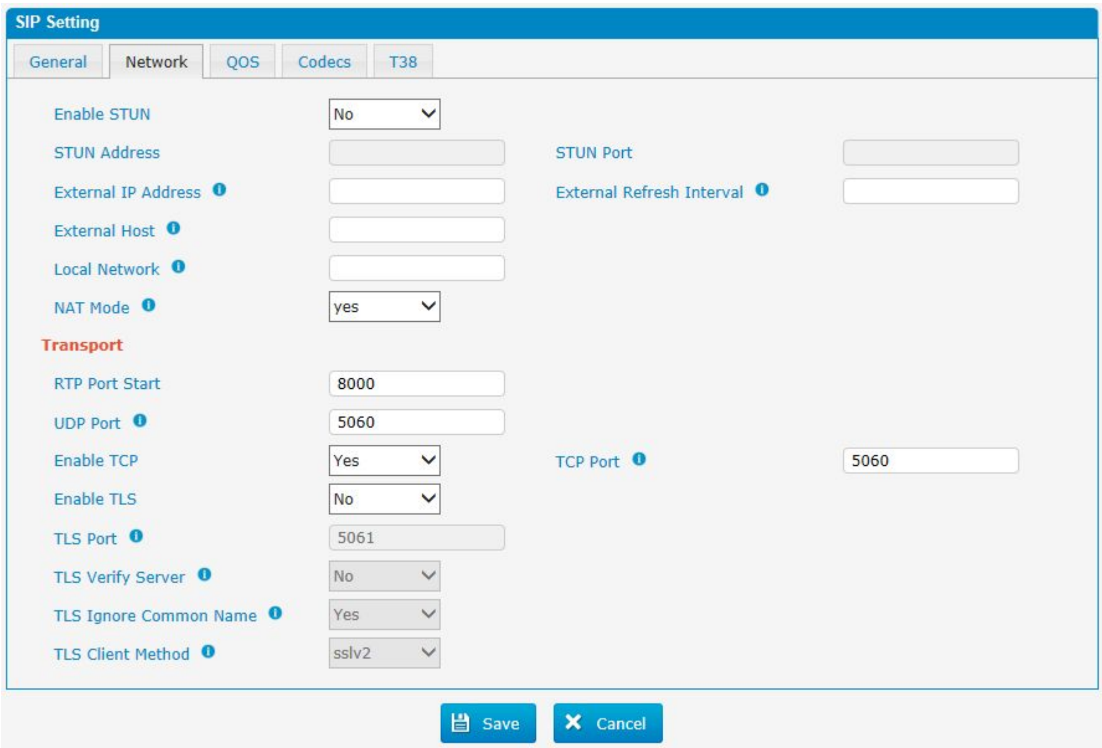


Table 3.8.1.2 Description of SIP Network Configuration

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

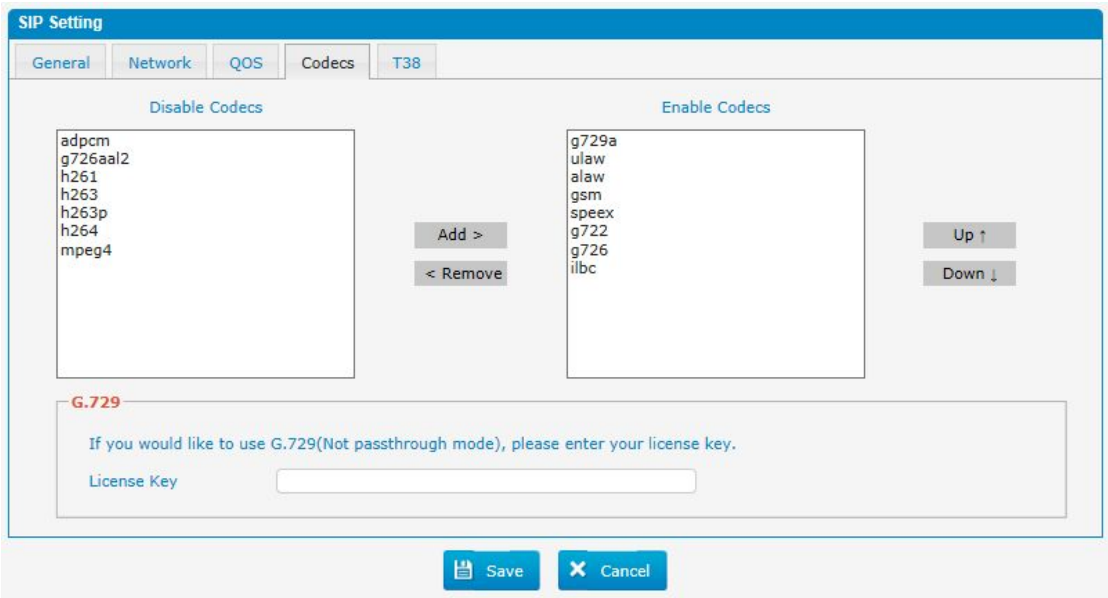
3.8.1.3 Qos

Figure 3.8.1.3 Qos

3.8.1.4 Codecs

We can choose the allowed codec in PBX, 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.

3.8.1.5 T.38

Figure 3.8.1.5 T.38

3.8.2 IAX Setting

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

Figure 3.8.2 IAX setting

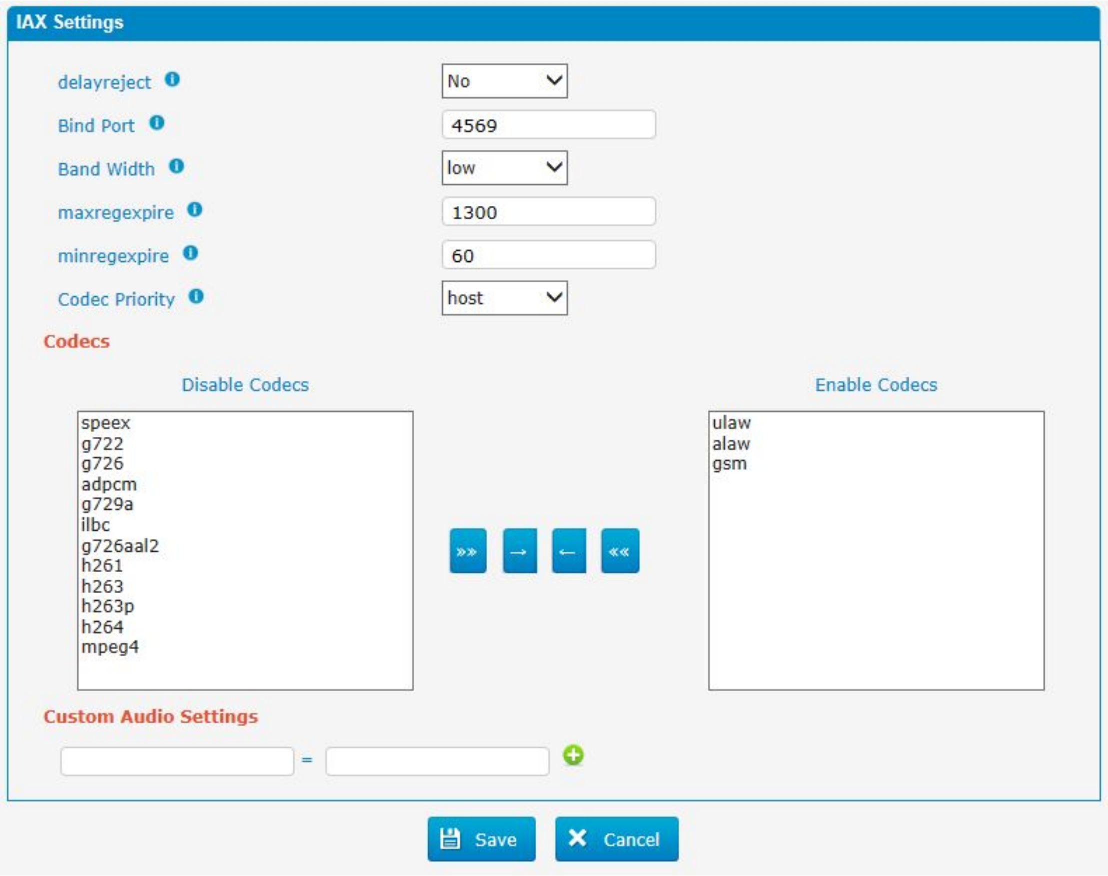


Table 3.8.2 Description of IAX setting

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

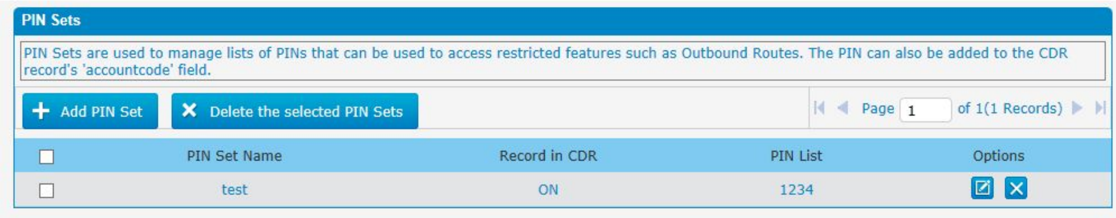


Figure 3.8.3a PIN Set Edit

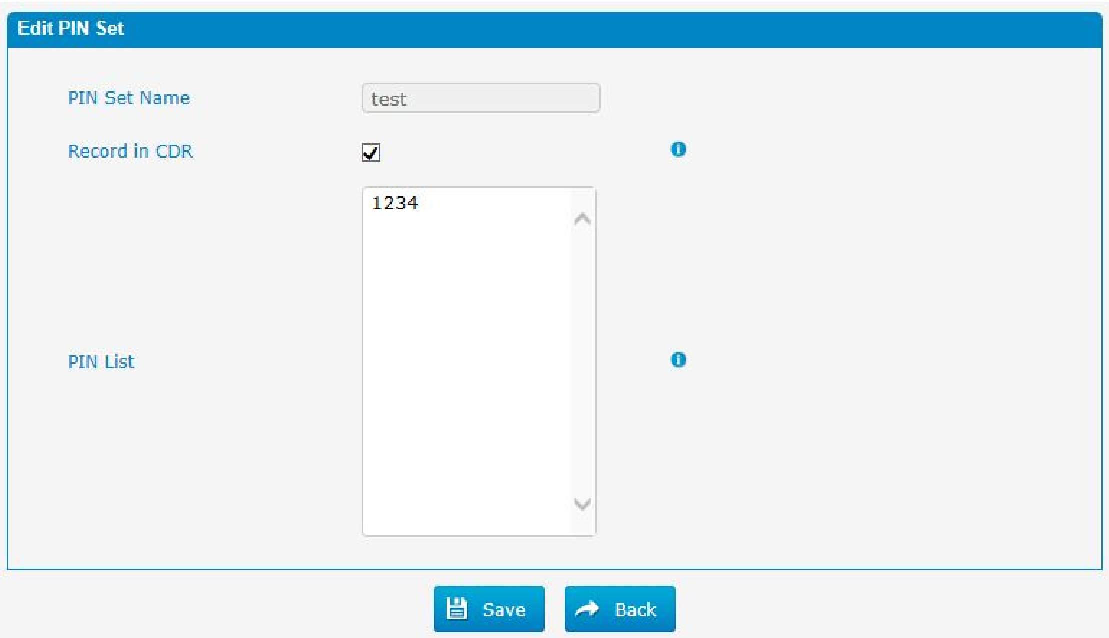


Table 3.8.3a Description of PIN Set Edit

Parameters	**Description**
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 PIN Users

Figure 3.8.4 PIN Users

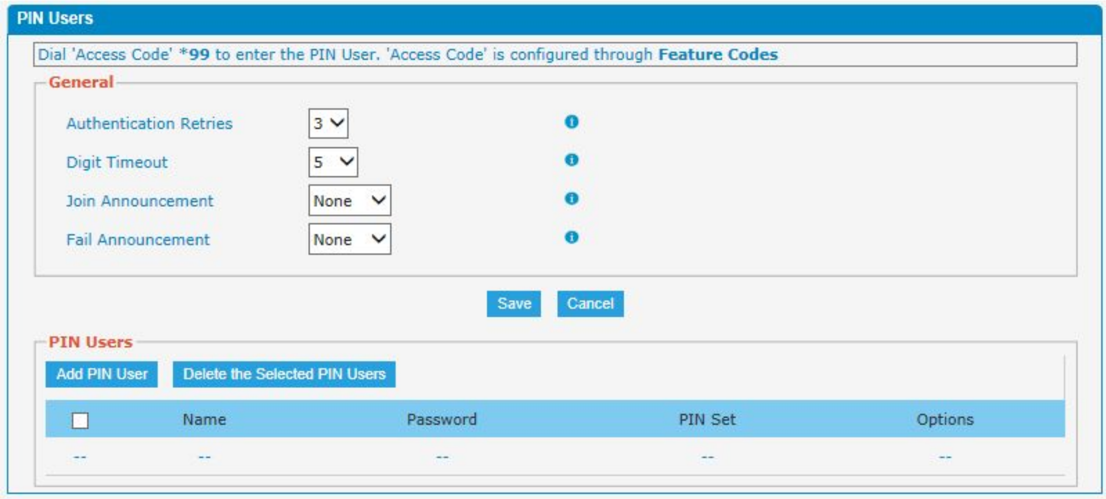


Table 3.8.4 Description of PIN Users

Parameters	**Description**
Authentication Retries	Number of times to retry when receiving an wrong password.
Digit Timeout	The maximum amount of time permitted between digits when the user is typing in an extension. Default of 5 seconds.
Join Announcement	Waiting for validation, the system will play the prompt.
Fail Announcement	After validation fails, the system will play the prompt.

Figure 3.8.4a PIN Users Add/Edit

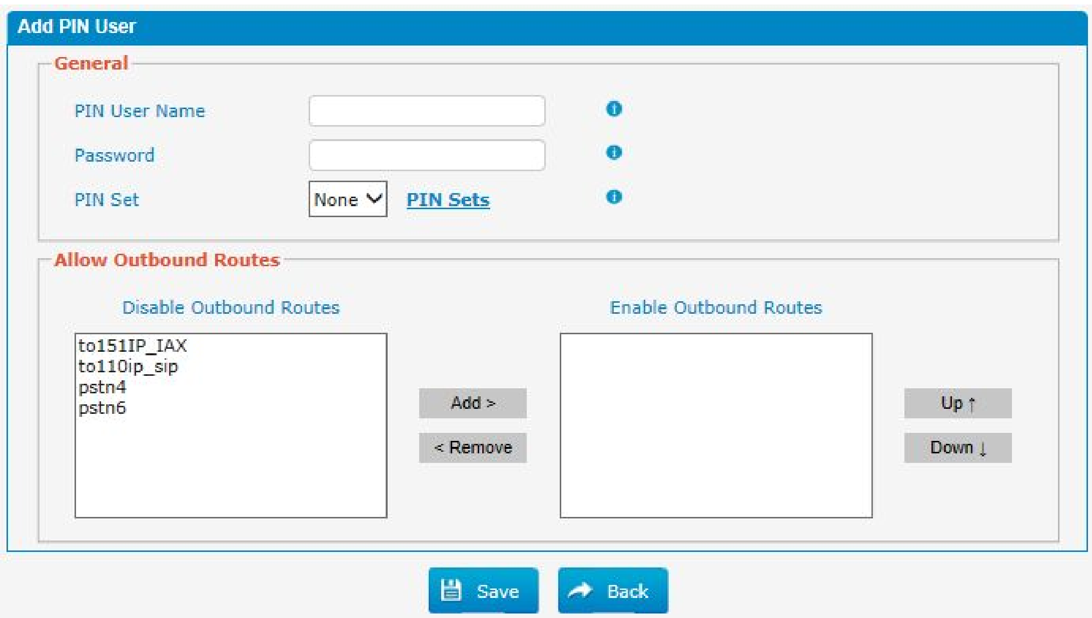


Table 3.8.4a Description of PIN Users Add/Edit

Parameters	**Description**
PIN User Name	A character-based name for this PIN list, e.g. "NEXOPIN" PBX can store a number of PIN Users. PIN Users may be used to keep track of calls in relation to particular activities or clients. They can also be used to keep track of calls by particular users or sets of users. ● PINs entered are checked against those stored by the system. If an invalid PIN is entered, the PIN is requested again. ● The system administrator can configure certain numbers or types of numbers to require entry of a PIN before users can continue making a call to such a number. ● The system administrator can also configure to require users to enter a PIN before making any external call.
Password	The password for this PIN User.
PIN Set	Click to add, delete or edit PIN list.
Allow Outbound Routes	PIN User can use those outbound route to make call out.

3.8.5 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 3.8.5 DISA

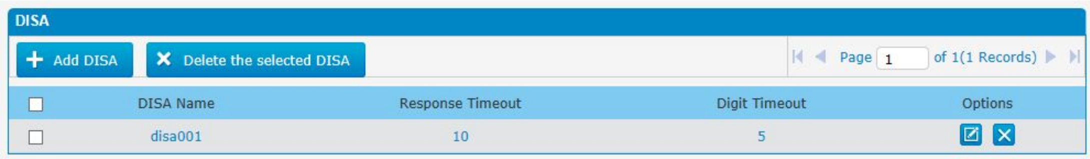


Figure 3.8.5 DISA Edit

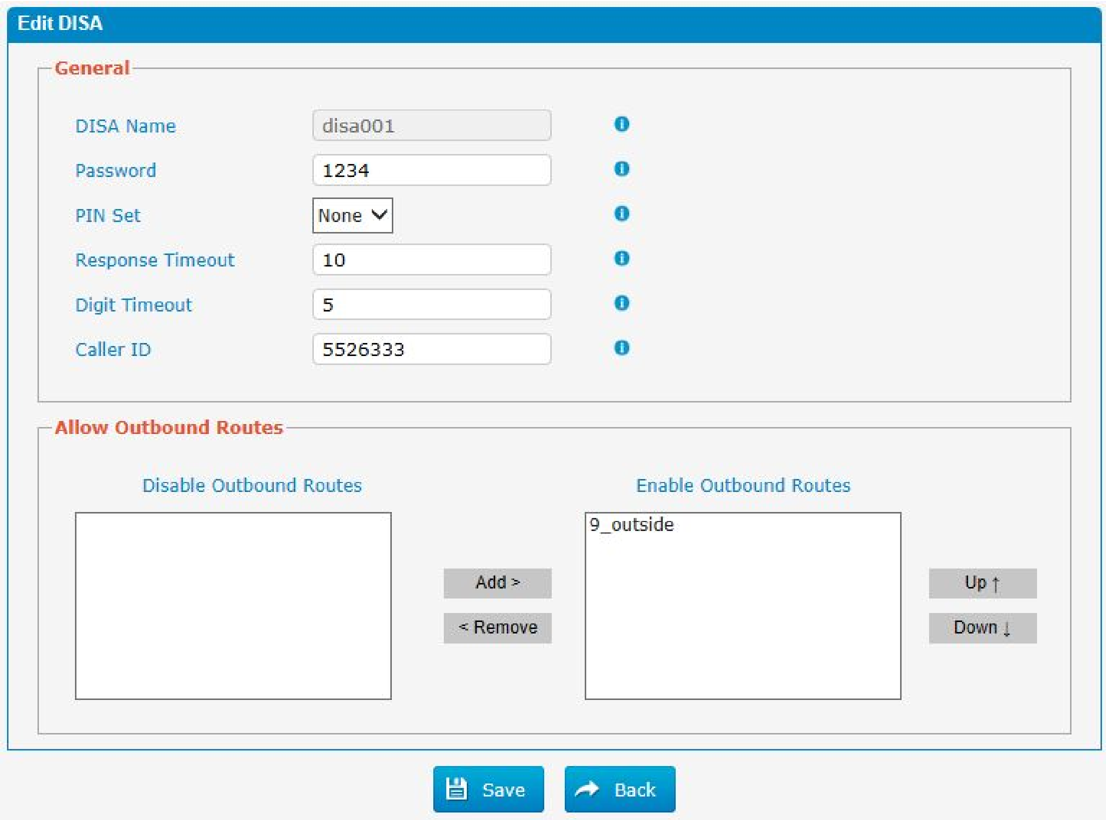


Table 3.8.5 Description of DISA Edit

Parameters	**Description**
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.
Caller ID	(Optional) When using this DISA, the users CallerID will be set to this. Format is "User Name" <5551234>.
Allow Outbound Routes	Used to set the outbound routes that can be accessed from this DISA.

3.8.6 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 3.8.6 Paging and Intercom

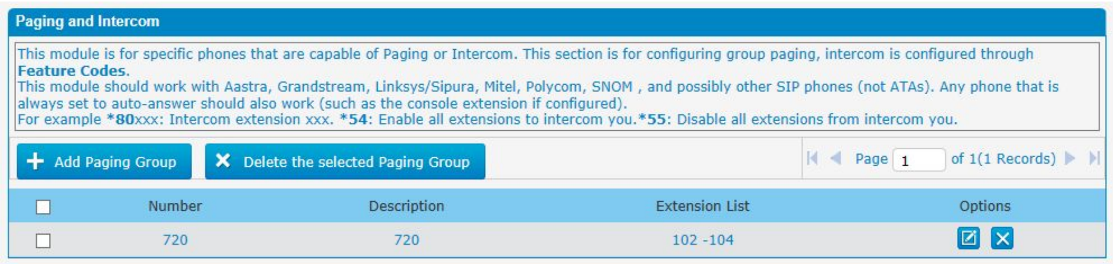
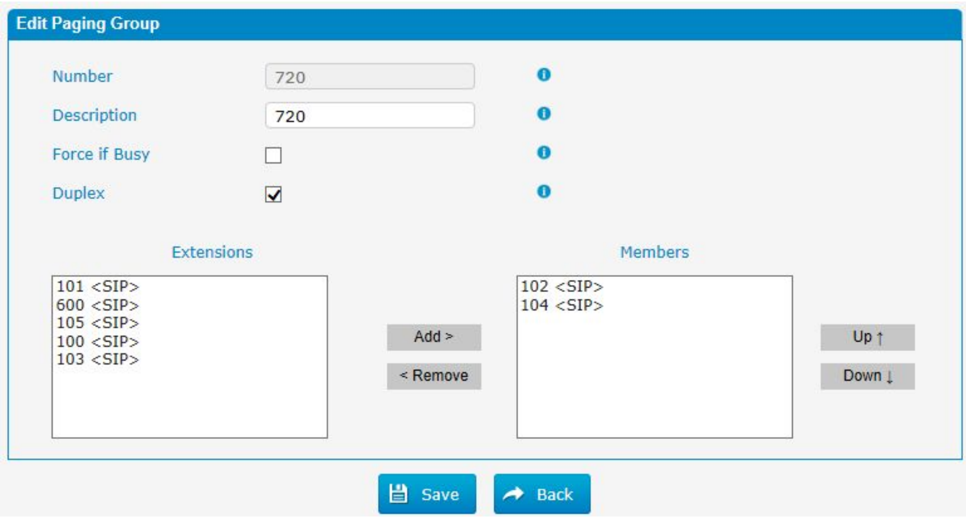


Figure 3.8.6a Paging and Intercom Edit/Add



Parameters	**Description**
Number	Define the numbered extension that may be dialed to reach this group.
Description	The description of this paging group.
Force if Busy	If selected, will not check if the device is in use before paging it.
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.
Members	Select members in this group.

3.9 Voice Management

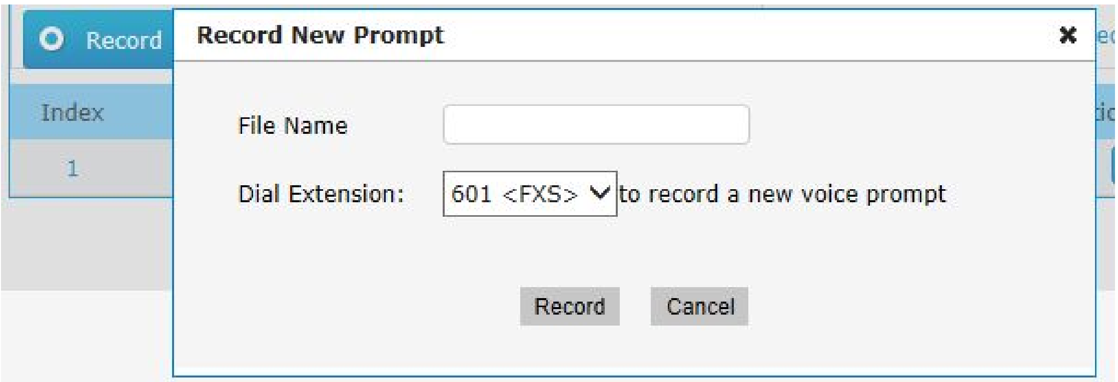
3.9.1 System Recordings

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

1. Record New Prompt

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).

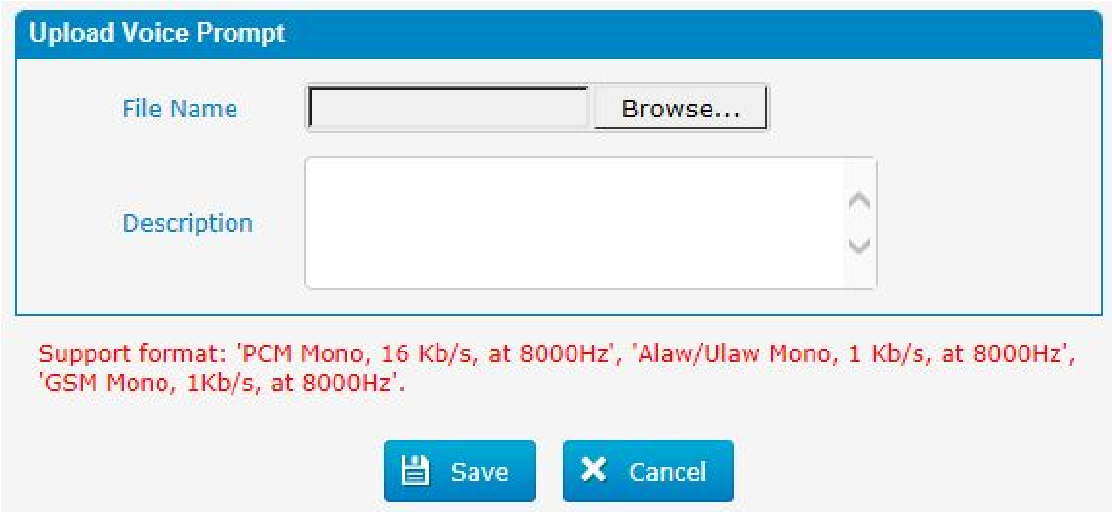
3) Click "Record". The selected extension will ring and you can pick up the phone to start recording.

2. Upload

Prompt Click

"Upload"

Figure 3.9.1b Upload Voice Prompt



The administrator can also upload prompts by doing the following:

1) Click "Upload Prompt".

2) Click "Browse" to choose the desired prompt.

3) Click "Save" to upload the selected prompt.

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

3.9.2 Music on Hold

Figure 3.9.2 Music on Hold

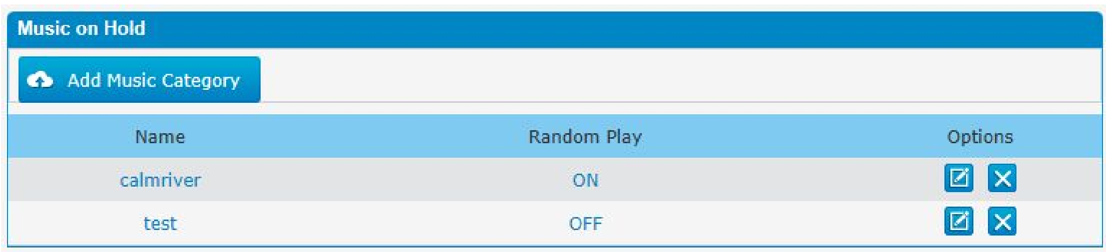
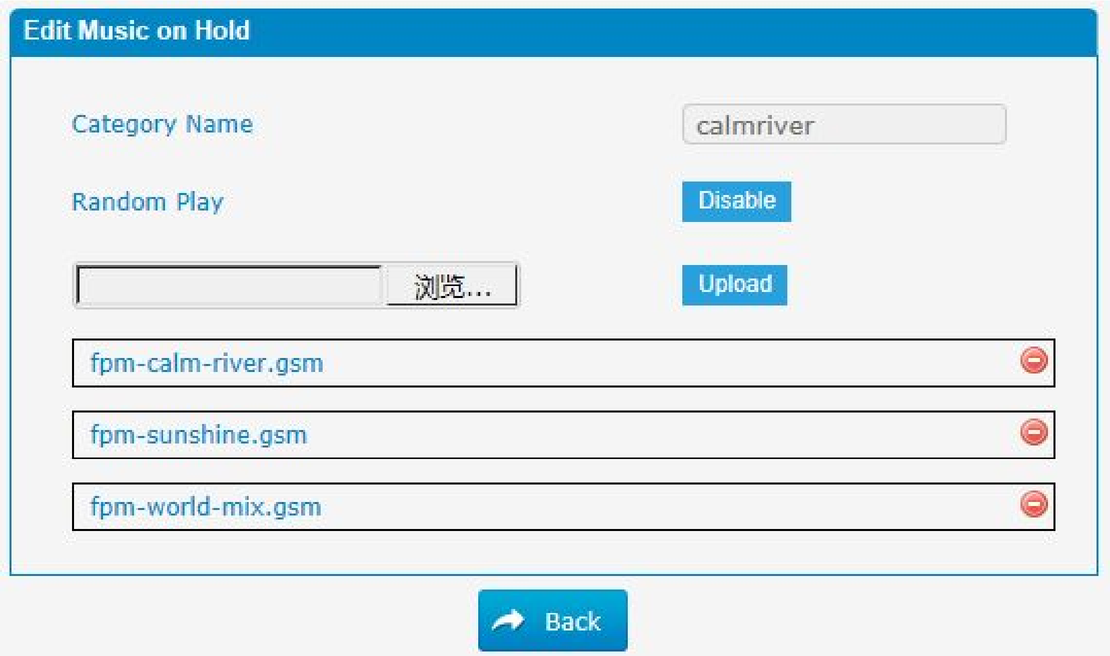


Figure 3.9.2a Music on Hold Edit



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

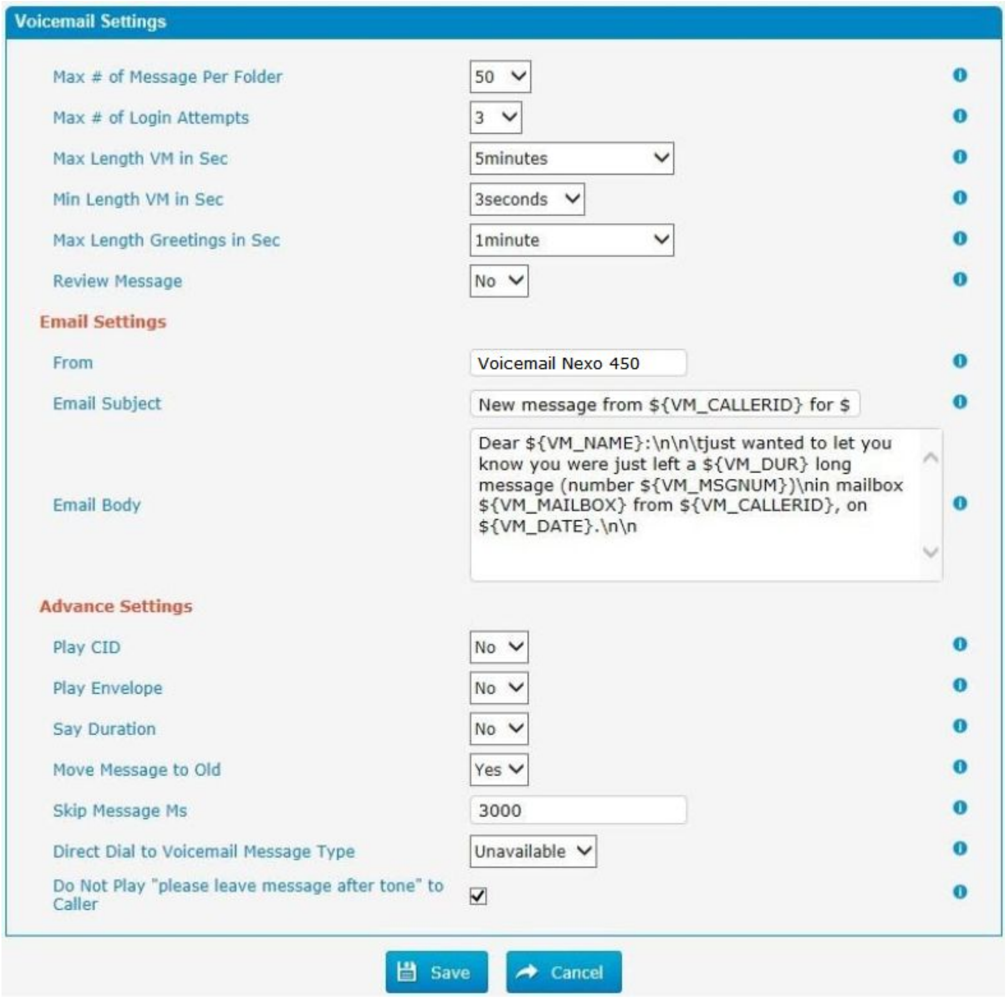


Table 3.9.3 Description of Voicemail Setting

Parameters	**Description**
Max # of Message 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	Set the maximum length of a single voicemail message.

Sec	
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)
From	Email from
Email Subject	Email subject
Email Body	Email body
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
Skip Message Ms	Specifies how many milliseconds to skip forward/back when the user skips forward or backward during message playback.
Direct Dial to Voicemail Message Type	Default message type to use when dialing direct to an extensions voicemail
Do Not Play "please leave message after tone" to Caller	Do Not Play "please leave message after tone" to Caller

3.9.4 System Prompts Settings

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

Figure 3.9.3 System Prompts Settings Upload

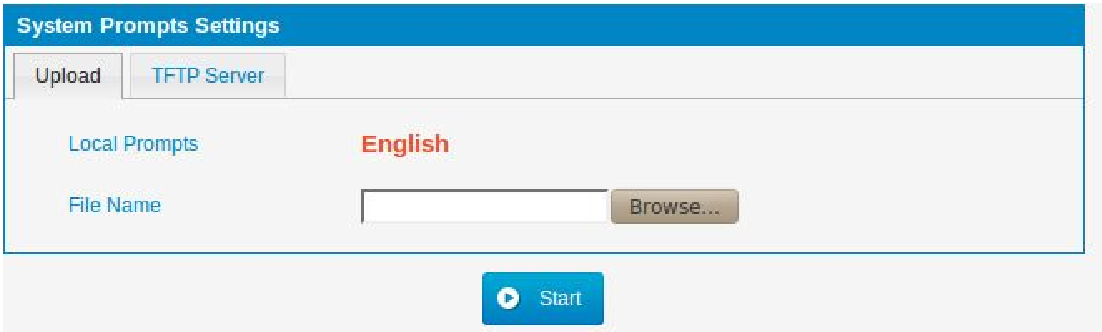


Figure 3.9.3a System Prompts Settings TFTP

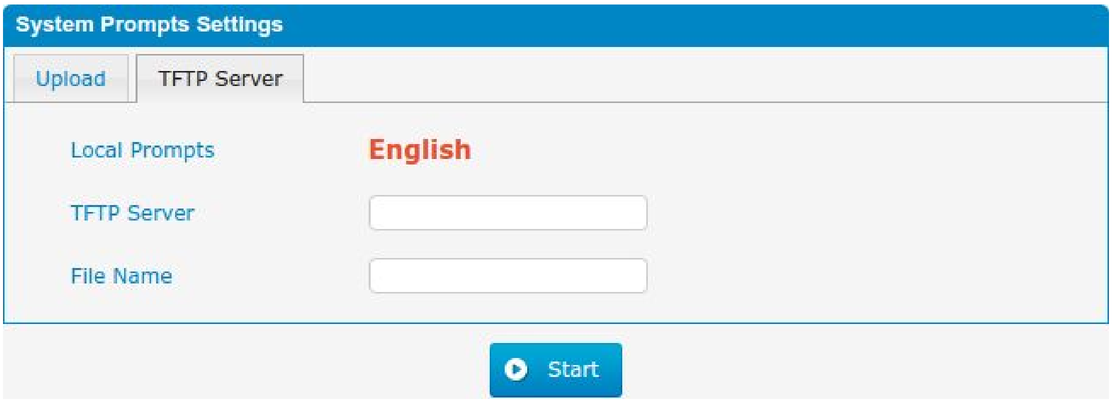


Table 3.9.3 Description of System Prompts Settings

Parameters	**Description**
File Name	Choose a country voice package,filename must to '.tar.gz' ending.
TFTP Server	Tftp service server.

3.10 System Preferences

3.10.1 Firewall Rules

Figure 3.10.1 Firewall Rules

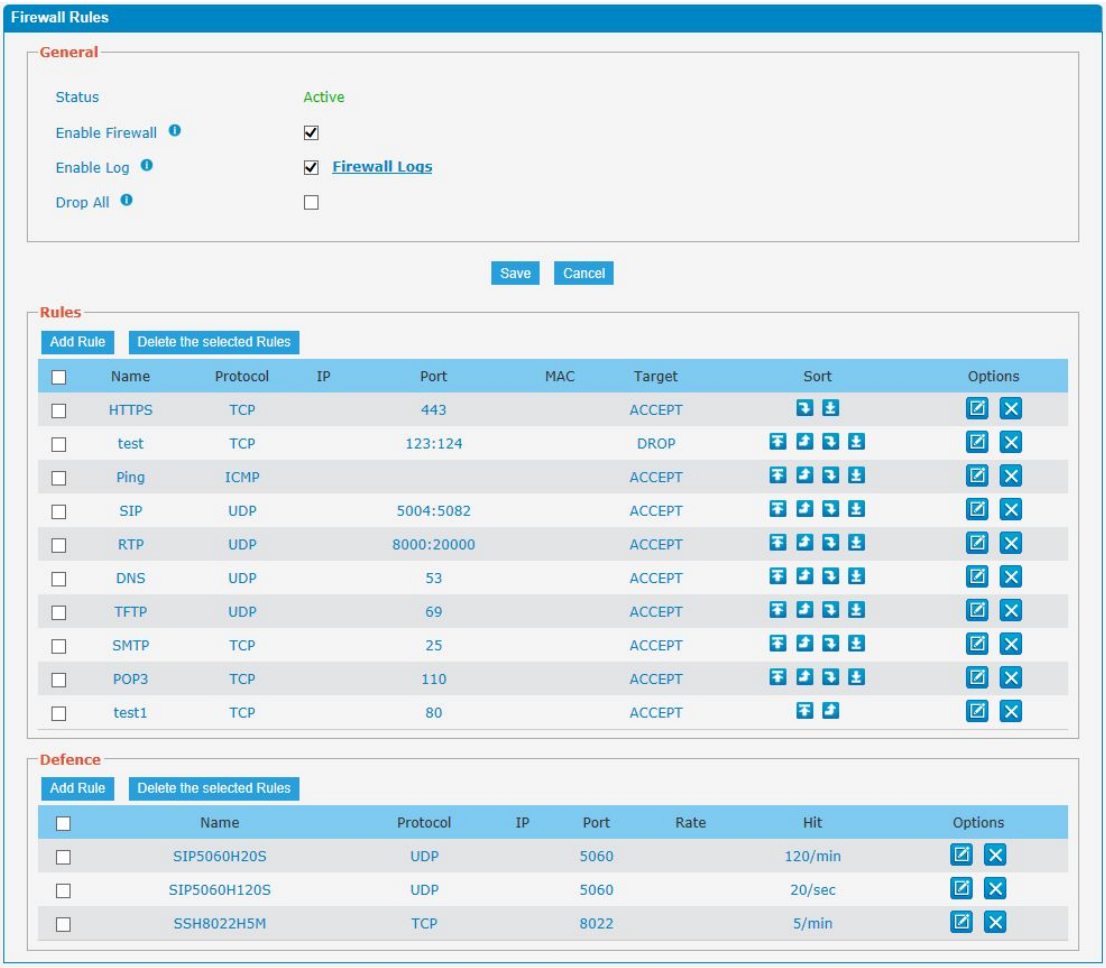


Figure 3.10.1a Firewall Rules Edit/Add

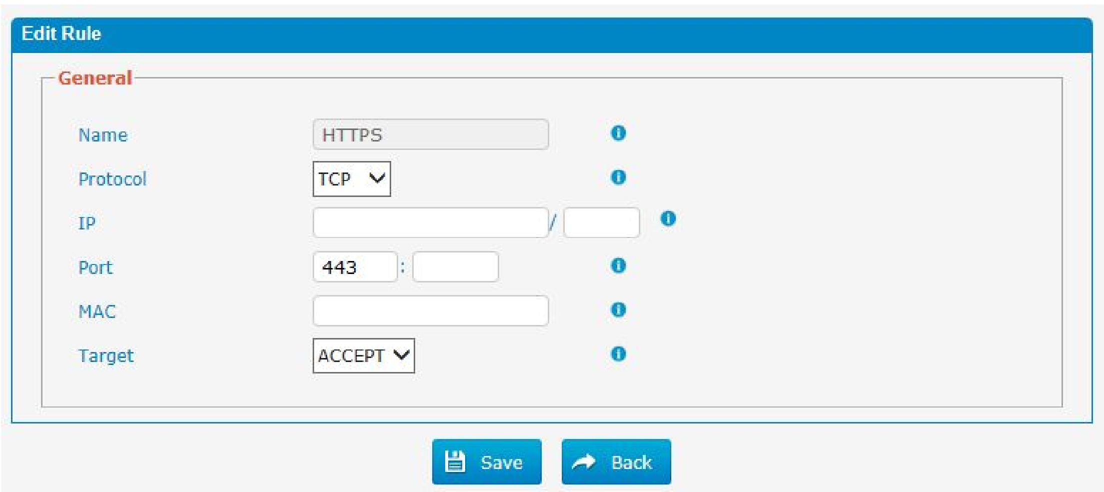


Table 3.10.1a Description of Firewall Rules

Parameters	Description
Name	A name for this rule. eg: HTTP.
Protocol	The protocols for this rule.
IP	The IP address for this rule. The format of IP address is:IP/mask ● Ex:192.168.6.88/32 for ip 192.168.6.88 ● Ex:192.168.6.0/24 for ip from 192.168.6.0 to 192.168.6.255
Port	Initial port should be on the left and end port should be on the right.The port must be equal to or greater than start port.
MAC	The format of MAC Address is XX:XX:XX:XX:XX:XX, X means 0~9 or A~F in hex, the A~F are not case sensitive.
Target	● ACCEPT:Accept the access from remote hosts ● DROP:Drop the access from remote hosts ● REJECT:Reject the access from remote hosts

Figure 3.10.1b Firewall Defence Edit/Add

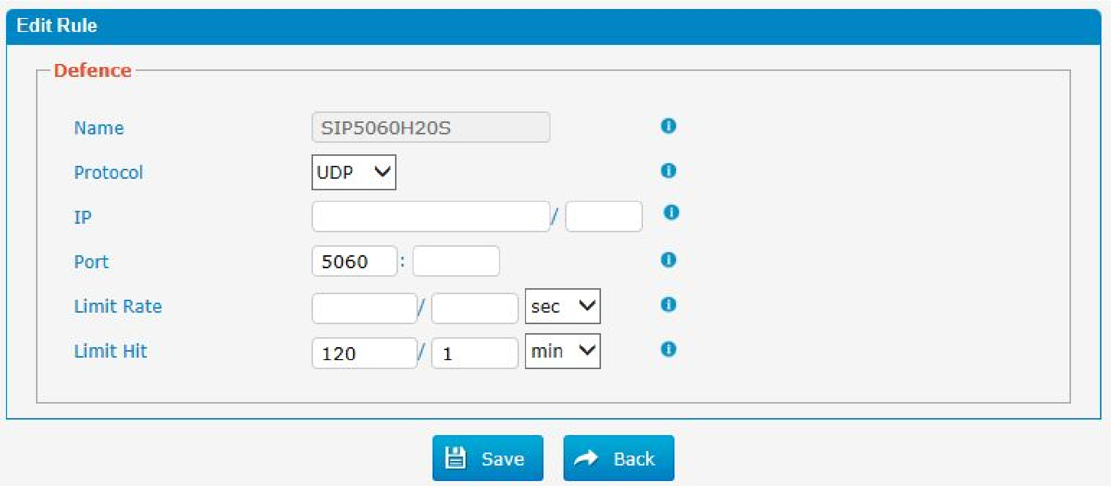


Table 3.10.1b Description of Firewall Defence Edit/Add

Parameters	Description
Name	A name for this rule. eg: HTTP.
Protocol	The protocols for this rule.
IP	The IP address for this rule. The format of IP address is:IP/mask ● Ex:192.168.6.88/32 for ip 192.168.6.88 ● Ex:192.168.6.0/24 for ip from

	192.168.6.0 to 192.168.6.255
Port	Initial port should be on the left and end port should be on the right.The port must be equal to or greater than start port.
Limit Rate	The maximum packets can be handled per unit time. eg:(IP:192.168.6.88/32 Protocol:UDP Rate:10/sec)means maximum 10 UDP packets from 192.168.6.88 can be handled per minute,and drop the redundant packets.
Limit Hit	The maximum connections can be handled per unit time. eg:(Port:8022 Protocol:TCP Hit:10/minute)means maximum 10 TCP connections to port 8022 can be handled per minute,the eleventh connection will be refused directly.

3.10.2 Security Info

Alert Settings, if the device is attacked, the system will notify users the alert via call or E-mail. the attack modes include IP attack and Web Login.

Figure 3.10.2 Alert Settings

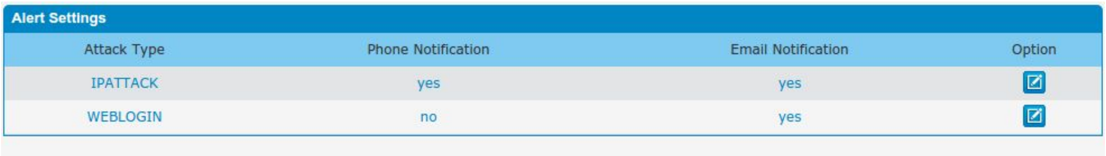


Figure 3.10.2a Alert Settings Edit

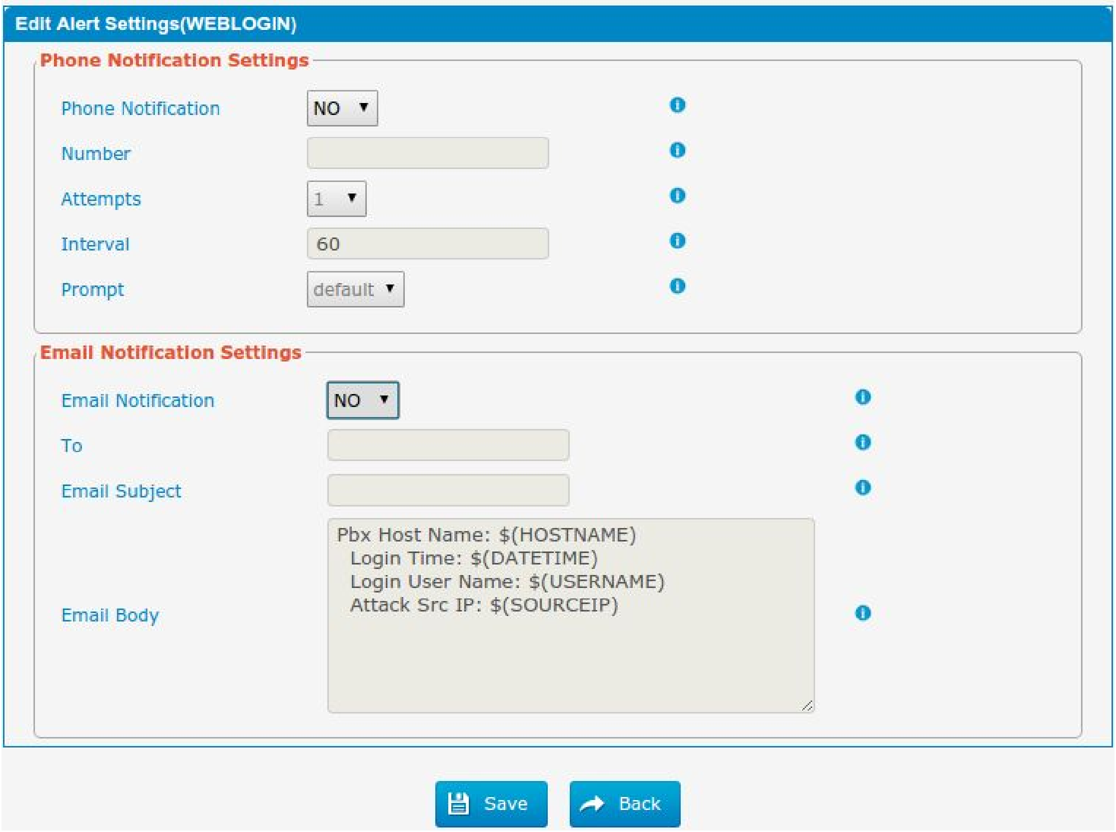


Table 3.10.2a Description of Alert Settings Edit

Parameters	Description
Phone Notification	Enable phone notification
Number	Multiple extensions and outbound phone numbers could be set for alert phone notification. Please separate them by ';', e.g. '103;9XXX'.
Attempts	The attempt times to dial a phone number when there is no answer.
Interval	The interval between each attempt to dial the phone number.Must be greater than 3 seconds.
Prompt	When answered, System will play this prompt.
Email Notification	Enable email notification
To	Multiple email addresses are allowed; please separate them by ';', e.g. XXXX@gmail.com; YYYY@hotmail.com.
Email Subject	Email subject
Email Body	Email Body, Until 511 characters

IP Blacklist, if the device is attacked by IP attack.system will add ip to firewall and Disable this IP access.

Figure 3.10.2b IP Blacklist

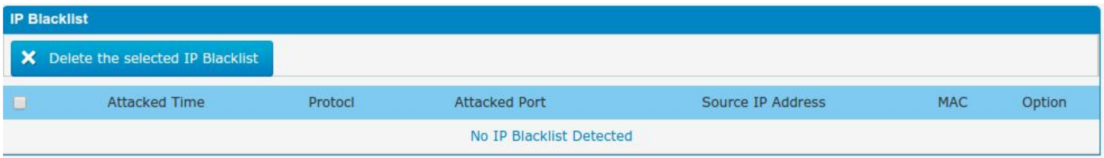


Table 3.10.2b Description of IP Blacklist

Parameters	Description
Date	IP Attack time
Protocol	Attack protocol type
IP	Attack ip
MAC Address	Attack MAC address
Dest Port	Attack destination port

3.10.3 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 3.10.3 Firmware Update Upload

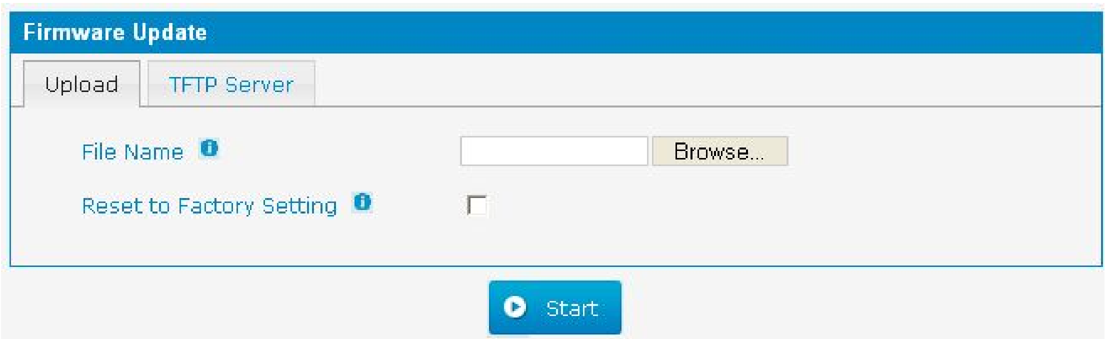


Figure 3.10.3a Firmware Update TFTP

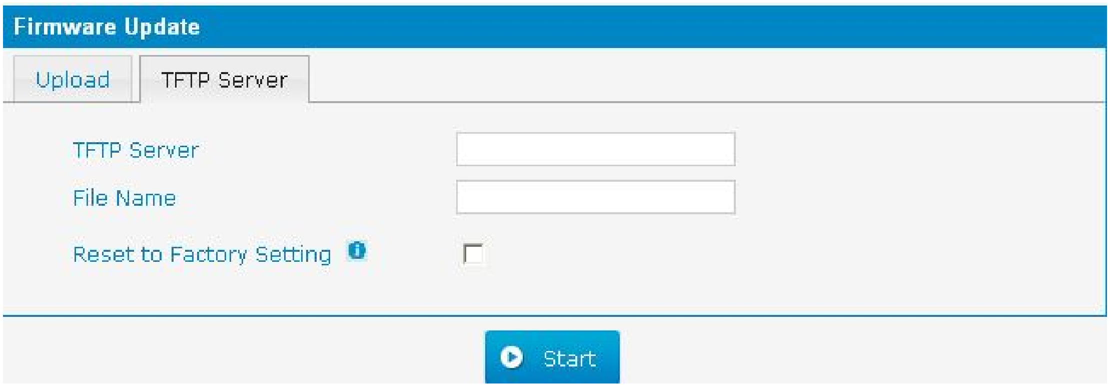


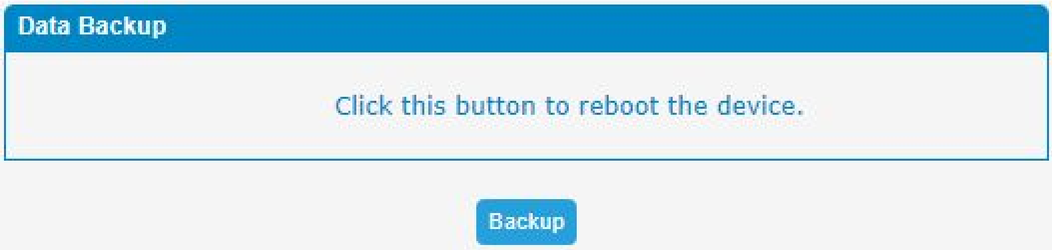
Table 3.10.3 Firmware Update

Parameters	Description
Firmware update	Send package file from your computer to the device
File name	Firmware name,file must to „.img" ending.
Reset to Factory Setting	Reset Configuration to Factory Defaults
Browse	Choose File

3.10.4 Data Backup

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

Figure 3.10.4



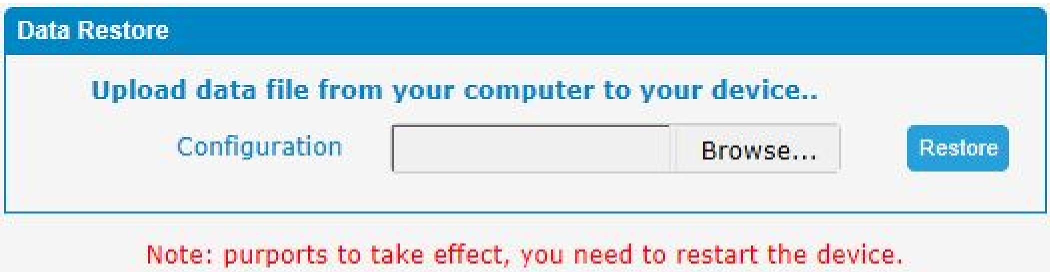
Click 'Backup' to download configuration file to your computer.
Notes:
1. Only configurations, custom prompts will be backed up.
2. When you have updated the firmware version, it"s not recommended to restore using old package.

3.10.5 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.

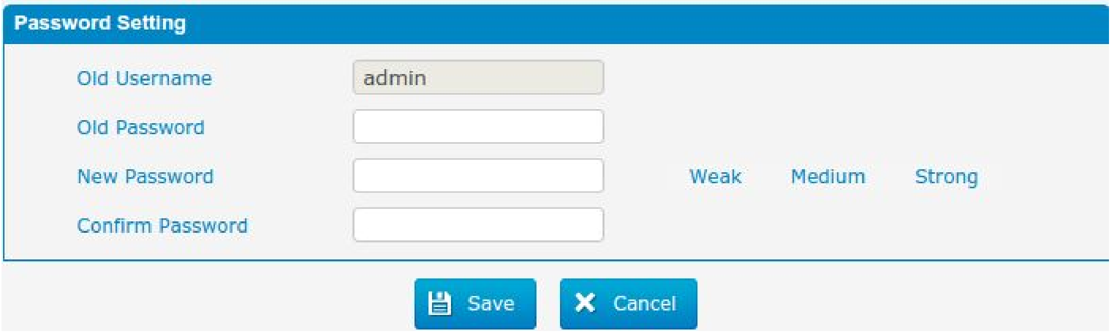
Figure 3.10.5



3.10.6 Password

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

Figure 3.10.6 Password Setting



3.10.7 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 3.10.7 Time & Date parameter

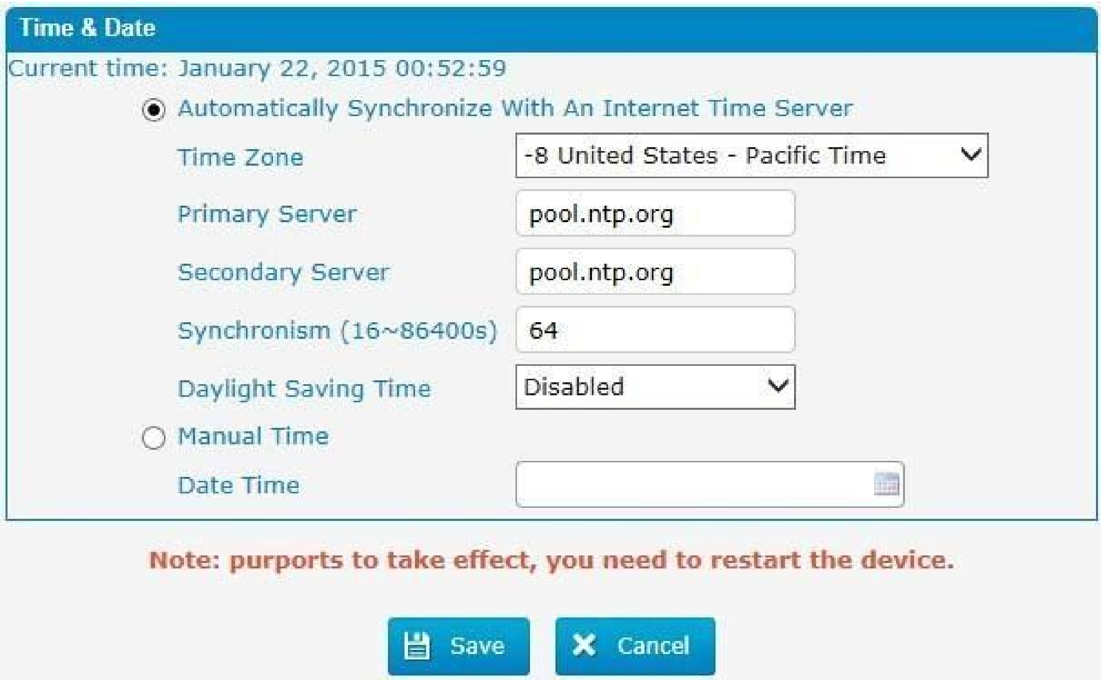


Table 3.10.7 Time & Date parameter

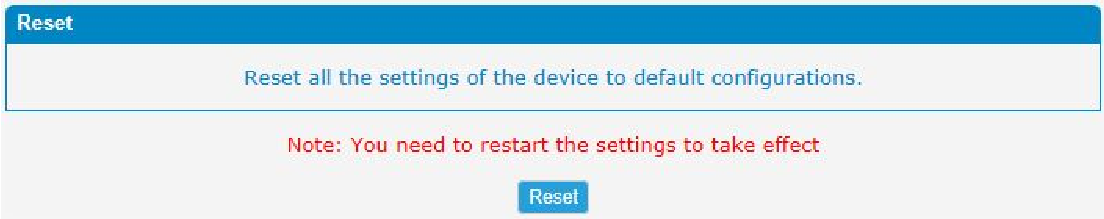
Parameters	**Description**
Time zone	You can choose your time zone here.
Primary server	Primary NTP Server Address

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
Manual Time	Manual setup time

3.10.8 Reset

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

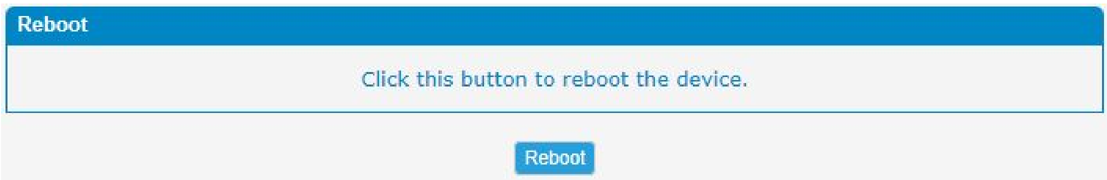
Figure 3.10.8 factory reset



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

3.10.9 Reboot

Figure 3.10.9 Reboot



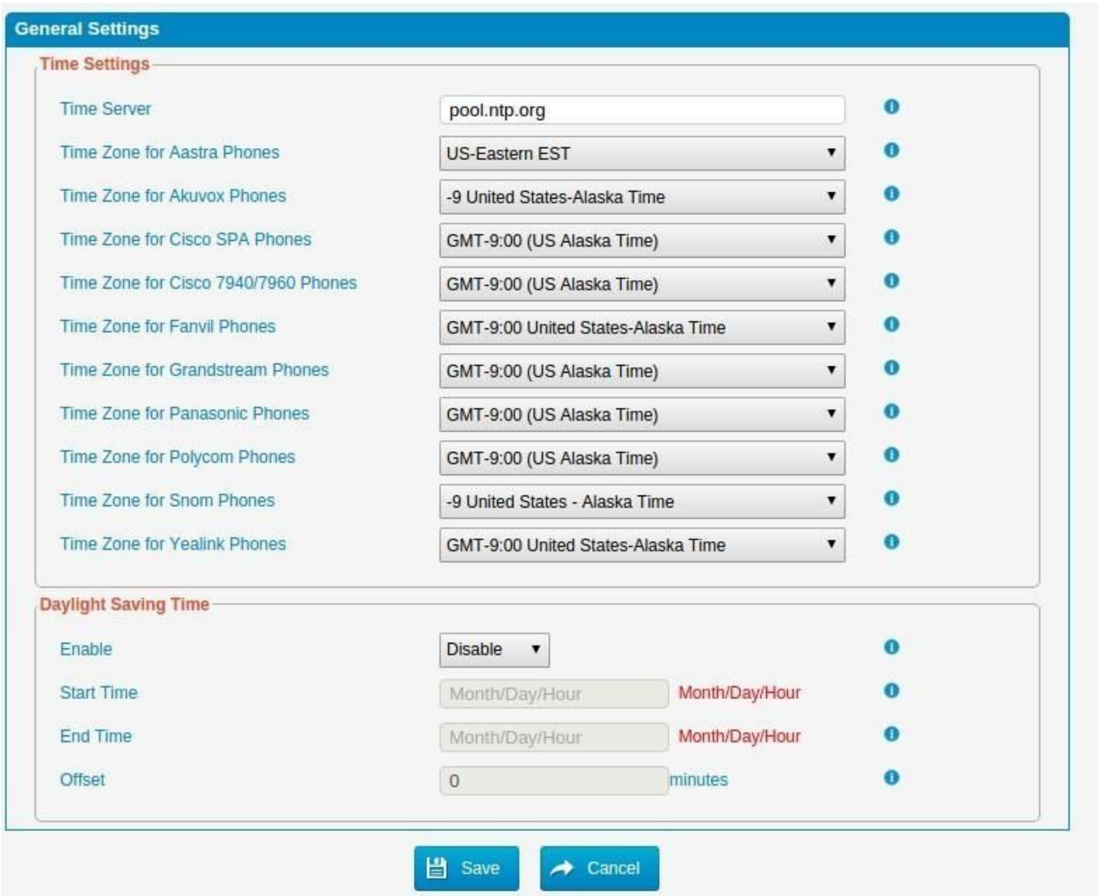
Warning: Rebooting the system will terminate all active calls!

3.11 Phone Provisioning

The Phone Provisioning provides users a method to Centralized config IP Phone.

3.11.1 General Settings

Figure 3.11.1 General Settings



3.11.2 Phones

Figure 3.11.2 Configured Phones

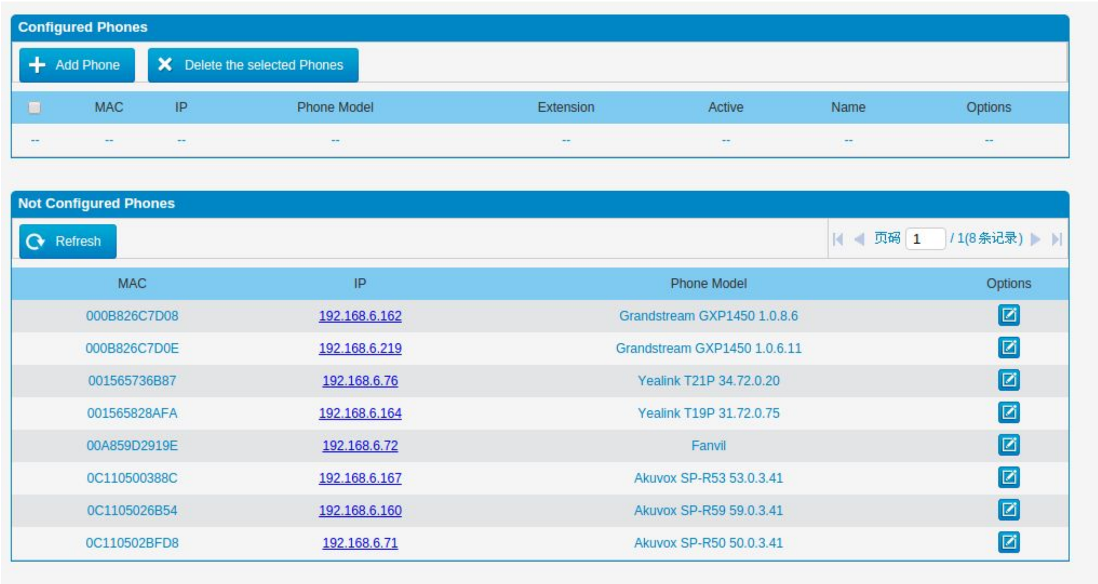
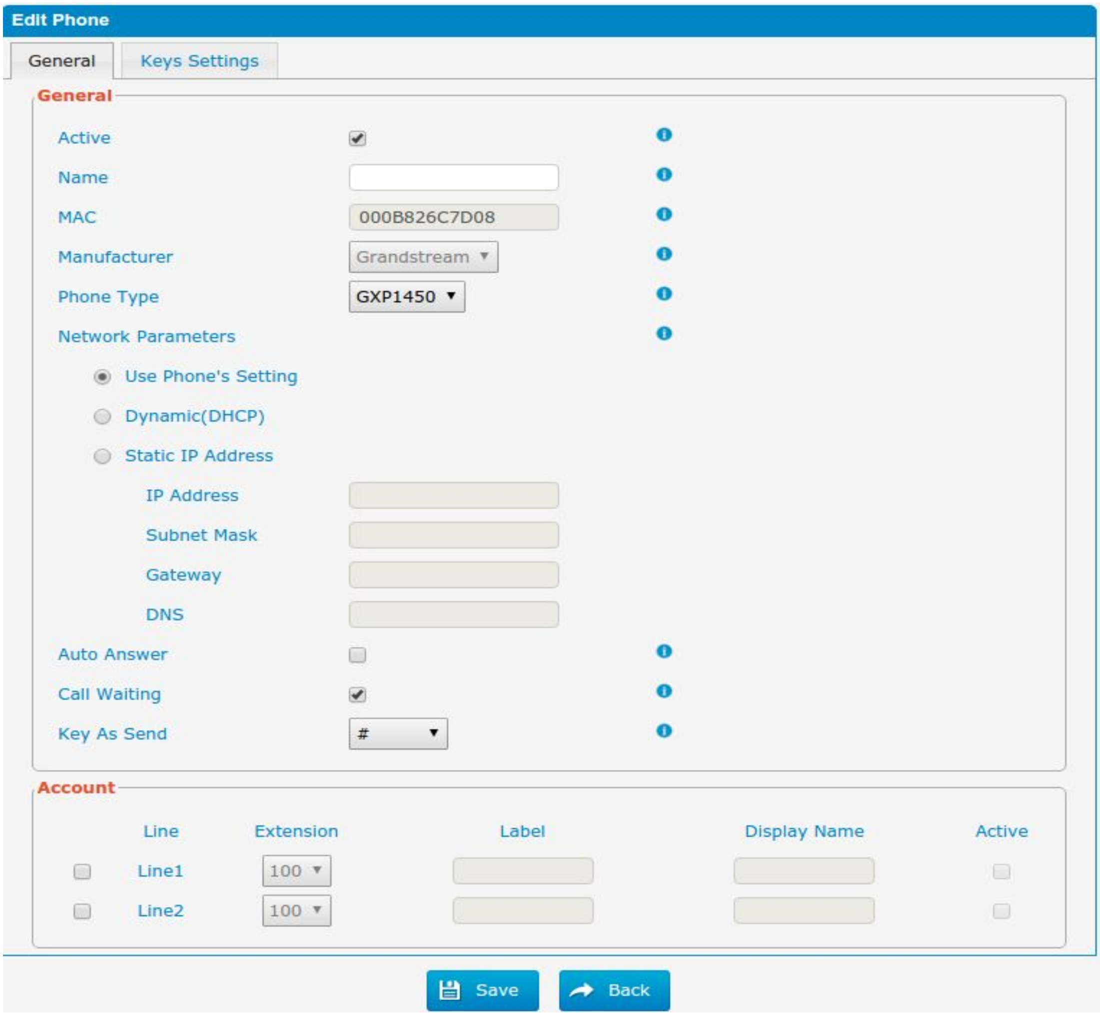


Figure 3.11.2a Edit Phone



3.12 Reports

3.12.1 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 3.12.1 CDR Report

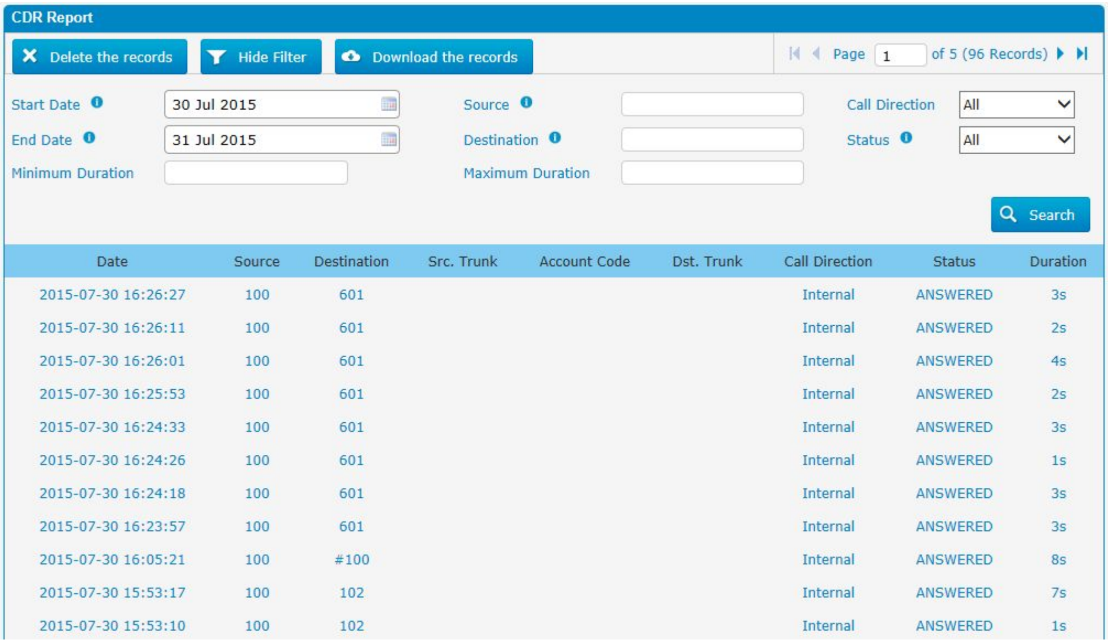


Table 3.12.1 CDR Report

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.

3.12.2 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 3.12.2 System logs Local

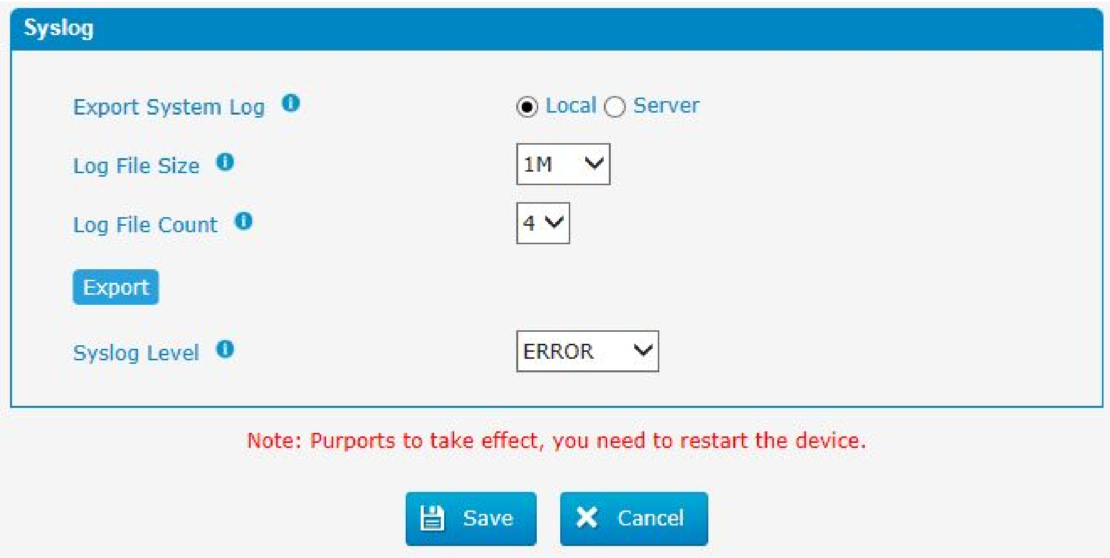


Figure 3.12.2a System logs Server

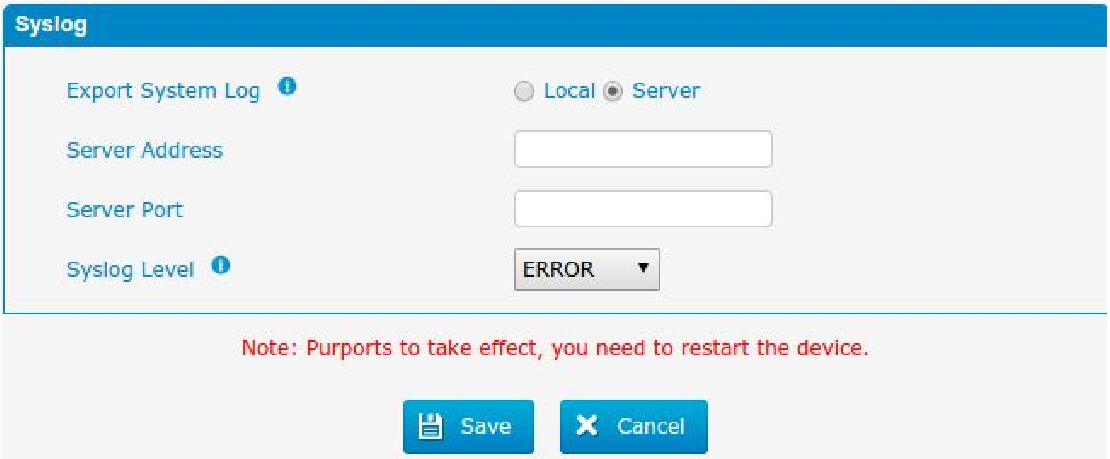


Table 3.12.2 Description of System logs

Parameters	Description
Export System Log	● Local: save log in local ● Server: save log in server
Log File Size	Max size before rotation
Log File Count	Rotated logs to keep (default: 4)

Syslog level	Syslog Level
Server Address	Server address
Server Port	Server port

3.12.3 Firewall Logs

Figure 3.12.3 Firewall logs

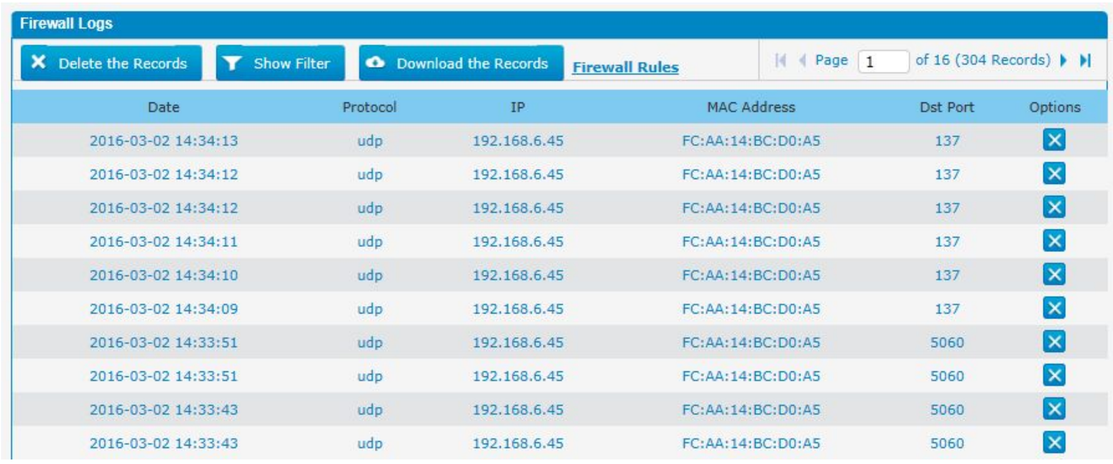


Table 3.12.3 Description of Firewall logs

Parameters	**Description**
Date	IP Attack time
Protocol	Attack protocol type
IP	Attack ip
MAC Address	Attack MAC address
Dest Port	Attack destination port

3.12.4 Trace Logs

Figure 3.12.4 DAHDI Monitor Tool

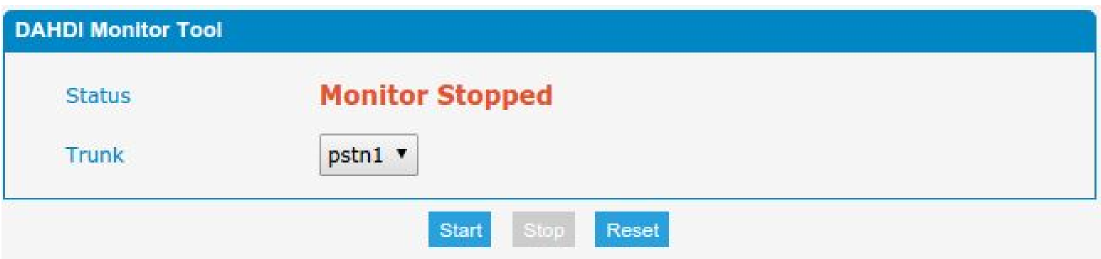


Table 3.12.4 Description of DAHDI Monitor Tool

Parameters	**Description**
Status	Display recording status of using this tool.
Trunk	Choose a Trunk to record.
Start	Start recording
Stop	Stop and download recordfile
Reset	Reset recording and Cancel the recording file

Figure 3.12.4a Asterisk Logs

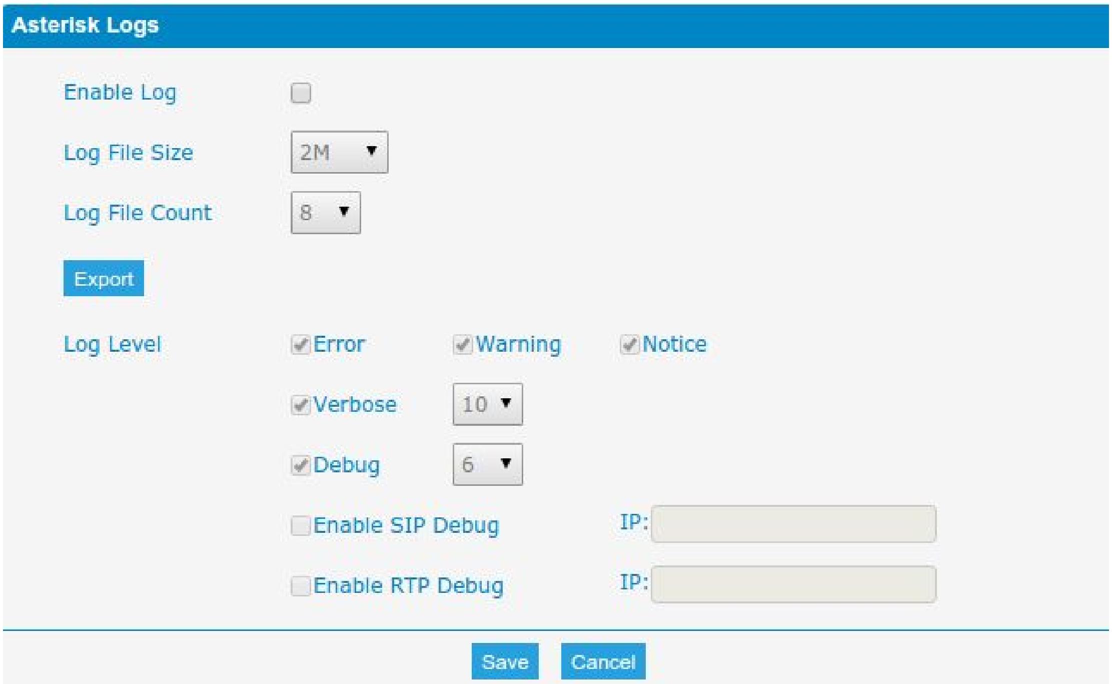


Table 3.12.4a Description of Asterisk Logs

Parameters	**Description**
Enable Log	Enable record asterisk log
Log File Size	Log file size
Log File Count	Rotated logs to keep (default: 8)
Log Level	Asterisk log level
Enable SIP Debug	Enable and set IP to enable sip debug
Enable RTP Debug	Enable and set IP to enable rtp debug

3.13 System tools

3.13.1 SMTP Parameter

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

Figure 3.13.1 SMTP Parameters

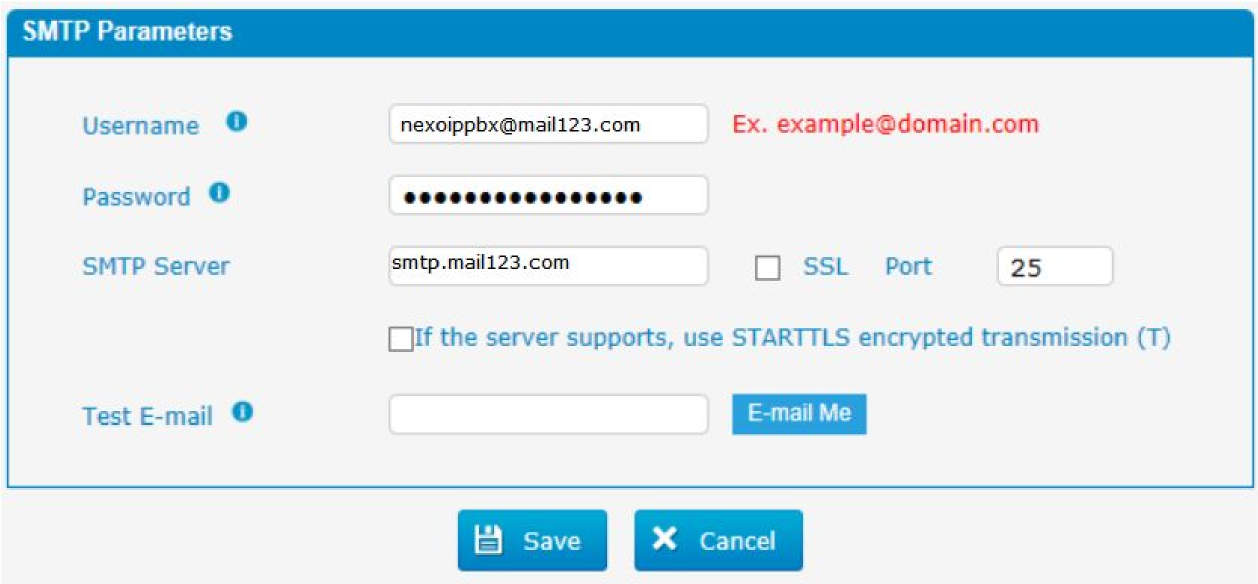


Table 3.13.1 SMTP Parameters

Parameters	Description
Username	The E-mail Address that PBX 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 PBX 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.

3.13.2 AMI Settings

The Asterisk Manager Interface(AMI) is a socket interface that you can use to get configuration and status information, request actions to be performed, and be notified about things happening to calls.

Figure 3.13.2 SMTP Parameters

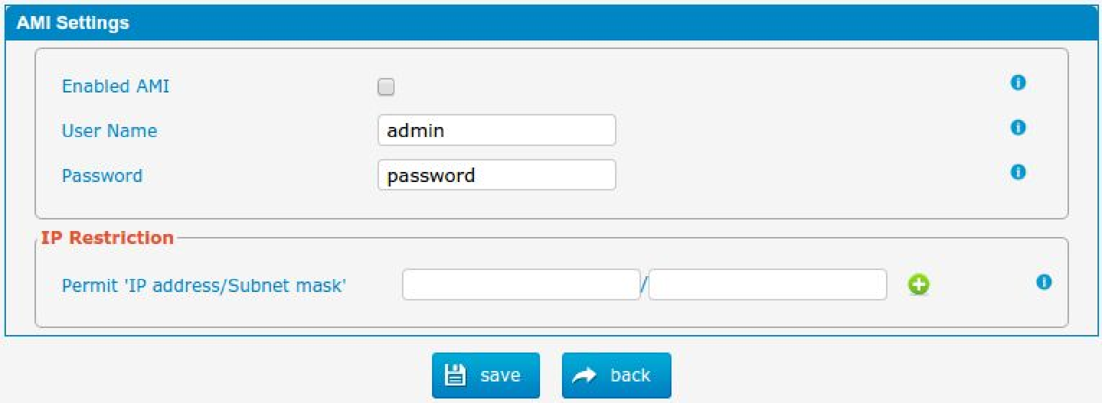
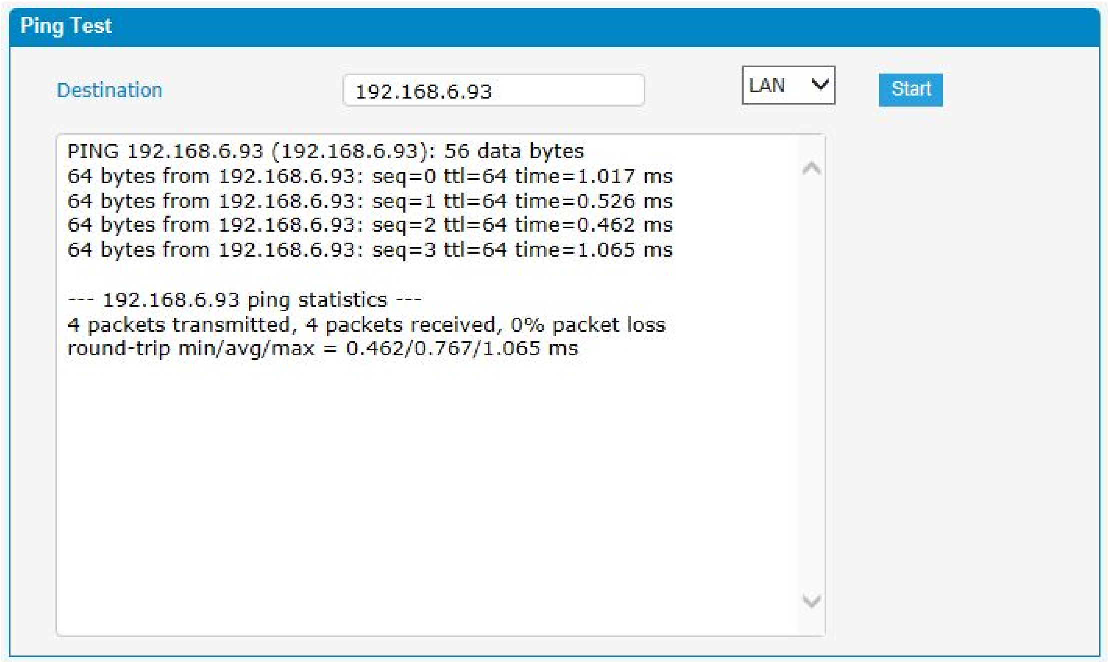


Table 3.13.2 Description of SMTP Parameters

Parameters	Description
Enable AMI	Enable AMI setttins. The Asterisk Manager Interface(AMI) is a socket interface that you can use to get configuration and status information, request actions to be performed, and be notified about things happening to calls.
Username	AMI user name,default "admin"
Password	AMI password,default "password"
IP Restriction	Set IP address and subnet mask that can connect to AMI

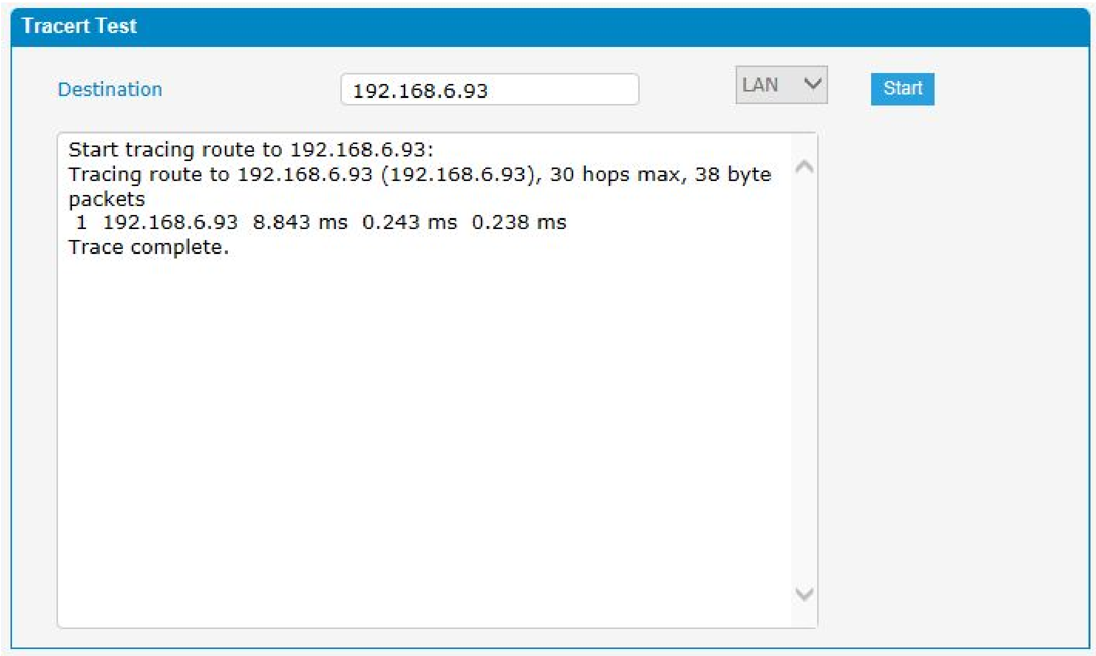
3.13.3 Ping

Figure 3.13.3 Ping



3.13.4 Tracert

Figure 3.13.4 Tracert



3.13.5 Packet Capture

Figure 3.13.5 Packet Capture

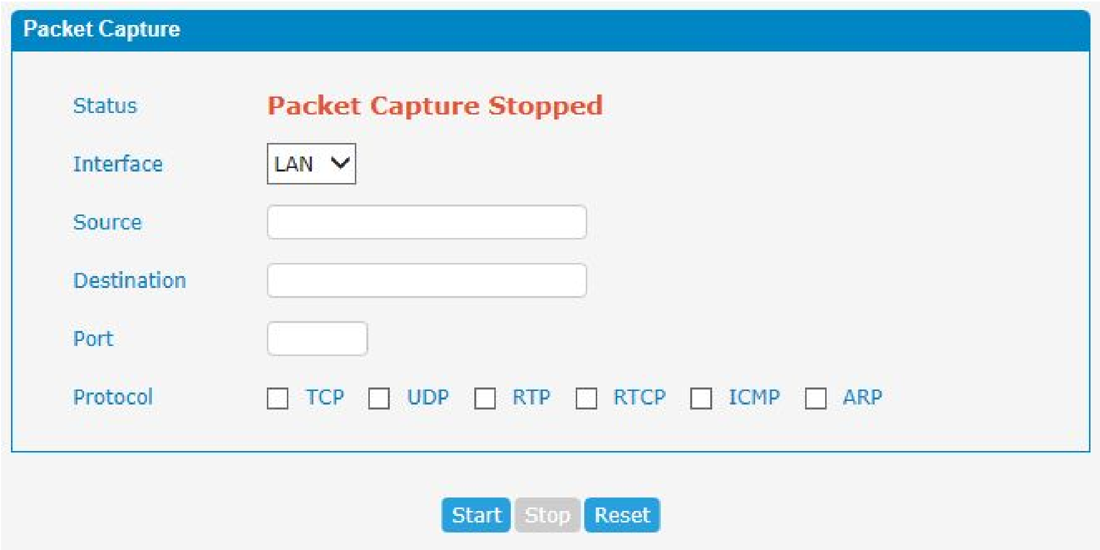


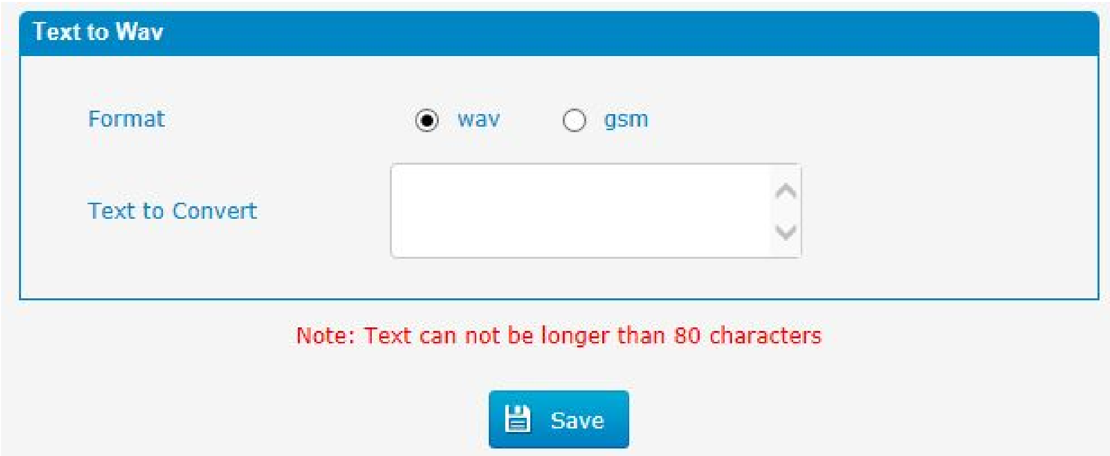
Table 3.13.5 Description of Packet Capture

Parameters	**Description**
Status	Packet capture status
Interface	Choose network interface, LAN/WAN
Souce	Capture souce Address
Destination	Capture destination Address
Port	Capture port
Protocol	Capture protocol

3.13.6 Text to Wav

PBX can Transfer text to wav.

Figure 3.13.6 Text to wav



3.13.7 Certificates

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

Figure 3.13.7 Certificates

Trusted Certificate

This certificate is a CA certificate. When selecting "TLS Verify Client" as "Yes", you should upload a CA. The relevant IP PBX 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 PBX. If IP PBX enables "TLS Verify server", you should also upload this certificate on IP PBX.