

# MUC1004/2008/2016

## IP PBX

# Administrator guide V1.1

Version 12.1.0.14



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# 1. Introduction

## 1.1 Overview

### MUC Series PBX—IP PBX for Small Business/Home Office

MUC1004/2008/2016 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

The appearance of MUC1004/2008/2016 shows as follow

Figure 1-3-1 Front view of MUC1004

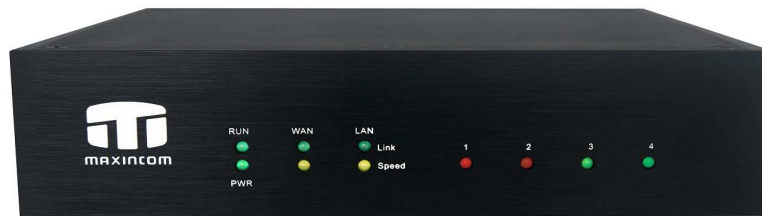


Figure 1-3-2 Front view of MUC2008



Figure 1-3-3 Front view of MUC2016



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)	<p>Red LED stands for FXO port</p> <p>Orange LED indicates presence of a BRI port.</p> <p>Green LED stands for FXS port</p> <p>Red LED blinks: FXO port isn't connected to PSTN line.</p> <p>Alternately blinks Red and Green: FXO port has an incoming call.</p> <p>Alternately blinks Red and Green fast: FXO port is in a call.</p> <p>Alternately blinks Green and Red: FXS port is ringing.</p> <p>Alternately blinks Green and Red fast: FXS port is in a call.</p>

Figure 1-3-4 Rear view of MUC1004





Figure 1-3-5 Rear view of MUC2008



Figure 1-3-6 Rear view of MUC2016



Table 1-3-2 Description of Rear view

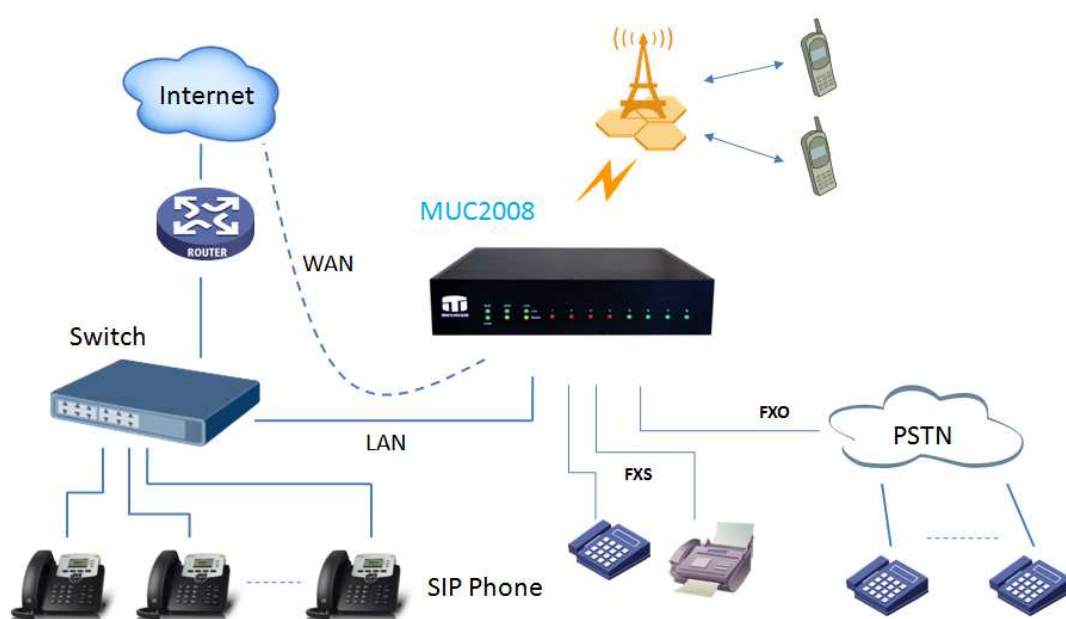
Index	Interface	Description
1	RST	Reset button to restore default IP and password or restore factory setting. Hold RST button 8 seconds, RUN LED being ON during this time
2	DC 12V	Power connector of DC power. Input: DC12V 3A/DC12

		1A(MUC1004 only)
3	USB	For the storage of call recording files
4	WAN,LAN	MUC2008 provides two 10/100 adaptive RJ45 Ethernet ports, marked as LAN and WAN. <b>-LAN port</b> :LAN port is for the connection to Local Area Network <b>-WAN port</b> : WAN port is the network 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)	<b>FXO port</b> (red light):For the connection of PSTN lines or FXS port of traditional PBX.MU2008 users could make or receive calls via FXO port. <b>FXS port</b> (green light): For the connection of analog phones. <b>BRI port</b> (orange port): For the connection of ISDN BRI lines. MU2008 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

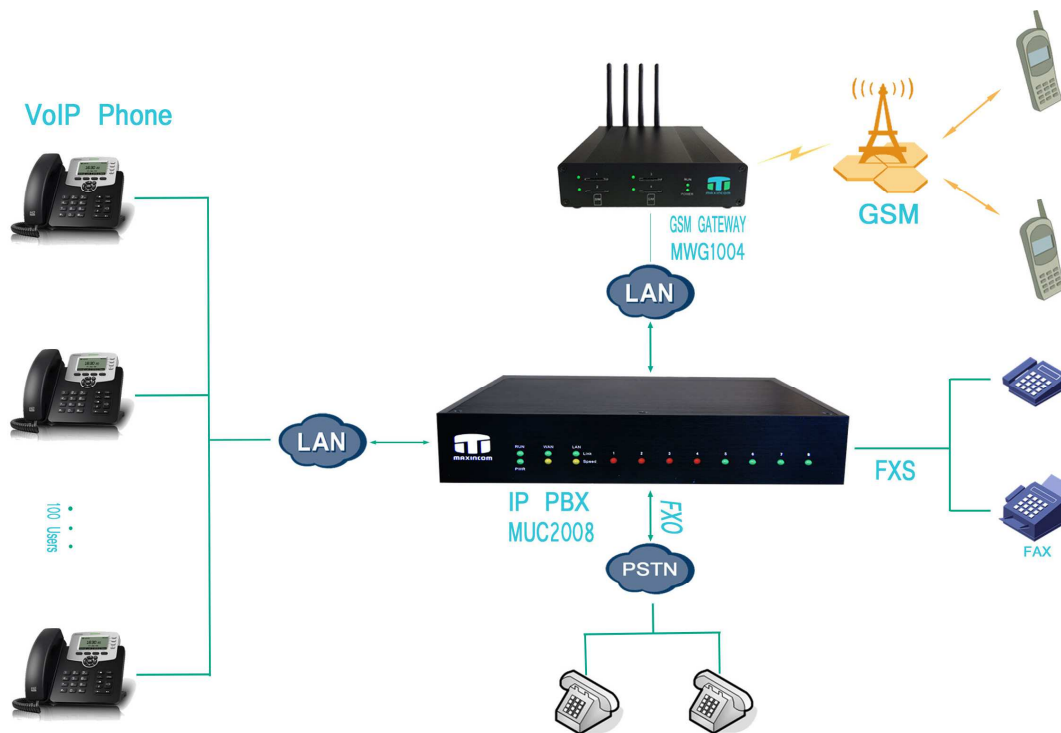
### Application 1

Figure 1.4.1



## Application 2

Figure 1.4.2



## 2. Installation Guide

### 2.1 Installation Notice

We use the MUC2008 device as an installation case as follows:

MUC2008 adapts 12VDC Power adapter, make sure AC power supply grounded well to ensure the reliability and stability;

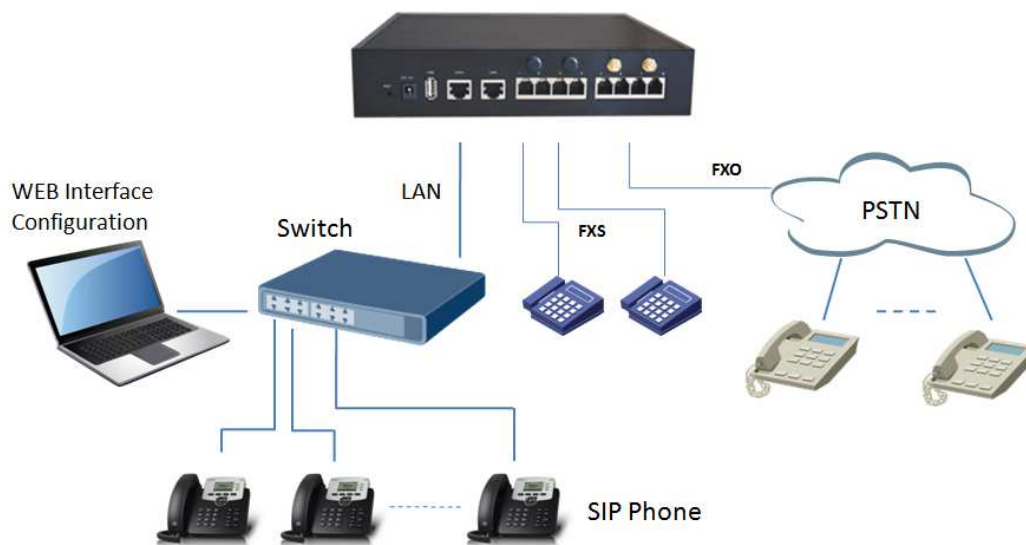
Notes: incorrect power connection may damage power adapter and device.

MUC2008 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.

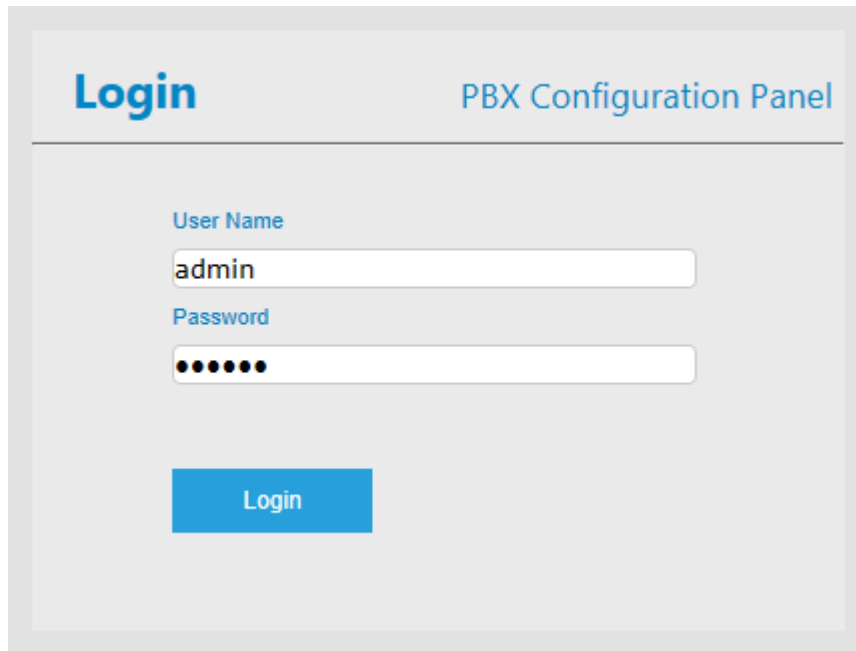
We use the MUC2008 device as a configuration case as follows:

#### 3.1 Access MUC2008 unit

Enter IP address of MUC2008 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



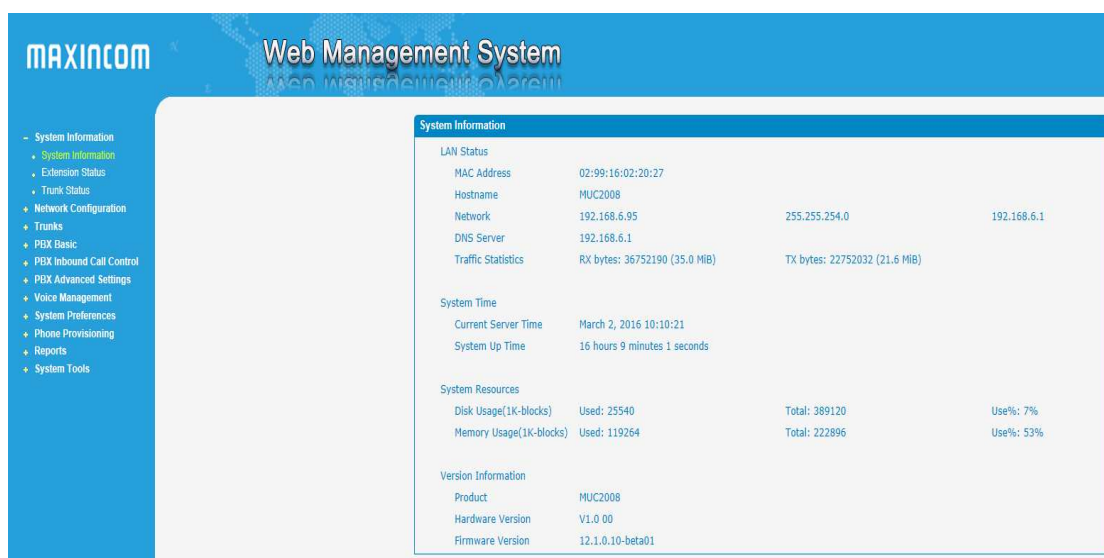
The login interface features a light gray background with a white central box. At the top left of the box is the word "Login" in blue. At the top right is "PBX Configuration Panel" in blue. Below these are two input fields: "User Name" with the text "admin" and "Password" with masked characters. A blue "Login" button is positioned below the password field.

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



The Web Management System interface has a blue header with the MAXINCOM logo and the title "Web Management System". On the left is a navigation tree with the following items: System Information (selected), System Information, Extension Status, Trunk Status, Network Configuration, Trunks, PBX Basic, PBX Inbound Call Control, PBX Advanced Settings, Voice Management, System Preferences, Phone Provisioning, Reports, and System Tools. The main content area displays "System Information" with the following data:

System Information			
<b>LAN Status</b>			
MAC Address	02:99:16:02:20:27		
Hostname	MUC2008		
Network	192.168.6.95	255.255.254.0	192.168.6.1
DNS Server	192.168.6.1		
Traffic Statistics	RX bytes: 36752190 (35.0 MIB)		TX bytes: 22752032 (21.6 MIB)
<b>System Time</b>			
Current Server Time	March 2, 2016 10:10:21		
System Up Time	16 hours 9 minutes 1 seconds		
<b>System Resources</b>			
Disk Usage(1K-blocks)	Used: 25540	Total: 389120	Use%: 7%
Memory Usage(1K-blocks)	Used: 119264	Total: 222896	Use%: 53%
<b>Version Information</b>			
Product	MUC2008		
Hardware Version	V1.0.00		
Firmware Version	12.1.0.10-beta01		

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

System Information			
LAN Status			
MAC Address	02:99:16:02:20:27		
Hostname	MUC2008		
Network	192.168.6.95	255.255.254.0	192.168.6.1
DNS Server	192.168.6.1		
Traffic Statistics	RX bytes: 36948829 (35.2 MiB)		TX bytes: 22900888 (21.8 MiB)
System Time			
Current Server Time	March 2, 2016 10:13:21		
System Up Time	16 hours 12 minutes 5 seconds		
System Resources			
Disk Usage(1K-blocks)	Used: 25540	Total: 389120	Use%: 7%
Memory Usage(1K-blocks)	Used: 123932	Total: 222896	Use%: 55%
Version Information			
Product	MUC2008		
Hardware Version	V1.0 00		
Firmware Version	12.1.0.10-beta01		

Table 3.3.1 System Information

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

### 3.3.2 Extensions Status

Figure 3.3.2 Extensions Status

Extension Status				
Free Busy Hold Unavailable Ringing				
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 Provider	test	SIP	udp	--	192.168.6.253
OK (3 ms)	Service Provider	6150	SIP	udp	--	192.168.6.150
OK (3 ms)	Service Provider	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



LAN Configuration

Network Parameters

☐ Dynamic(DHCP) ⓘ
   
☒ Static IP Address ⓘ
   

Hostname

MUC2008

IP Address

192.168.6.95

Subnet Mask

255.255.254.0

Gateway

192.168.6.1

IP Address2

Subnet Mask2

MTU

1500

DNS Server

☐ Dynamic DNS Address
   
☒ Static DNS Address
   

Primary DNS Server

192.168.6.1

Secondary DNS Server

Note: Purports to take effect, you need to restart the device.

Save

Cancel

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

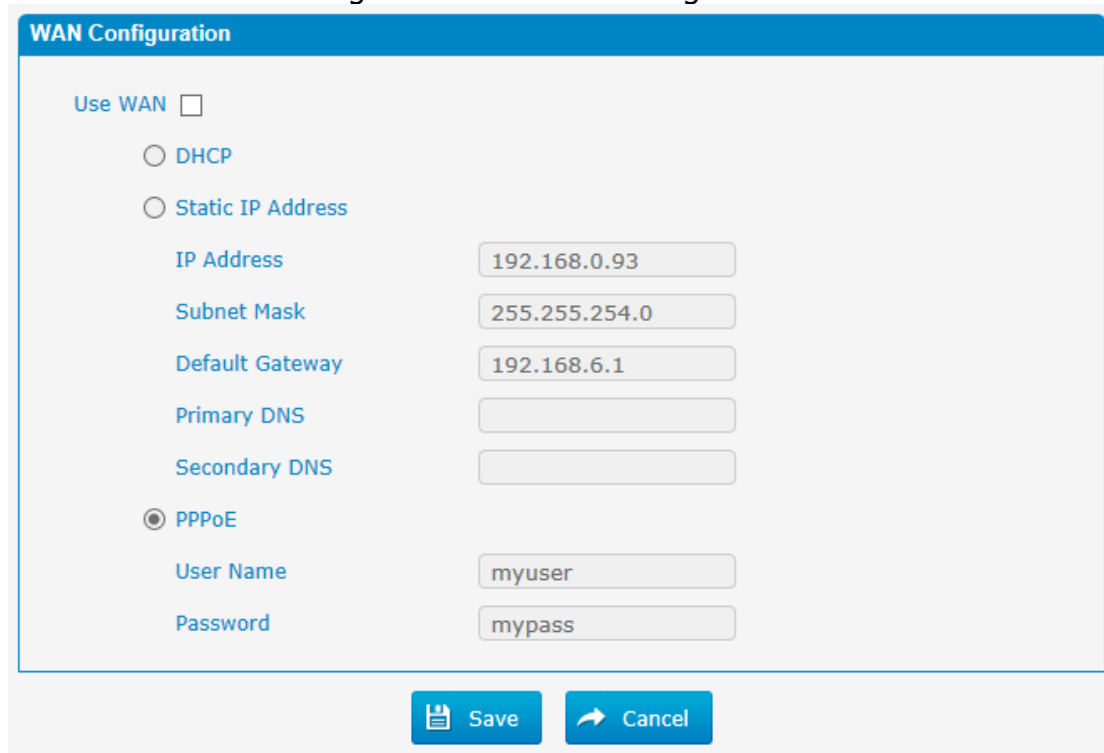


Table 3.4.1.2 Description of WAN Configuration

Parameter	Description
Use WAN	Enable 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

VLAN

VLAN Parameters(LAN)

☒ No.1
 

IP Address

192.168.8.125

Subnet Mask

255.255.255.0

Gateway

192.168.6.1

☐ No.2
 

IP Address

Subnet Mask

Gateway

VLAN Parameters(WAN)

☒ No.1
 

IP Address

192.168.0.92

Subnet Mask

255.255.255.0

Gateway

192.168.6.1

☐ No.2
 

IP Address

Subnet Mask

Gateway

Note: Purports to take effect, you need to restart the device.

Save

Cancel

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:

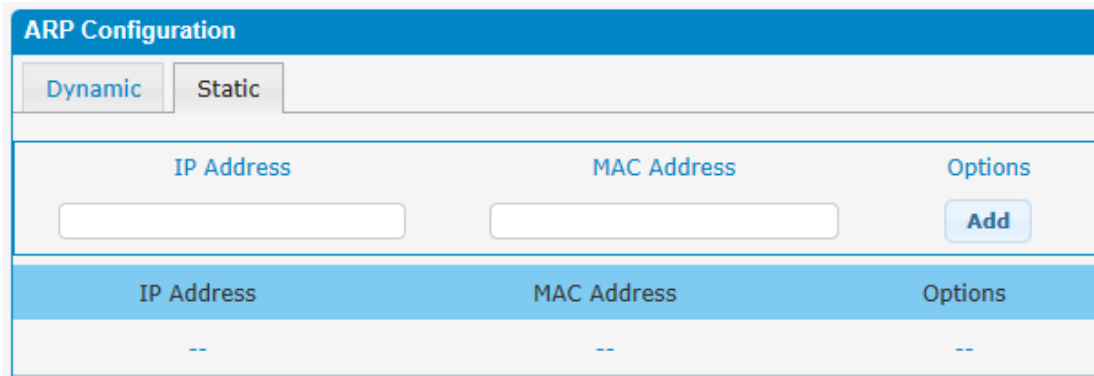
- Protect your network against ARP spoofing
- Prevent network confusion as a result of misconfigured network device

Click "Dynamic ARP" to check ARP buffer

Figure 3.4.3a Dynamic ARP

ARP Configuration	
Dynamic	Static
IP Address	MAC Address
192.168.6.252	00:0c:29:58:79:b1
192.168.6.210	00:15:65:73:6b:87
192.168.6.110	f4:b5:49:01:38:96
192.168.6.6	74:d4:35:95:03:8d
192.168.6.202	00:15:65:73:65:db
192.168.6.2	74:d4:35:d4:12:8c
192.168.6.51	78:a5:04:bd:0c:f7

Figure 3.4.3 Add ARP

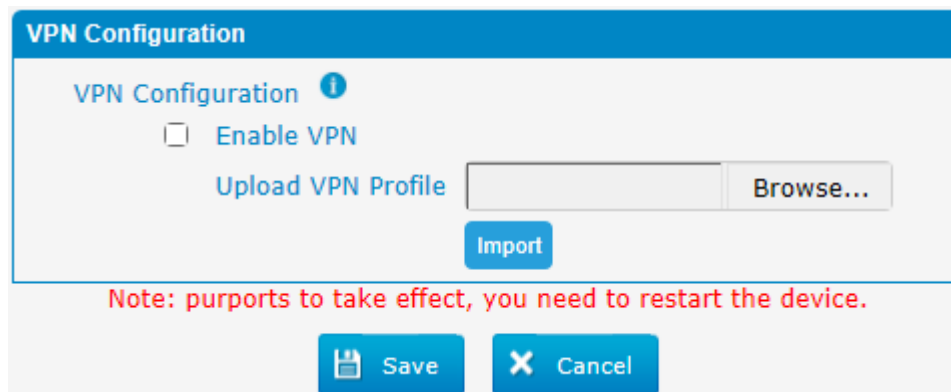


The ARP Configuration interface features a blue header bar with the title "ARP Configuration". Below the header, there are two tabs: "Dynamic" (selected) and "Static". The main area contains a table with three columns: "IP Address", "MAC Address", and "Options". The "IP Address" and "MAC Address" columns have input fields, and the "Options" column has an "Add" button. Below the table, there is a row of three dashes "--" under each column header.

### 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



The VPN Configuration interface has a blue header bar with the title "VPN Configuration". Below the header, there is a section titled "VPN Configuration" with an information icon. It contains a checkbox labeled "Enable VPN". Below this, there is a label "Upload VPN Profile" followed by a text input field and a "Browse..." button. An "Import" button is located below the input field. A red note states: "Note: purports to take effect, you need to restart the device." At the bottom, there are "Save" and "Cancel" buttons.

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

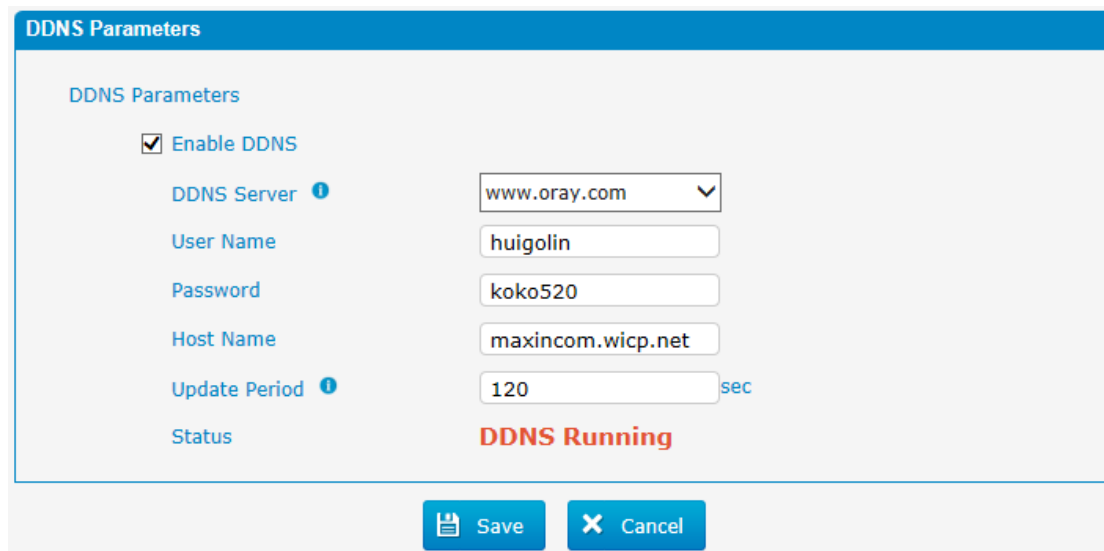


Table 3.4.5 Description of DDNS Server

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

Static Route				
Routing Table		Static Routing Rules		
Destination IP Address	Subnet Mask	Gateway	Metric	Interface
0.0.0.0	0.0.0.0	192.168.6.1	0	LAN
192.168.6.0	255.255.255.0	0.0.0.0	0	LAN

## 2) Static Route Rules

You can add new static route rules here.

Figure 3.4.6a Static Routing Rules

Static Route					
Routing Table		Static Routing Rules			
Destination IP Address	Subnet Mask	Gateway	Metric ⓘ	Interface	Options
<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	LAN ▼	<input type="button" value="Add"/>
192.168.7.0	255.255.255.0	192.168.6.1	--	LAN	<input type="button" value="X"/>

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

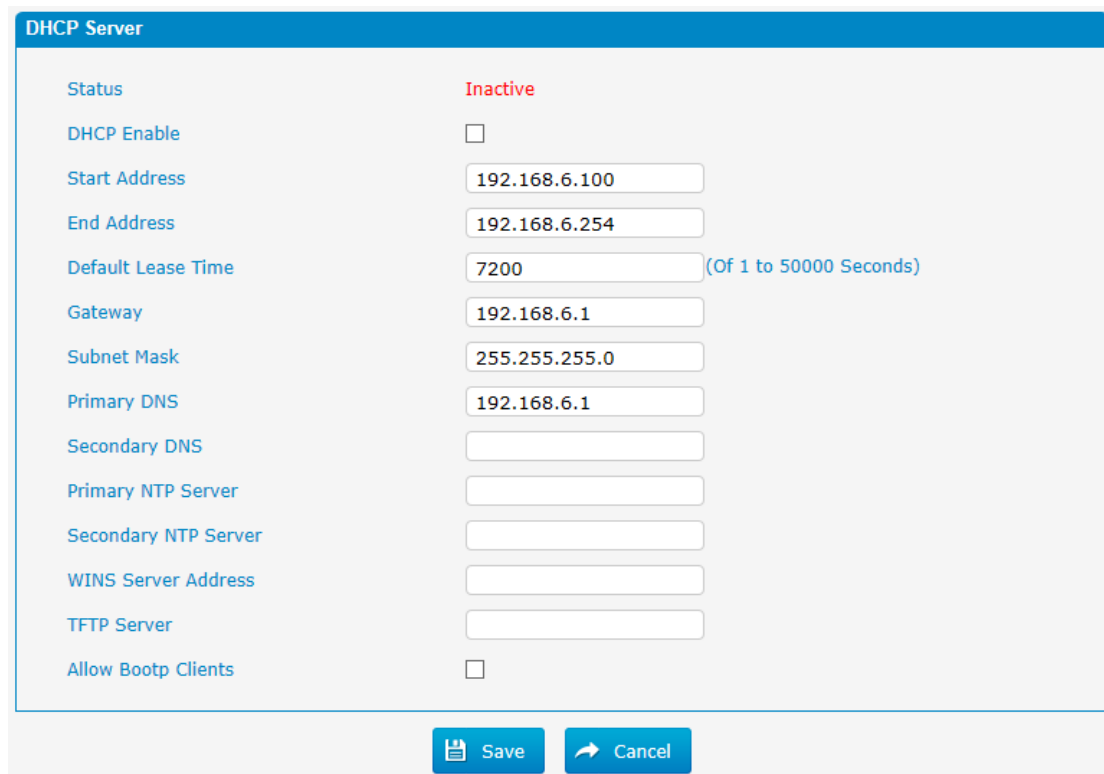


Table 3.4.7 Description of DHCP Server

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	Specify the destination network portion.
Address	
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

Analog Trunk				
			Page 1 of 1(2 Records)	
Trunk Name	Port	Rxgain	Ring Detect Timeout	Options
pstn113	3	40%	8001	
pstntest112	4	40%	8002	

Figure 3.5.1a Analog Trunks Edit

Analog Trunk Edit

General

Port

3

Trunk Name

pstn113

Rxgain

40%

Answer On Polarity Detection

No

CID Settings

CID Detection

No

CID Start

Ring

CID Signalling

Bell - USA

Ring Detect Timeout

8001

ms

Hangup

Busy Detection

Yes

Busy Count

4

Busy Interval

1

Busy Pattern

Frequency Detection

No

Busy Frequency

Hangup On Polarity Detection

No

Save

Back

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".
Rx gain	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:

	<p>Bell_USA, DTMF).</p> <p>Polarity: Start when a polarity reversal is started (Caller ID Signaling: V23_UK, V23_JP, DTMF).</p> <p>Before Ring: Start before a ring is received (Caller ID Signaling: DTMF).</p>
CID Signalling	<p>This option defines the type of Caller ID signaling to use. It can be set to one of the following:</p> <ul style="list-style-type: none"> <li>● Bell_USA: bell202 as used in the United States</li> <li>● v23_UK: suitable in the UK</li> <li>● v23_Japan: suitable in Japan</li> <li>● v23-Japan pure: suitable in Japan</li> <li>● DTMF: suitable in Denmark, Sweden, and Holland</li> </ul>
Busy Detection	<p>Busy Detection is used to detect far end hang-up or for detecting a busy signal. Select "Yes" to turn this feature on.</p>
Busy Count	<p>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.</p>
Busy Interval	<p>The busy detection interval</p>
Busy Pattern	<p>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 &lt;Busy Count&gt; 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.</p>
Frequency Detection	<p>Used for Frequency Detection (Enable detecting the busy signal frequency or not).</p>
Busy Frequency	<p>If the Frequency Detection is enabled, you must specify the local frequency.</p>
Hang-up Polarity Reversal Detection	<p>The call will be considered as "hang up" on a polarity reversal.</p>

Figure 3.5.1b GSM Trunks



Gsm Trunk					
					Page 1 of 1(1 Records)
Trunk Name	Port	Type	Tx Gain	Rx Gain	Options
GSM1	1	GSM	40%	40%	

Figure 3.5.1c GSM Trunks Edit

Gsm Trunk Edit(1)


Port

1




Trunk Name

GSM1




Mobile Number




CLIR

No




Rx Gain

40%




Tx Gain

40%




Call Progress Tone

No




DTMF Detect Mode

Echo Before




DTMF Detect Sensitive

Yes



PIN



Note: If you failed to enter your correct PIN code 3 times in succession, the SIM card will be blocked.

Save

Back

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 ring back for this trunk.
DTMF Detect Mode	Set default dtmf mode 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



IP Trunk				
+ Add			Page 1 of 1(1 Records)	
Trunk Name	Type	Hostname/IP	Transport	Options
test	SIP	192.168.6.252	udp	 

Figure 3.5.2a Add IP Trunk

Add

Trunk Name  ⓘ  
Type SIP ⓘ  
Outbound Caller ID  ⓘ  
Maximum Channels  ⓘ  
Hostname/IP  ⓘ  
Port  ⓘ  
Transport UDP ⓘ  
DTMF Mode rfc2833 ⓘ  
Qualify Yes ⓘ  
Allowed Audio Codecs  ⓘ

DOD Settings

DOD	Associated Extension	Option
<div> DOD <input type="text"/> Associated Extension <span>100</span> ⓘ <div>+ Add DOD</div> <div>+ Add Bulk DOD</div> </div>		

Save

Back

Figure 3.5.2b Add Bulk Dod

**Add Bulk DOD**

**Add Bulk DOD**

Extensions List

- 100 <SIP>
- 101 <SIP>
- 102 <SIP>
- 103 <SIP>
- 104 <SIP>
- 105 <SIP>
- 106 <SIP>
- 107 <SIP>
- 108 <SIP>
- 109 <SIP>
- 110 <SIP>
- 111 <SIP>
- 112 <IAX>
- 603 <FXS>
- 604 <FXS>

Add >

< Remove

Selected Extensions

Up ↑

Down ↓

Begin

Save Cancel

Table 3.5.2 Description of IP Trunk

Parameters	Description
IP Trunk	Add remote IP of Soft switch, 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, SIP info, In-band, 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 Settings	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





VoIP Trunk					
+ Add			Page 0 Of 0		
Index	Description	Type	Hostname/IP	Transport	Options
30	5646546	SIP	192.168.6.4	udp	 
31	123123	SIP	192.168.6.4	udp	 

Figure 3.5.3a Add VoIP Trunk

VoIP Trunk Edit(voiptrunkto12)

Trunk Name
voiptrunkto12

Type
SIP

Outbound Caller ID

Maximum Channels

Hostname/IP
192.168.6.90 : 5060

User Name
102

Password

Authorization Name
102

Domain
192.168.6.90

From User
102

Transport
UDP

DTMF Mode
rfc2833

RTP Encryption(SRTP)
No

Qualify
Yes

Allowed Audio Codecs
ulaw,alaw,gsm

Send outbound via:
☒ Domain
☐ Proxy Address

DOD Settings

DOD	Associated Extension	Option
603	100	✕
123660	101	✕
123661	102	✕
123662	103	✕
123663	104	✕
123664	105	✕

DOD
Associated Extension
100
+ Add DOD
+ Add Bulk DOD

Save
Back



Figure 3.5.3b Add Bulk DOD

**Add Bulk DOD**

**Add Bulk DOD**

Extensions List

- 100 <SIP>
- 101 <SIP>
- 102 <SIP>
- 103 <SIP>
- 104 <SIP>
- 105 <SIP>
- 106 <SIP>
- 107 <SIP>
- 108 <SIP>
- 109 <SIP>
- 110 <SIP>
- 111 <SIP>
- 112 <IAX>
- 603 <FXS>
- 604 <FXS>

Add >

< Remove

Selected Extensions

Up ↑

Down ↓

Begin

Save Cancel

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, SIP info, In-band, 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 Settings	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%	<a href="#">Detail</a>
2	602	602	602	40%	40%	<a href="#">Detail</a>

VoIP Extensions						
<a href="#">+ Add Extension</a>		<a href="#">X Delete the selected Extensions</a>		Page 1 of 1(5 Records)		
<input type="checkbox"/>	Extension Number	Register Name	Type	Display Name	Caller ID	Options
<input type="checkbox"/>	100	100	SIP	100	100	<a href="#">Detail</a> <a href="#">X</a>
<input type="checkbox"/>	101	101	SIP	101	101	<a href="#">Detail</a> <a href="#">X</a>
<input type="checkbox"/>	102	102	SIP	102	102	<a href="#">Detail</a> <a href="#">X</a>
<input type="checkbox"/>	105	105	SIP	105	105	<a href="#">Detail</a> <a href="#">X</a>
<input type="checkbox"/>	600	600	SIP	600	600	<a href="#">Detail</a> <a href="#">X</a>

Figure 3.6.1.1a Fxs Extensions Edit

Edit FXS Extension

General

Voicemail

Options

Other

User Information

Extension Type

FXS

Port

1

Extension Number

601

Display Name

601

Caller ID

601

Outbound CID

Emergency CID

Save

Back

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 caller ID Settings.

Figure 3.6.1.1b Fxs Extensions Voicemail

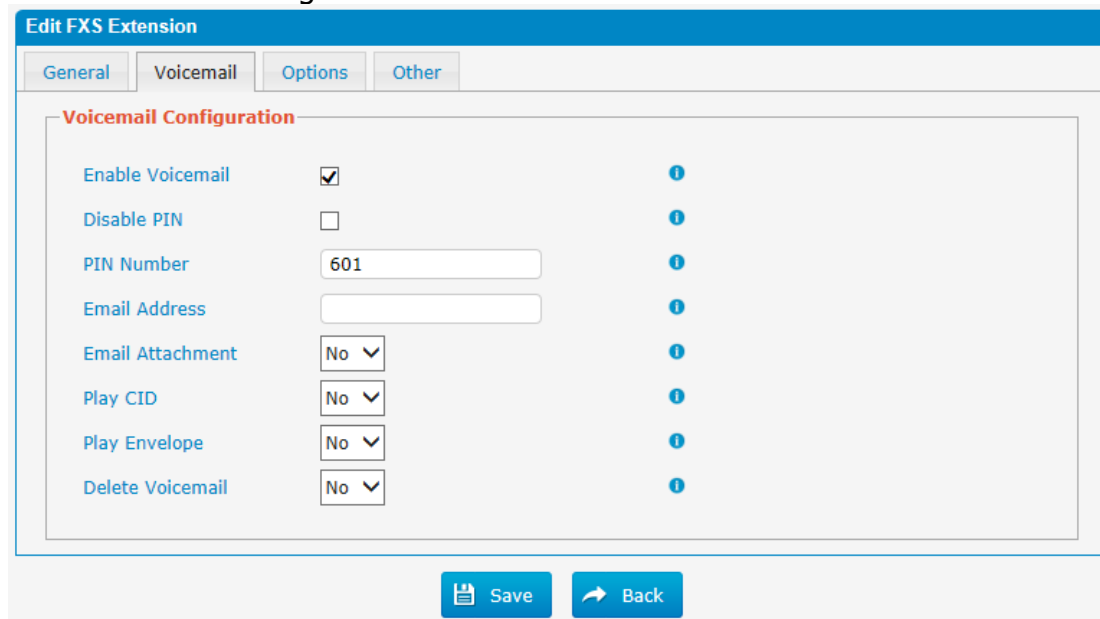


Table 3.6.1.1b Description of FXS Extensions Voicemail

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. <b>Note:</b> 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 not the Voicemail system will play the message envelope (date/time) before playing the voicemail message.
Delete Voicemail	The message will be deleted from the Voice mailbox (after having been emailed).

Figure 3.6.1.1c FXS Extensions Options

Edit FXS Extension(604)

General
Voicemail
Options
Other

### Call Forward

☐ Always
☒ On Unavailable
☒ When Busy

Send Call to:
☒ Voicemail
☐ Number
☐ Hang Up

### Volume Settings

RX Gain
40%
TX Gain
40%

### Mobility Extension

☐ Enable MobileExten
☐ Enable RingAll

Mobile Num
Outbound Prefix

### Options

Maximum Call Duration
Ring Time
Default
Call Waiting
Disable
Pinless Dialing
Disable
Call Group
Pickup Group
Do Not Disturb
☐

Note: **SMTP Parameter** must be configured correctly before Voicemail to E-mail will work.

Save
Back

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	<ul style="list-style-type: none"> <li>● 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.</li> <li>● 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.</li> </ul>
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	<p>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).</p> <p><b>Note:</b> *8 is the default setting, it can be changed under Feature Codes -&gt; General -&gt; Call Pickup.</p>
Do Not Disturb	Do Not Disturb

Figure 3.6.1.1d Fxs Extensions Other

Edit FXS Extension

General
Voicemail
Options
Other

Spy Setting

Allow Being Spied
Enable

Spy Modes
Disable

Flash

Hook Flash Detection
1000

Web Login

Enable
☒

Login Name
601

Password

Weak

Save

Back

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	<p>There are 4 spy modes available:</p> <ul style="list-style-type: none"> <li>● General spy: you have the permission to use the following 3 modes.</li> <li>● Quiet spy: you can only hear the call, but can't talk.</li> <li>● Whisper spy: you can hear the call, and can talk with the monitored extension.</li> <li>● Barge spy: you can hear the call and talk with them both.</li> </ul>
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 settings.

### 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						
<a href="#">+ Add Extension</a>		<a href="#">X Delete the selected Extensions</a>			Page 1 of 1(12 Records)	
<input type="checkbox"/>	Extension Number	Register Name	Type	Display Name	Caller ID Number	Options
<input type="checkbox"/>	100	100	SIP	100		<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	101	101	SIP	101	101	<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	102	102	SIP	102	102	<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	103	103	SIP	103	103	<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	104	104	SIP	104	104	<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	105	105	SIP	105	105	<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	106	106	IAX	106	106	<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	107	107	IAX	107	107	<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	108	108	IAX	108	108	<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	555	555	SIP	555	555	<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	556	556	SIP	556	556	<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	600	600	SIP	600	600	<a href="#">Edit</a> <a href="#">Delete</a>

Figure 3.6.1.2a VoIP Extensions Edit/Add

Edit VoIP Extension

General

Voicemail

Options

Other

User Information

Extension Type

SIP

Extension Number

100

Display Name

100

Caller ID

100

Outbound CID

Emergency CID

Authentiaction

Register Name

100

Password

••••••••

••••••••

Medium

VoIP Setting

Transport

UDP

RTP Encryption(SRTP)

No

DTMF Mode

RFC2833

Qualify

Yes

NAT

Save

Back



Table 3.6.1.2a Description of VoIP Extensions Edit/Add

Parameters	Description
Extension Type	<p>Extension type: SIP, IAX or SIP/IAX.</p> <ul style="list-style-type: none"> <li>● SIP—The extension sends and receives calls using the VoIP protocol SIP.</li> <li>● IAX—The extension sends and receives calls using the VoIP protocol IAX.</li> </ul>
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	<p>Overrides the caller ID when dialing out a trunk. Any setting here will override the common outbound caller ID set in the trunks admin.</p> <p>Format: "caller name" &lt;#####&gt;</p> <p>Leave this field blank to disable the outbound caller ID feature for this user</p>
Emergency CID	This Caller ID will always be set when dialing out an outbound route flagged as emergency. The emergency CID overrides all other caller ID 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

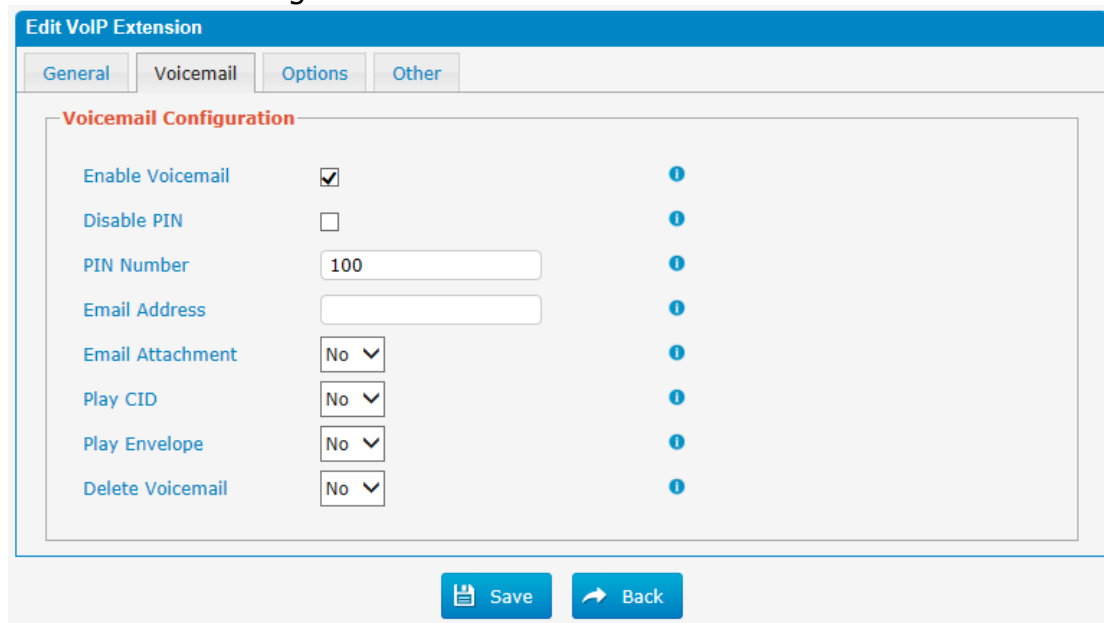


Table 3.6.1.2b Description of VoIP Extensions Voicemail

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. <b>Note:</b> 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 not the Voicemail system will play the message envelope (date/time) before playing the voicemail message.
Delete Voicemail	The message will be deleted from the Voice mailbox (after having been emailed).

Figure 3.6.1.2c VoIP Extensions Options

Edit VoIP Extension(100)

General
Voicemail
Options
Other

**Call Forward**

☐ Always
☒ On Unavailable
☒ When Busy

☒ Voicemail
☐ Number
☐ Hang Up

Send Call to:

**Mobility Extension**

☐ Enable MobileExten
☐ Enable RingAll

Mobile Num
Outbound Prefix

**Options**

Maximum Call Duration
Ring Time
Default
Call Waiting
Disable
Pinless Dialing
Disable
Allow Re-invite
yes
Call Group
Pickup Group
Do Not Disturb
☐

Note: **SMTP Parameter** must be configured correctly before Voicemail to E-mail will work.

Save
Back

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	<ul style="list-style-type: none"> <li>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.</li> <li>Enable Ring All: when someone calls the associated extension, your mobile phone will ring together, what you need is set outbound route and set Outbound Prefix number.</li> </ul>
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	<p>Re-Invite policy for this device.</p> <ul style="list-style-type: none"> <li>● yes: Allow RTP media direct.</li> <li>● no: Deny re-invites.</li> <li>● nonat: Allow reinvoke when local, deny reinvoke when NAT.</li> <li>● update: Use UPDATE instead of INVITE.</li> <li>● update,nonat: Use UPDATE when local, deny when NAT.",</li> </ul>
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	<p>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).</p> <p><b>Note:</b> *8 is the default setting, it can be changed under Feature Codes -&gt; General -&gt; Call Pickup.</p>
Do Not Disturb	Do Not Disturb

Figure 3.6.1.2d VoIP Extensions Other

Edit VoIP Extension(100)

General

Voicemail

Options

Other

Spy Setting

Allow Being Spied

Disable

Spy Modes

Disable

IP Restriction

Deny

Permit

Web Login

Enable

☒

Login Name

100

Password

...

Weak

Fax Configuration

Associated Email

Note: SMTP Parameter must be configured correctly before Voicemail to E-mail will work.

Save

Back

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	<p>There are 4 spy modes available:</p> <ul style="list-style-type: none"> <li>● General spy: you have the permission to use the following 3 modes.</li> <li>● Quiet spy: you can only hear the call, but can't talk.</li> <li>● Whisper spy: you can hear the call, and can talk with the monitored extension.</li> <li>● Barge spy: you can hear the call and talk with them both.</li> </ul>
IP Restriction	<p>IP Restriction Settings</p> <p>Default leave it blank on "IP Restriction" configuration. it indicate that registration of remote extension is allowed(remote extension IP Address is not deny)</p>

	<p><b>Deny:</b> IP Address range to deny access to,in the form of network/netmask, e.g.0.0.0.0/0.0.0.0</p> <p><b>Permit:</b> 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</p>
Web Login	Extension web login settings.
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

<b>General</b>			
Call Pickup	*8	<input checked="" type="checkbox"/>	Enable ▼
Call Trace	*69	<input checked="" type="checkbox"/>	Enable ▼
Directed Call Pickup	*08	<input checked="" type="checkbox"/>	Enable ▼
Attended Transfer	*2	<input checked="" type="checkbox"/>	Enable ▼
Blind Transfer	##	<input checked="" type="checkbox"/>	Enable ▼
One Touch Record	*1	<input checked="" type="checkbox"/>	Enable ▼
<b>Call Forward</b>			
Call Forward All Activate	*72	<input checked="" type="checkbox"/>	Enable ▼
Call Forward All Deactivate	*73	<input checked="" type="checkbox"/>	Enable ▼
Call Forward Busy Activate	*90	<input checked="" type="checkbox"/>	Enable ▼
Call Forward Busy Deactivate	*91	<input checked="" type="checkbox"/>	Enable ▼
Call Forward No Answer Activate	*52	<input checked="" type="checkbox"/>	Enable ▼
Call Forward No Answer Deactivate	*53	<input checked="" type="checkbox"/>	Enable ▼
Call Forward to Voicemail	*900	<input checked="" type="checkbox"/>	Enable ▼
Call Forward to Number	*901	<input checked="" type="checkbox"/>	Enable ▼
Call Forward Hang Up	*902	<input checked="" type="checkbox"/>	Enable ▼
<b>Call Waiting</b>			
Call Waiting - Activate	*70	<input checked="" type="checkbox"/>	Enable ▼
Call Waiting - Deactivate	*71	<input checked="" type="checkbox"/>	Enable ▼
<b>Do-Not-Disturb (DND)</b>			
DND Activate	*78	<input checked="" type="checkbox"/>	Enable ▼
DND Deactivate	*79	<input checked="" type="checkbox"/>	Enable ▼
DND Toggle	*76	<input checked="" type="checkbox"/>	Enable ▼
<b>Speed Dial</b>			
Speed Dial Prefix	*0	<input checked="" type="checkbox"/>	Enable ▼
<b>Voicemail</b>			
Voicemail Main Menu	*97	<input checked="" type="checkbox"/>	Enable ▼
Dial Voicemail	*98	<input checked="" type="checkbox"/>	Enable ▼
Direct Dial Prefix	#	<input checked="" type="checkbox"/>	Enable ▼
<b>Parking Lot</b>			
Call Parking	*85	<input checked="" type="checkbox"/>	Enable ▼
<b>ChanSpy</b>			
Quiet Mode	*93	<input checked="" type="checkbox"/>	Enable ▼
Whisper Mode	*94	<input checked="" type="checkbox"/>	Enable ▼
Barge Mode	*95	<input checked="" type="checkbox"/>	Enable ▼
<b>Paging and Intercom</b>			
Intercom Prefix	*80	<input checked="" type="checkbox"/>	Enable ▼
User Intercom Allow	*54	<input checked="" type="checkbox"/>	Enable ▼
User Intercom Disallow	*55	<input checked="" type="checkbox"/>	Enable ▼
<b>PIN User</b>			
Access Code	*99	<input checked="" type="checkbox"/>	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	E.g. 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

Speed Dial			
Dial 'speeddial prefix *0 + Source Number' to turn into the 'Destination Number', speeddial prefix is configured through <a href="#">Feature Codes</a> .			
<input type="button" value="+ Add Speed Dial"/> <input type="button" value="X Delete the selected Speed Dial"/>		Page 1 of 1(1 Records)	
<input type="checkbox"/>	Description	Source Number	Destination Number Options
<input type="checkbox"/>	60001	61	60001 <input type="button" value="edit"/> <input type="button" value="delete"/>

Figure 3.6.3a Speed Dial Add

#### Add Speed Dial

Description
60002

Source Number
62

Destination Number
60002

Table 3.6.3 Description of Speed Dial

Parameters	Description
Source Number	The speed dial number.
Destination Number	<p>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.</p> <p>The prefix of Speed dial is setting on "feature codes"</p> <p><b>Note:</b> Don't forget to add the outbound dial prefix if you would like to dial the speed dial number through trunk.</p>

### 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

Outbound Routes							
<a href="#">+ Add Route</a>		<a href="#">X Delete the selected Routes</a>			Page 1 of 1(1 Records)		
<input type="checkbox"/>	Route Name	Route CID	Dial Patterns	Emergency	Office Hours Mode	Sort	Options
<input type="checkbox"/>	9_outside		9.	No	None		<a href="#">Edit</a> <a href="#">Delete</a>

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

Edit Outbound Route

Settings

Dial Patterns

Office Hours

Route Settings

Route Name

9\_outside

Route CID

Route Password

PIN Set

test

PIN Sets

Route Type

☐ Emergency
☐ Intra-Company

Allow Extensions

Disable Extensions

600 <SIP>

105 <SIP>

100 <SIP>

Add >

< Remove

Enable Extensions

101 <SIP>

102 <SIP>

602 <FXS>

601 <FXS>

Up ↑

Down ↓

Allow Trunks

Disable Trunks

test <SPS>

Add >

< Remove

Enable Trunks

frompstn1222 <FXO>

frompstn23333333 <FXO>

Up ↑

Down ↓

Save

Back

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 override extension cid
Route Passed	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	<ul style="list-style-type: none"> <li>● Emergency</li> <li>● Intra-Company</li> </ul>
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

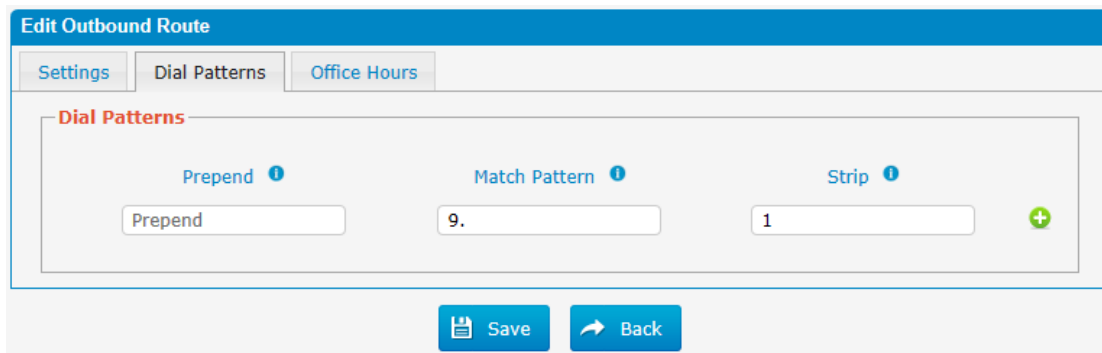


Table 3.6.4b Description of Outbound Routes Edit

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: <b>X</b> : Any Digit from 0-9 <b>Z</b> : Any Digit from 1-9 <b>N</b> : Any Digit from 2-9 <b>[12345-9]</b> : Any digit in the brackets (in this example,


	<p>1,2,3,4,5,6,7,8,9)</p> <p>The "." Character will match any remaining digits. For example, "9011." will match any phone number that starts with "9011", excluding "9011" itself.</p> <p>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.</p> <p>Example 1: <b>1[5-8]6</b> will match 156,166,176,186.</p> <p>Example 2: <b>1NXXNXXXXXX</b> will match a phone number starting with a 1, followed by a 3-digit area code, and then 6-digit number.</p>
Strip	<p>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.</p>
Add 	<p>Add multiple dial patterns in this outbound route.</p>

Figure 3.6.4c Outbound Routes Edit

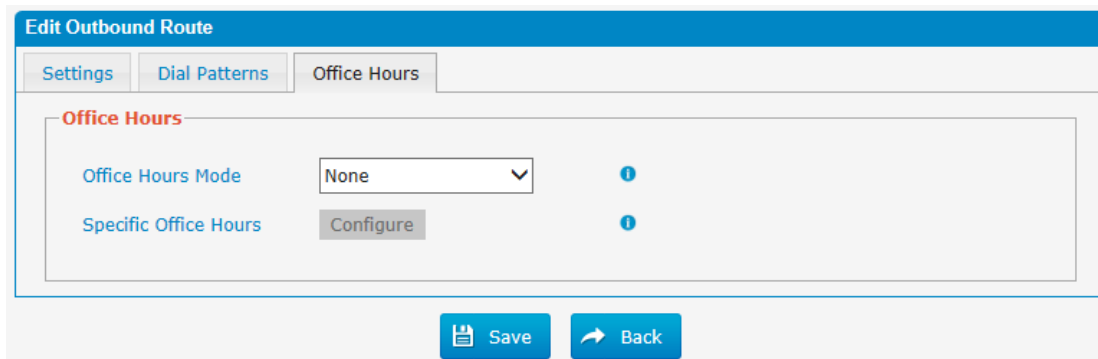


Table 3.6.4c Description of Outbound Routes Edit

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.
Specific Office Hours	Configure specific office hour

### 3.6.5 Parking Lot

Figure 3.6.5 Parking Lot

Parking Lot

Call parking feature code (\*85) is configured through [Feature Codes](#)

General

Parking Lot Extension

700

Parking Lot Starting Position

701

Number of Slots

8

(701-708)

Options

Parking Timeout(sec)

60

Alert Info

Find Slot

Next

Parked Music Class

calmriver

Transfer Capability

Caller

Re-Parking Capability

Caller

Destination for Orphaned Parked Calls

Destination

End Call

Save

Cancel

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 Starting Position	The starting position 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

Time Groups

Day/Night Mode Control

Status: **Normal Mode**

Switch To Day Mode

Switch To Night Mode

Switch To Normal Mode

Use Default?

Feature Status

\*34

☒

Enable

\*35

☒

Enable

\*034

☒

Enable

Global Time Groups

Configure Time Groups

Configure Holidays

Save

Cancel





### 3.6.7 General Preferences

Figure 3.6.7 General Preferences

General

General

Select Language

English

Max Account of Calls

0

Global Max Call Duration

7200

Ring Timeout

30

Country Tonezone

United States / North America

Music on Hold

calmriver

Follow Me Play Music on Hold

Ring

FXO Mode

FCC

Feature Digit Timeout

4000ms

Services

FTP Parameter

Enable FTP

Port

21

SSH Parameter

Enable SSH

Port

22

Web Parameter

Enable HTTP

Port

80

Enable HTTPS

Port

443

Extension Parameters

Extension Number

100 - 588

IVR Extensions

620 - 639

Conference Extensions

740 - 749

Queue Extensions

820 - 839

Ring Group Extensions

920 - 939

Paging Group Extensions

720 - 729

Save

Cancel

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 Tone zone	Please select your country or nearest neighboring country to enable the default dial tone, busy tone, and ring tone for your region.
Music on Hold	Select MOH music
Follow Me Play Music on Hold	Music of follow me Ring: normal ring back tone Default: default MOH music None: silence
FXO Mode	FXO country mode
Feature Digit 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 Extensions	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

Inbound Routes						
+ Add Route		X Delete the selected Routes			Page 1 of 1(1 Records)	
<input type="checkbox"/>	Route Name	DID Number	Caller ID Number	Enable Callback	Sort	Options
<input type="checkbox"/>	default	6xxx	136060	OFF		 

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

Figure 3.7.1a Inbound Routes Edit

Edit Route(default)

Settings

Advanced Setting

General

Route Name

default

DID Number

Extension

Caller ID Number

Alert Info

Incoming Trunks

All Trunks

252IP\_sip <SPS>

to151IP\_IAX <SPX>

to110VoIP\_IAX <IAX>

to110VoIP\_SIP <SIP>

pstn5 <FXO>

pstn6 <FXO>

GSM1 <GSM>

Add >

< Remove

Allow Trunks

pstn8 <FXO>

Up ↑

Down ↓

Time Conditions

Time Groups Mode

None

Specific Time Groups

Configure

Day Destination

IVR

Night Destination

End Call

<620> Welcome

Holidays Settings

Holiday Mode

None

Specific Holiday

Configure

Holiday Destination

End Call

Fax Detection

Destination

No Detect

Save

Back

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: <b>X</b> : Any Digit from 0-9 <b>Z</b> : Any Digit from 1-9

	<p><b>N:</b> Any Digit from 2-9</p> <p><b>[12345-9]:</b> Any digit in the brackets (in this example, 1, 2, 3, 4, 5, 6, 7, 8, 9)</p> <p>The "." Character will match any remaining digits. For example, "9011." will match any phone number that starts with "9011", excluding "9011" itself.</p> <p>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.</p> <p>Example 1: <b>NXXXXXX</b> will match any 7-digit phone number.</p> <p>Example 2: <b>1NXXNXXXXXX</b> will match a phone number starting with a 1, followed by a 3-digit area code, and then 6-digit number.</p>
Extension	<p>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.</p>
Caller ID Number	<p>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.</p> <p>The following patterns may be used:</p> <p><b>X:</b> Any Digit from 0-9</p> <p><b>Z:</b> Any Digit from 1-9</p> <p><b>N:</b> Any Digit from 2-9</p> <p><b>[12345-9]:</b> Any digit in the brackets (in this example, 1, 2, 3, 4, 5, 6, 7, 8, 9)</p> <p>The "." Character will match any remaining digits. For example, "9011." will match any phone number that starts with "9011", excluding "9011" itself.</p> <p>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.</p> <p>Example 1: <b>NXXXXXX</b> will match any 7 digits phone number.</p> <p>Example 2: <b>1NXXNXXXXXX</b> will match a phone number</p>

	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	<p>Select time group's mode.</p> <ul style="list-style-type: none"> <li>● None: Disable office hours for this route.</li> <li>● Global office hours: It is configured through general preferences.</li> <li>● Specific office hours: Use the specific office hours settings.</li> </ul>
Specific Time Groups	Set specific time groups
Day Destination	<ul style="list-style-type: none"> <li>● End Calls</li> </ul> <p>Route the incoming calls to end calls, the system will auto hang up the call.</p> <ul style="list-style-type: none"> <li>● Extension</li> </ul> <p>Route the incoming calls to a specific extension.</p> <ul style="list-style-type: none"> <li>● Voicemail</li> </ul> <p>Route the incoming calls to extension's voicemail.</p> <ul style="list-style-type: none"> <li>● IVR</li> </ul> <p>Route the incoming calls to a specific IVR.</p> <ul style="list-style-type: none"> <li>● Ring Group</li> </ul> <p>Route the incoming calls to a specific Ring Group.</p> <ul style="list-style-type: none"> <li>● Conference Room</li> </ul> <p>Route the incoming calls to a specific Conference Room.</p> <ul style="list-style-type: none"> <li>● DISA</li> </ul> <p>Route the incoming calls to a specific DISA.</p> <ul style="list-style-type: none"> <li>● Queues</li> </ul> <p>Route the incoming calls to a specific Queue.</p> <ul style="list-style-type: none"> <li>● Outbound Routes</li> </ul> <p>Route the incoming calls to a specific outbound route. This function is mainly used for the connection of two branches.</p> <p>For example: Company A locates headquarters in the USA</p>
Night Destination	

	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. <ul style="list-style-type: none"> <li>● Select which defined Holiday to use.</li> <li>● None: Disable holiday for this route.</li> <li>● Global holiday: It is configured through general preferences.</li> <li>● Specific holiday: Use the holiday settings.</li> </ul>
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

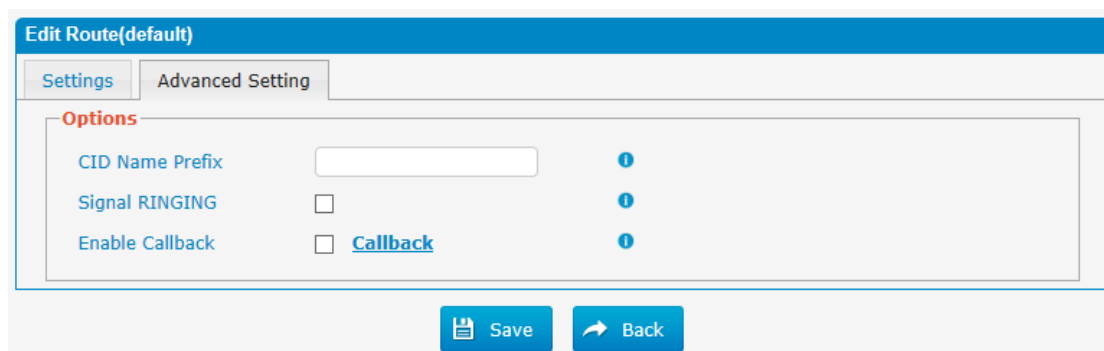


Table 3.7.1b Description of Inbound Routes Edit

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

Blacklist	
<a href="#">+ Add</a>	Page 1 of 1 (1 Records)
Number	Options
5608344	<a href="#">X</a>

We can add a number to blacklist

Figure 3.7.2a Blacklist Add

Blacklist Add	
Number	<input type="text" value="5984624"/>
<a href="#">Save</a> <a href="#">Back</a>	

### 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

IVR					
<a href="#">+ Add IVR</a>	<a href="#">X Delete the selected IVR</a>	Page 1 of 1(1 Records)			
<input type="checkbox"/>	Number	Description	Timeout	Call Direct Extensions	Options
<input type="checkbox"/>	620	Welcome	3	Yes	<a href="#">Edit</a> <a href="#">X</a>

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

Figure 3.7.3a IVR Add

Add IVR

General

IVR Number

621

IVR Description

621

Announcement

default

Enable Direct Dial

No

Timeout

3

Invalid Retries

3

Invalid Destination

End Call

Timeout Retries

3

Timeout Destination

End Call

CID Name Prefix

IVR Entries

Key

digits pressed

Destination

==choose one==



Delete

Save

Back

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 played 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 prefix 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 answered in the order they were received to deliver top tier customer service.

Figure 3.7.4 Queue



Queues							
Dial 'Queue number + *' to log in or 'Queue number + *' to log out the queue. For example, if the queue number is '820', then the agent can dial '820*' to log in or '820*' to log out							
+ Add Queue		X Delete the selected Queues			Page 1 of 1(1 Records)		
<input type="checkbox"/>	Queue Number	Queue Name	Password	Strategy	Timeout Queue	Timeout Agents	Options
<input type="checkbox"/>	820	Queue820		ringall	Unlimited	10	 

Figure 3.7.4a Queue General

Edit Queue

General

Options

Advanced Settings

General

Queue Number

820

Queue Name

Queue820

Queue Password

Max Time Caller in Queue

Unlimited

Agents Timeout

10seconds

CID Name Prefix

Queue820-

Ring Strategy

ringall

Restrict Dynamic Agents

No

Static Agents

Extensions

601 <FXS>  
602 <FXS>  
600 <SIP>  
100 <SIP>  
103 <SIP>

Add >  
< Remove

Allow Members

101 <SIP>  
102 <SIP>  
105 <SIP>

Up ↑  
Down ↓

Dynamic Agents

Extensions

601 <FXS>  
602 <FXS>  
101 <SIP>  
102 <SIP>  
600 <SIP>  
105 <SIP>  
100 <SIP>  
103 <SIP>

Add >  
< Remove

Allow Members

Up ↑  
Down ↓

Save

Back

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 prefix name
Alert Info	Alert info can be used for distinctive ring with SIP devices.
Ring Strategy	<p>This option sets the Ringing Strategy for this Queue. The options are</p> <ul style="list-style-type: none"> <li>● ring All: Ring all available Agents simultaneously until one answers.</li> <li>● least Recent: Ring the Agent which was least recently called.</li> <li>● fewest Calls: Ring the Agent with the fewest completed calls.</li> <li>● random: Ring a Random Agent.</li> <li>● memory: Round Robin with Memory, Remembers where it left off in the last ring pass.</li> <li>● Linear: Rings agents in the order specified, for dynamic agents in the order they logged in.</li> </ul>
Restrict Dynamic Agents	Restrict 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

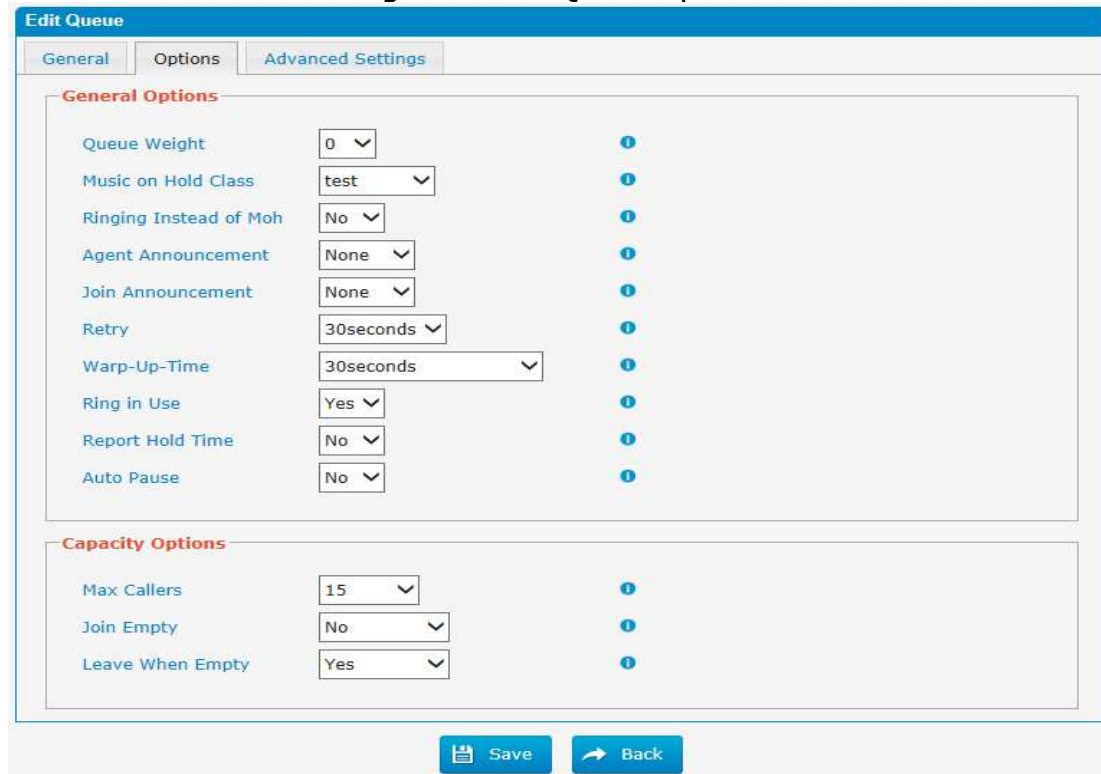


Table 3.7.4b Description of Queue Options

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, <ul style="list-style-type: none"> <li>● Yes: Callers can join a call queue without agents or only unavailable agents</li> <li>● No: Callers cannot join a queue when there are no agents in the queue.</li> </ul>
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. <ul style="list-style-type: none"> <li>● Yes: Callers are forced out of a queue when no agents are logged in.</li> <li>● No: Callers will remain in a queue with no agents.</li> </ul>

Figure 3.7.4c Queue Advanced Settings

Edit Queue

General
Options
Advanced Settings

**Caller Position Announcements**

Frequency
1minute,15seconds
Announce Position
Yes
Announce Hold Time
Yes

**Periodic Announcements**

Prompt
default
Frequency
30seconds

**Events,Stats**

Event When Called
No
Member Status Event
No
Service Level
1minute

**Fail Over Destination**

Destination
End Call

Save
Back

Table 3.7.4c Description of Queue Advanced Settings

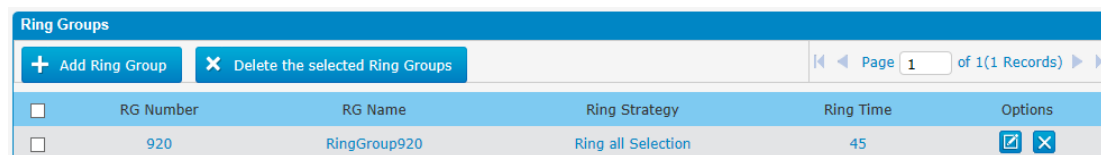
Parameters	Description
Frequency	How often to announce queue position and estimated hold time. <b>Note:</b> "0 seconds" means disabling the announcement.
Announce Position	Announce position of caller in the queue
Announce Hold Time	Enabling this option causes PBX to announce the hold time to the caller 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 Groups





Ring Groups					
+ Add Ring Group		X Delete the selected Ring Groups		Page 1 of 1(1 Records)	
<input type="checkbox"/>	RG Number	RG Name	Ring Strategy	Ring Time	Options
<input type="checkbox"/>	920	RingGroup920	Ring all Selection	45	 



Figure 3.7.5a Ring Groups Edit

Edit Ring Group

General

RG Number
920

RG Name
RingGroup920

Ring Strategy
Ring all Selection

Ring Time
45

Music on Hold
calmriver

Ring Instead Of Moh
☐

CID Name Prefix
RingGroup920-

Alert Info

Ring Group Members

Extensions

600 <SIP>  
105 <SIP>  
100 <SIP>  
103 <SIP>

Add >

< Remove

Members

101 <SIP>  
102 <SIP>  
601 <FXS>  
602 <FXS>

Up ↑

Down ↓

Destination If No Answer

Destination
End Call

Save

Back

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: <ul style="list-style-type: none"> <li>● Ring All Simultaneously: Ring all available Extensions simultaneously.</li> <li>● Ring Sequentially: Ring each extension in the group one</li> </ul>

	at a time.
Ring Time	<p>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.</p> <p>2. If the strategy is "Ring Sequentially", it means the number of seconds to ring a single extension before moving onto the next one.</p>
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, i.e.: 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



Conferences						
+ Add Conference		X Delete the selected Conferences		Page 1 of 1(1 Records)		
<input type="checkbox"/>	Room Number	Room Name	User PIN	Admin PIN	Participants	Options
<input type="checkbox"/>	740	Conference740		740	Unlimited	OFF  

Figure 3.7.6a Conferences Edit/Add

Edit Conference

General

Conference Number

740

Conference Name

Conference740

User PIN

Admin PIN

740

Room Options

Join Prompt

None

Max Participants

Unlimited

Allow Menu

☒

Music on Hold

☒

Music on Hold Class

calmriver

Quiet Mode

☐

User Count

☐

User join/leave

☒

Leader Wait

☐

Save

Back

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 Participants	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.
Music 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.

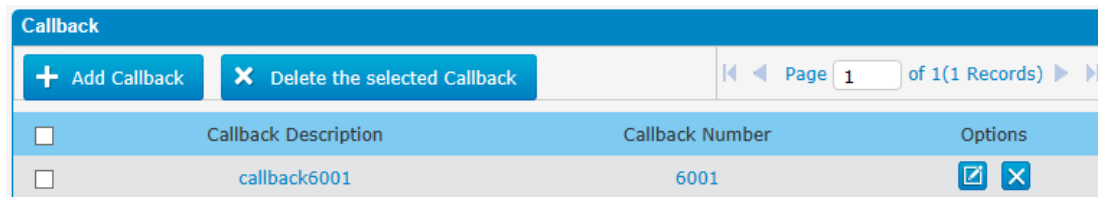
### 3.7.7 Callback

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



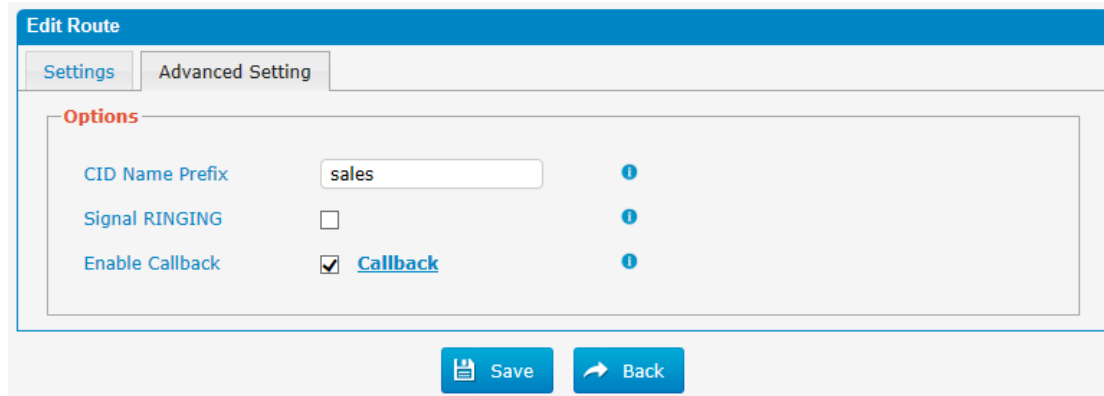
Callback		
<a href="#">+ Add Callback</a>	<a href="#">X Delete the selected Callback</a>	Page 1 of 1(1 Records)
<input type="checkbox"/>	Callback Description	Callback Number Options
<input type="checkbox"/>	callback6001	6001 <a href="#">✎</a> <a href="#">✕</a>

Follow the steps below to use this function.

Step 1: Enable Callback.

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

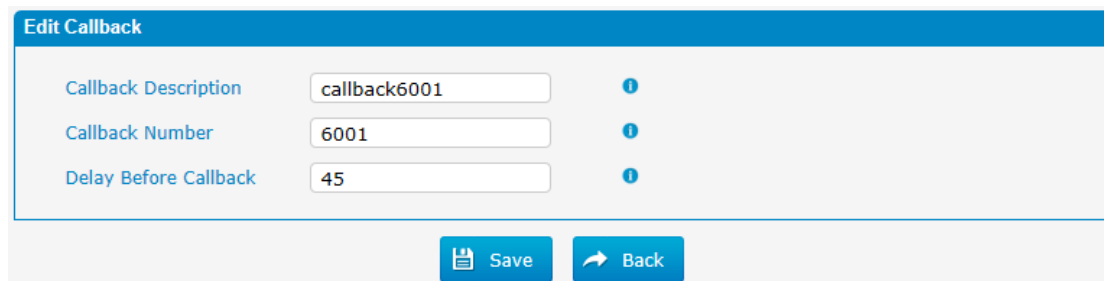
Figure 3.7.7a Inbound route Callback settings



Edit Route		
<div>Settings   <b>Advanced Setting</b></div>		
<b>Options</b>		
CID Name Prefix	sales	?
Signal RINGING	<input type="checkbox"/>	?
Enable Callback	<input checked="" type="checkbox"/> <a href="#">Callback</a>	?
<div>Save   Back</div>		

Step 2: Create Callback number.

Figure 3.7.7b Callback Edit/Add



Edit Callback		
Callback Description	callback6001	?
Callback Number	6001	?
Delay Before Callback	45	?
<div>Save   Back</div>		

## 3.8 PBX Advanced Settings

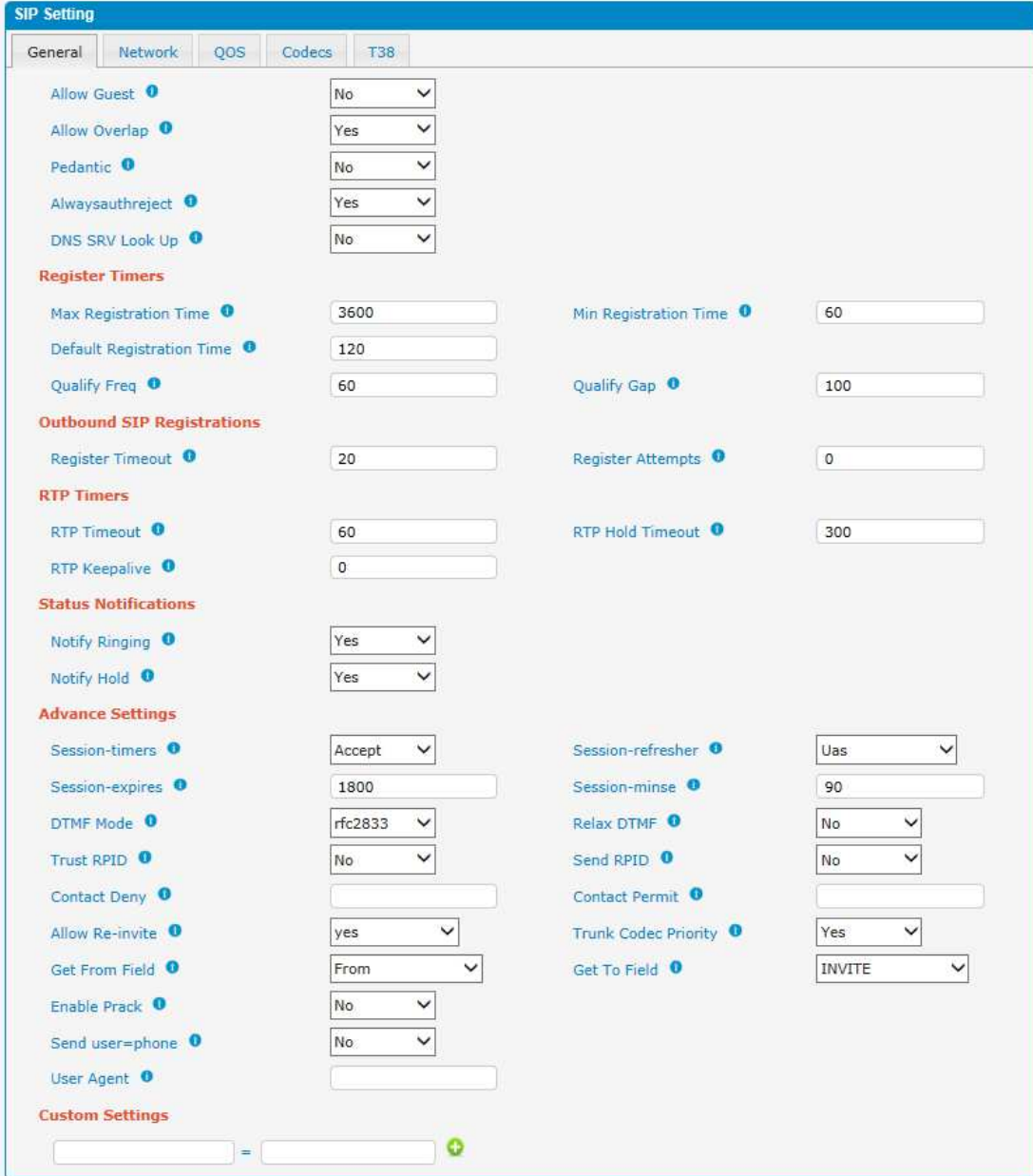
### 3.8.1 SIP settings

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



**SIP Setting**

General | Network | QOS | Codecs | T38

**General**

Allow Guest: No

Allow Overlap: Yes

Pedantic: No

Alwaysauthreject: Yes

DNS SRV Look Up: No

**Register Timers**

Max Registration Time: 3600

Default Registration Time: 120

Qualify Freq: 60

Min Registration Time: 60

Qualify Gap: 100

**Outbound SIP Registrations**

Register Timeout: 20

Register Attempts: 0

**RTP Timers**

RTP Timeout: 60

RTP Keepalive: 0

RTP Hold Timeout: 300

**Status Notifications**

Notify Ringing: Yes

Notify Hold: Yes

**Advance Settings**

Session-timers: Accept

Session-expire: 1800

DTMF Mode: rfc2833

Trust RPID: No

Contact Deny:

Allow Re-invite: yes

Get From Field: From

Enable Prack: No

Send user=phone: No

User Agent:

Session-refresher: Uas

Session-minse: 90

Relax DTMF: No

Send RPID: No

Contact Permit:

Trunk Codec Priority: Yes

Get To Field: INVITE

**Custom Settings**

= +

Save Cancel

Table 3.8.1.1 Description of SIP General Setting

Parameters	Description
Allow guest	Whether allow anonymous registration extension. Default: no. It's recommended to be disabled for security.
Allow overlap	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

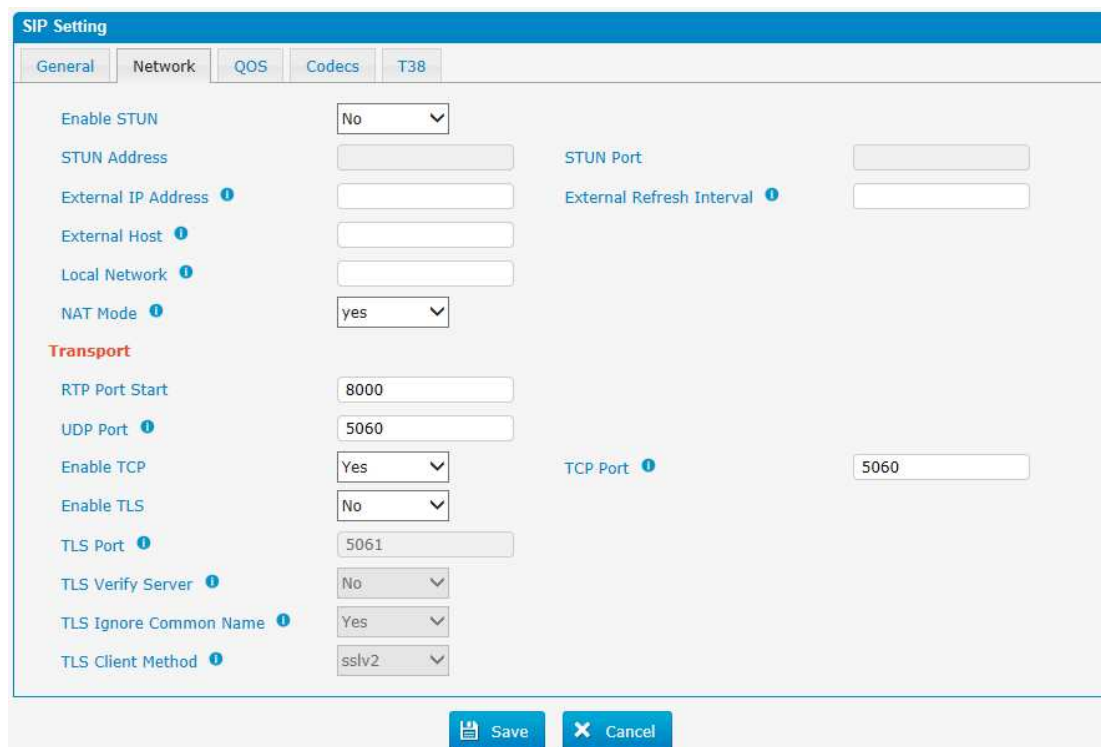
	NOT FOUND". It's recommended to be enabled for security.
DNS SRV Look Up	Please enable this option when your SIP trunk contains more than one IP address.
Maxexpiry	Maximum duration (in seconds) of a SIP registration. Default is 3600 seconds.
Minexpiry	Minimum duration (in seconds) of a SIP registration. Default is 60 seconds.
Defaultexpiry	Default Incoming/Outgoing Registration Time: Default duration (in seconds) of incoming/outgoing registration.
Qualifyfreq	How often 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 handling
Trustpid	If Remote-Party-ID should be trusted
Sendrpip	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 would 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



The screenshot shows the 'SIP Setting' window with the 'Network' tab selected. The configuration includes:

- Enable STUN:** No
- STUN Address:** (empty text field)
- STUN Port:** (empty text field)
- External IP Address:** (empty text field)
- External Refresh Interval:** (empty text field)
- External Host:** (empty text field)
- Local Network:** (empty text field)
- NAT Mode:** yes
- Transport Section:**
  - RTP Port Start:** 8000
  - UDP Port:** 5060
  - Enable TCP:** Yes
  - Enable TLS:** No
  - TCP Port:** 5060
  - TLS Port:** 5061
  - TLS Verify Server:** No
  - TLS Ignore Common Name:** Yes
  - TLS Client Method:** sslv2

Buttons at the bottom: Save, Cancel.

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



The interface shows the 'SIP Setting' window with the 'QOS' tab selected. It contains two columns of settings:

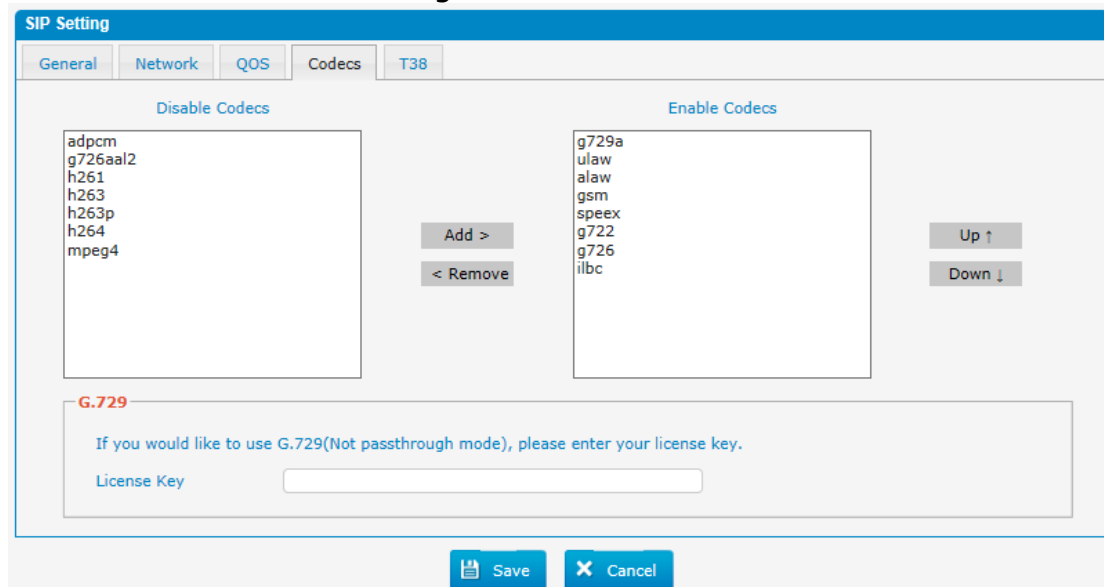
Setting	Value	Setting	Value
Tos SIP:	CS3	Cos SIP:	3
Tos Audio:	EF	Cos Audio:	5
Tos Video:	AF41	Cos Video:	4

At the bottom, there are 'Save' and 'Cancel' buttons.

### 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](http://en.wikipedia.org/wiki/List_of_codecs)

Figure 3.8.1.4 codecs



The interface shows the 'SIP Setting' window with the 'Codecs' tab selected. It features two lists:

- Disable Codecs:** adpcm, g726aal2, h261, h263, h263p, h264, mpeg4
- Enable Codecs:** g729a, ulaw, alaw, gsm, speex, g722, g726, ilbc

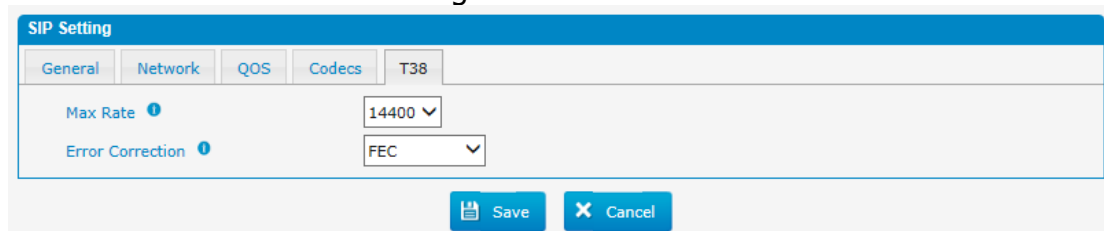
Buttons 'Add >' and '< Remove' are between the lists. 'Up ↑' and 'Down ↓' buttons are next to the 'Enable Codecs' list. Below the lists, there is a section for 'G.729' with a message: 'If you would like to use G.729(Not passthrough mode), please enter your license key.' and a 'License Key' input field.

At the bottom, there are 'Save' and 'Cancel' buttons.

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



The interface shows the 'SIP Setting' window with the 'T38' tab selected. It contains two settings:

Setting	Value
Max Rate	14400
Error Correction	FEC

At the bottom, there are 'Save' and 'Cancel' buttons.

### 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's supported by the asterisk-based IPPBX.

Figure 3.8.2 IAX setting

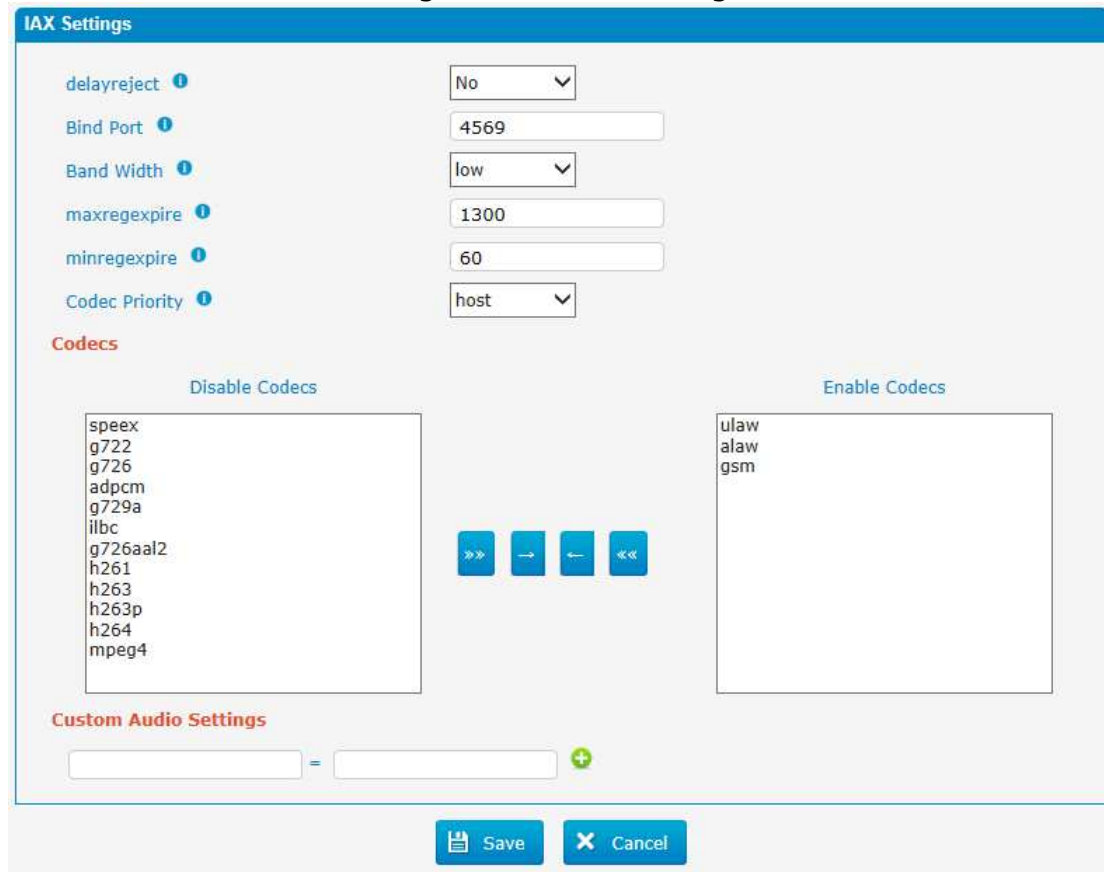


Table 3.8.2 Description of IAX setting

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



PIN Sets				
PIN Sets are used to manage lists of PINs that can be used to access restricted features such as Outbound Routes. The PIN can also be added to the CDR record's 'accountcode' field.				
+ Add PIN Set		X Delete the selected PIN Sets		Page 1 of 1(1 Records)
<input type="checkbox"/>	PIN Set Name	Record in CDR	PIN List	Options
<input type="checkbox"/>	test	ON	1234	 

Figure 3.8.3a PIN Set Edit

Edit PIN Set

PIN Set Name

test

Record in CDR

☒

PIN List

1234

Save

Back

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

### 3.8.4 PIN Users

Figure 3.8.4 PIN Users

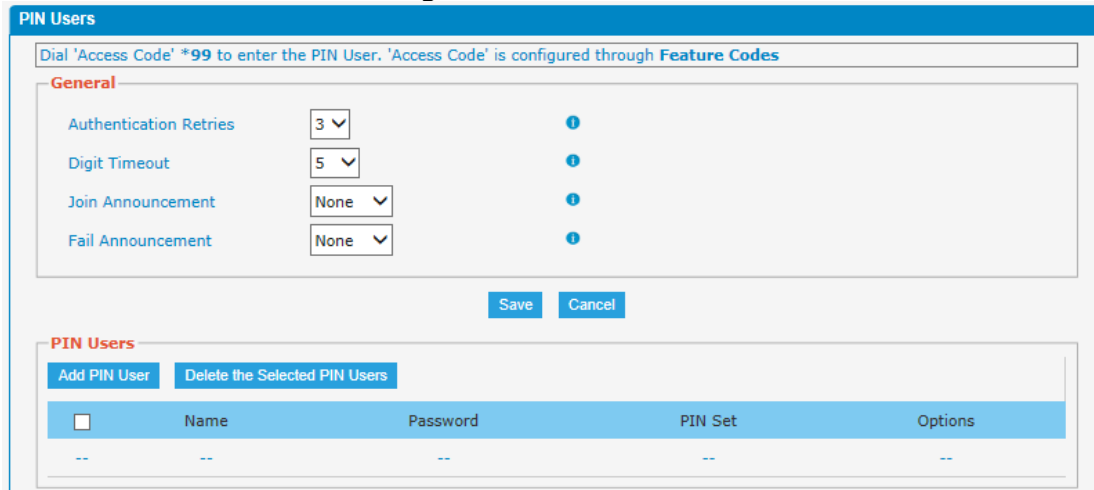


Table 3.8.4 Description of PIN Users

Parameters	Description
Authentication Retries	Number of times to retry when receiving a 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

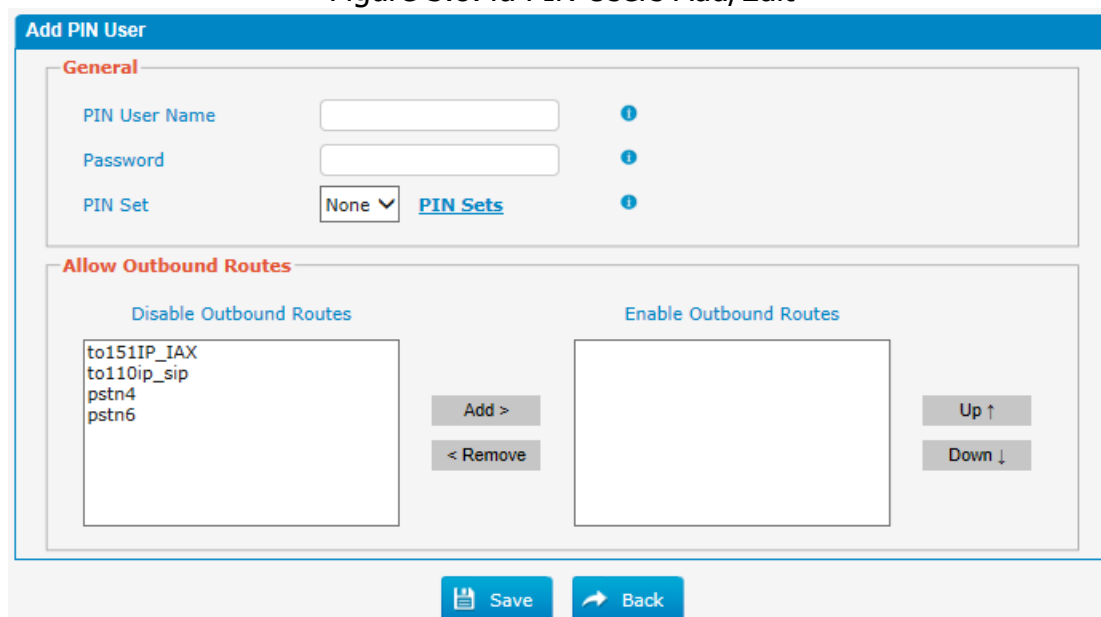


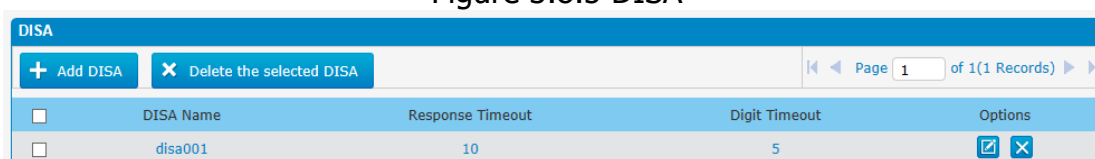
Table 3.8.4a Description of PIN Users Add/Edit

Parameters	Description
PIN User Name	<p>A character-based name for this PIN list, e.g. "MUCPIN"</p> <p>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.</p> <ul style="list-style-type: none"> <li>● PINs entered are checked against those stored by the System. If an invalid PIN is entered, the PIN is requested again.</li> <li>● 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.</li> <li>● The system administrator can also configure to require users to enter a PIN before making any external call.</li> </ul>
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



DISA			
<input type="button" value="+ Add DISA"/> <input type="button" value="X Delete the selected DISA"/>		Page 1 of 1(1 Records)	
<input type="checkbox"/>	DISA Name	Response Timeout	Digit Timeout
<input type="checkbox"/>	disa001	10	5

Figure 3.8.5 DISA Edit

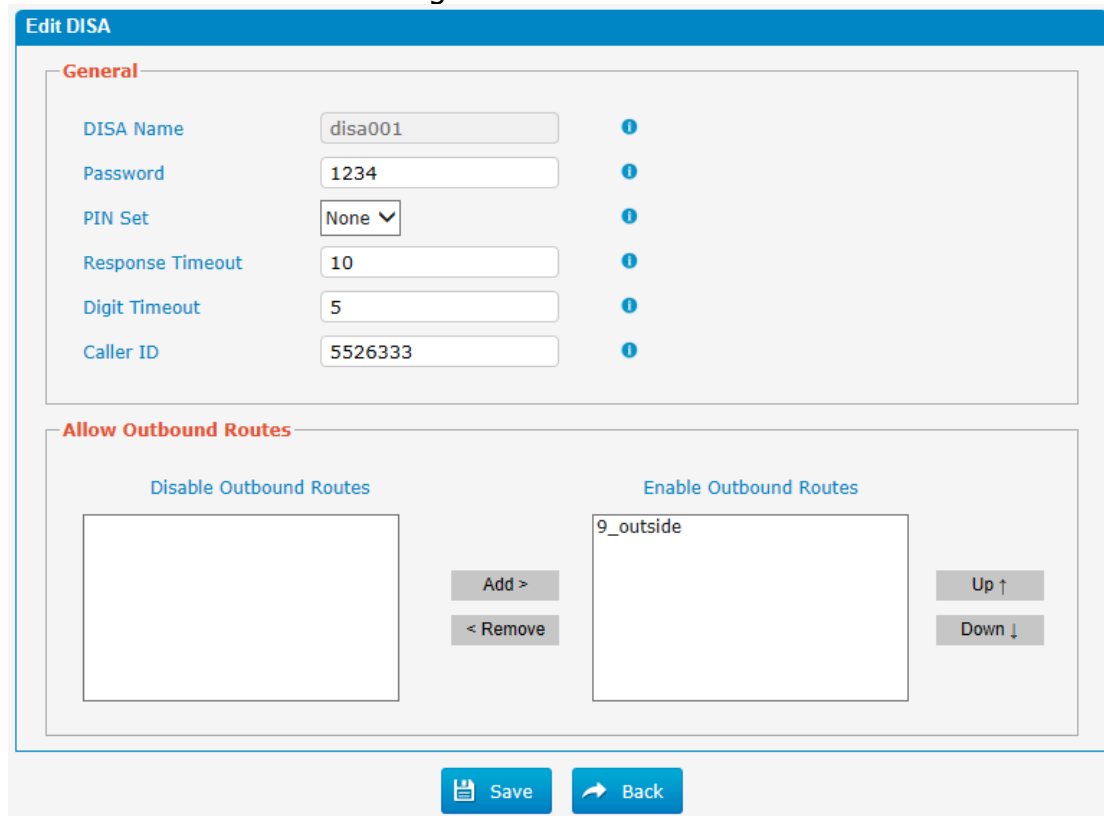


Table 3.8.5 Description of DISA Edit

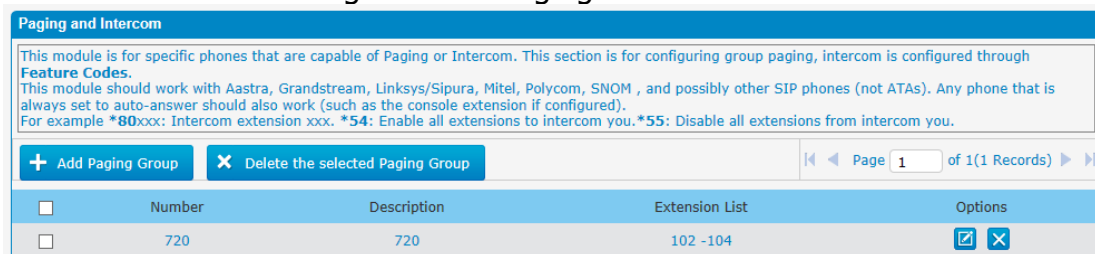
Parameters	Description
DISA Name	Give this DISA application a name to help you identify it.
Password	The password for this DISA.
PIN Set	Optional: select a PIN set to use. If using this option, leave the password field blank
Response Timeout	The maximum amount of time the system will wait before hanging 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



**Paging and Intercom**

This module is for specific phones that are capable of Paging or Intercom. This section is for configuring group paging, intercom is configured through **Feature Codes**.  
This module should work with Aastra, Grandstream, Linksys/Sipura, Mitel, Polycom, SNOM, and possibly other SIP phones (not ATAs). Any phone that is always set to auto-answer should also work (such as the console extension if configured).  
For example \*80xxx: Intercom extension xxx. \*54: Enable all extensions to intercom you. \*55: Disable all extensions from intercom you.

+ Add Paging Group    X Delete the selected Paging Group    Page 1 of 1(1 Records)



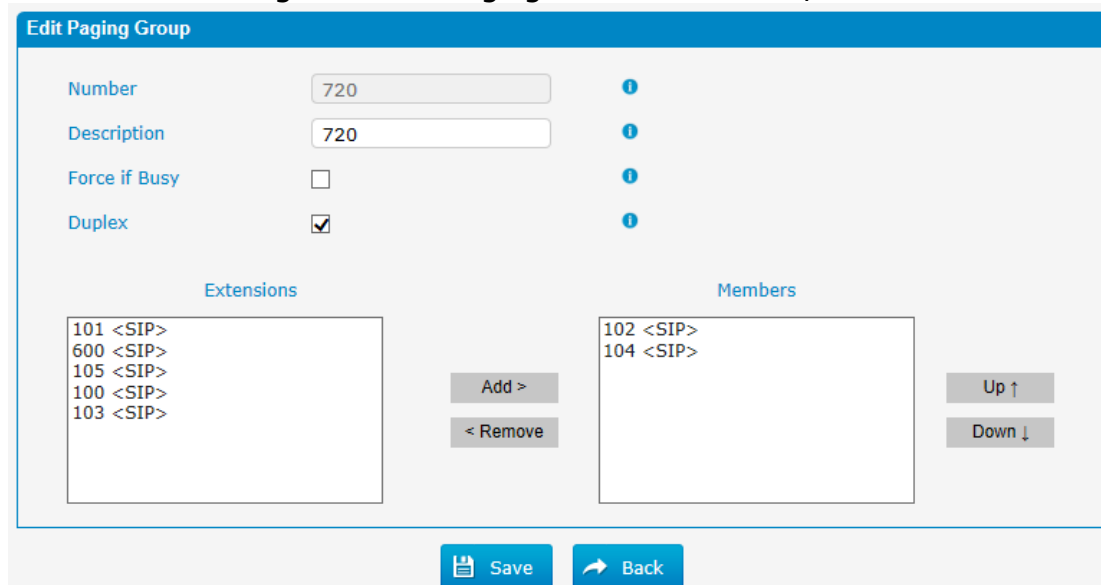
<input type="checkbox"/>	Number	Description	Extension List	Options
<input type="checkbox"/>	720	720	102 -104	 

Figure 3.8.6a Paging and Intercom Edit/Add



**Edit Paging Group**

Number: 720 ⓘ

Description: 720 ⓘ

Force if Busy: ☐ ⓘ

Duplex: ☒ ⓘ

**Extensions**

101 <SIP>  
600 <SIP>  
105 <SIP>  
100 <SIP>  
103 <SIP>

Add >

< Remove

**Members**

102 <SIP>  
104 <SIP>

Up ↑

Down ↓

Save    Back

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



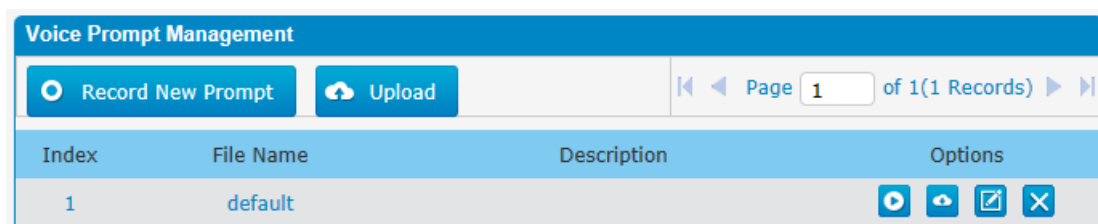
	in the group to talk and be heard by all.
Members	Select members in this group.

## 3.9 Voice Management

### 3.9.1 System Recordings

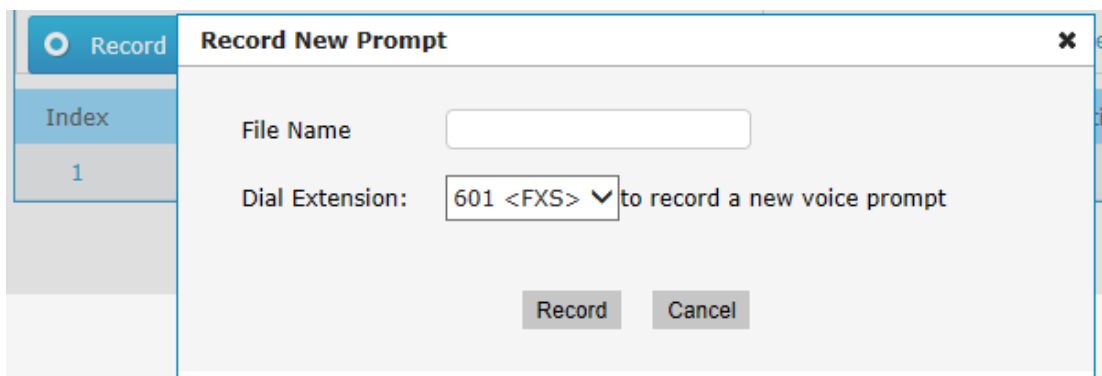
We can record or upload the prompts in this page; you can also play it directly to confirm if it's a valid one, you can also download it and save it as a backup.

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

Upload Voice Prompt

File Name

Browse...

Description

Support format: 'PCM Mono, 16 Kb/s, at 8000Hz', 'Alaw/ULaw Mono, 1 Kb/s, at 8000Hz', 'GSM Mono, 1Kb/s, at 8000Hz'.

Save

Cancel

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.

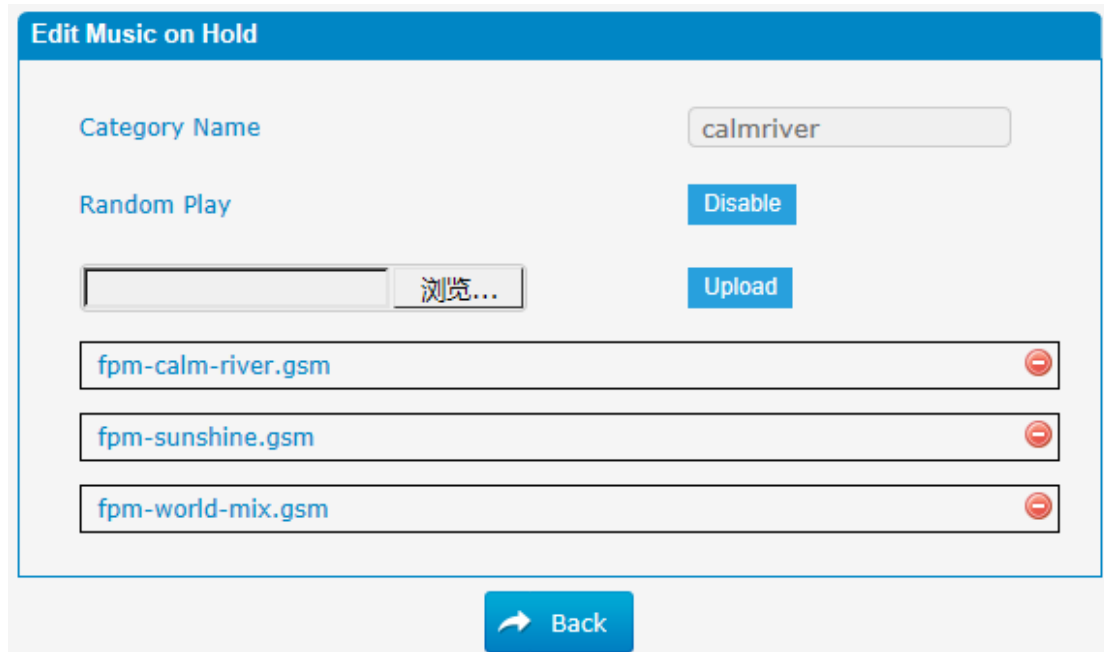
**Note:** The file size must not be larger than 1.8 MB, and the file must be WAV format.

## 3.9.2 Music on Hold

Figure 3.9.2 Music on Hold

Music on Hold		
<div>Add Music Category</div>		
Name	Random Play	Options
calmriver	ON	<div></div> <div></div>
test	OFF	<div></div> <div></div>

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

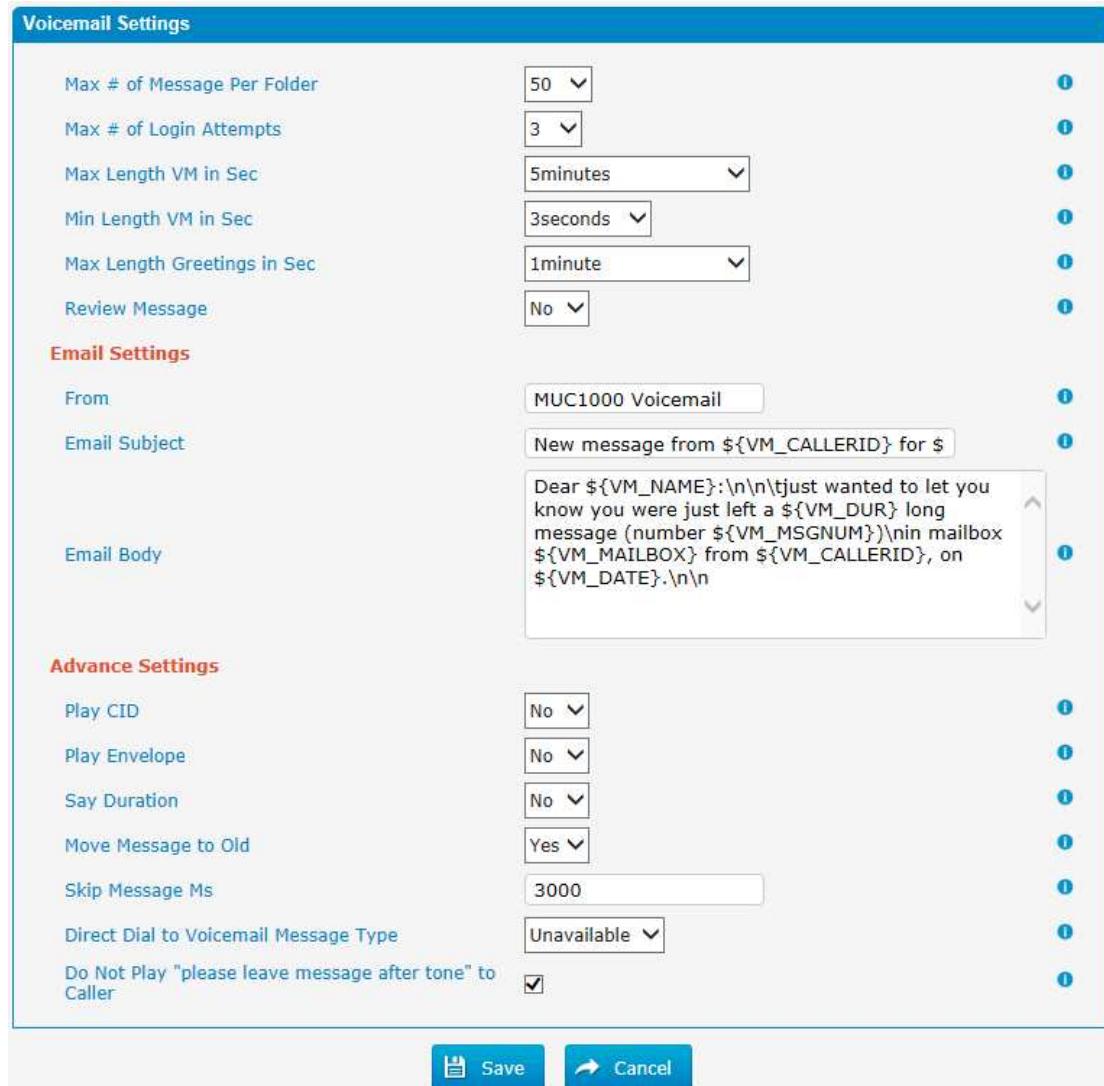


Table 3.9.3 Description of Voicemail Setting

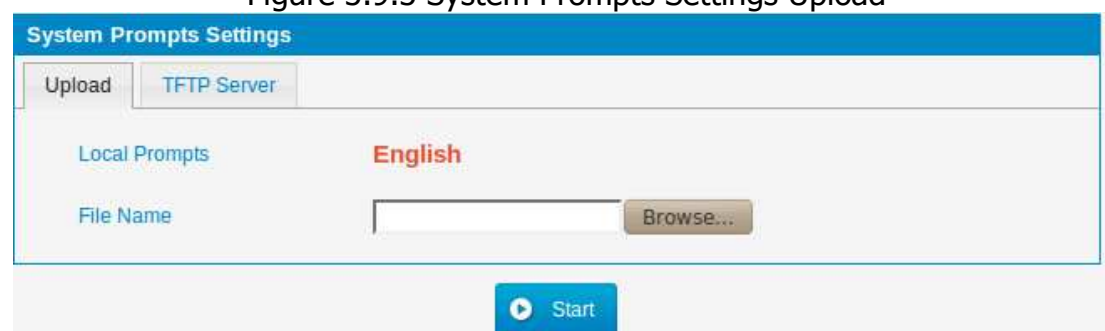
Parameters	Description
Max # of Message Per Folder	Set the maximum number of messages that can be stored in a single voicemail box.
Max # of Login Attempts	Max number of failed login attempts
Max Length VM in	Set the maximum length of a single voicemail message.

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



The screenshot shows the 'System Prompts Settings' interface. At the top, there are two tabs: 'Upload' and 'TFTP Server'. Below the tabs, the language is set to 'English'. There is a section for 'Local Prompts' with a 'File Name' input field and a 'Browse...' button. At the bottom, there is a 'Start' button with a play icon.

Figure 3.9.3a System Prompts Settings TFTP

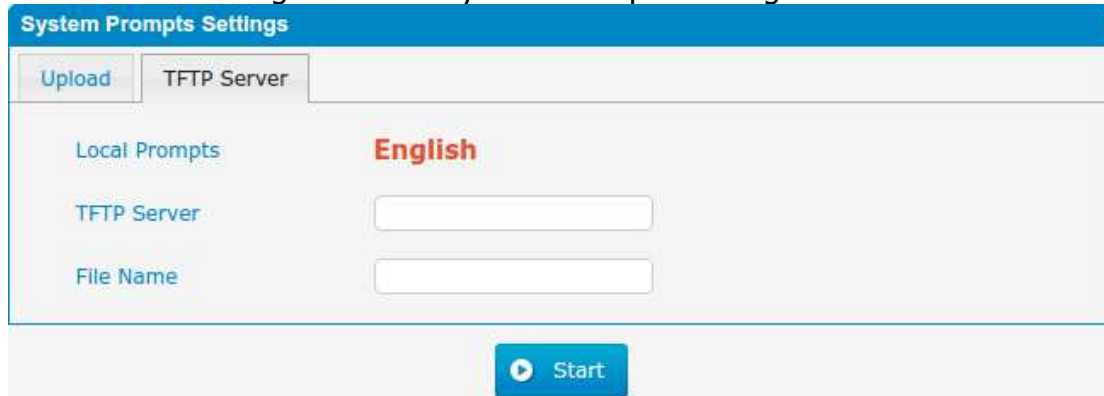


Table 3.9.3 Description of System Prompts Settings

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

Firewall Rules

General

Status

Active

Enable Firewall

☒

Enable Log

☒

Firewall Logs

Drop All

☐

Save

Cancel

Rules

Add Rule

Delete the selected Rules

	Name	Protocol	IP	Port	MAC	Target	Sort	Options
<input type="checkbox"/>	HTTPS	TCP		443		ACCEPT	↕	
<input type="checkbox"/>	test	TCP		123:124		DROP	↕	
<input type="checkbox"/>	Ping	ICMP				ACCEPT	↕	
<input type="checkbox"/>	SIP	UDP		5004:5082		ACCEPT	↕	
<input type="checkbox"/>	RTP	UDP		8000:20000		ACCEPT	↕	
<input type="checkbox"/>	DNS	UDP		53		ACCEPT	↕	
<input type="checkbox"/>	TFTP	UDP		69		ACCEPT	↕	
<input type="checkbox"/>	SMTP	TCP		25		ACCEPT	↕	
<input type="checkbox"/>	POP3	TCP		110		ACCEPT	↕	
<input type="checkbox"/>	test1	TCP		80		ACCEPT	↕	

Defence

Add Rule

Delete the selected Rules

	Name	Protocol	IP	Port	Rate	Hit	Options
<input type="checkbox"/>	SIP5060H20S	UDP		5060		120/min	
<input type="checkbox"/>	SIP5060H120S	UDP		5060		20/sec	
<input type="checkbox"/>	SSH8022H5M	TCP		8022		5/min	

Figure 3.10.1a Firewall Rules Edit/Add

Edit Rule

General

Name

HTTPS

Protocol

TCP

IP

Port

443

MAC

Target

ACCEPT

Save

Back

Table 3.10.1a Description of Firewall Rules

Parameters	Description
Name	A name for this rule. e.g.: HTTP.
Protocol	The protocols for this rule.
IP	The IP address for this rule. The format of IP address is: IP/mask <ul style="list-style-type: none"> <li>● Ex:192.168.6.88/32 for ip 192.168.6.88</li> <li>● Ex:192.168.6.0/24 for ip from 192.168.6.0 to 192.168.6.255</li> </ul>
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	<ul style="list-style-type: none"> <li>● ACCEPT:Accept the access from remote hosts</li> <li>● DROP:Drop the access from remote hosts</li> <li>● REJECT:Reject the access from remote hosts</li> </ul>

Figure 3.10.1b Firewall Defense Edit/Add

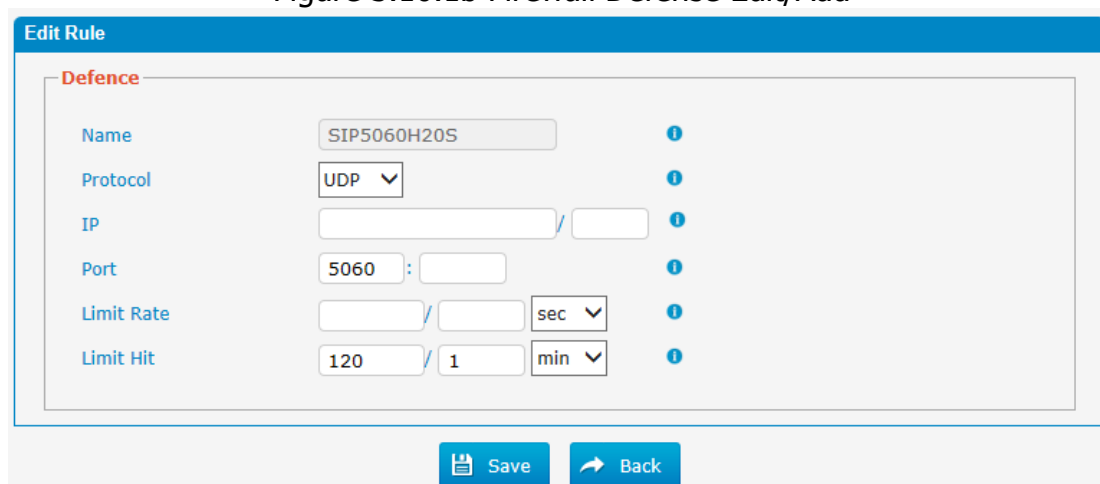


Table 3.10.1b Description of Firewall Defense Edit/Add

Parameters	Description
Name	A name for this rule. e.g.: HTTP.
Protocol	The protocols for this rule.
IP	The IP address for this rule. The format of IP address is: IP/mask <ul style="list-style-type: none"> <li>● Ex:192.168.6.88/32 for ip 192.168.6.88</li> <li>● Ex:192.168.6.0/24 for ip from 192.168.6.0 to 192.168.6.255</li> </ul>
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. e.g.:(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

Alert Settings			
Attack Type	Phone Notification	Email Notification	Option
IPATTACK	yes	yes	<input checked="" type="checkbox"/>
WEBLOGIN	no	yes	<input checked="" type="checkbox"/>

Figure 3.10.2a Alert Settings Edit

Edit Alert Settings(WEBLOGIN)

**Phone Notification Settings**

Phone Notification
NO

Number

Attempts
1

Interval
60

Prompt
default

**Email Notification Settings**

Email Notification
NO

To

Email Subject

Email Body

Pbx Host Name: \${HOSTNAME}  
Login Time: \${DATETIME}  
Login User Name: \${USERNAME}  
Attack Src IP: \${SOURCEIP}

Save
Back

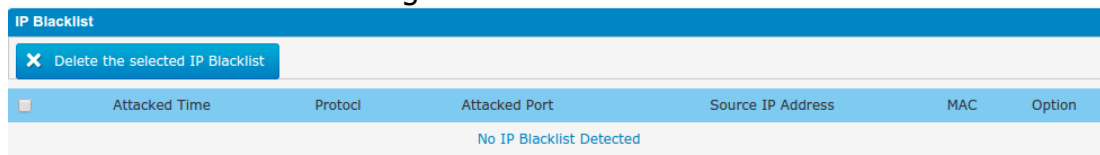
Table 3.10.2a Description of Alert Settings Edit

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



IP Blacklist					
X Delete the selected IP Blacklist					
Attacked Time	Protocol	Attacked Port	Source IP Address	MAC	Option
No IP Blacklist Detected					

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

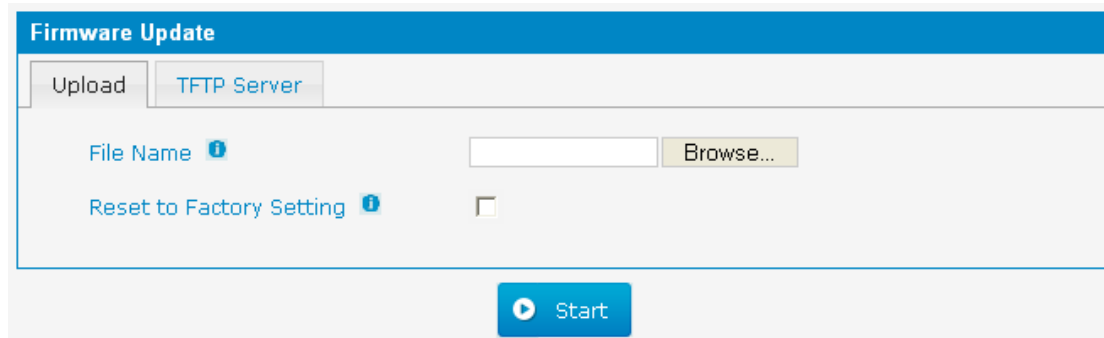
Upgrading of the firmware is possible through the Administrator Web interface using a TFTP Server or an Upload  
Enter your TFTP Server IP address and firmware file location, then click start to update the firmware

Notes:

1. If enabled "Reset configuration to Factory Defaults", System will restore to factory default settings.

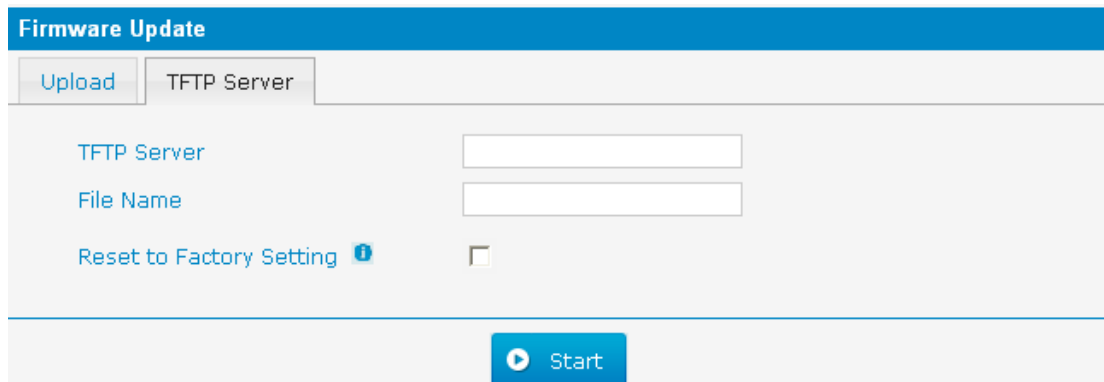
2. When update the firmware, please don't turn off the power. Or the system will get damaged.

Figure 3.10.3 Firmware Update Upload



The interface shows the 'Firmware Update' section with two tabs: 'Upload' and 'TFTP Server'. The 'Upload' tab is active. It contains a 'File Name' field with an information icon, a 'Browse...' button, and a 'Reset to Factory Setting' checkbox with an information icon. A 'Start' button is located at the bottom right.

Figure 3.10.3a Firmware Update TFTP



The interface shows the 'Firmware Update' section with two tabs: 'Upload' and 'TFTP Server'. The 'TFTP Server' tab is active. It contains a 'TFTP Server' field, a 'File Name' field, and a 'Reset to Factory Setting' checkbox with an information icon. A 'Start' button is located at the bottom right.

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.

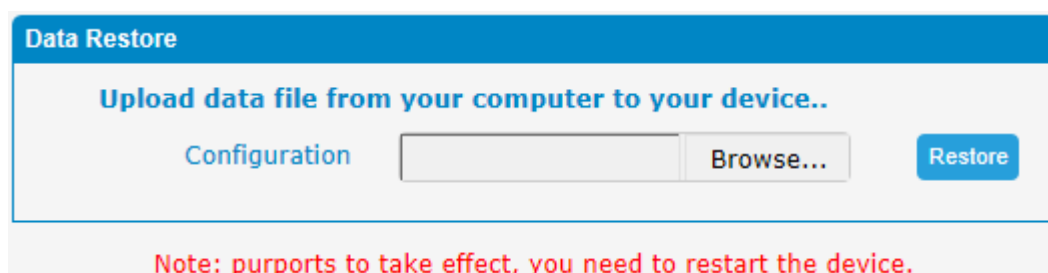
### 3.10.5 Data Restore

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.

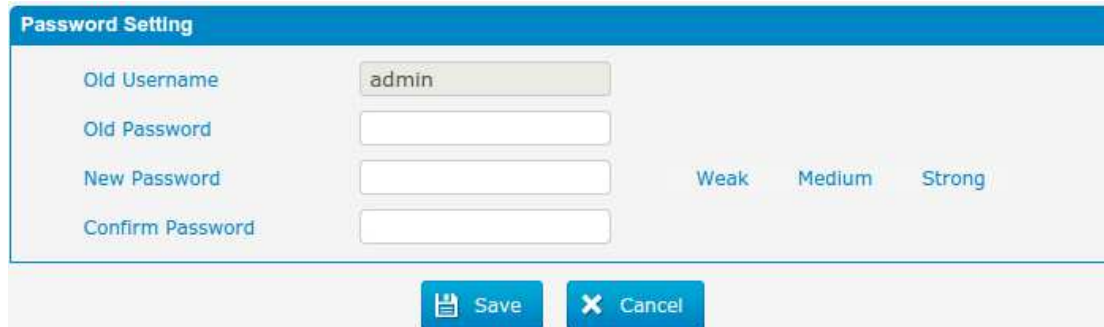
Figure 3.10.5



### 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



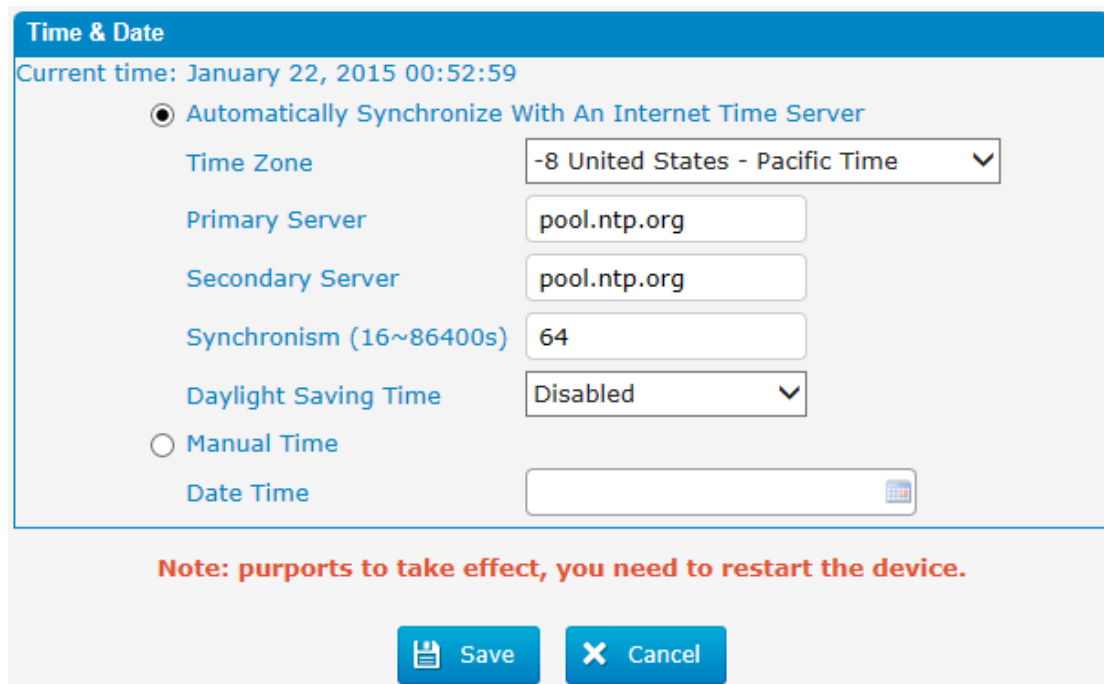
The Password Setting form has a blue header bar with the title "Password Setting". Below the header, there are four input fields: "Old Username" (containing "admin"), "Old Password", "New Password", and "Confirm Password". To the right of the "New Password" field, there are three radio buttons labeled "Weak", "Medium", and "Strong". At the bottom of the form, there are two buttons: "Save" (with a floppy disk icon) and "Cancel" (with an "X" icon).

### 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



The Time & Date form has a blue header bar with the title "Time & Date". Below the header, it shows the "Current time: January 22, 2015 00:52:59". There are two radio buttons: "Automatically Synchronize With An Internet Time Server" (selected) and "Manual Time". Under "Automatically Synchronize", there are five input fields: "Time Zone" (a dropdown menu showing "-8 United States - Pacific Time"), "Primary Server" (containing "pool.ntp.org"), "Secondary Server" (containing "pool.ntp.org"), "Synchronism (16~86400s)" (containing "64"), and "Daylight Saving Time" (a dropdown menu showing "Disabled"). Under "Manual Time", there is a "Date Time" input field with a calendar icon. At the bottom of the form, there is a red note: "Note: purports to take effect, you need to restart the device." and two buttons: "Save" (with a floppy disk icon) and "Cancel" (with an "X" icon).

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

Reset all the settings of the device to default configurations.

Note: You need to restart the settings to take effect

Reset

Reset to Factory Defaults

Click this button to reset Factory Default settings

### 3.10.9 Reboot

Figure 3.10.9 Reboot

Reboot

Click this button to reboot the device.

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

General Settings

Time Settings

Time Server

pool.ntp.org

Time Zone for Aastra Phones

US-Eastern EST

Time Zone for Akuvox Phones

-9 United States-Alaska Time

Time Zone for Cisco SPA Phones

GMT-9:00 (US Alaska Time)

Time Zone for Cisco 7940/7960 Phones

GMT-9:00 (US Alaska Time)

Time Zone for Farvil Phones

GMT-9:00 United States-Alaska Time

Time Zone for Grandstream Phones

GMT-9:00 (US Alaska Time)

Time Zone for Panasonic Phones

GMT-9:00 (US Alaska Time)

Time Zone for Polycom Phones

GMT-9:00 (US Alaska Time)

Time Zone for Snom Phones

-9 United States - Alaska Time

Time Zone for Yealink Phones

GMT-9:00 United States-Alaska Time

Daylight Saving Time

Enable

Disable

Start Time

Month/Day/Hour

Month/Day/Hour

End Time

Month/Day/Hour

Month/Day/Hour

Offset

0

minutes

Save

Cancel

### 3.11.2 Phones

Figure 3.11.2 Configured Phones

Configured Phones							
+ Add Phone		X Delete the selected Phones					
MAC	IP	Phone Model	Extension	Active	Name	Options	
--	--	--	--	--	--	--	

Not Configured Phones			
Refresh		页码 1 / 1(8条记录)	
MAC	IP	Phone Model	Options
000B826C7D08	<a href="#">192.168.6.162</a>	Grandstream GXP1450 1.0.8.6	<input checked="" type="checkbox"/>
000B826C7D0E	<a href="#">192.168.6.219</a>	Grandstream GXP1450 1.0.6.11	<input checked="" type="checkbox"/>
001565736B87	<a href="#">192.168.6.76</a>	Yealink T21P 34.72.0.20	<input checked="" type="checkbox"/>
001565828AFA	<a href="#">192.168.6.164</a>	Yealink T19P 31.72.0.75	<input checked="" type="checkbox"/>
00A859D2919E	<a href="#">192.168.6.72</a>	Farvill	<input checked="" type="checkbox"/>
0C110500388C	<a href="#">192.168.6.167</a>	Akuvox SP-R53 53.0.3.41	<input checked="" type="checkbox"/>
0C1105026B54	<a href="#">192.168.6.160</a>	Akuvox SP-R59 59.0.3.41	<input checked="" type="checkbox"/>
0C110502BFD8	<a href="#">192.168.6.71</a>	Akuvox SP-R50 50.0.3.41	<input checked="" type="checkbox"/>

Figure 3.11.2a Edit Phone

Edit Phone					
General		Keys Settings			
<b>General</b>					
Active	<input checked="" type="checkbox"/>				
Name	<input type="text"/>				
MAC	000B826C7D08				
Manufacturer	Grandstream				
Phone Type	GXP1450				
Network Parameters					
<input checked="" type="radio"/> Use Phone's Setting <input type="radio"/> Dynamic(DHCP) <input type="radio"/> Static IP Address					
IP Address	<input type="text"/>				
Subnet Mask	<input type="text"/>				
Gateway	<input type="text"/>				
DNS	<input type="text"/>				
Auto Answer	<input type="checkbox"/>				
Call Waiting	<input checked="" type="checkbox"/>				
Key As Send	#				
<b>Account</b>					
Line	Extension	Label	Display Name	Active	
<input type="checkbox"/> Line1	100	<input type="text"/>	<input type="text"/>	<input type="checkbox"/>	
<input type="checkbox"/> Line2	100	<input type="text"/>	<input type="text"/>	<input type="checkbox"/>	
Save		Back			

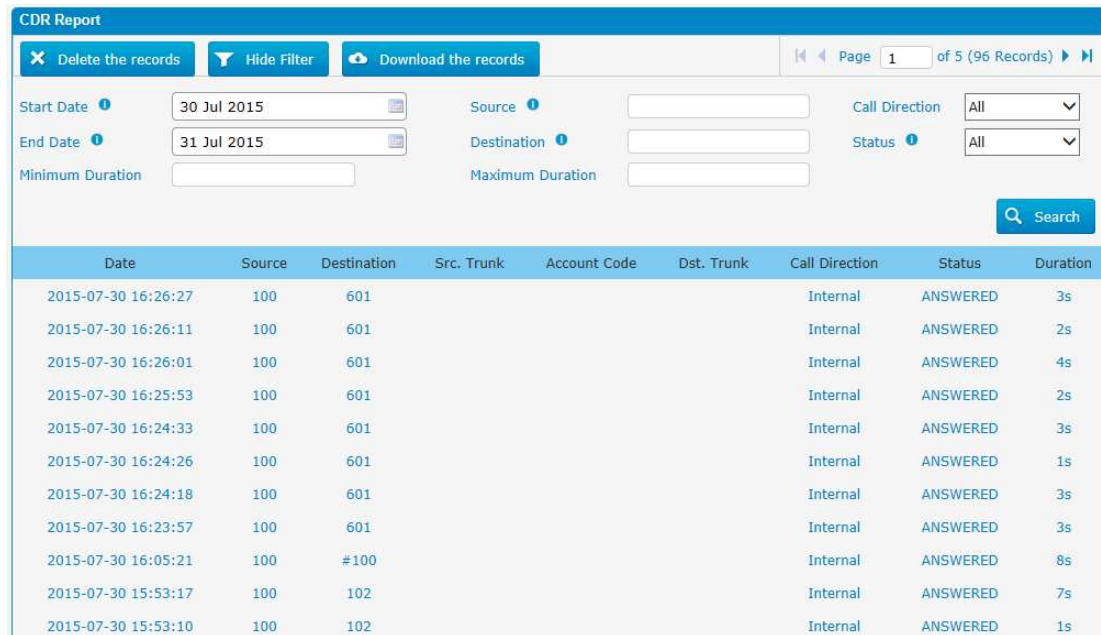


## 3.12 Reports

### 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



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

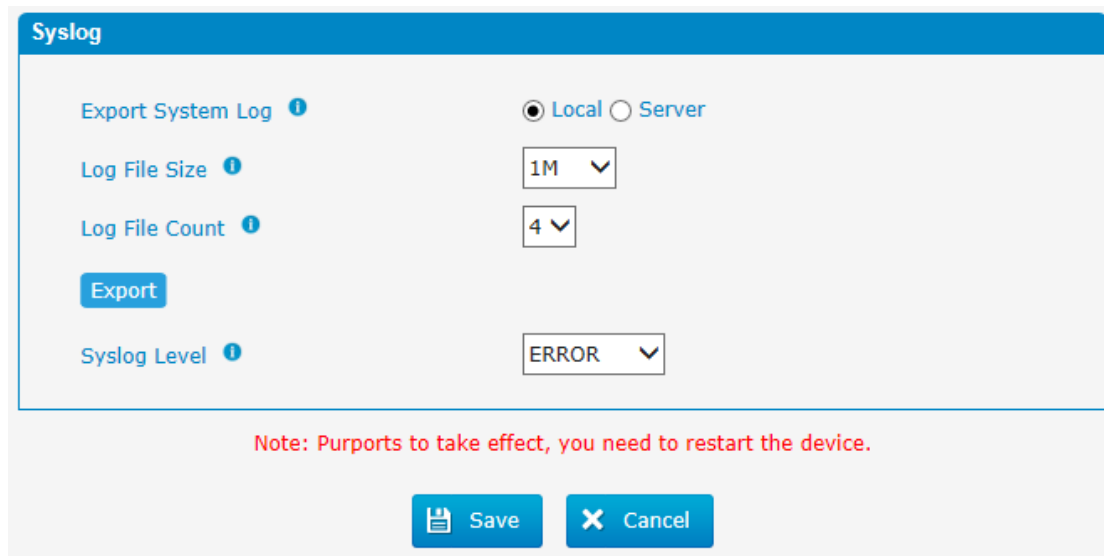
Table 3.12.1 CDR Report

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

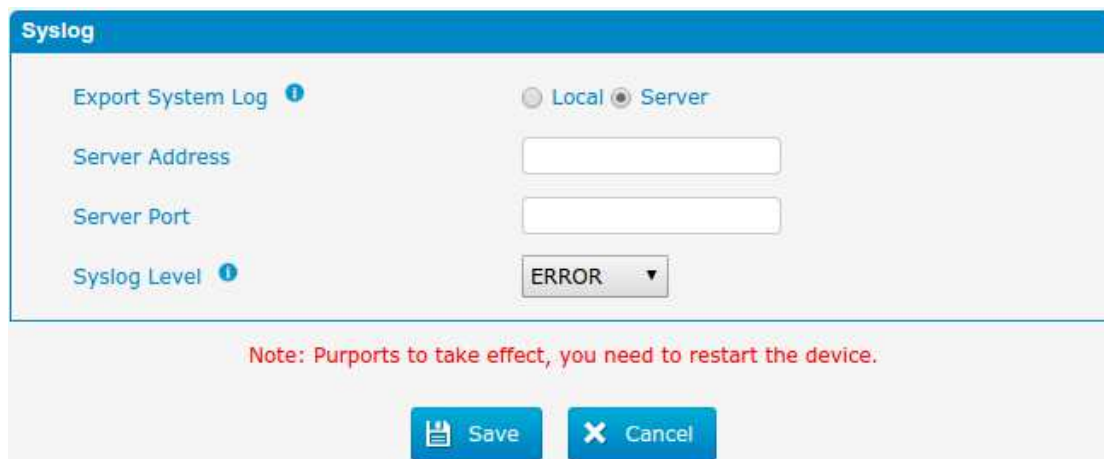
Syslog is a standard for network device data logging. It allows separation of the software that generates messages from the system that stores them and the software that reports and analyzes them. It also provides devices which would otherwise be unable to communicate a means to notify administrators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



The screenshot shows the 'Syslog' configuration window. It has a blue header bar with the title 'Syslog'. Below the header, there are several settings: 'Export System Log' with a blue information icon, 'Log File Size' with a blue information icon, 'Log File Count' with a blue information icon, 'Export' button, and 'Syslog Level' with a blue information icon. The 'Export System Log' section has two radio buttons: 'Local' (selected) and 'Server'. The 'Log File Size' is set to '1M' with a dropdown arrow. The 'Log File Count' is set to '4' with a dropdown arrow. The 'Export' button is blue. The 'Syslog Level' is set to 'ERROR' with a dropdown arrow. Below the settings, there is a red note: 'Note: Purports to take effect, you need to restart the device.' At the bottom, there are two buttons: 'Save' (blue) and 'Cancel' (blue with a red X).

Figure 3.12.2a System logs Server



The screenshot shows the 'Syslog' configuration window. It has a blue header bar with the title 'Syslog'. Below the header, there are several settings: 'Export System Log' with a blue information icon, 'Server Address', 'Server Port', 'Syslog Level' with a blue information icon, and 'ERROR' dropdown. The 'Export System Log' section has two radio buttons: 'Local' and 'Server' (selected). The 'Server Address' and 'Server Port' are empty text input fields. The 'Syslog Level' is set to 'ERROR' with a dropdown arrow. Below the settings, there is a red note: 'Note: Purports to take effect, you need to restart the device.' At the bottom, there are two buttons: 'Save' (blue) and 'Cancel' (blue with a red X).

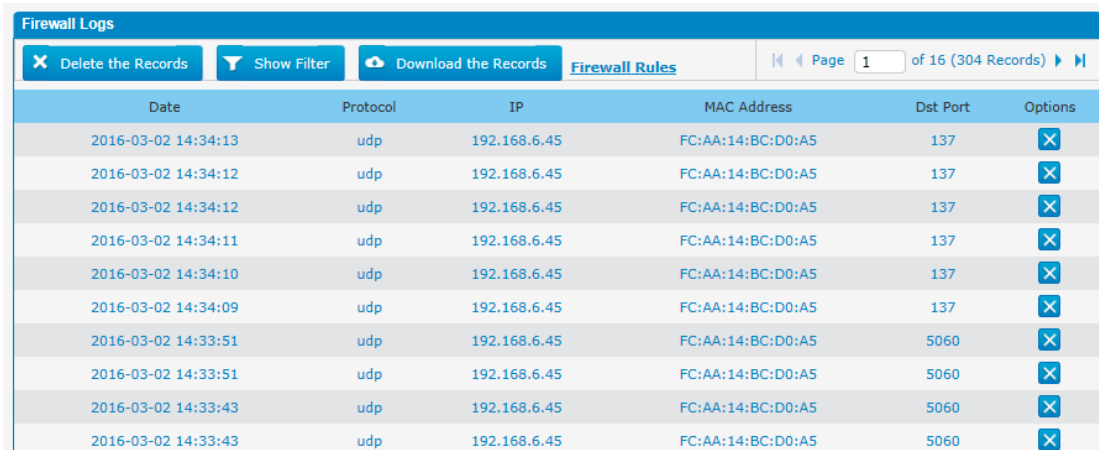
Table 3.12.2 Description of System logs

Parameters	Description
Export System Log	<ul style="list-style-type: none"> <li>● Local: save log in local</li> <li>● Server: save log in server</li> </ul>
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



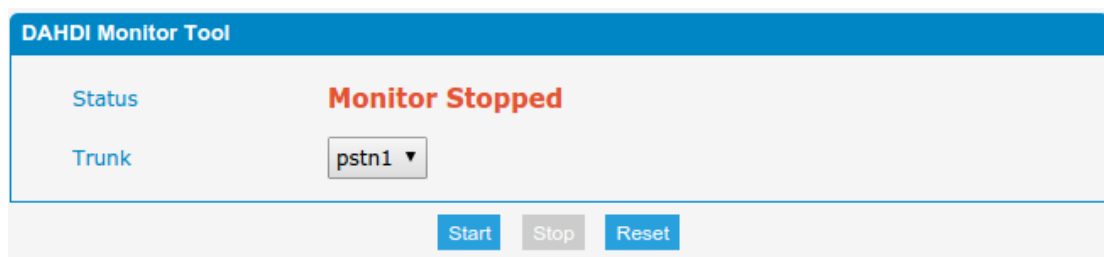
Date	Protocol	IP	MAC Address	Dst Port	Options
2016-03-02 14:34:13	udp	192.168.6.45	FC:AA:14:BC:D0:A5	137	X
2016-03-02 14:34:12	udp	192.168.6.45	FC:AA:14:BC:D0:A5	137	X
2016-03-02 14:34:12	udp	192.168.6.45	FC:AA:14:BC:D0:A5	137	X
2016-03-02 14:34:11	udp	192.168.6.45	FC:AA:14:BC:D0:A5	137	X
2016-03-02 14:34:10	udp	192.168.6.45	FC:AA:14:BC:D0:A5	137	X
2016-03-02 14:34:09	udp	192.168.6.45	FC:AA:14:BC:D0:A5	137	X
2016-03-02 14:33:51	udp	192.168.6.45	FC:AA:14:BC:D0:A5	5060	X
2016-03-02 14:33:51	udp	192.168.6.45	FC:AA:14:BC:D0:A5	5060	X
2016-03-02 14:33:43	udp	192.168.6.45	FC:AA:14:BC:D0:A5	5060	X
2016-03-02 14:33:43	udp	192.168.6.45	FC:AA:14:BC:D0:A5	5060	X

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



**DAHDI Monitor Tool**

Status **Monitor Stopped**

Trunk pstn1 ▼

Start Stop Reset

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

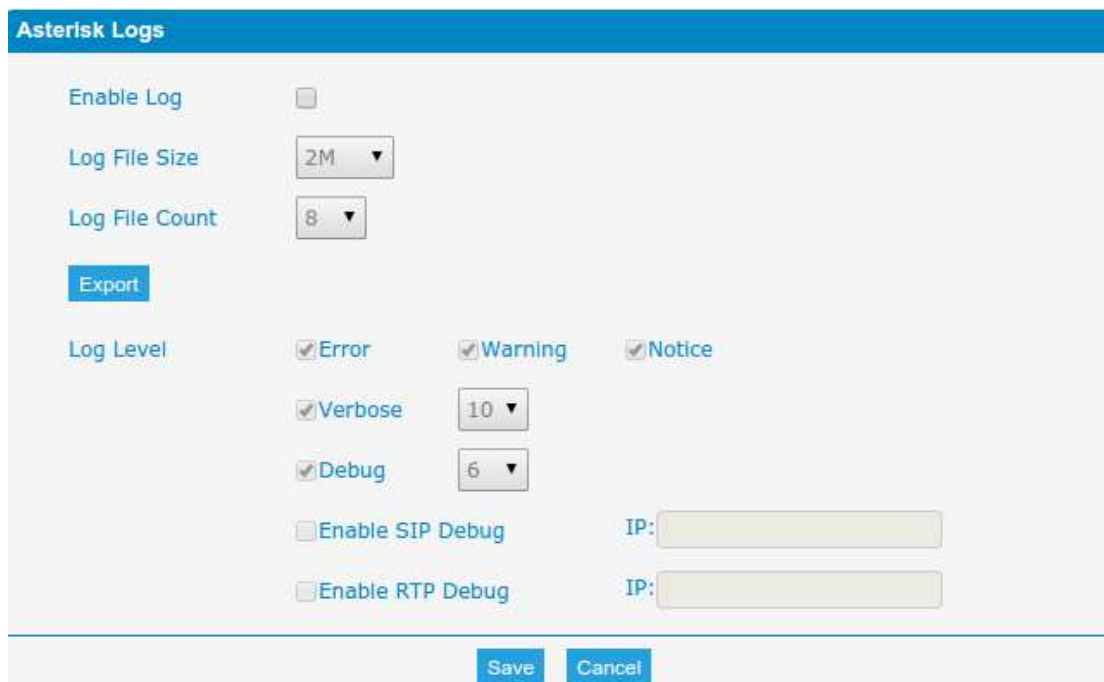


Table 3.12.4a Description of Asterisk Logs

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

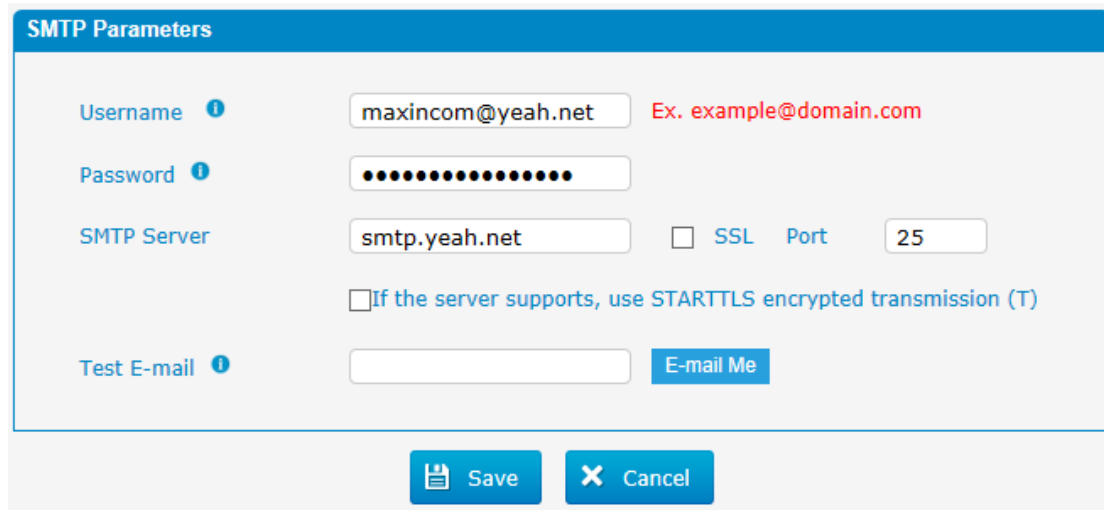


Table 3.13.1 SMTP Parameters

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

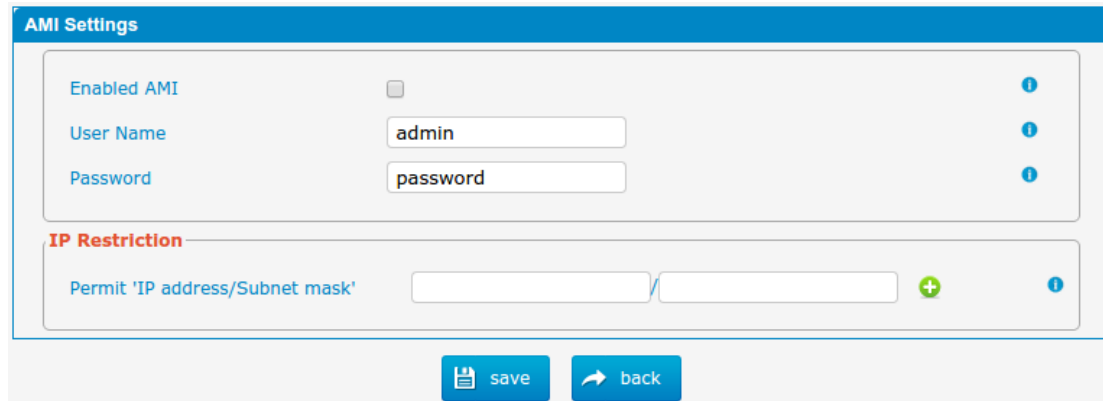
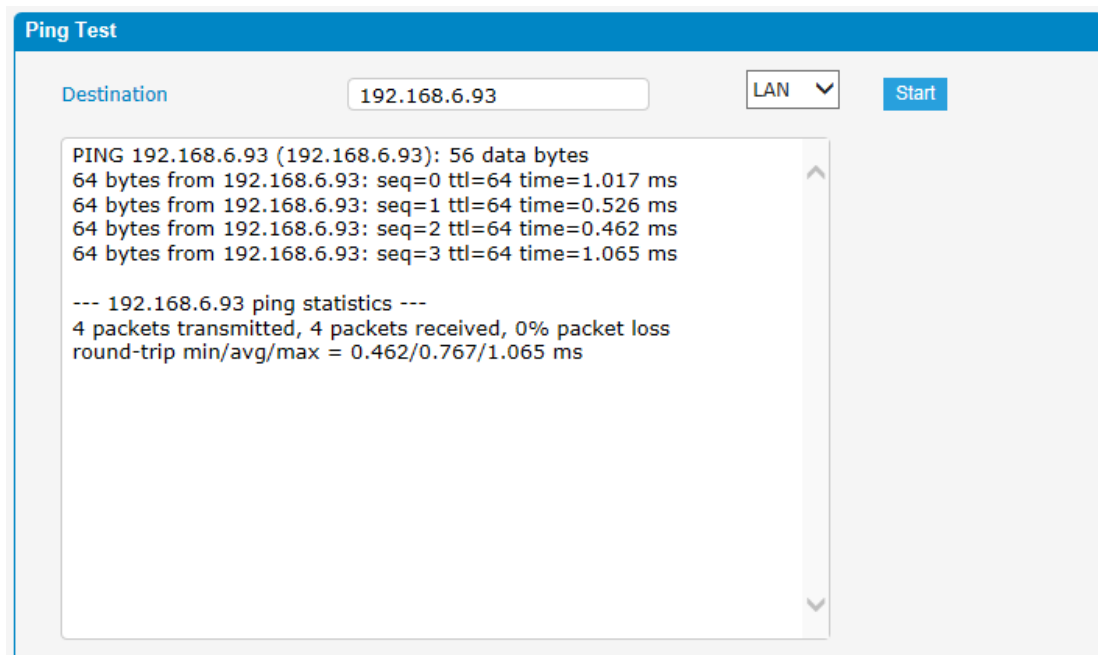


Table 3.13.2 Description of SMTP Parameters

Parameters	Description
Enable AMI	Enable AMI 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.
Username	AMI user name, default 'admin'
Password	AMI password, default 'password'
IP Restriction	Set IP address and subnet mask that can connect to AMI

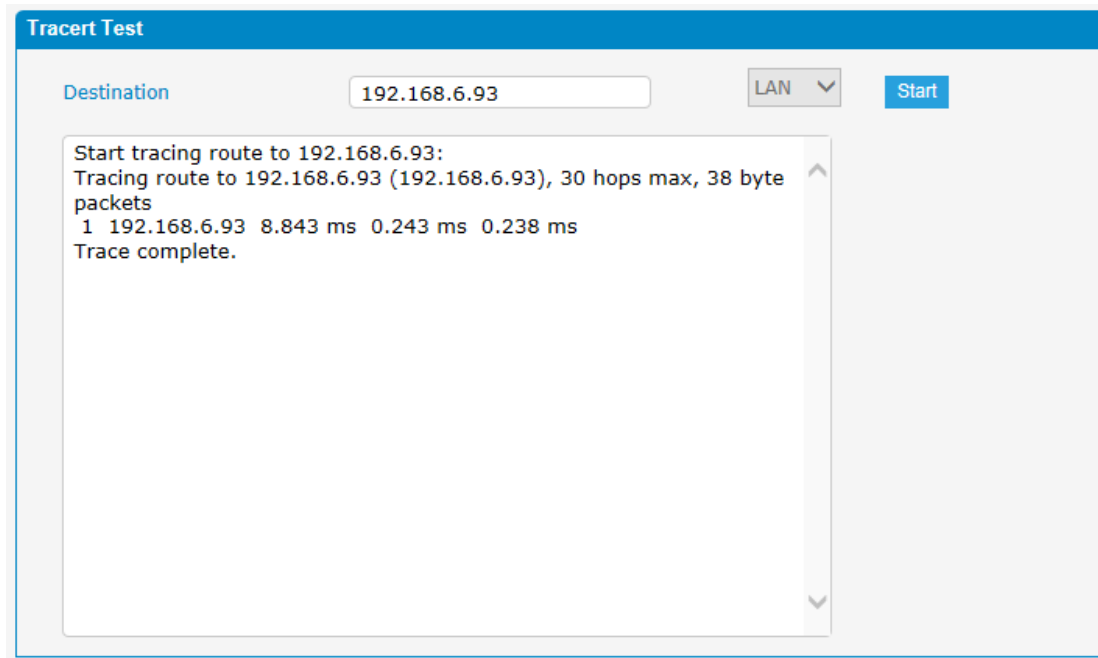
### 3.13.3 Ping

Figure 3.13.3 Ping



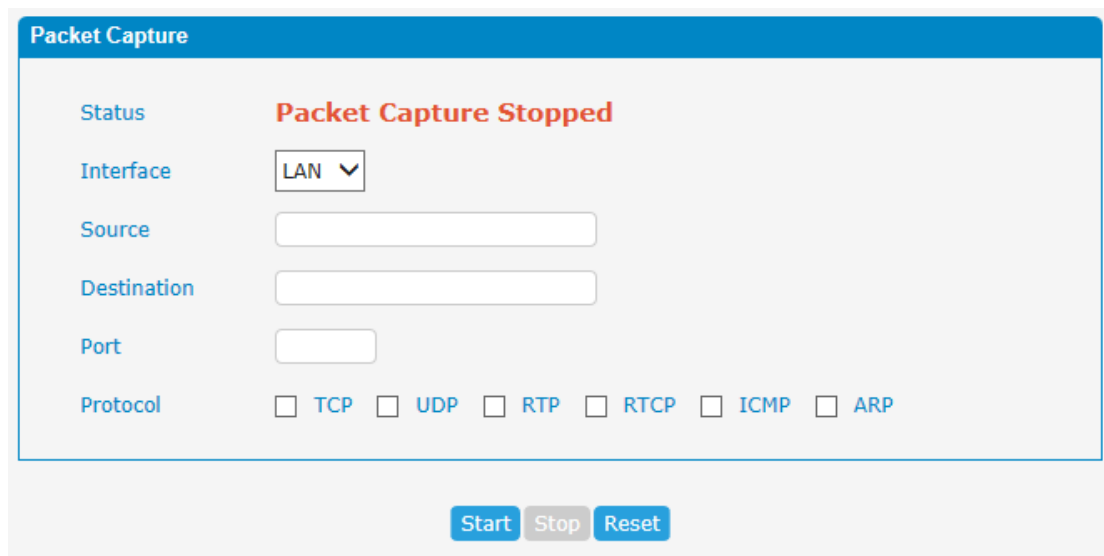
### 3.13.4 Tracer

Figure 3.13.4 Tracer



### 3.13.5 Packet Capture

Figure 3.13.5 Packet Capture



**Packet Capture**

Status: **Packet Capture Stopped**

Interface: LAN ▼

Source:

Destination:

Port:

Protocol: ☐ TCP ☐ UDP ☐ RTP ☐ RTCP ☐ ICMP ☐ ARP

Start Stop Reset

Table 3.13.5 Description of Packet Capture

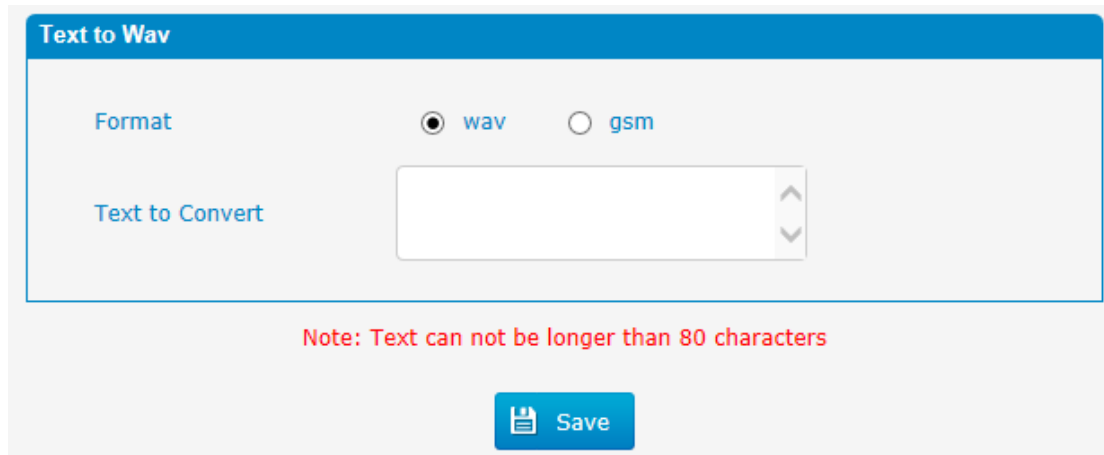
Parameters	Description
Status	Packet capture status
Interface	Choose network interface, LAN/WAN
Source	Capture source Address
Destination	Capture destination Address
Port	Capture port
Protocol	Capture protocol



### 3.13.6 Text to Wav

PBX can Transfer text to wav.

Figure 3.13.6 Text to wav

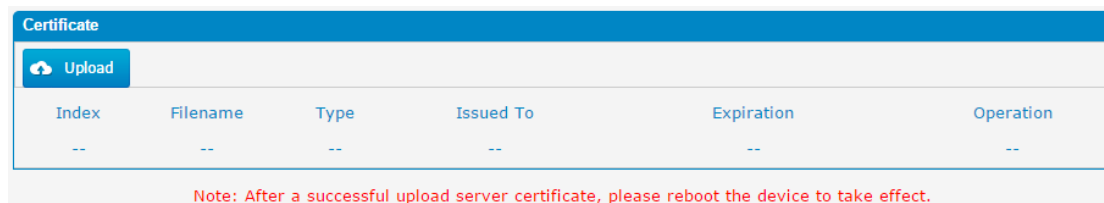


The interface for 'Text to Wav' conversion. It features a blue header bar with the title 'Text to Wav'. Below the header, there is a 'Format' section with two radio buttons: 'wav' (selected) and 'gsm'. Underneath, there is a text input field labeled 'Text to Convert'. A red note below the input field states: 'Note: Text can not be longer than 80 characters'. At the bottom, there is a blue 'Save' button with a floppy disk icon.

### 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



The 'Certificate' management interface. It has a blue header bar with the title 'Certificate'. Below the header, there is an 'Upload' button with a cloud icon. Underneath, there is a table with the following columns: Index, Filename, Type, Issued To, Expiration, and Operation. The table currently contains one row with dashes in all columns. A red note below the table states: 'Note: After a successful upload server certificate, please reboot the device to take effect.'

Index	Filename	Type	Issued To	Expiration	Operation
--	--	--	--	--	--

#### Trusted Certificate

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

#### Gateway Certificate

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