XonTel Technology Co.

شركة زونتل تكنولوجي
للتجارة العامة ذ.م.م

XonTel SMB PBX

User Manual

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Contents

1 Overview

1.1 Introduction

The XonTel SMB PBX delivers a multi-functional business office telephony system designed for small to medium enterprises. The series integrates functions such as IP phone, fax, and voice recording, and is compatible with multiple service platforms such as Cisco Call Manager, Avaya, Huawei and Asterisk, and terminals. The products are highly reliable, easy to install and deploy, and offer a brand-new experience in mobile offices and communications.

The XonTel SMB PBX delivers a full-featured IP Telephony solution. By supporting intelligent communication functions such as mobile phone extensions, instant multi-party conferences, call history, it not only facilitates seamless communication between enterprise employees and customers, but also provides a solid basis for enterprises to analyze core business data.

The XonTel SMB PBX comes with an asterisk-based system, the PBX software, offering not only full PBX functionality, but also a new feature that enables new stability for your unified communication systems.

It can seamlessly integrate VoIP trunks and your existing PSTN lines with 4 FXO ports and 1 FXS port analog connections and 2 Ethernet ports. They are developed with a wide selection of codes and signaling protocols, including G711 (alaw/ulaw), G722, OPUS, AMR-NB/WB, SILK, G723.1 G726, G729, GSM, ADPCM, ILBC, H263, H263P, H264, VP8.

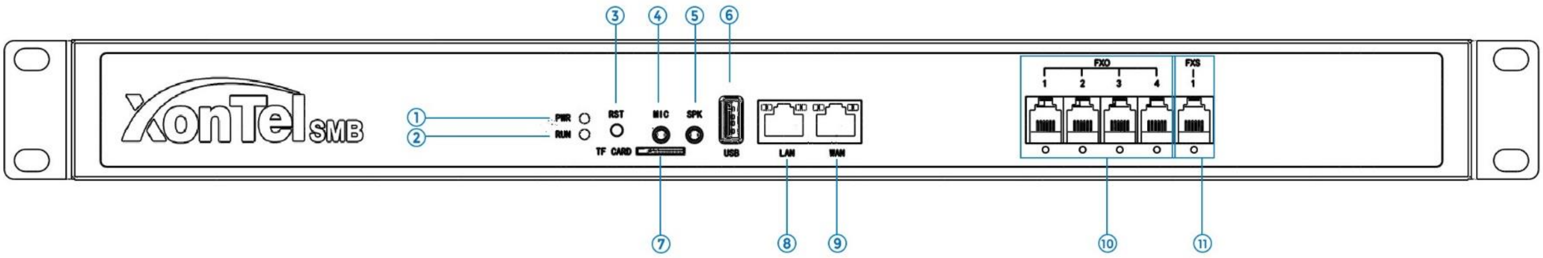
Taking full advantages of open source platform, the XonTel SMB appliances support industry standard SIP trunks, IAX2 trunks, analog PSTN trunks, and analog station trunks.

1.2 Specifications

Extensions	300 (expandable up to 800)
Max Concurrent Calls	300 concurrent calls
Voicemail	Unlimited
FXS Ports	1
FXO Ports	4
Protocol	SIP (RFC3261), IAX2
Transport Protocol	UDP, TCP, TLS
Encryption Protocol	SRTP, ZRTP
Codec	G711 a/μ law, G722, OPUS, AMR-NB/WB, SILK, G723.1 G726, G729, GSM, ADPCM, iLBC, H263, H263P, H264, VP8
DTMF	In-band, RFC2833, SIP INFO
LAN	1 × 10/100Mbps
WAN	1 × 10/100Mbps
IP Services	Static IP, DHCP, VPN, Firewall, PPPoE, Bridge
Recording	86,000 mins (.gsm); 9,500 mins (.wav)
Firewall	Yes
T.38 Fax	Yes
External Storage	SD Card, up to 128GB
USB	1
MIC Interface	1
SPK Interface	1
Power Supply	100-240V AC
Operating Temperature	0°C to 50 °C
Power Consumption	2.1-18.5 W
Storage Temperature	-20 °C ~ 65 °C
Dimension (Without handle)	440mm*251mm*44mm
Weight	216g

1.3 Hardware specifications

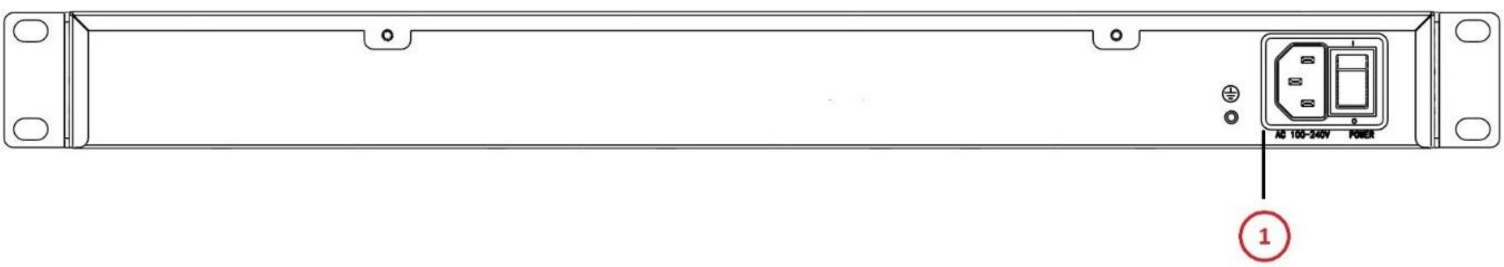
A. Front View



Number	Description
1	Power indicator
2	System indicator
3	Reset button
4	Microphone
5	Speaker
6	USB interface
7	TF card slot
8	LAN interface
9	WAN interface
10	FXO ports
11	FXS port

LED Indicator	Color	Status
Power Indicator	Always Green	Supply Power
System Indicator	Green and Flash(1s)	Work Normally
Module Indicators (FXO)	Red and Flash(1s)	Normal
	Always Red	Connected
	Red and Flash(0.1s)	Communicating
Module Indicators (FXS)	Always Green	Normal
	Green and Flash(0.1s)	Communicating
During initialization, all LED indicators will flash successively		

B. Back View



Number	Description
1	Power Supply 100-240V AC

1.4 Features

PBX

- Supported codecs: ADPCM, G.711(A-Law & u-Law), G.722, OPUS, AMR-NB/WB, SILK, G723.1 G726, G729, GSM, ADPCM, iLBC, H263, H263P, H264, VP8
- Support for analog interfaces such as FXS/FXO(PSTN/POTS)
- SIP and IAX2 support
- Incoming and outgoing routes with support for dial pattern matching
- Hardware detection interface
- Support for paging and intercom
- Web-based operator panel
- DISA (Direct Inward System Access)
- Call detail record(CDR) report
- Billing and consumption report
- Distributed Dial Plan with DUNDi
- Call recording, Call parking, call queues, Voicemail, Conference
- Echo canceller
- Callback support
- Flexible and configurable IVR
- Support for PIN sets
- Support for time conditions
- VoIP provider configuration
- Support for follow-me
- Support for ring group
- Support for video-phones
- Channel usage reports

Email

- Mail server with multi-domain support
- Based in Postfix for high email volume
- Remote SMTP Module
- Web based email client
- Support for quotas
- Antispam support
- Support for mail relay
- Email list management

FAX

- Fax to email application
- Fax visor with downloaded PDFs
- Can be integrated with Winprint Hylafax
- Fax send through Web Interface
- Fax to email customization
- Access control for fax client
- Backup/restore support via Web
- Automatic Backup Restore
- Server shutdown from the web
- DHCP server for dynamic IP
- Access control to the interface based on ACLs

General

- System resources monitor
- Backup Restore Validation
- Network configurator
- Heartbeat Module
- Configurable server date, time and timezone

1.5 Compatible Endpoints

Any SIP compatible IP Phone (Desktop Phones and Soft Phones for Windows, Linux, iOS and also Android platforms). Desktop phone examples include: Cisco, Avaya, Huawei, Grandstream, Yealink, Polycom, Snom, etc. Soft Phone examples include 3CX, Linphone, X-Lite, Zoiper etc.

- IAX compatible endpoints, for example Zoiper softphone.
- Analog Phones and Fax Machines
- Web Extensions (WebRTC)

1.6 Log in to the Web GUI

- **Step 1**

  Use a CAT6 cable to connect the device **WAN** port to the local network where the PC is connected, or connect the device **WAN** port directly to the PC.

- **Step 2**

  Dial "**89" to obtain device IP address by an analog telephone, the device defaults to a fixed **IP address**: **172.16.101.1/255.255.255.0**

- **Step 3**

  Make sure that the PC and the device are on the same network segment.

- **Step 4**

  Enter the device IP address in the browser address bar (**172.16.101.1**);

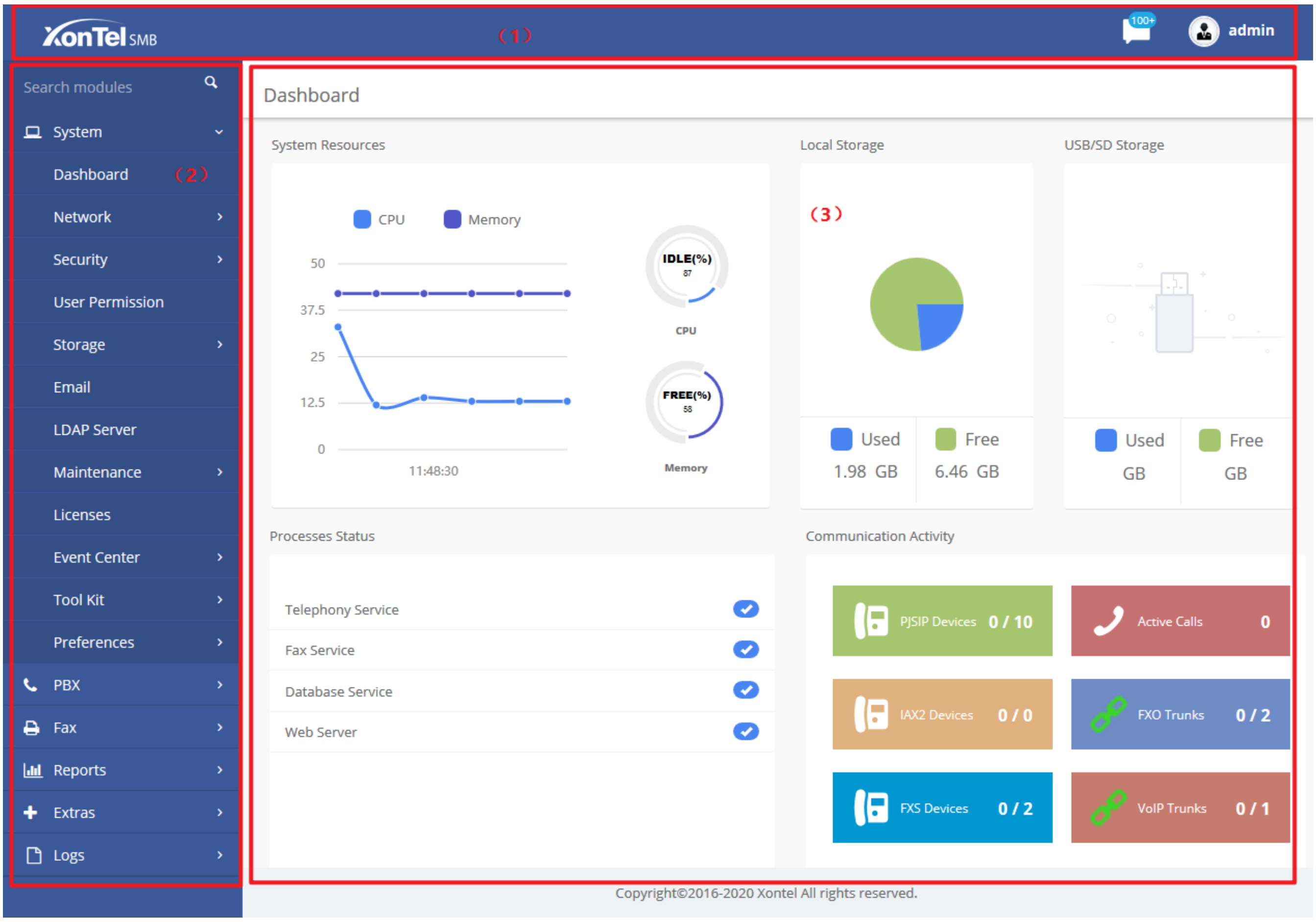
- **Step 5**

  You can enter the login interface for device configuration by selecting your role and entering a password on the login interface. The default administrator **username** is **admin** and **password** is **xontel**.

Login interface

1.7 Web GUI overview

The web management interface of the SMB PBX includes three areas: System button area, Menu bar and Configuration area.



Web GUI layout

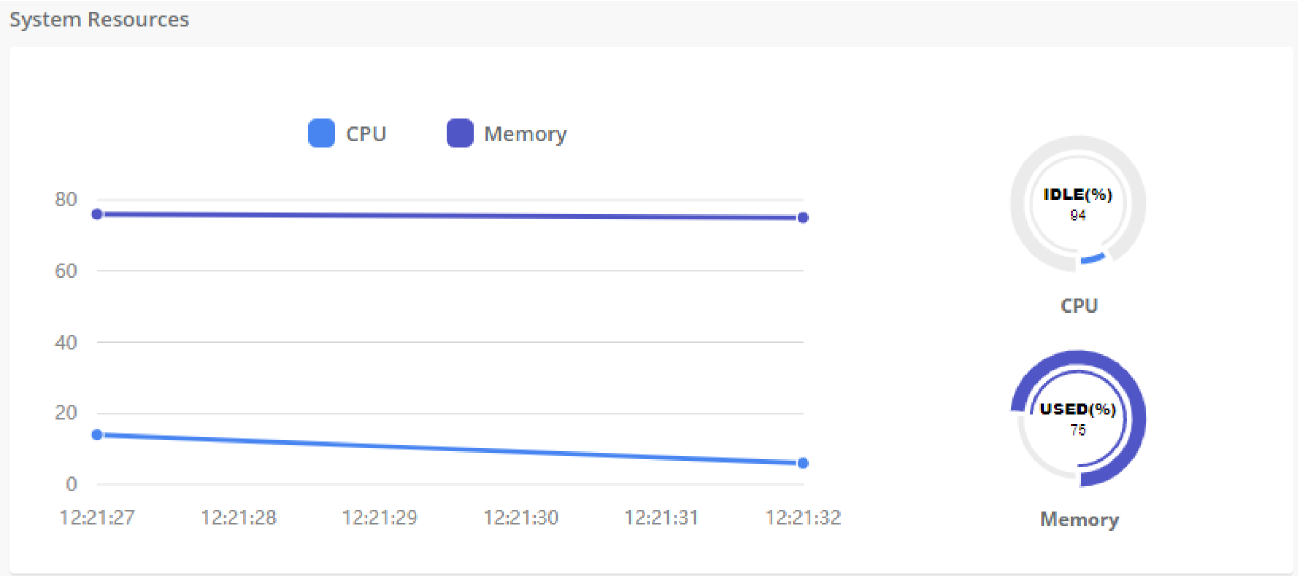
Item	Description
(1) System button area	Contains buttons such as Change Password, Reboot, Logout, Skins, Language, etc., and the event notification bar, displays the current login user.
(2) Menu bar	Displays submenus for your selection when the mouse pointer is moved onto a menu. The selection result is displayed in the configuration area.
(3) Configuration area	View or modify configurations.

2 System

2.1 Dashboard

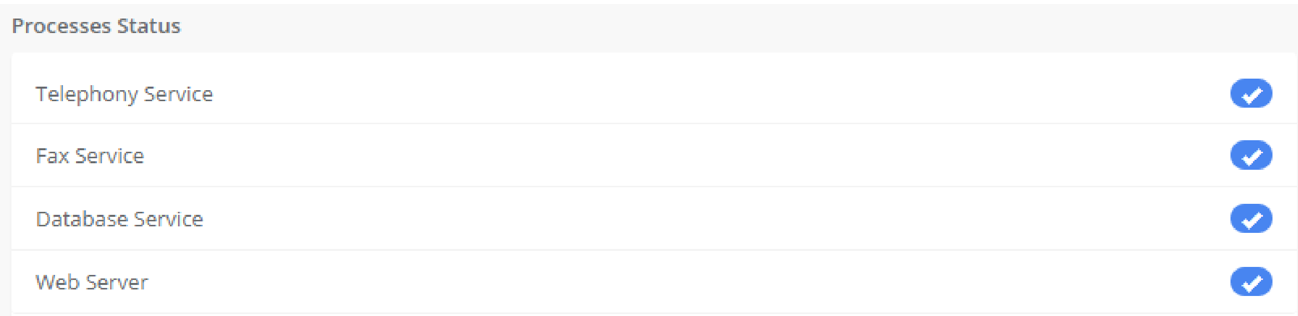
The option **Dashboard** of menu **System** in SMB PBX is a visualization tool that shows a general view of the system and gives a faster access to administrative actions in order to allow the user an easy administration of the server such as "System Resources", "Processes Status", "Hard Drives". Below a short description of each one.

System Resources: Here shows general information about the system where PBX is running. It allows to check out the history of CPU and Memory usage over the time.



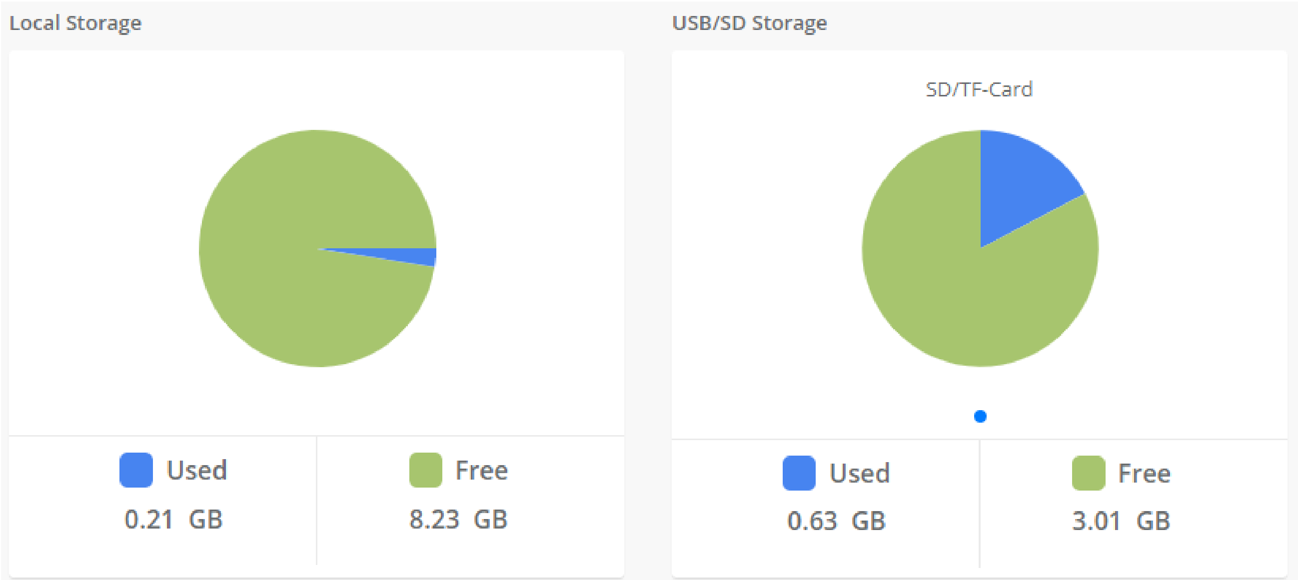
System Resource

Processes Status: It shows the enabled and disabled processes. Here you can start, stop and restart these processes.



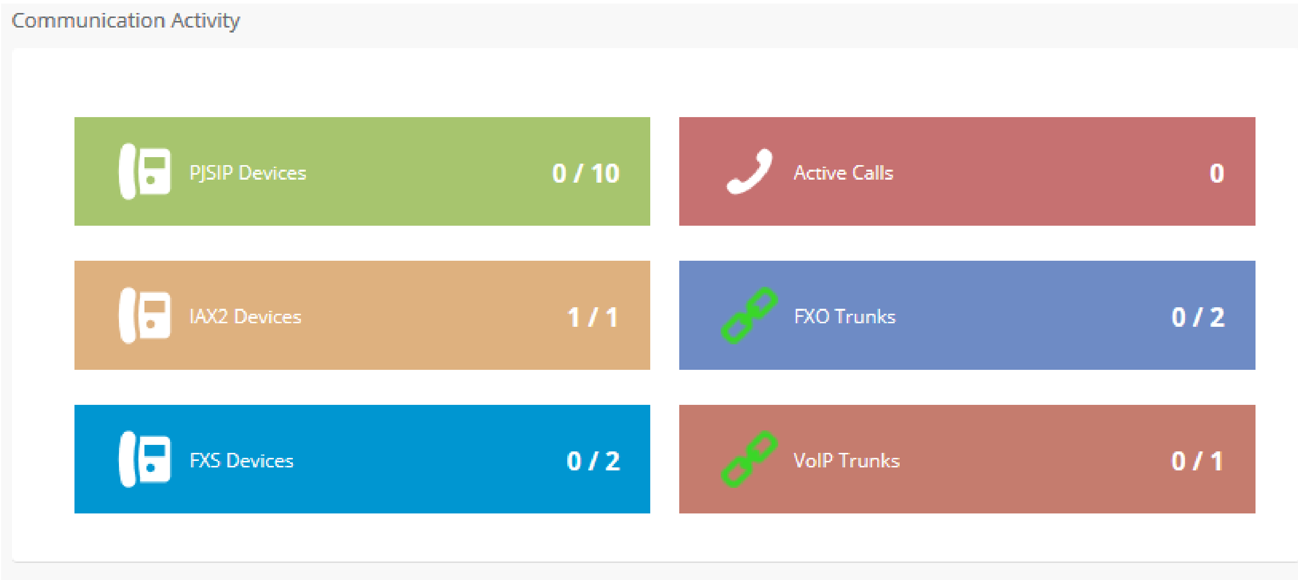
Processes Status

Hard Drives: Hard Drives shows the free and used space of the hard drives installed on your server.



Hard Drives

Communication Activity: This applet shows the number of extensions, trunks and calls currently on sip server.



Communication Activity

2.2 Network

2.2.1 Network Parameters

The option **Network Parameters** of the Menu **Network** in SMB PBX lets us view and configure the network parameters of the server.

Navigate to **System > Network > Network Parameters** to set network parameters according to the installed network environment.

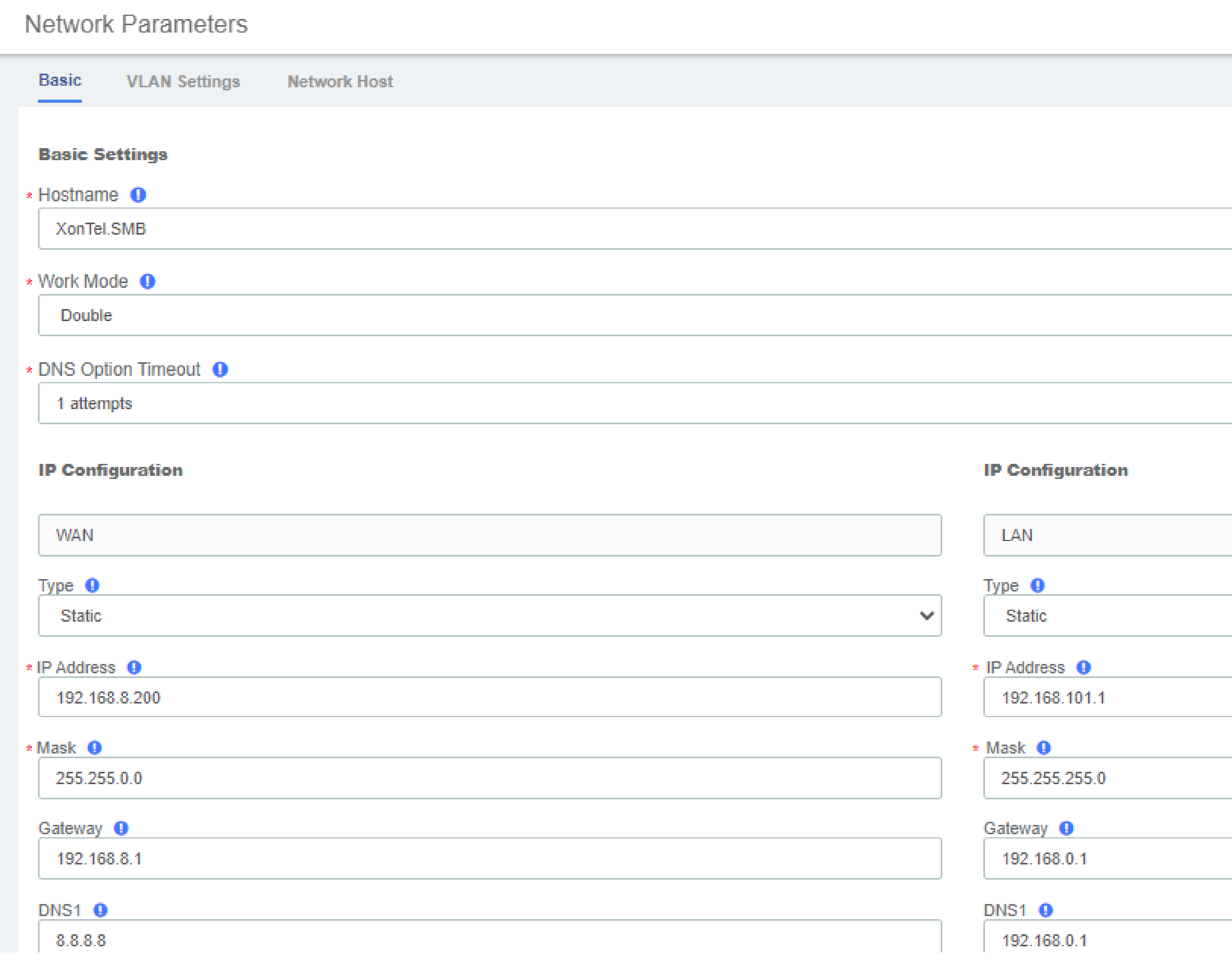
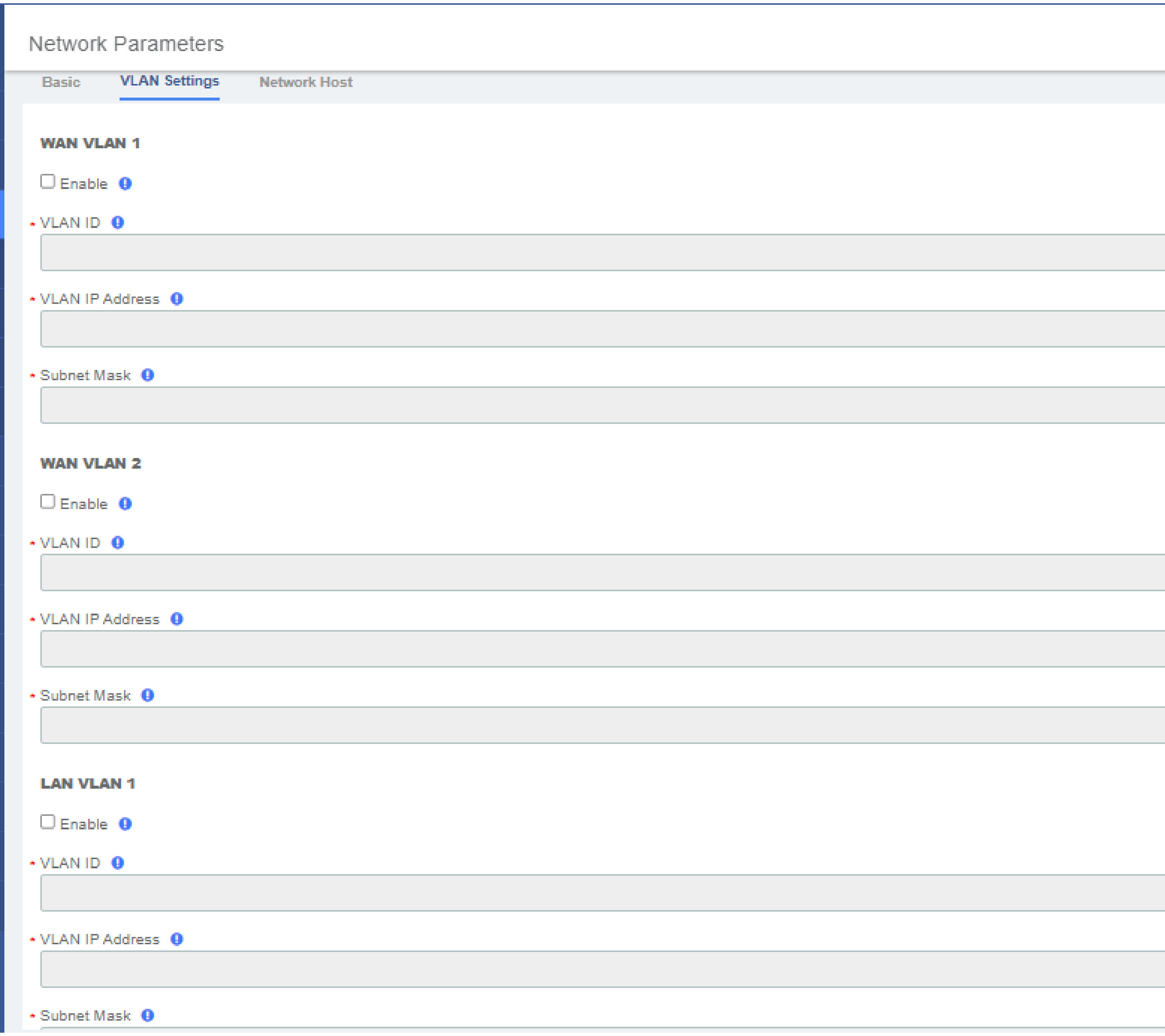


Figure 2-2-1 Network Parameters Interface

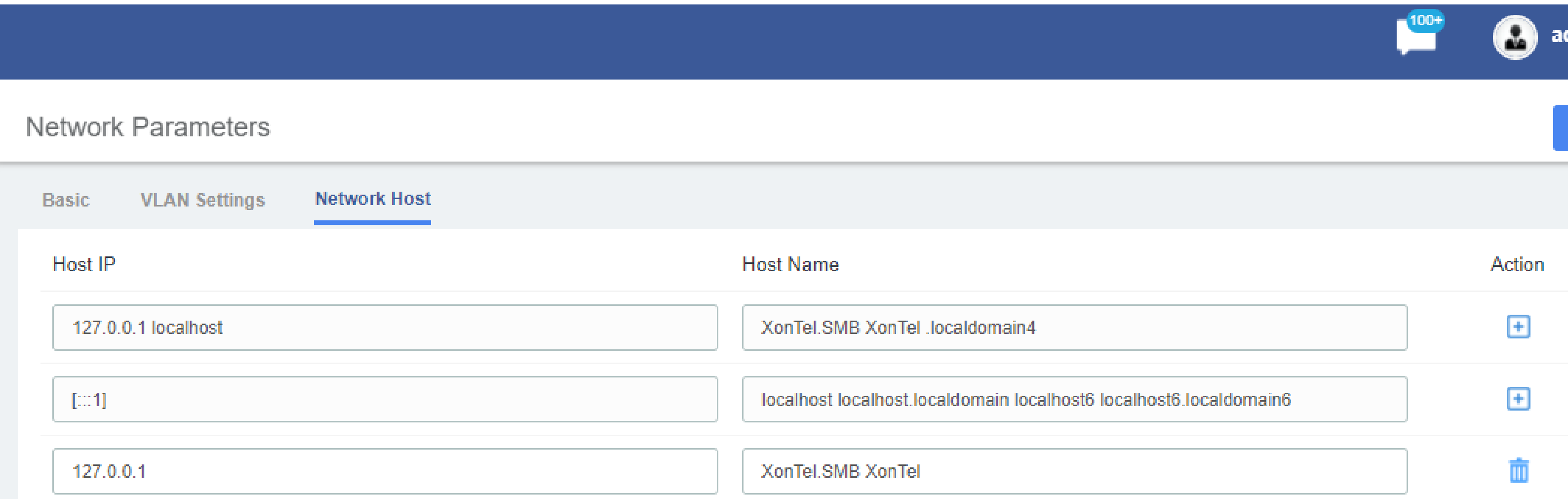
Item	Description
Basic Settings	
Hostname	used to uniquely identify the device in the specific network and it must add dots between words. For example: pbx.subdomain.com
Work Mode	Select the Ethernet mode. The options are: **Single**: only LAN port will be used for uplink connection. WAN port will be disabled. **Double**: Both ports can be used for uplink connection. Users will need to assign the default interface and configure. **Bridge**: LAN port interface will be used for uplink connection. WAN port interface will be used as bridge for PC connection.
DNS Option Timeout	Sets the amount of time the resolver will wait for a response from a remote name server before retrying the query via a different name server. The default timeout is 5. The value for this option is silently capped to 30.
IP Configuration	
Type	Set the network type, There three types: DHCP, Static and PPPOE. PPPOE only use in double mode.
IP Address	IP Address assigned to the Interface
Mask	The Network Mask assigned to the Interface
Gateway	IP Address of the Port of Connection (Default Gateway)
DNS1	IP Address of the Primary Domain Name Server (DNS)
DNS2	IP Address of the Secondary or Alternative Domain Name Server (DNS)
MAC	Physical Address of the network Interface
Status	Shows the physical status of the Interface, if it’s connected or not

Default Route	Mainly used in **Double** work mode to determine the default exit for network traffic
IP Address 2	The second IP assigned to the Interface
Netmask 2	The network mask for the second IP

Also you can configure VLAN in your PBX network interfaces (LAN and WAN) as shown below



You can also resolve DNS name to an IP address as shown below.



2.2.2 VPN Client

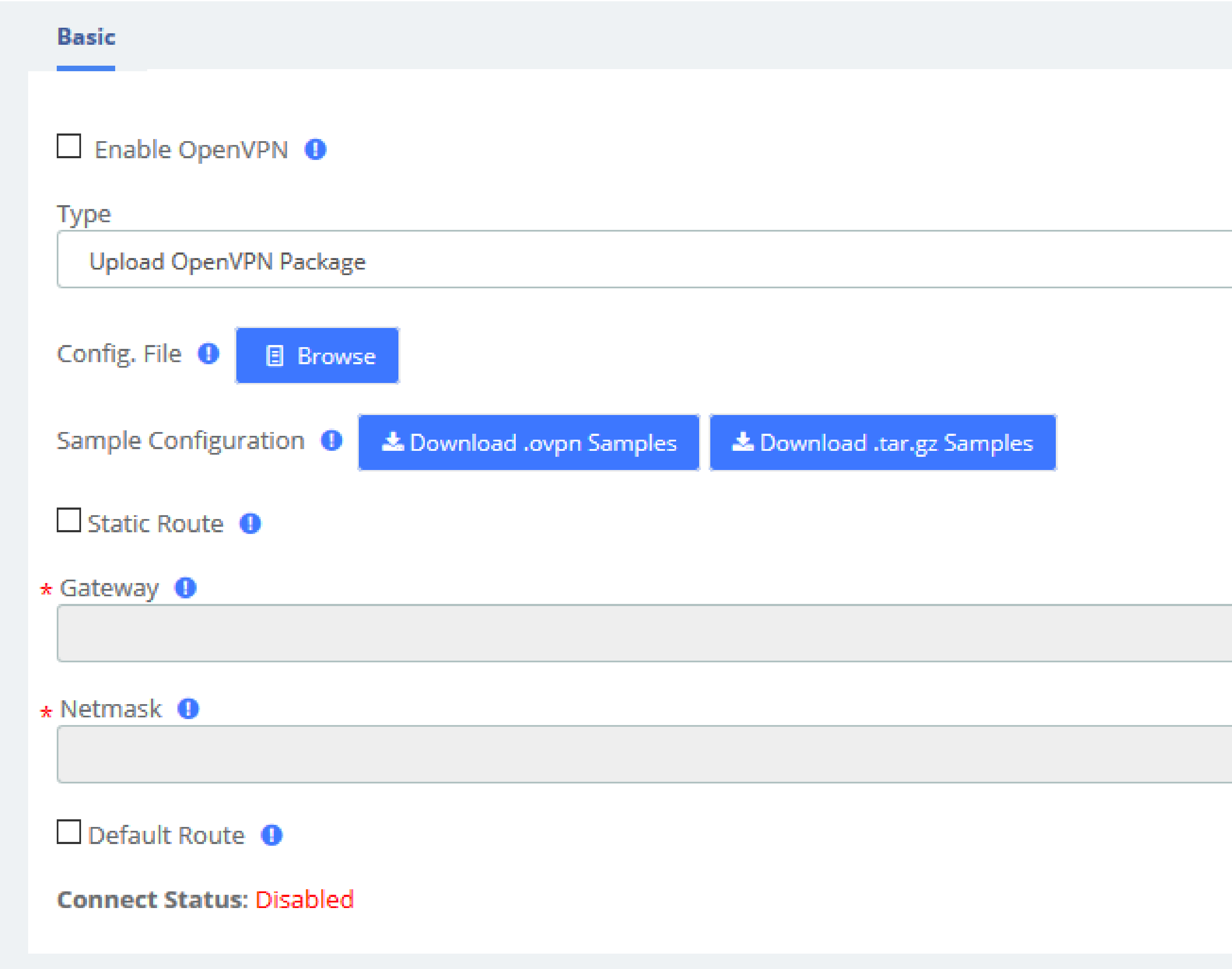
The **VPN Client** module of the menu **Network** lets us connect to the VPN Server.

Navigate to **System** > **Network** > **VPN Client**, chose client type and enter the Server IP Address, switching the Enable to on and save changes. Then the Server will assign this client an IP address.

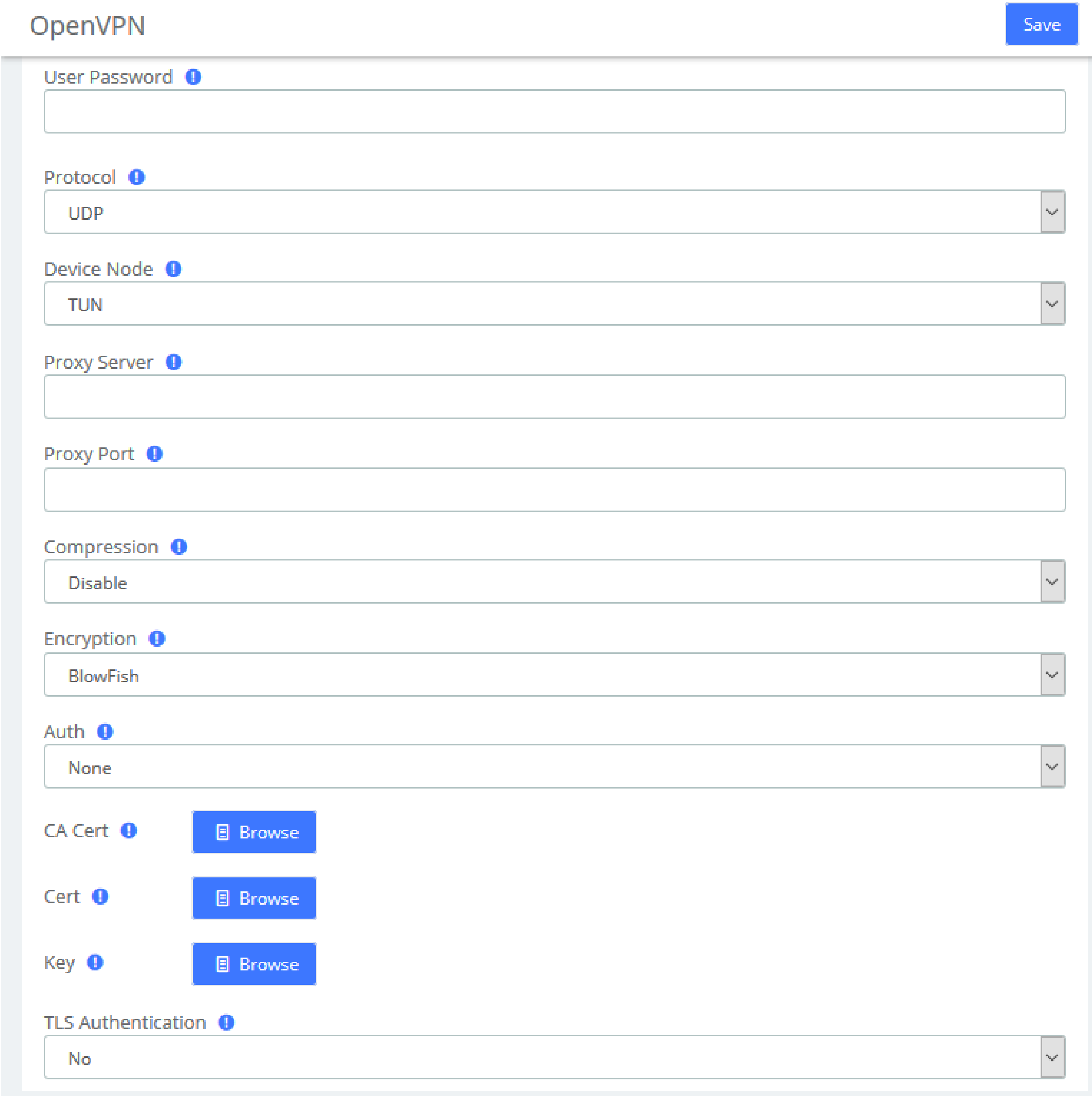
The SMB PBX offers four common VPN connections: OpenVPN, N2N, L2TP and SSTP, allowing users to establish virtual private networks, encrypt communications, and enable remote access.

OpenVPN

You can choose to directly upload the configuration package file (.ovpn format) for the connection.



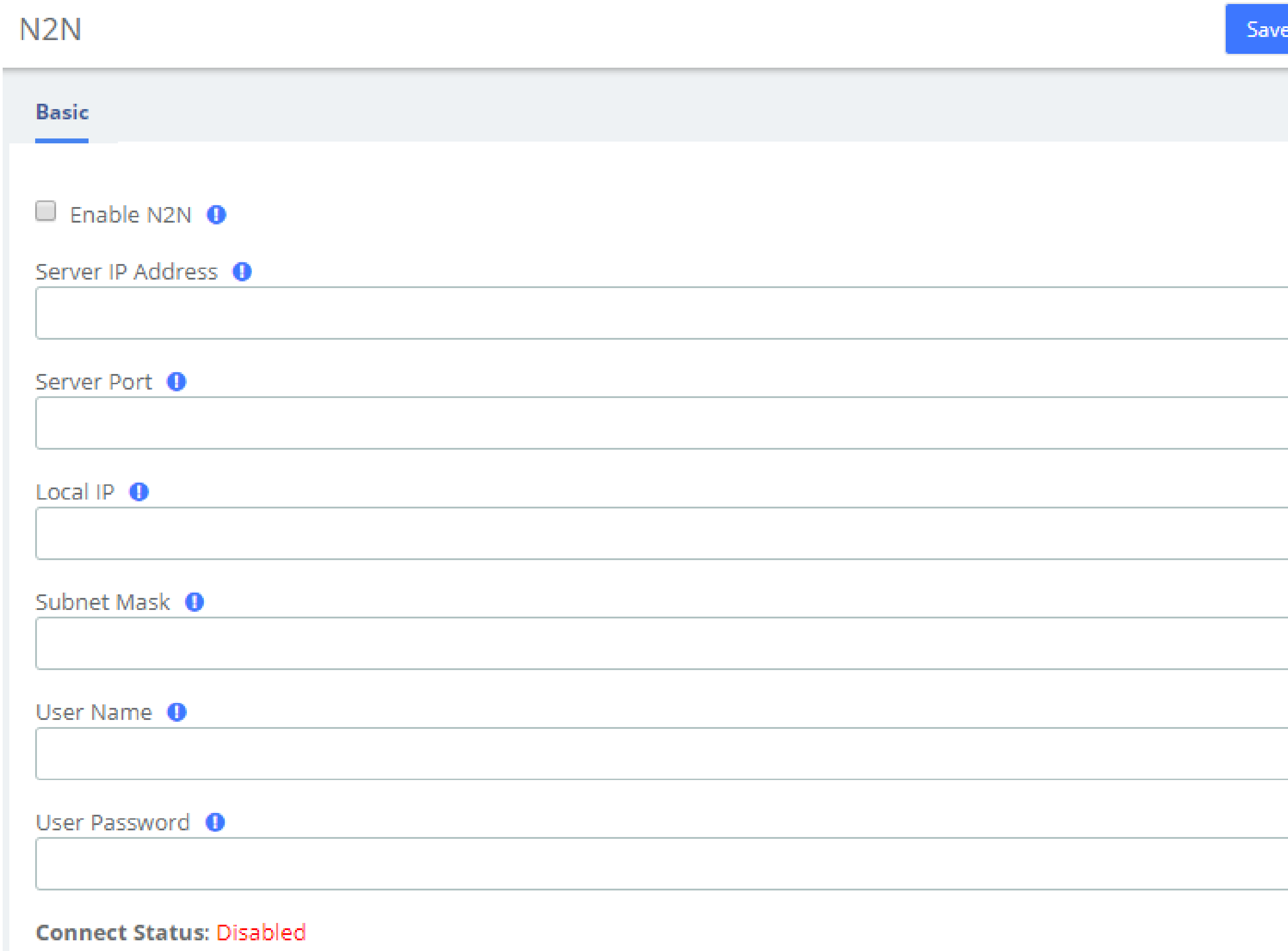
It is also possible to manually configure the server information and upload files such as certificates to connect.



OpenVPN/Manual Configuration

N2N

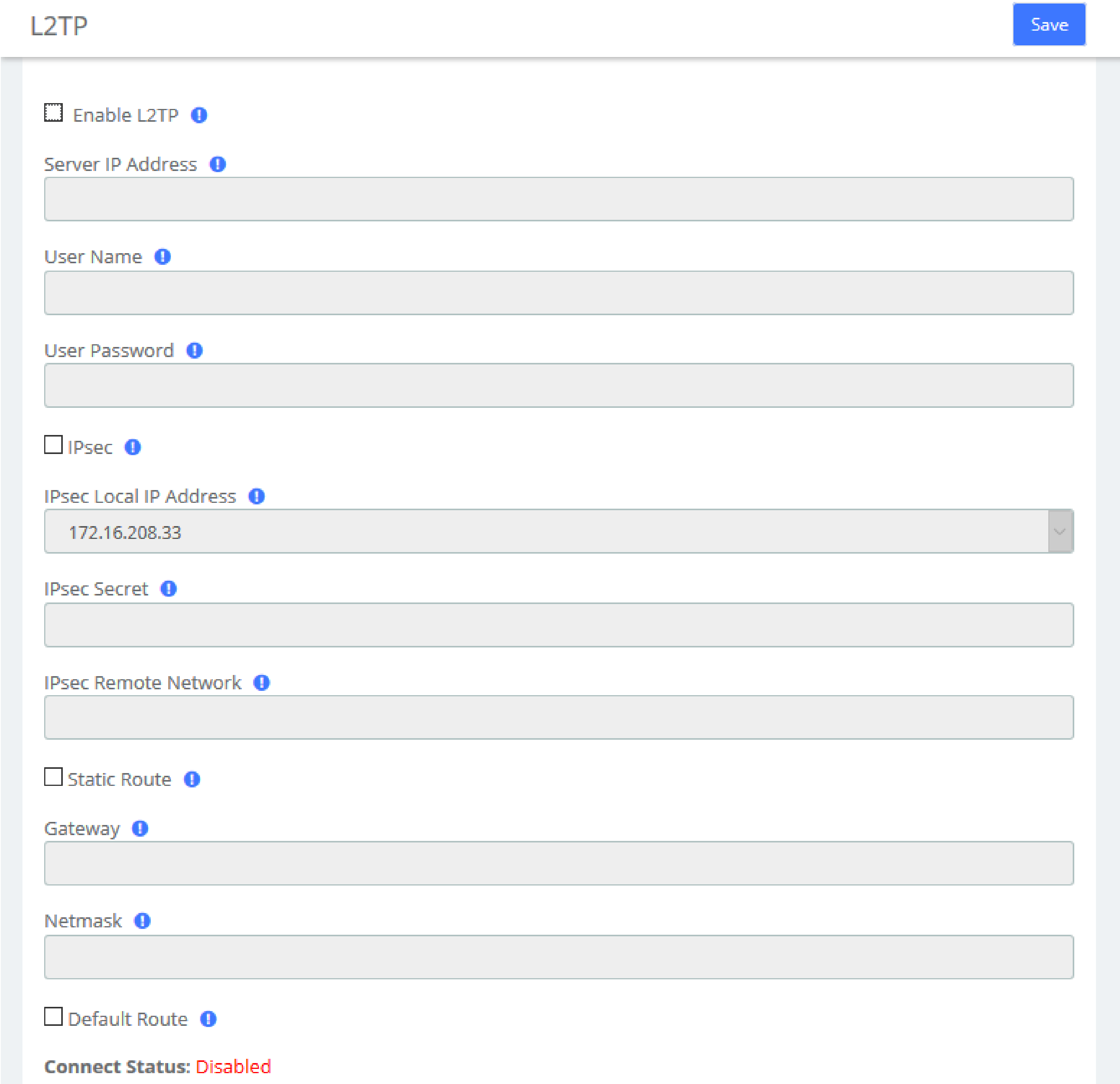
Enter the server and user information and click the **Save** button to connect.



N2N Interface

L2TP

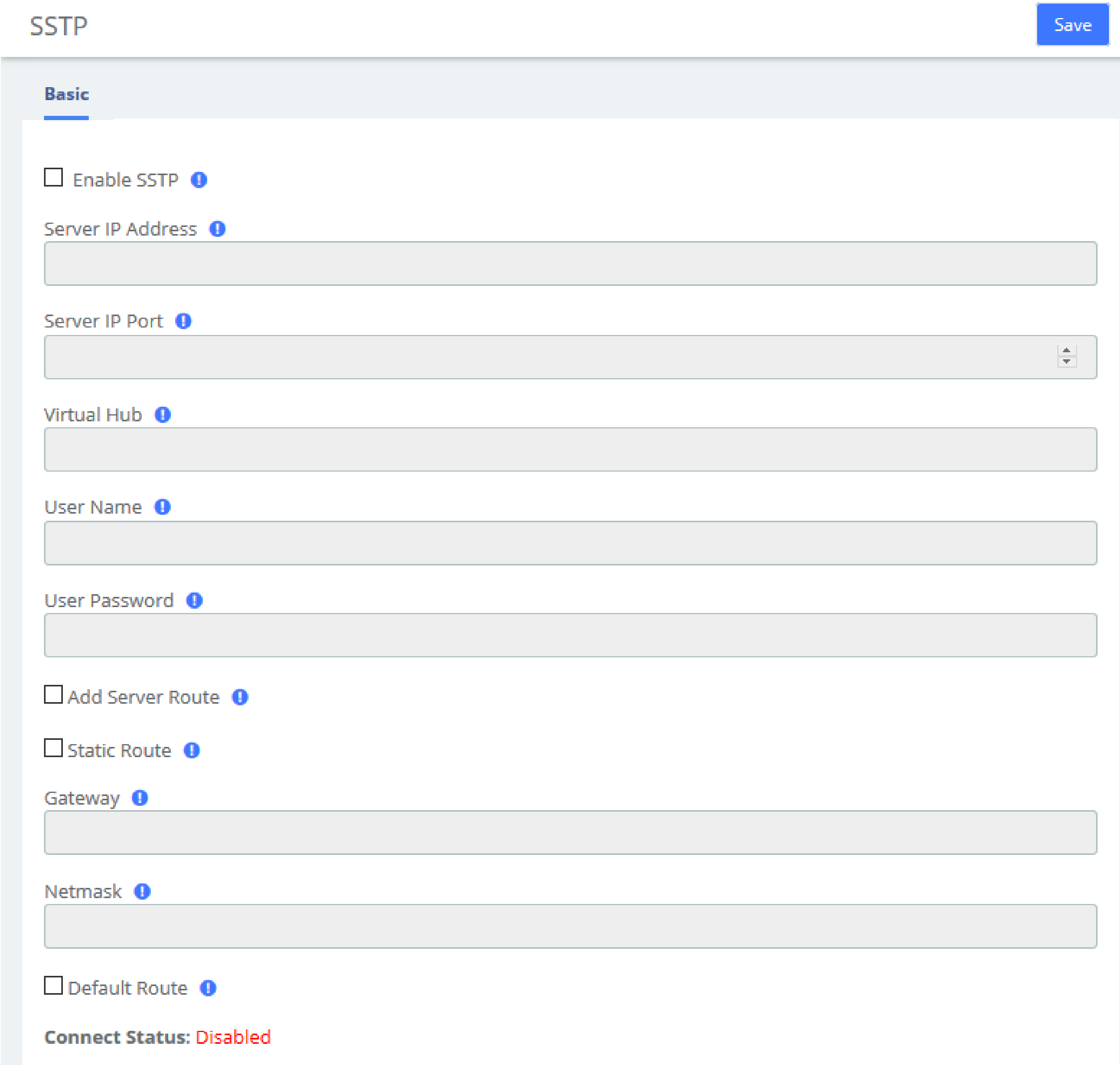
Enter the corresponding information and click the **Save** button to connect.



L2TP Interface

SSTP

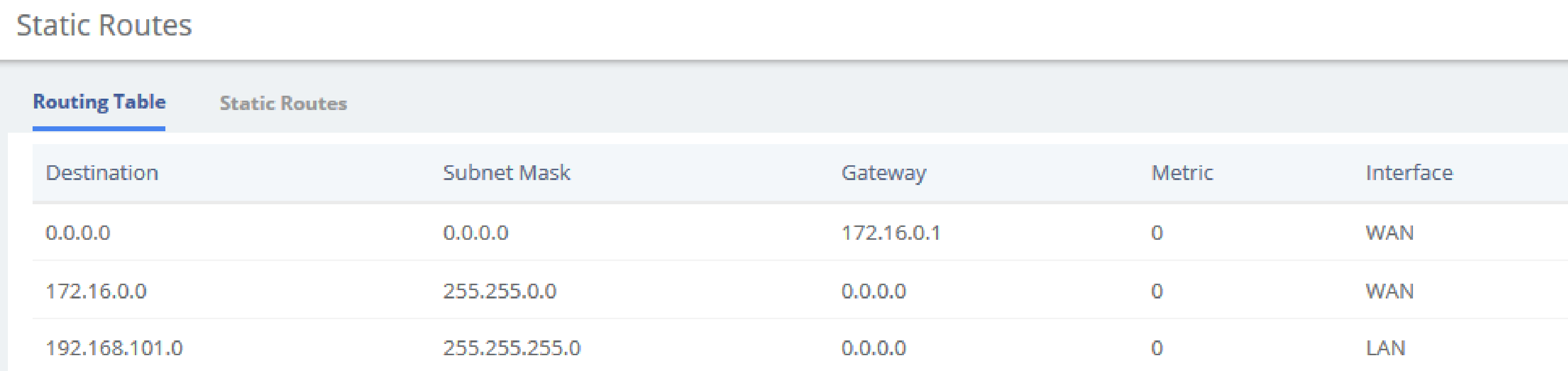
Enter the corresponding information and click the **Save** button to connect.



SSTP Interface

2.2.3 Static Routes

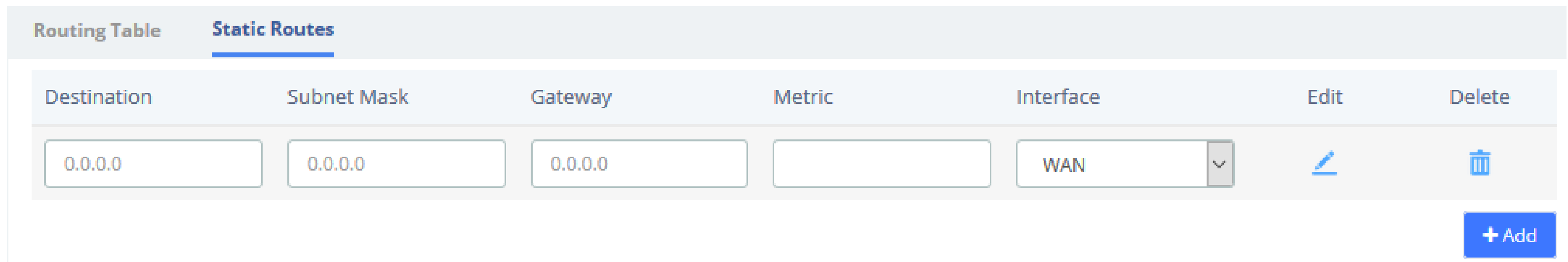
The Static Routes module of the menu “Network” lets users view and add the static routes.



Static Routes

Routing Table | Static Routes

Destination	Subnet Mask	Gateway	Metric	Interface
0.0.0.0	0.0.0.0	172.16.0.1	0	WAN
172.16.0.0	255.255.0.0	0.0.0.0	0	WAN
192.168.101.0	255.255.255.0	0.0.0.0	0	LAN



Routing Table | Static Routes

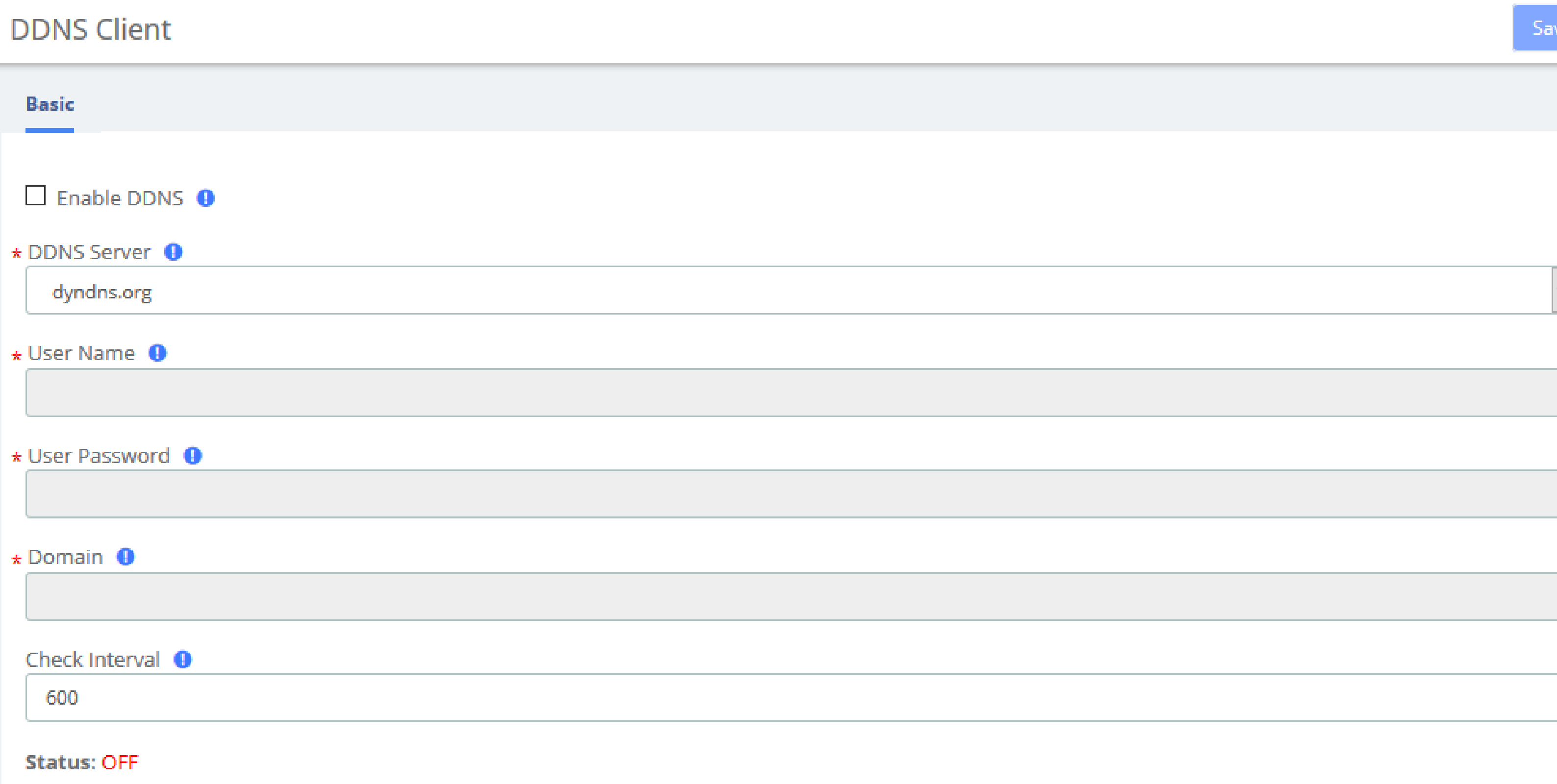
Destination	Subnet Mask	Gateway	Metric	Interface	Edit	Delete
0.0.0.0	0.0.0.0	0.0.0.0		WAN		

+Add

Item	Description
Destination	Identified the destination of IP packet.
Subnet Mask	Identified the segment where the destination host or router locates with destination.
Gateway	Also named Next Hop Router, defined the next hop server the packets send to.
Metric	Used to make routing decisions, contains any number of values that help the router determine the best route among multiple routes to a destination.
Interface	The Ethernet LAN/WAN interface, defined the interface used to send packet for the specific destination.

2.2.4 DDNS Client

Select DDNS server, enter user name, password and other information, then click **Save** to make DDNS take effect.



DDNS Client Interface

2.2.5 DHCP

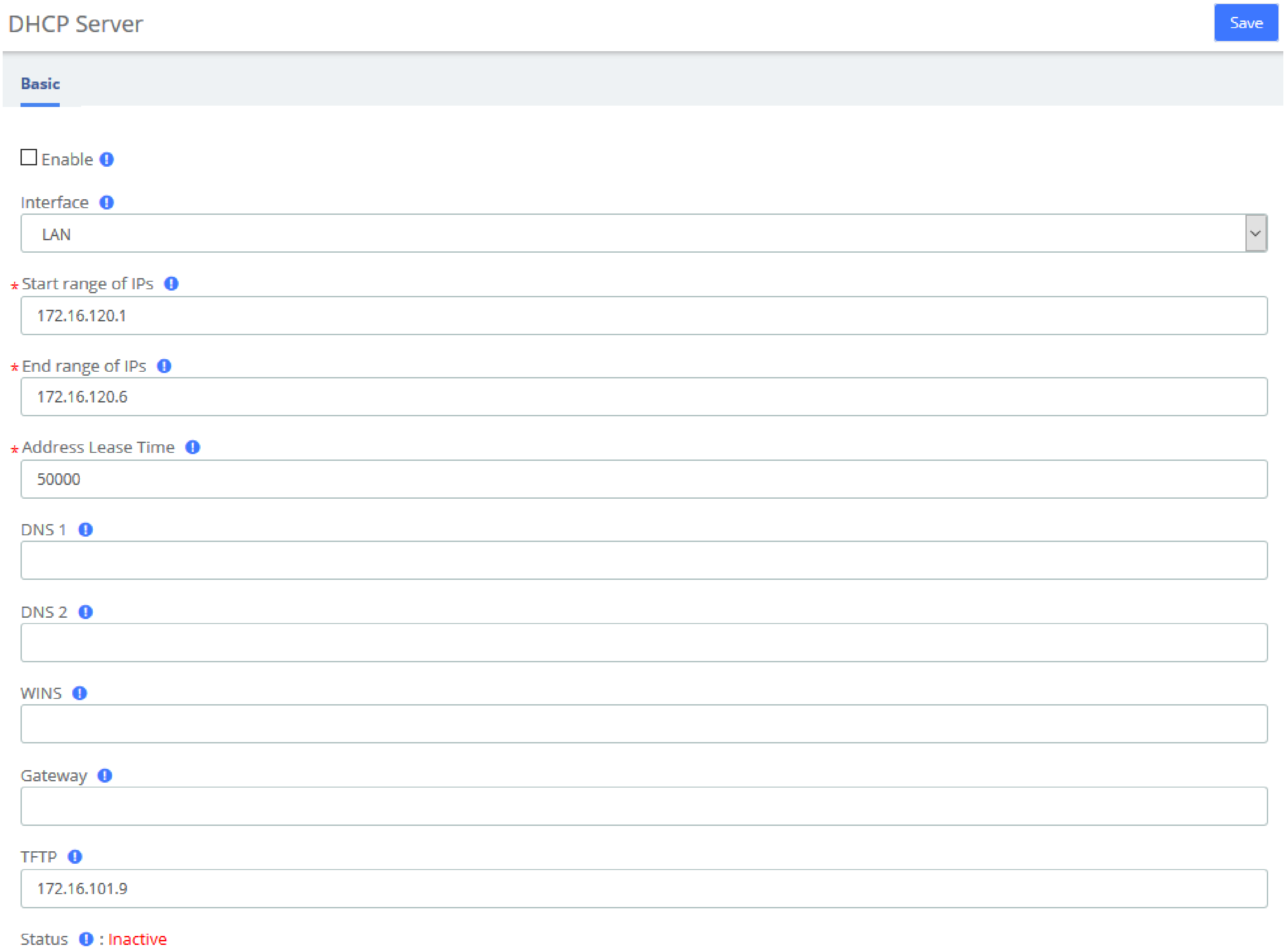
DHCP Server

DHCP (Dynamic Host Configuration Protocol) is a standardized network protocol used on Internet Protocol (IP) networks for dynamically distributing network configuration parameters, such as IP addresses for interfaces and services.

With DHCP, computers/IP phones request IP addresses and networking parameters automatically from WAN/LAN port which saves administrators a lot of time when compared with having to configure these settings manually.

The option "DHCP Server" allows configuring PBX’s DHCP service so it can assign IP addresses in the network.

Navigate to **System** > **Network** > **DHCP** > **DHCP Server**:



Item	**Description**
Enable	It indicates if the DHCP service is enabled or disabled.
Start range of IPs	This will be the beginning of the IP range that the server will provide.
End range of IPs	This will be the ending of the IP range that the server will provide.
Address Lease Time	Amount of time the IP address will be assigned to devices in the network.
DNS 1	This address is the Primary DNS that the server will provide.
DNS 2	This address is the Secondary DNS that the server will provide.

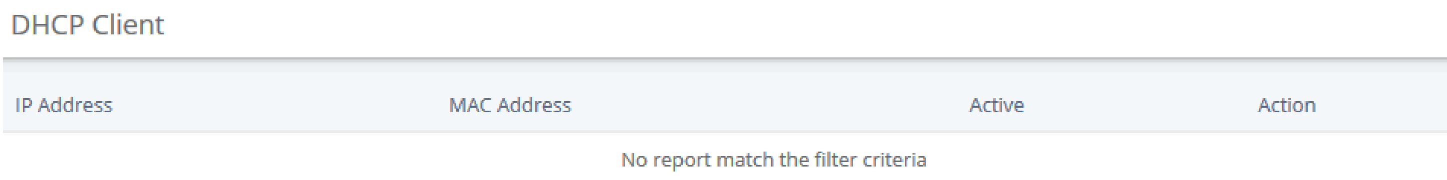
WINS	It is the IP of the WINS Server that will be given to Windows machines.
Gateway	This is the address the server will provide as Gateway.

To save changes just click on the button Save . The service can be started by checking on ☐ Enable .

DHCP Client

This module shows a list of DHCP clients and leased IP addresses.

Navigate to **System** > **Network** > **DHCP** > **DHCP Client** List and you will see a list of all devices receiving their IP address from the system.

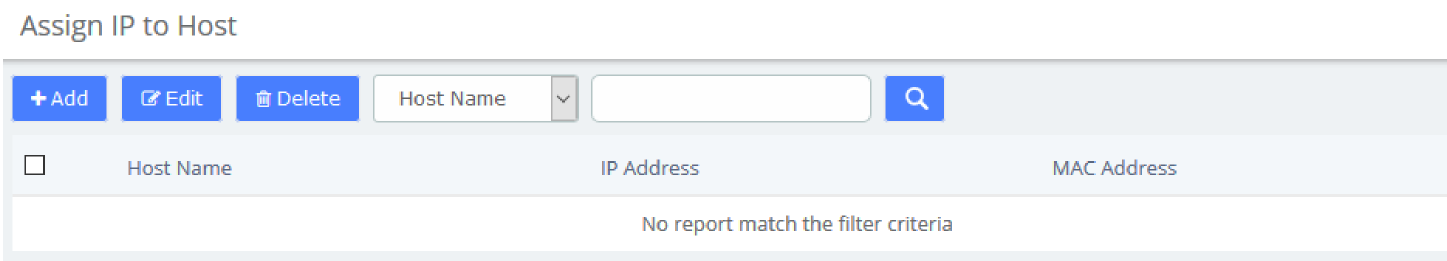


To see the leased time of each address, click on "View Details".

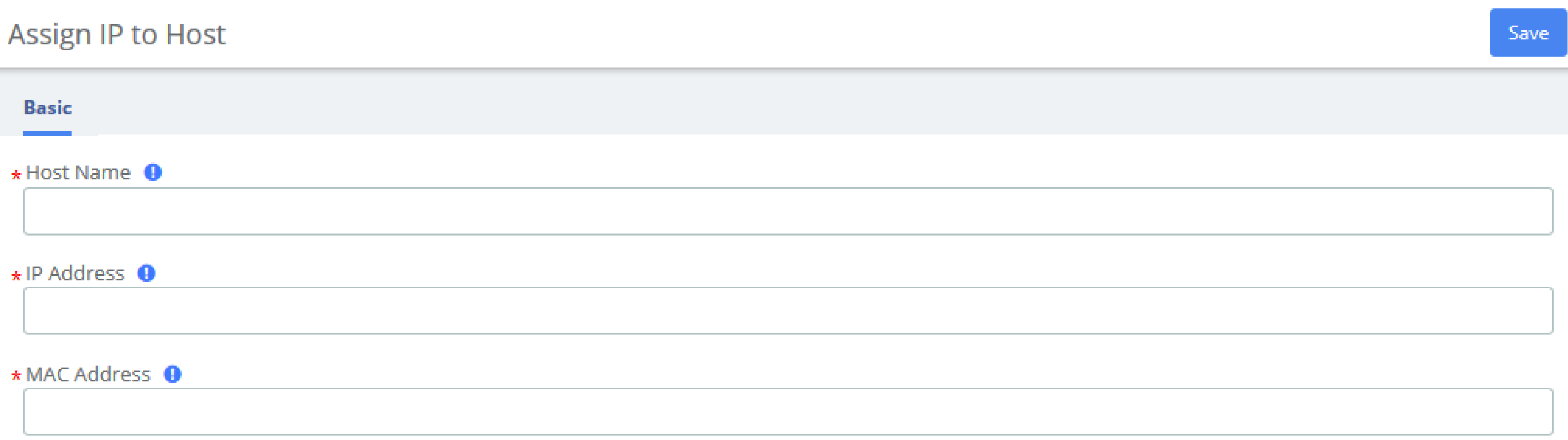
Assign IP to Host

With this option you can assign an IP address to a specific device through MAC address. When the device requests an IP address, the DHCP server will provide it according to the MAC address. All the associations created by the user are shown in a list.

Navigate to **System** > **Network** > **DHCP** > **Assign IP to Host**.



To create a new association, click +Add button. Fill out the required information and click on Save button.



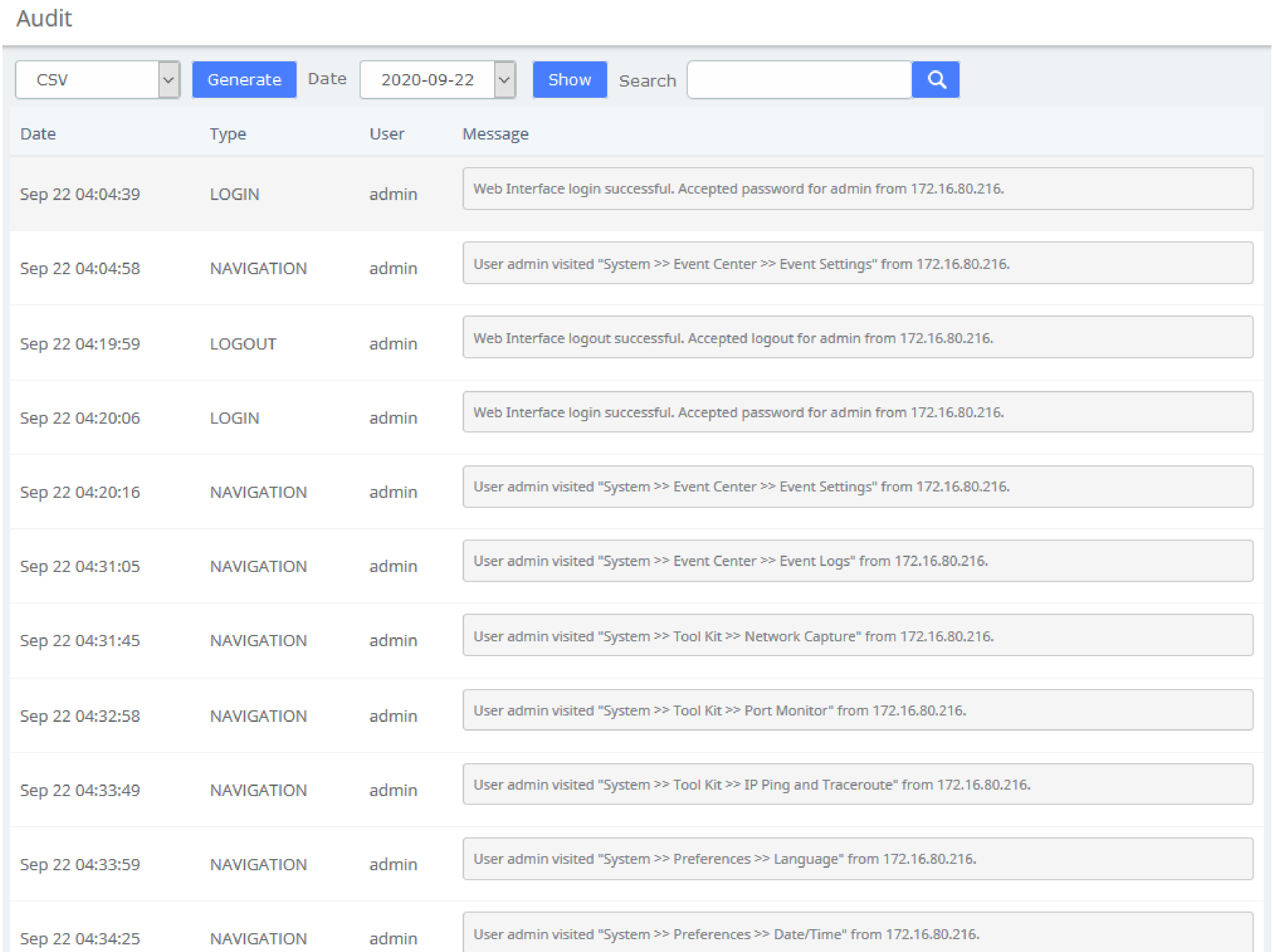
Add Assign IP Address

Item	Description
Host Name	Name that you want to assign to the device
IP Address	IP Address you want to use for the device
MAC Address	MAC number of the device

2.3 Security

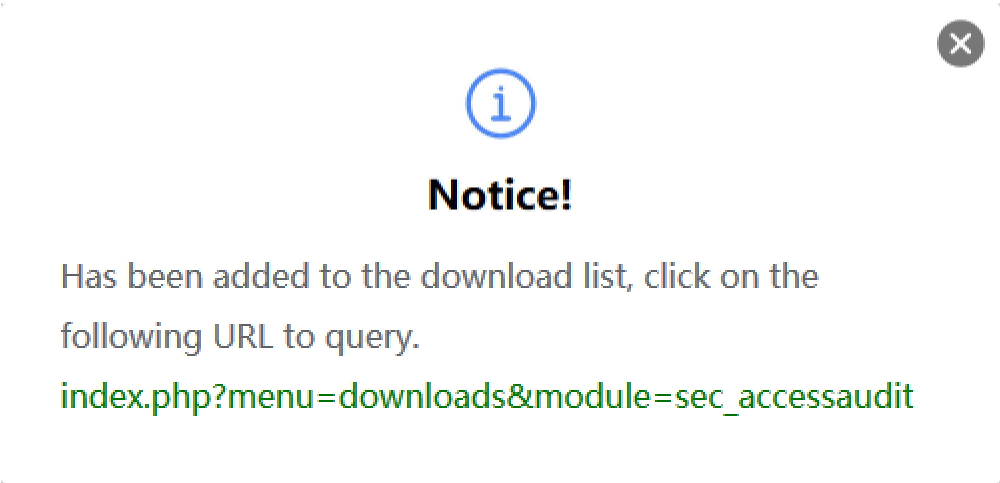
2.3.1 Audit

The module **Audit** of the menu **Security** shows a list of all the users that have logged in the system with the date, the username, the source IP address and other details. The results can be filtered by date and string. The coincidences with the string will be highlighted in the results.

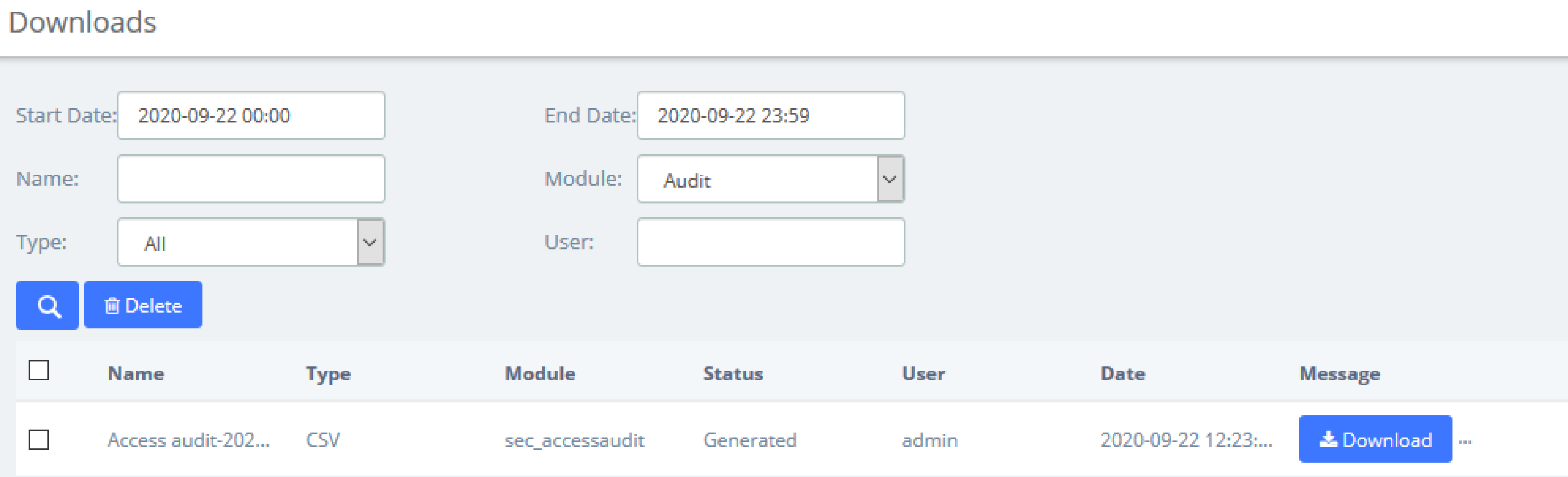


Date	Type	User	Message
Sep 22 04:04:39	LOGIN	admin	Web Interface login successful. Accepted password for admin from 172.16.80.216.
Sep 22 04:04:58	NAVIGATION	admin	User admin visited "System >> Event Center >> Event Settings" from 172.16.80.216.
Sep 22 04:19:59	LOGOUT	admin	Web Interface logout successful. Accepted logout for admin from 172.16.80.216.
Sep 22 04:20:06	LOGIN	admin	Web Interface login successful. Accepted password for admin from 172.16.80.216.
Sep 22 04:20:16	NAVIGATION	admin	User admin visited "System >> Event Center >> Event Settings" from 172.16.80.216.
Sep 22 04:31:05	NAVIGATION	admin	User admin visited "System >> Event Center >> Event Logs" from 172.16.80.216.
Sep 22 04:31:45	NAVIGATION	admin	User admin visited "System >> Tool Kit >> Network Capture" from 172.16.80.216.
Sep 22 04:32:58	NAVIGATION	admin	User admin visited "System >> Tool Kit >> Port Monitor" from 172.16.80.216.
Sep 22 04:33:49	NAVIGATION	admin	User admin visited "System >> Tool Kit >> IP Ping and Traceroute" from 172.16.80.216.
Sep 22 04:33:59	NAVIGATION	admin	User admin visited "System >> Preferences >> Language" from 172.16.80.216.
Sep 22 04:34:25	NAVIGATION	admin	User admin visited "System >> Preferences >> Date/Time" from 172.16.80.216.

The results of the search can be downloaded in different formats such as PDF, XML and CSV by clicking on the Generate button.



By clicking on the URL above, you can jump to the **Reports > Downloads** page. Click the **Download** button to download the generated file.



Download Audit Content

2.3.2 Weak Keys

The module **Weak Keys** of the menu **Security** lets us identify the keys that are not enough strength for the extensions created in SMB PBX (SIP and IAX2). This module shows all the extensions but you can filter the results by entering a specific extension number or part of it.

Weak Keys

CSV | Generate | Search | Extension

Extension	Description	Registration Password Status	User Password Status
107	107	Strong	Strong
106	106	Strong	Strong
102	102	Strong	Strong
103	103	Strong	Strong
104	104	Strong	Strong
105	105	Strong	Strong
101	101	Strong	Strong
108	108	Strong	Strong
109	109	Strong	Strong
110	110	Strong	Strong
112	112	Strong	Strong
1301	Virtual_Fax_1301		-

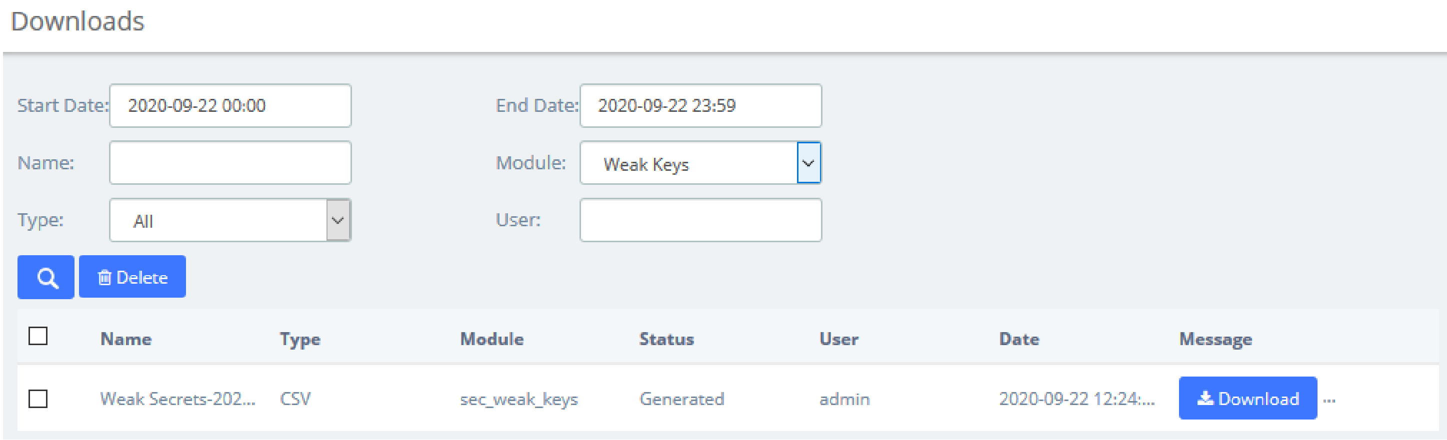
Weak keys interface

You can generate the results in different formats such as PDF, XML and CSV by clicking on the Generate button.

Notice!

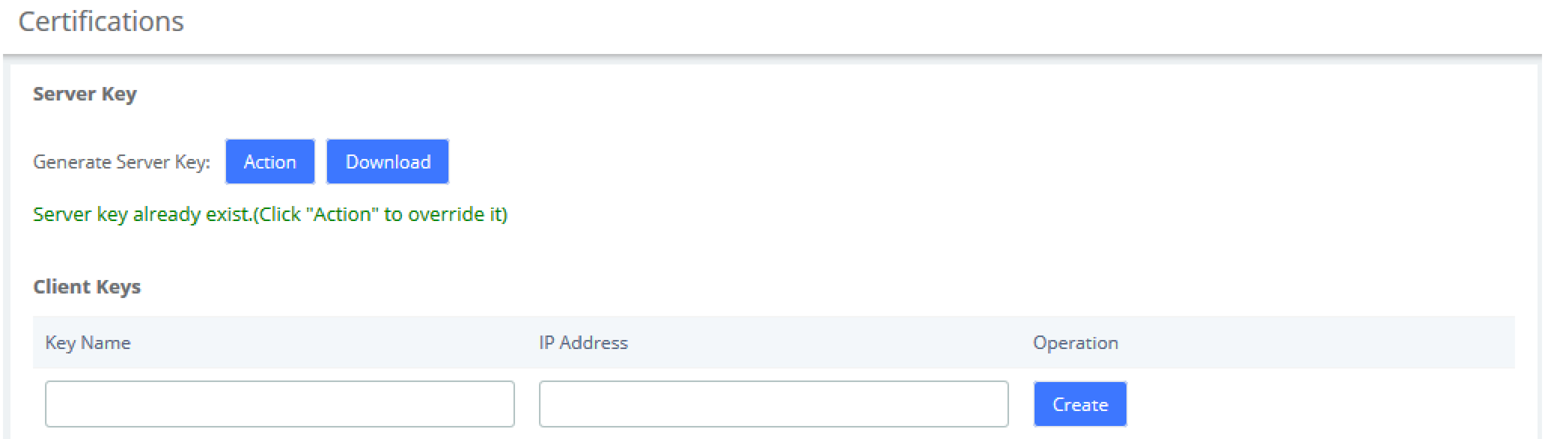
Has been added to the download list, click on the following URL to query.
index.php?menu=downloads&module=sec_weak_keys

By clicking on the URL above, you can jump to the **Reports** > **Downloads** page. Click the **Download** button to download the corresponding file.



2.3.3 Certifications

The **Certifications** module of the **Security** menu greatly enhances the security of the device. The SMB PBX supports TLS encrypted calling (SIP), which requires SIP phone support.



Clicking **Action** to generates the Server Key, which will overwrite the original certificate if it already exists. Click **Download** to download the Server Key (including the asterisk.pem and ca.crt files).

Note: After regenerating the Server Key, the original Client Keys will be invalid and will need to be recreated in the Client Key.

Enter the Key Name and IP Address in the Client Key to **Create** the certificate.

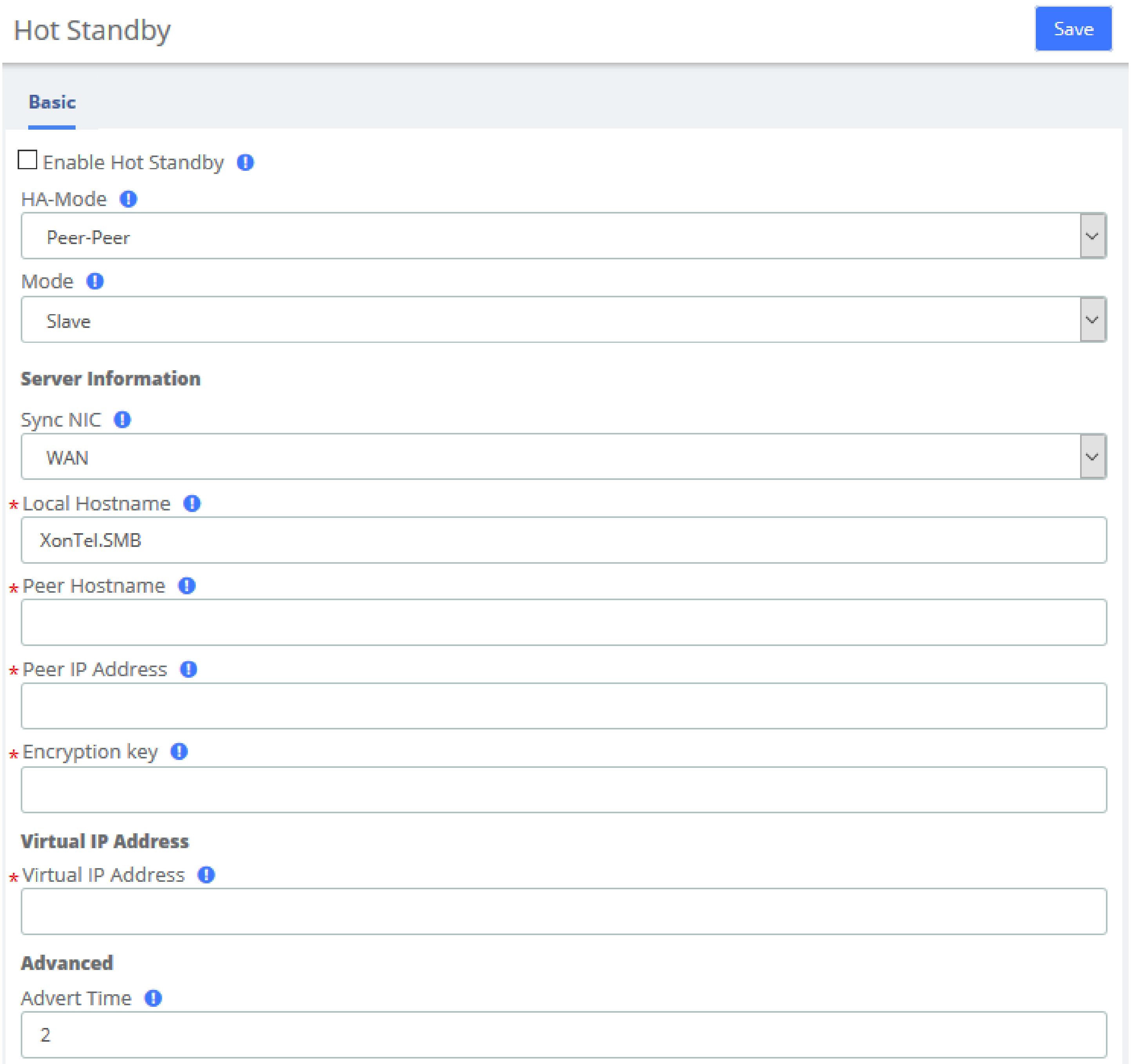
Note that if the device changes its IP, the corresponding client key will need to be generated again.

Download the Client Key (including [Key Name].pem and ca.crt), please import the Client Key into your SIP phone for encrypted transmission using TLS.

After mutual authentication between the client and the server, the phone can make encrypted calls. The specific parameters of the Certification module can be set in the column of **Transports** > **TLS** under **PBX**>**Settings**>**SIP Settings**.

2.3.4 Hot Standby

Hot Standby is a highly reliable application of software and hardware combination. The Hot Standby system consists of two identical PBX devices and control software system. The two devices appear as a single system in the network, and externally as an independent network IP, and control and management in the mode of a single system. The system mirrors the data and operational status of the two devices (including hard disk data and memory data), enables hot backup between the master and slave devices and seamless switching. Thus, providing stable and reliable services for users and achieving the high availability solution of dual-unit systems.



Options	Description
HA-Mode	Peer-Peer hot standby mode
Mode	The default is slave mode. The device that turns on the hot standby firstly is the master server.
Sync NIC	Network adapter which is used to heartbeat
Local Hostname	Hostname of the local host
Peer Hostname	Hostname of the peer host
Peer IP Address	IP address of the peer host
Encryption key	Master-slave devices must use the same password to communicate properly
Virtual IP Address	Master and slave devices share the IP addresses and this parameter must be identical on both sides
Advert Time	Heartbeat send interval. This parameter must be consistent in master/slave nodes. Default 1s.

2.3.5 Firewall

XonTel SMB PBX system has been preconfigured with a built-in firewall that protects your IP phone system from unauthorized access, phone calls and other attacks. It allows building Firewall rules to control the packets that send and receive by the PBX devices. To manage the firewall, navigate to web menu **Security->Firewall**.

The firewall has built-in default rules: accept all internal traffic, block all traffic from outside, and block all ports. After checking the **Enable Firewall**, click the **Save** button and the firewall will be turned on. If you don't want to be pinged by another device, you can check the **Disable Ping**.

Once the firewall is enabled, you can create, delete, modify, disable and reorder firewall rules. Click the **Save** button after each operation or it will be invalid in the system. Click the **Save** button every time a new or edited rule is completed, and then the list will automatically display your changes, otherwise they are invalid in the system.

Firewall

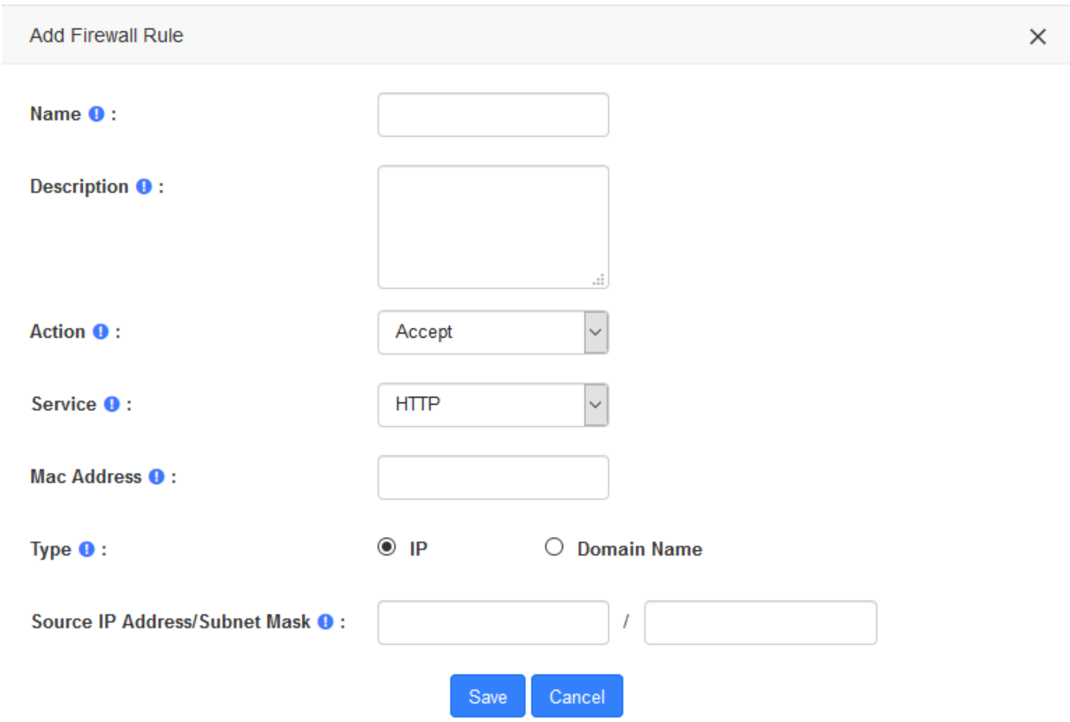
+ Add | Delete | Export | Import   Firewall is not enabled   ☐ Enable Firewall   ☐ Disable Ping   Save

☐	Name	Action	Protocol	Service	Source IP Address/Subnet Mask	Port	Edit	Delete	Move
☐	Allow Office IP 1	Accept	BOTH	Custom	185.36.179.9/255.255.255.255	1:65535			
☐	Allow Office IP 2	Accept	BOTH	Custom	195.39.130.18/255.255.255.255	1:65535			
☐	Accept Class B	Accept	BOTH	Custom	172.16.0.0/255.255.0.0	1:65535			
☐	Accept Class A	Accept	BOTH	Custom	10.0.0.0/255.0.0.0	1:65535			
☐	Accept Class C	Accept	BOTH	Custom	192.168.0.0/255.255.0.0	1:65535			
☐	Block SSH	Reject	TCP	SSH	0.0.0.0/0.0.0.0	13505:13505			
☐	Block WEB	Reject	TCP	HTTP	0.0.0.0/0.0.0.0	80:80			
☐	Block HTTPS	Reject	TCP	HTTPS	0.0.0.0/0.0.0.0	443:443			
☐	Block AMI	Reject	TCP	AMI	0.0.0.0/0.0.0.0	5038:5038			

Adding a New Rule

Click the button + Add to fill out the form to create a new firewall rule. The form will vary depending on the parameter selected for **Service**. You can simply select the **Action** and **Service** type, or customize the **Service** and set the port range.

In the **Source IP Address/Subnet Mask** field, you must enter an IP address in the format x. x.x.x/y, where y is the subnet mask and should be a number between 0 and 32. If you enter the default IP address (0.0.0.0), the subnet mask will be 0.


Add Firewall Rule

Name :

Description :

Action : Accept

Service : HTTP

Mac Address :

Type : ◉ IP ○ Domain Name

Source IP Address/Subnet Mask : /

Save Cancel

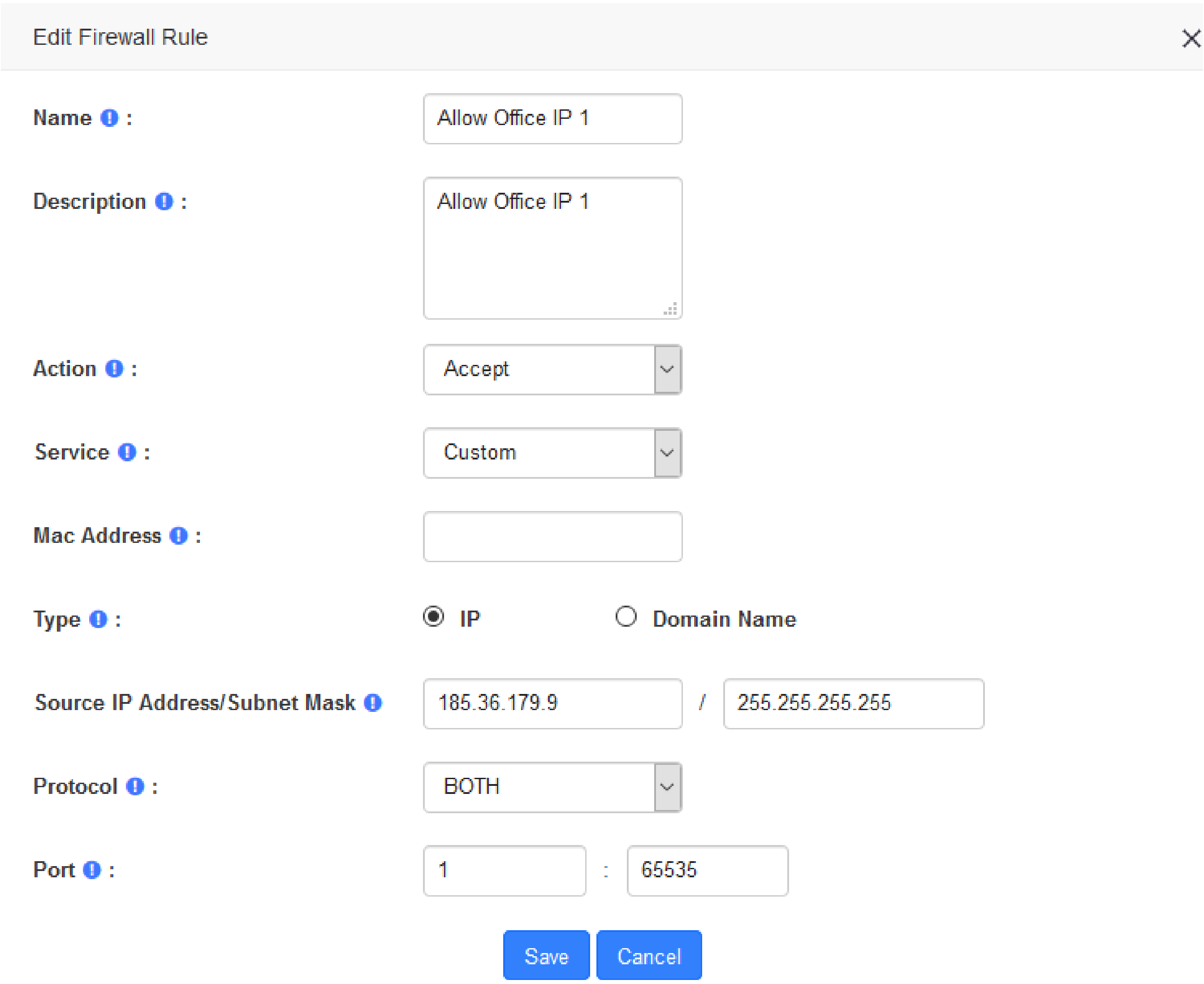
Options	Description
Name	Give this rule a descriptive name to help you identify it.
Description	A brief description of this rule. For example: accept a specific host to access the web interface for configuration.
Action	**Accept**: The device will accept access to the specified address. **Deny**: The PBX will deny the connection from the specified address and will send an error message to the other side informing them that the device has denied the connection. **Ignore**: The device will ignore the connection from the specified address, drop the data directly, and do not give any feedback. To improve the security of your PBX system, you can use **Ignore** actions to avoid malicious attacks to detect the server information of your device.
Service	Optional or customizable system services are available. By selecting a service, the default port for that service is selected. Of course, you can also customize the firewall service by selecting "Custom" and filling in the "Protocol" and "Port" options.
MAC Address	The MAC address format is:XX:XX:XX:XX:XX:XX:XX:XX:XX:XX.
Type	Select the type that matches this rule, either an IP address or a domain name.
Source IP Address/ Subnet Mask	The IP address format is: IP address/subnet mask, subnet mask needs to be written in full format, the short format is not supported. For example, 192.168.5.100/255.255.255.255 means that the rule applies to 192.168.5.100; 192.168.5.0/255.255.255.255.0 means that the rule applies to IP between 192.168.5.0 and 192.168.5.255.
Domain Name	Appears when "Domain Name" is selected for Type. The firewall rules will match the domain name filled in here.
Protocol	Appears when the service is selected "Custom", selects the protocol that applies to this rule, selects UDP, TCP and BOTH (UDP and TCP)
Port	Appears when the service selects "Custom" to specify the ports for this rule, which can specify port groups and individual ports.

	When specifying a port group, the left side is the start port and the right side is the end port (included), e.g. "5060:5070" means to specify ports 5060 to 5070 (including 5070). When specifying a single port, just fill in the same port number on the left as on the right. For example, "5060:5060" means that port 5060 is specified.

Once the rule is created, click the **Save** button and the new rule will appear in the list. Be sure to save the changes, otherwise, they will not take effect in the system.

Editing a Rule

To edit an existing rule, click on the icon corresponding to the rule. Here you can modify parameters of the rule.



Deleting a Rule

To delete a rule, just select the corresponding checkbox and click on the button. Be sure to save the changes or they will not work in the system.

Reordering the Rules

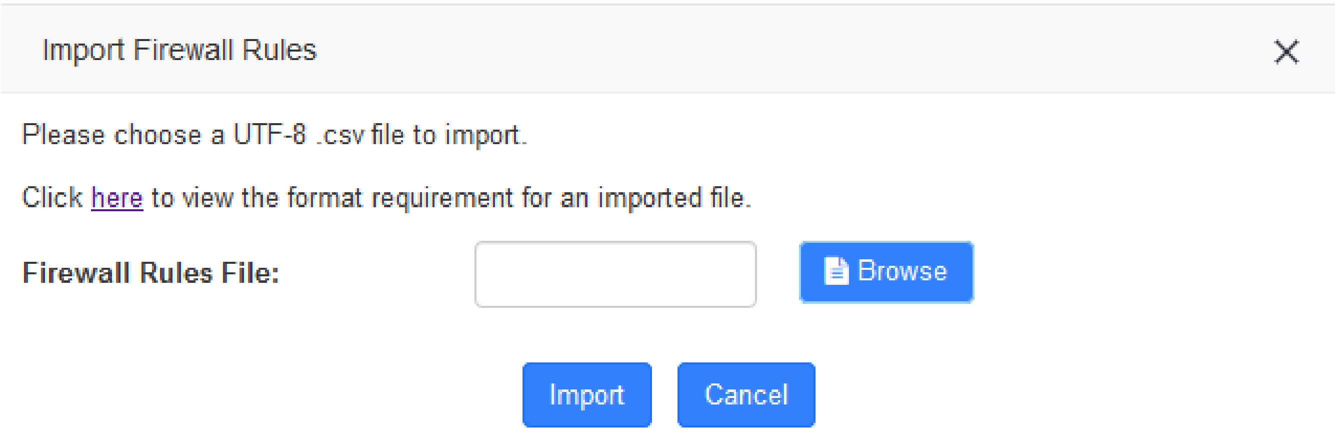
You can modify the order of the rules by clicking on the blue arrows in the column Move. If you click on the button of a rule, this rule will go up one position and if you click on the button, it will go down one position. If you click on the arrow, the rule will rise to the highest position which is the highest priority. Similarly, the arrows move the rule to the lowest position. Make sure you save the changes, so they will take effect in the system after modifying the position of the rules.

Export rules

Firewall rules now support exporting CSV files, just click the Export button and the browser will automatically download the exported CSV file. Note that please allow browser pop-ups.

Import rules

The firewall now supports importing CSV files to create rules in bulk, click the Import button and a popup will appear as follows



Click **Browse** to select the edited CSV file, then click **Import** to successfully import. The importation instructions are as follows.

Parameter	Importance	Restriction	Default Value
Name	Required	• The following characters are NOT allowed: & " ' \ < > ` | • The maximum length is 127.	N/A
Description	Optional system services and custom services	• The following characters are NOT allowed: & " ' \ < > ` | • The maximum length is 511.	N/A
Service	Optional system services and custom services	Permitted value: HTTP HTTPS SSH AMI SIP-UDP SIP-TCP SIP-TCP SIP-TLS SIP-RTP WEBRTC IAX2 LDAP MYSQL Custom	Custom
Action	Required	Permitted value: Accept Reject Drop	Accept
Protocol	Required	Permitted value: udp tcp both	udp
MAC Address	Optional system services and custom services	MAC address format required.	N/A
Type	Required	Permitted value: IP Domain	IP
Source IP Address/Subnet Mask	Required if Type is IP	IP format required.	N/A
Domain	Required if Type is domain	Domain format required.	N/A
Port	Required	The valid port range is 0-65535.	N/A

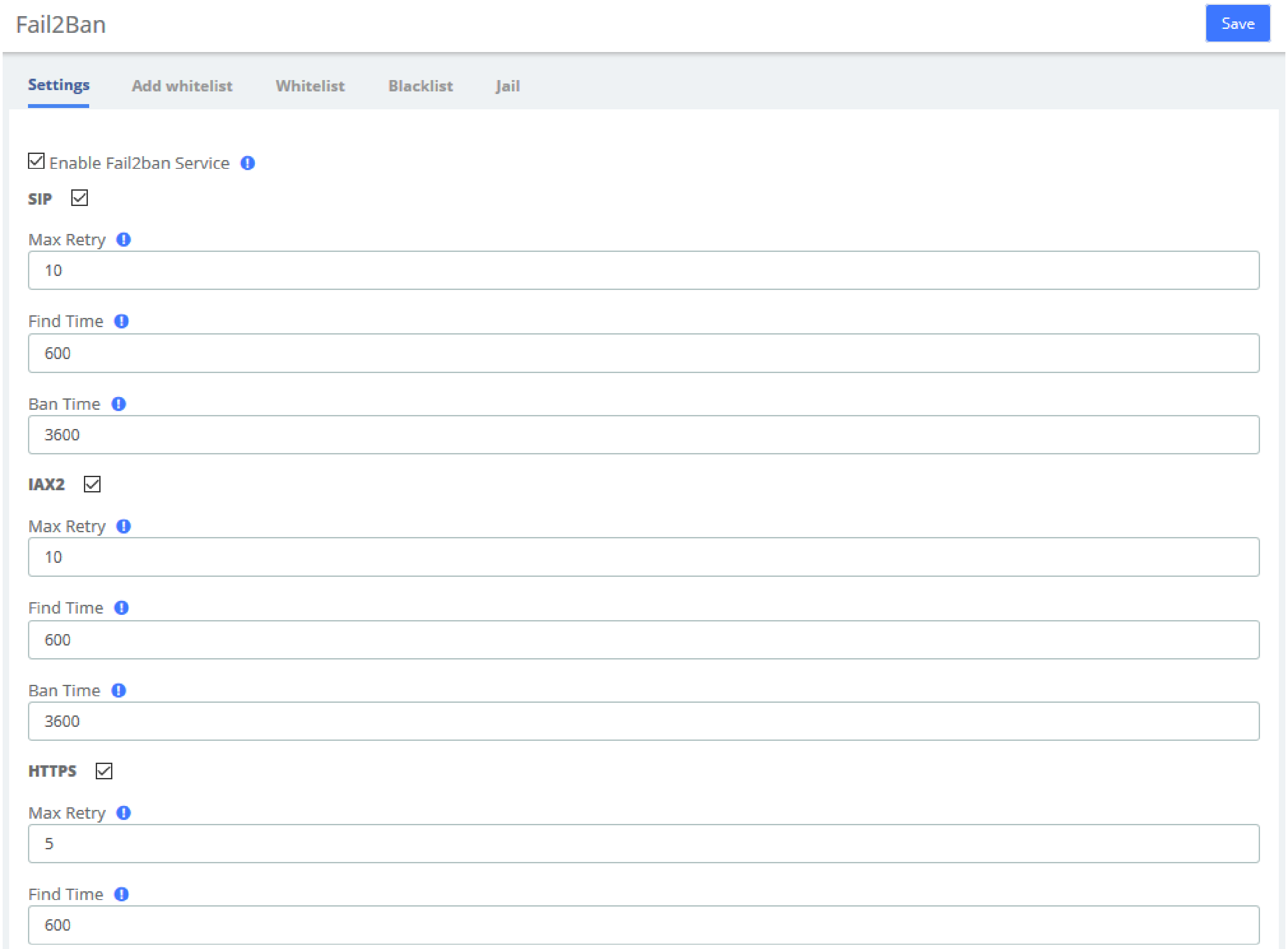
After opening Import Parameters - Firewall Rules, click Example and the browser will automatically download the template of the CSV file. Note that please allow browser pop-ups.

2.3.6 Fail2Ban

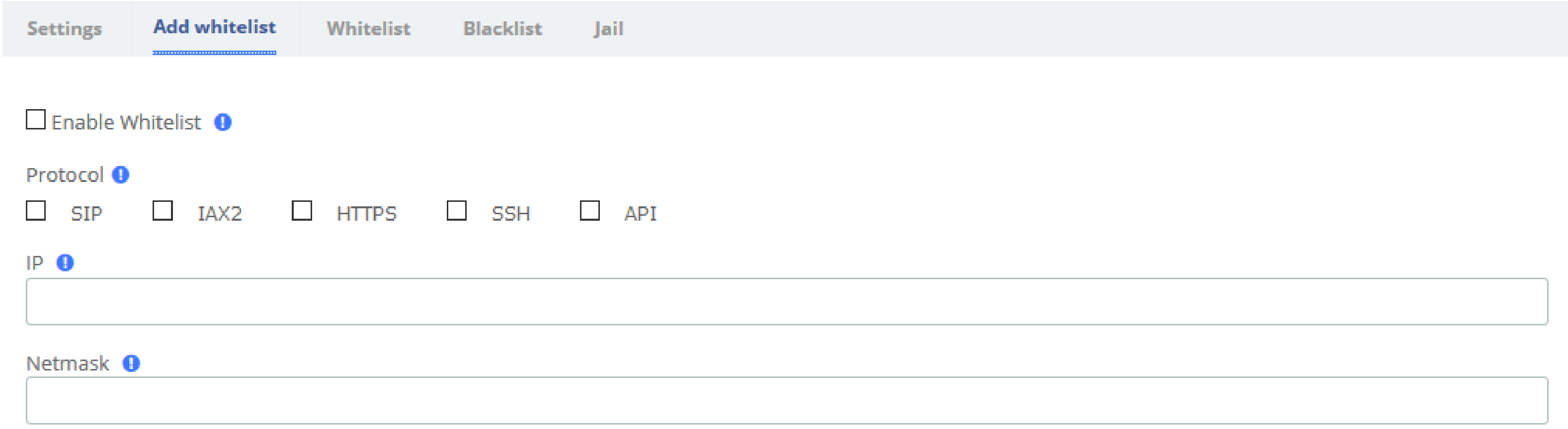
Fail2ban scans log files (e.g. /var/log/apache/error_log) and ban IPs that show the malicious signs -- too many password failures, seeking for exploits, etc. Generally, Fail2Ban is used to update firewall rules to reject the IP addresses for a specified amount of time, any arbitrary other action (e.g. sending an email) could also be configured.

Fail2Ban is able to reduce the rate of incorrect authentications attempts however it cannot eliminate the risk that weak authentication presents. Configure the longest password or passphrase permissible if you really want to protect services.

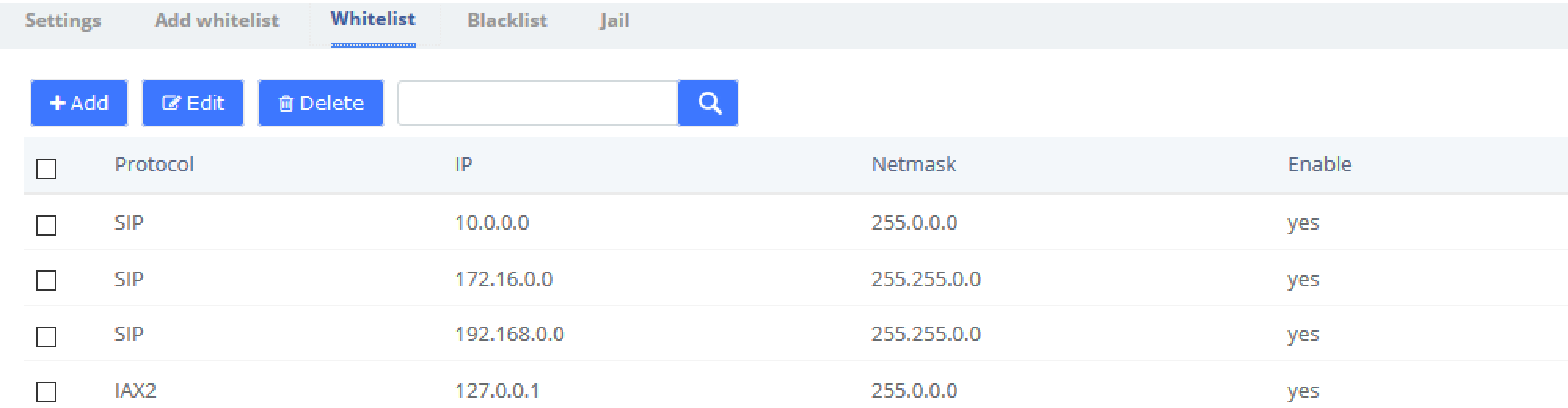
The module "**Fail2Ban**" allows configuring Fail2ban service so it can prevent the PBX from malicious attacks. Navigate to **System** > **Security** > **Fail2Ban** to configure rules.



Max Retry limits the authentication attempts. **Find Time** defines the time duration from the first attempt to the last attempt which reaches the "Max Retry" limitation. **Ban Time** is the time in seconds the PBX system will block the IP which exceeds max retry. Ban Time don't take effect on any whitelisted addresses.



Add whitelist allows you to add a trusted IP addresses or network addresses to the system IP whitelist.



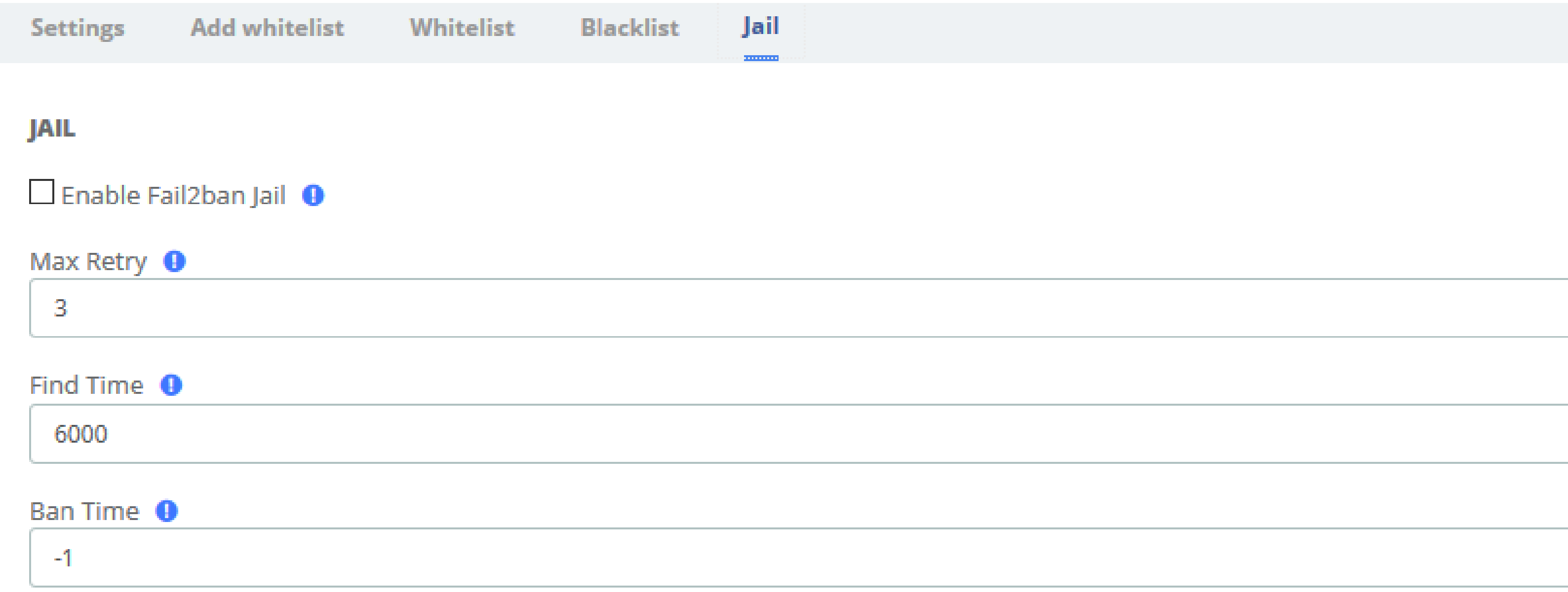
Fail2Ban whitelist

The IPs in the whitelist will always be treated as trusted IP's and will not be filtered by the firewall rules.

Fail2Ban blacklist

If mistakenly disabled, you can log in to that device with another IP and enter the blacklist to unblock it.

Jail is generally used for permanent bans, or "top bans", which are disabled by default. When running Fail2Ban Jail, if an IP has already been banned, and the IP continue to try to access and reach **Max Retry** within the set **Find Time**, then it will be blocked for longer time, this time is set by **Ban Time**, if Ban Time is set to -1, then it means permanent blocking.

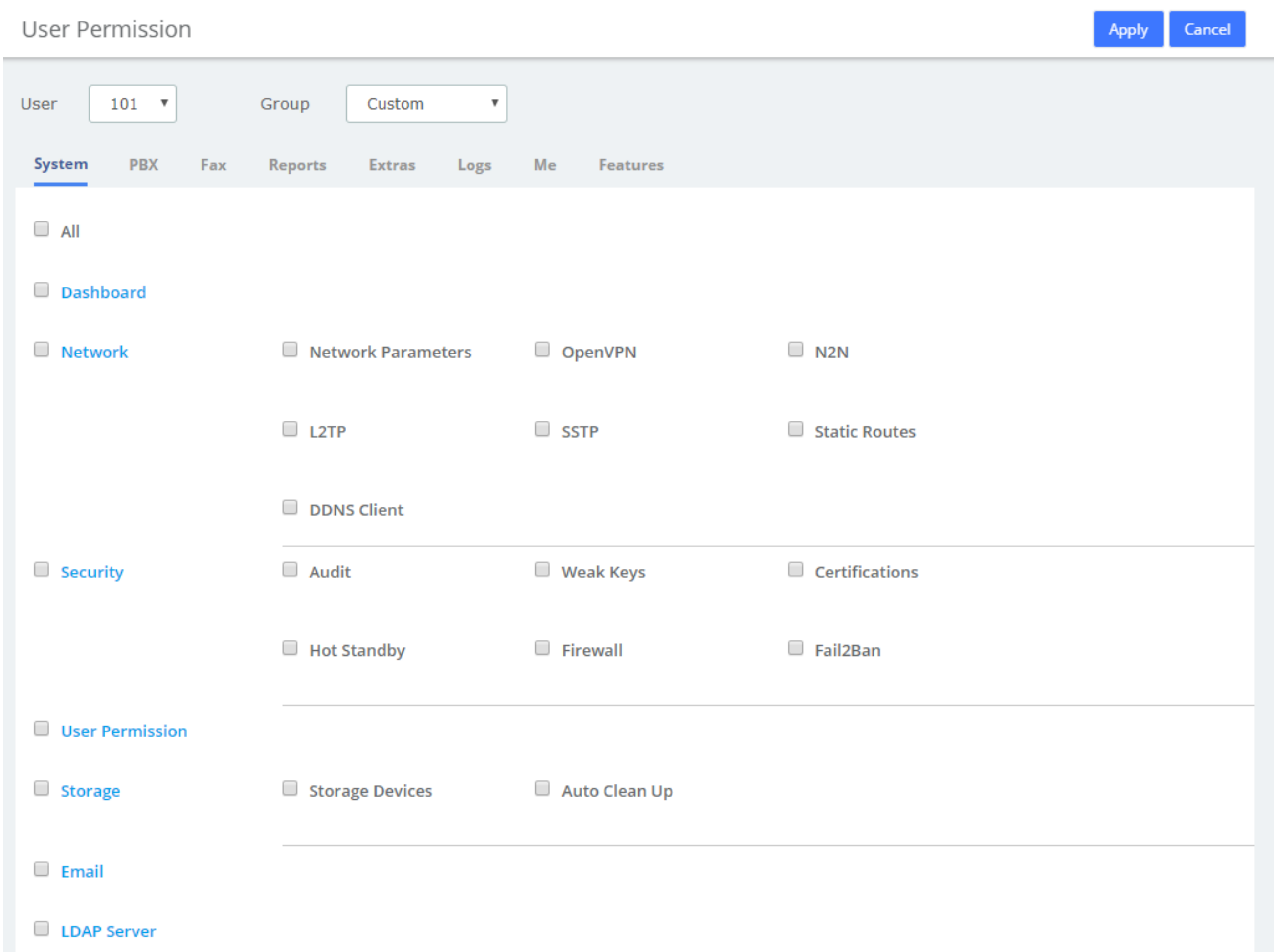


Fail2Ban Jail

2.4 User Permission

System > Users Permission allow the creation and modification of permissions for users accessing the web interface. An extension that has been granted access can log into the system using the SIP extension number/login password. It should be noted that by default, the user permissions give the Me module permissions for all extensions to log in and use some simple features.

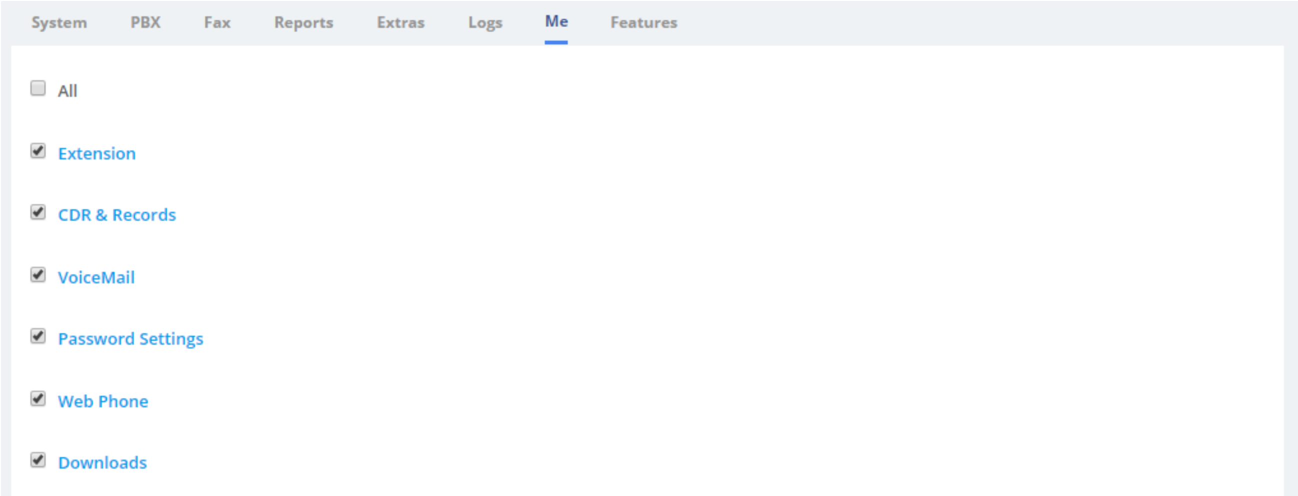
Click the Create button to grant permission to the specified extension, then Click Apply to save the configuration.



In the **User** drop-down box, you can select the corresponding extension, and in the **Group** drop-down box, you can select Custom/Administrator. If you set the user group as Custom, you can check the desired function module to give the user web privileges; if you select the user group as Administrator, all function privileges are enabled by default. Note that if some permissions are unchecked at this point, they will automatically become the Custom group after saving, in other words, the Administrators group will have all permissions at all times.

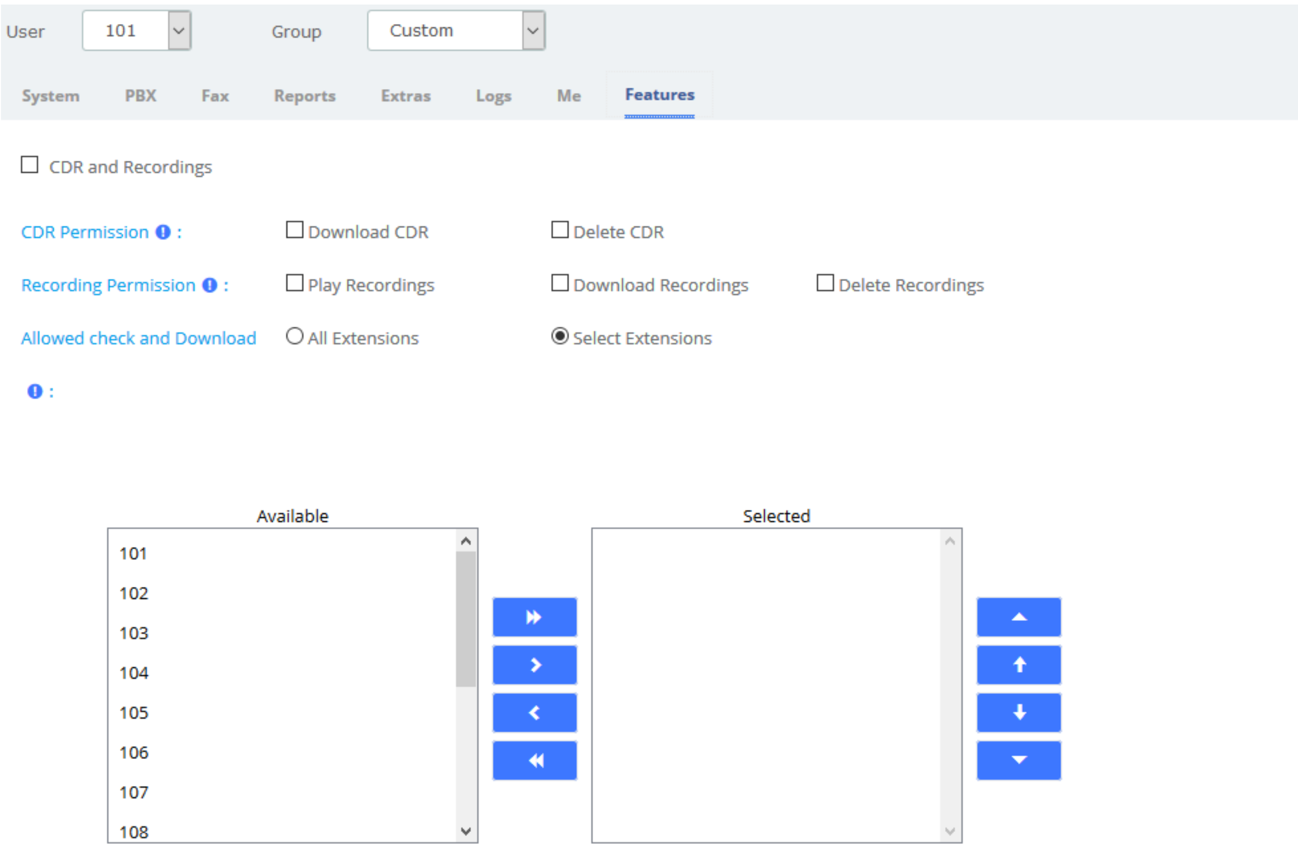
In addition to **Features** and **Me** modules, the other permissions correspond to the function menu on the left side of the page.

Me Bar provides basic permissions after extension user login and does not recommend modifications. See **8 Me Bar** for details.



User Permission/Me

The **Feature** provides enablement of some features associated with the extensions that are also used in the **Me Bar**.



User Permission/Features

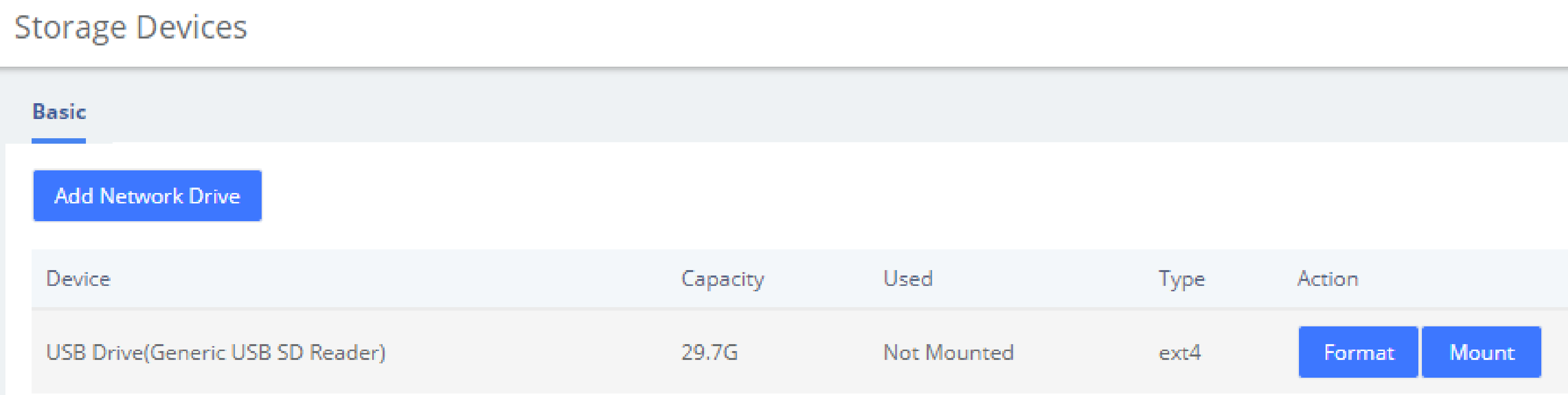
Type	Option	Description
CDR Permission	Download CDR	Allow the extension to download the CDR associated with this extension in the Me module
	Delete CDR	Allows the extension to delete the CDR associated with this extension in the Me module
Recording Permission	Play Recordings	Allows extensions to play recordings in the Me module
	Download Recordings	Allows extensions to download recordings in the Me module
	Delete Recordings	Allow extensions to delete recordings in the Me module
Allowed check and Download	Allows extensions to view downloads from other extensions in the "Downloads" section of the Me Bar	

2.5 Storage

2.5.1 Storage Devices

In this module, users can format or mount external storage devices such as TF/SD cards plugged into PBX devices, or add network storage. It should be noted that the system only allows one external device to be set as the primary storage device, which means that when one external storage device is mounted, other devices cannot be mounted at the same time. The large files such as audio files generated by the system will be automatically stored in the mounted external device.

Click **System**>**Storage**>**Storage Device**.



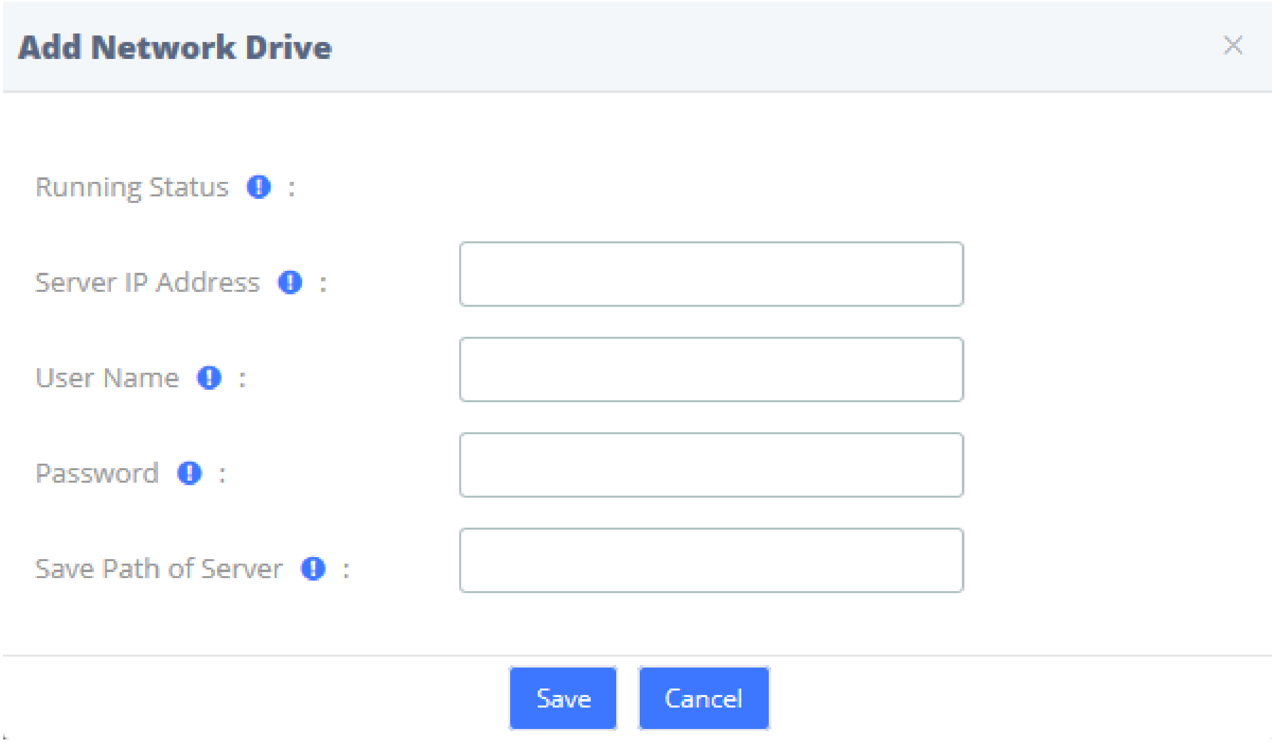
Storage Devices Interface

Click Format to format the inserted device. For TF/SD/U disk devices, only EXT4 or FAT file systems can be mounted. For non-EXT4/FAT file systems, please format them.

Click Mount to mount the device that has been inserted. At that time, large files such as recordings generated by the system will be automatically stored on the device. The Add Network Drive button will change to gray, and the **Unmount** button will appear.

Click Unmount to unmount the mounted device. **Add Network Dive** at that time will return to normal and click is valid.

Click Add Network Drive to add network storage, as shown in the following figure.

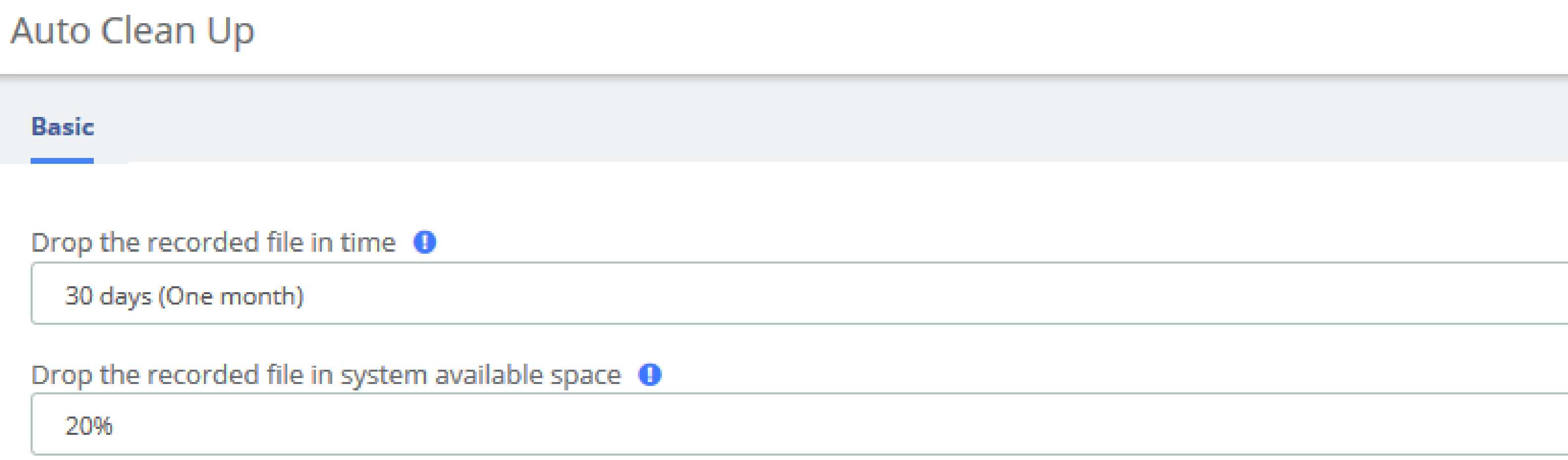


Add Network Drive

Currently network storage only supports **CIFS** services. Enter the Network Drive information, click **Save**, and you can mount it successfully.

2.5.2 Auto Clean Up

The option **Auto Clean Up** of the menu **Storage** allows you to configure the clean-up frequency.



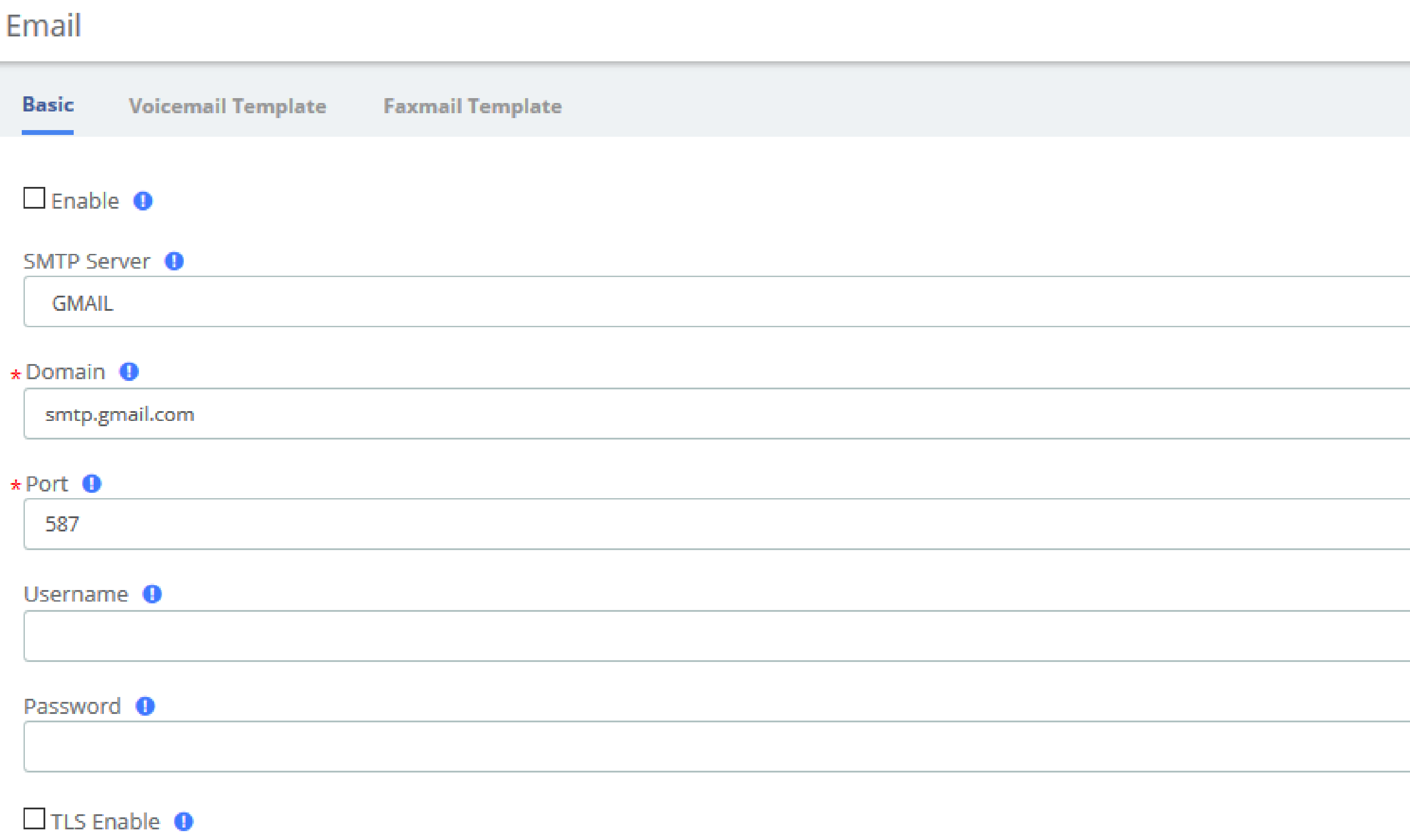
Auto Clean Up Interface

2.6 Email

The **Email** is mainly used in conjunction with **Event Center**, and by setting the remote SMTP configuration parameters of the mailbox, you can enable the Email service, send event reminder email and fax email, and provide you with timely and accurate information. It can also be combined with **Voicemail to Email**, allowing you to check your voice messages anytime, anywhere.

Note that there is no built-in SMTP server in the PBX system, but an external SMTP server is used.

The fields for configuring Email are shown below.

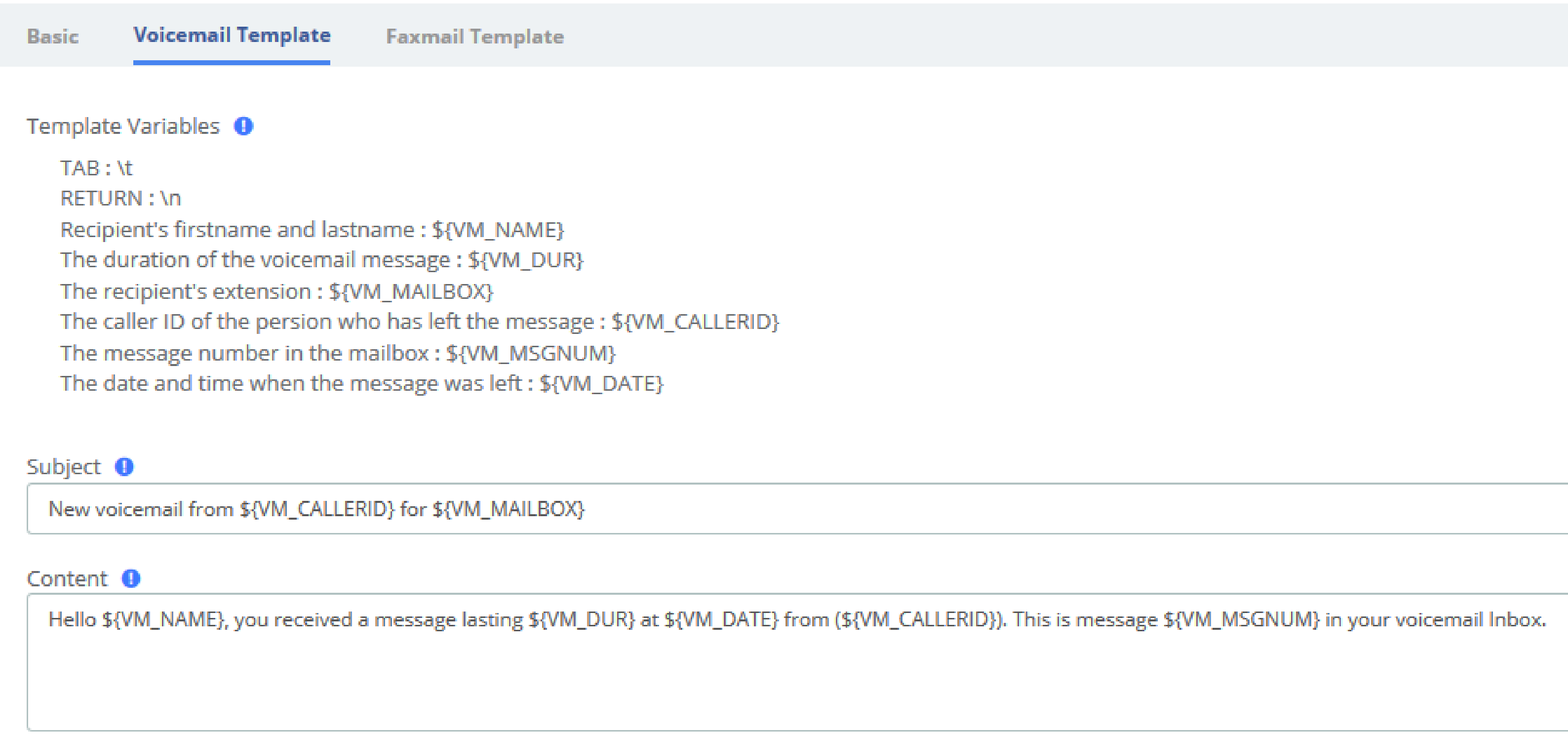


Item	Definition
Enable	Decide whether to turn on SMTP service
SMTP Server	SMTP server type. Multiple server types are built in, associated with Domain, or can be customized by selecting "other"
Domain	SMTP server address. It is automatically filled according to the SMTP server. When the SMTP

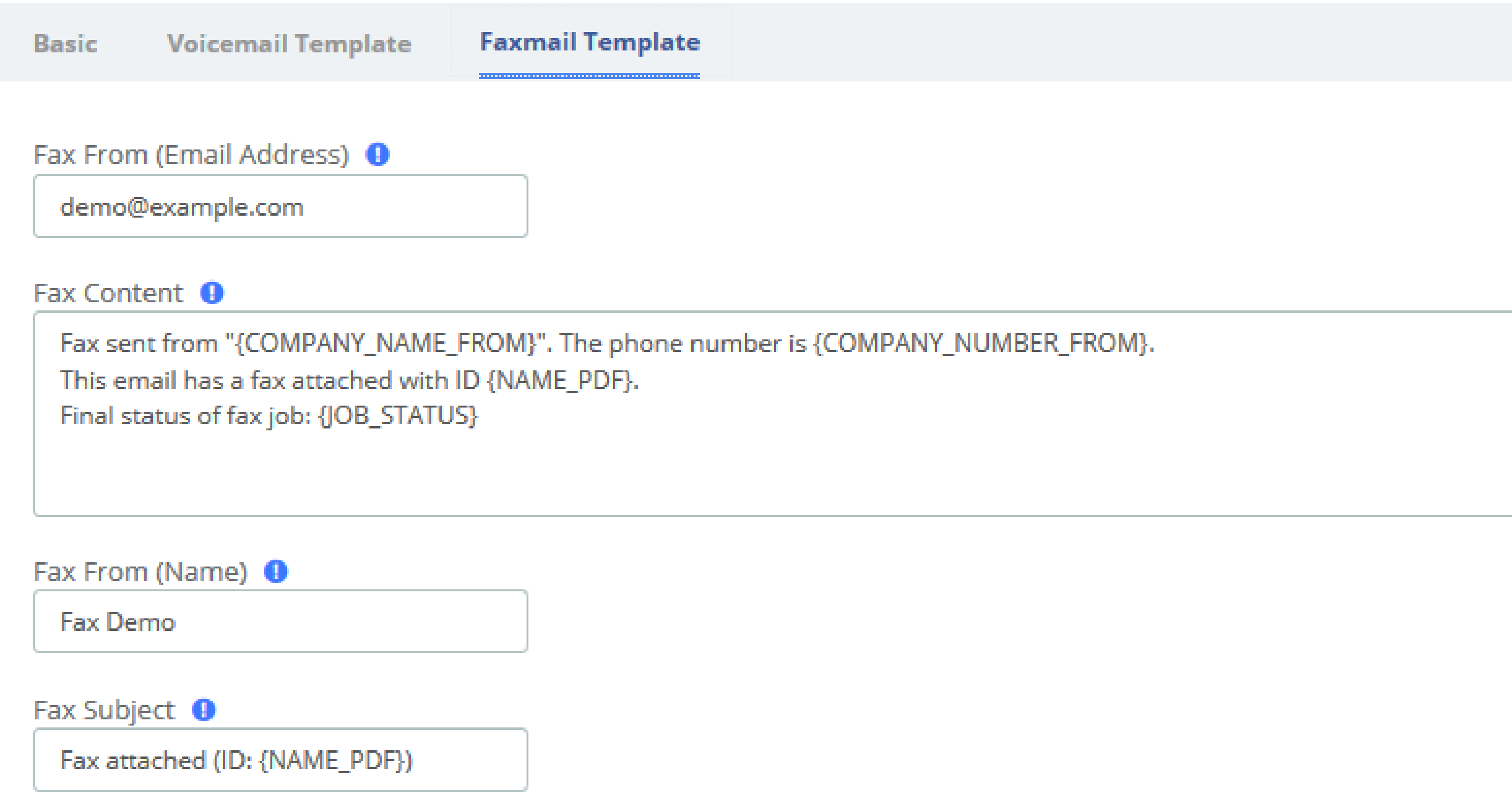
	Server selects "other", it needs to be filled manually.
Port	Port to establish the connection with SMTP Server. Common ports are 25, 465 (SSL), 587 (SSL)
Username	Username of email account from SMTP Server.
Password	Password of email account from SMTP Server
TLS Enable	To enable certificates of TLS (Transport Layer Security). Generally, this check is required when using ports that require SSL encryption, such as 465/587. If checked when using a port that does not require encryption, it will cause the send to fail

After setting Email, if you want to send a test email to check whether the Email function is enabled correctly, please click **Save** and then click **Test**, and a dialog box will pop up for sending.

The **Voicemail Template** and **Fax mail Template** options edit the Voicemail and Fax mail Template. After filling in the template variables in the Subject or Content according to the example shown above, they will be replaced with the corresponding parameter values when the actual email is sent.



Email/Voicemail Template

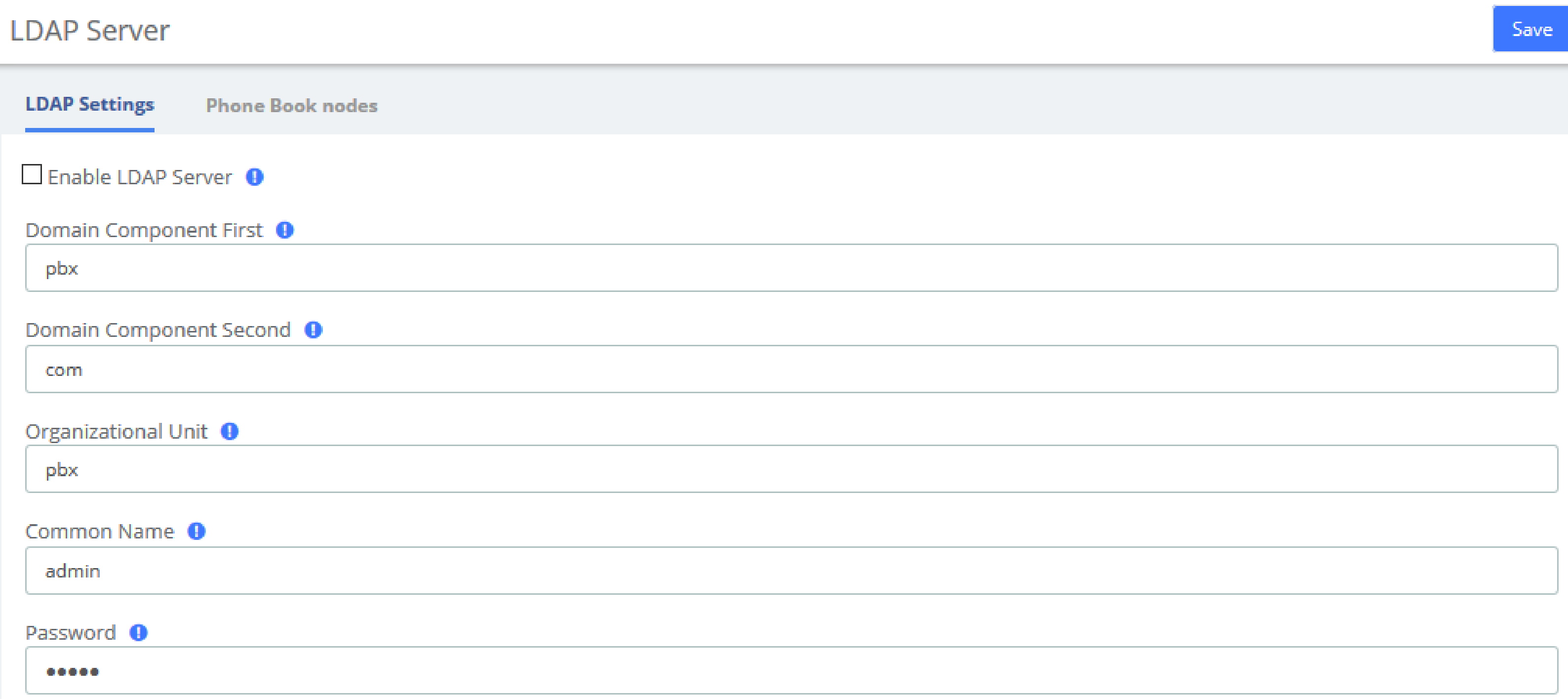


Email/Faxmail Template

2.7 LDAP Server

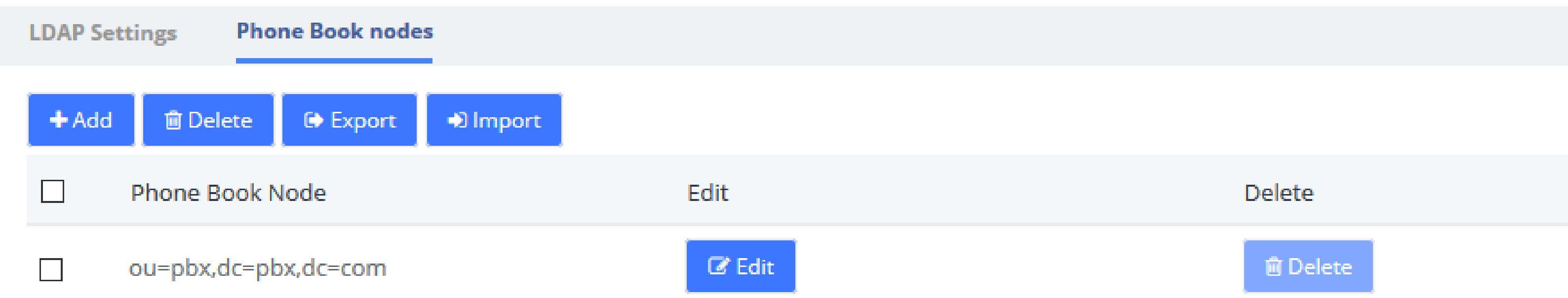
LDAP (Lightweight Directory Access Protocol) is a protocol for accessing directory services. It is generally used as a phone book on PBX. Based on the available LDAP services, it meets the requirements for fast search of phone directories. You can set up PBX as a server.

If you want to use LDAP service, just check the Enable LDAP service saving checkbox, and use the default configuration for the rest of the content. Once LDAP is set up, you can search the LDAP directory and find contacts on your IP phone.



The PBX has a built-in default phonebook node that contains all extensions on the system, which cannot be deleted or edited.

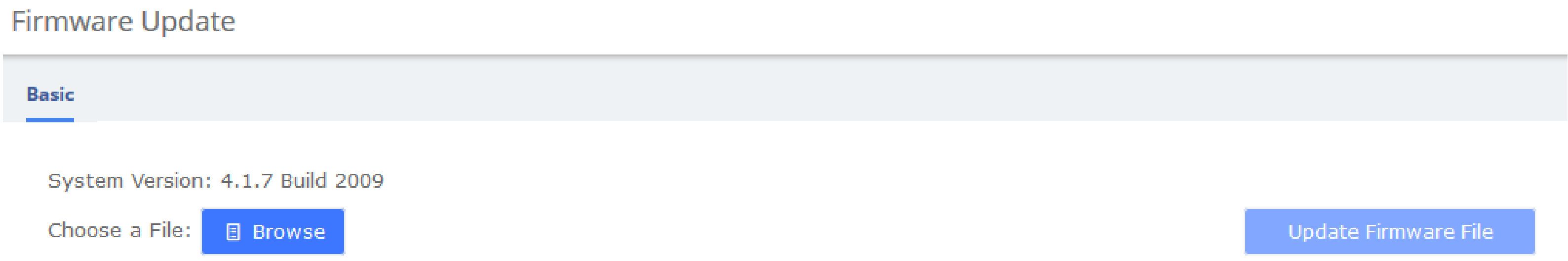
Of course, you can also manually add a phone book node, click the **+Add** button, enter the phone book name and save. Click **Edit** to add your contact information.



2.8 Maintenance

2.8.1 Firmware Update

The option **Firmware Update** of the menu **Maintenance** allows you to update the firmware version by uploading firmware file you download from the official website.

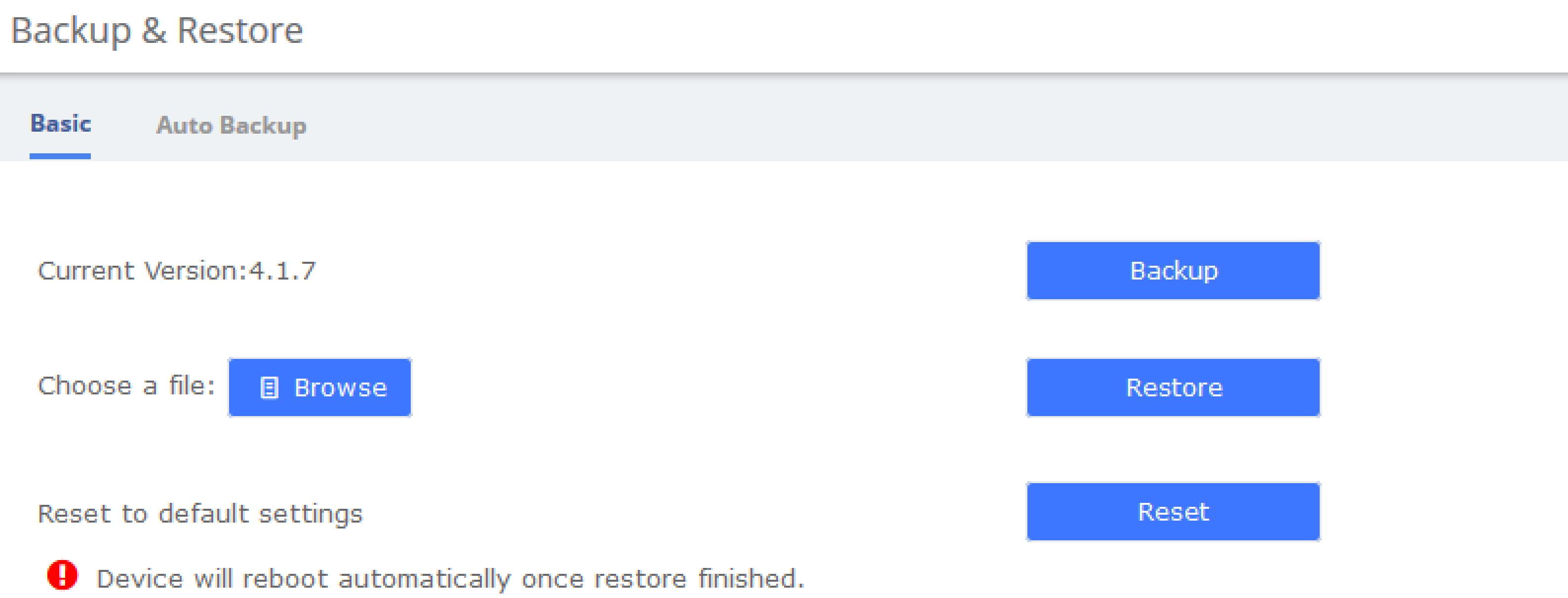


2.8.2 Backup & Restore

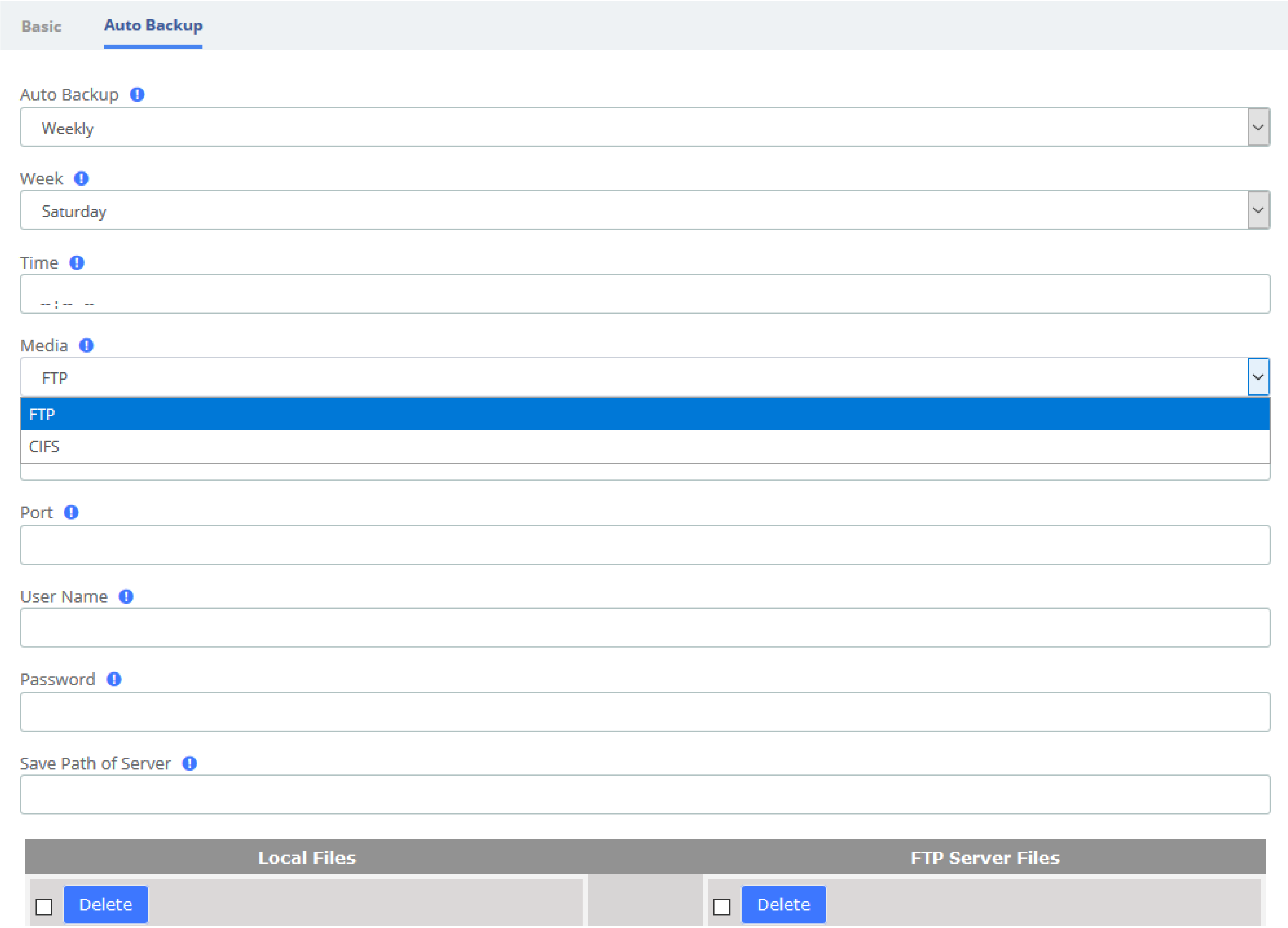
The **Backup & Restore** option in the **System** menu allows you to back up and restore the configuration of the PBX system.

If you have already made a backup before that, you can click **Browse** to select your backup file, upload it and select **Restore** to restore the backup. When you restore a backup, you will be asked if you want to keep the IP address of your current system. If you choose no, the IP address of your system will be changed to the IP address of your backup after restoration. You can also click Reset to restore the factory defaults.

Please note that both the restore backup and reset operations are not reversible.

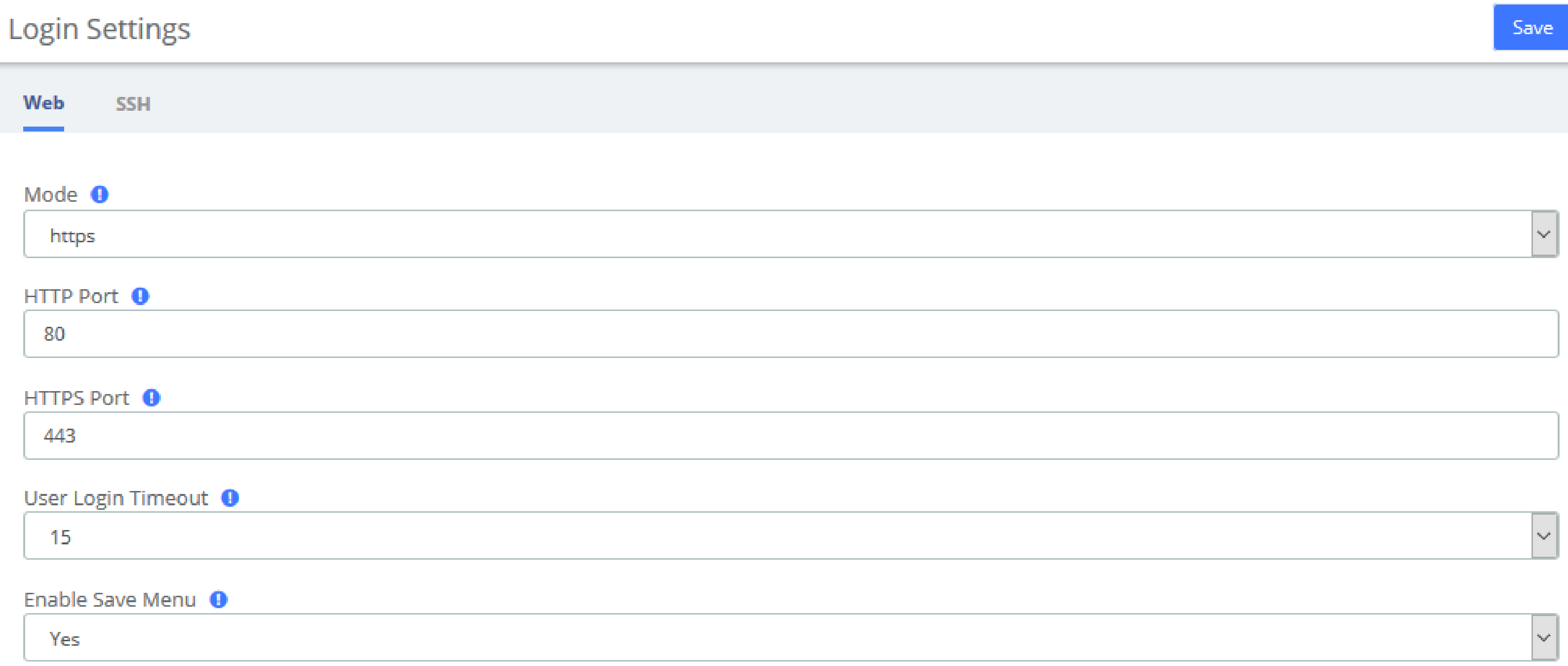


To enable Auto Backup, navigate to **System > Maintenance > Backup & Restore > Auto Backup**, change the disable option to the frequency you want. There are three media you could select to back up your config file: USB/SD Card, FTP and CIFS.



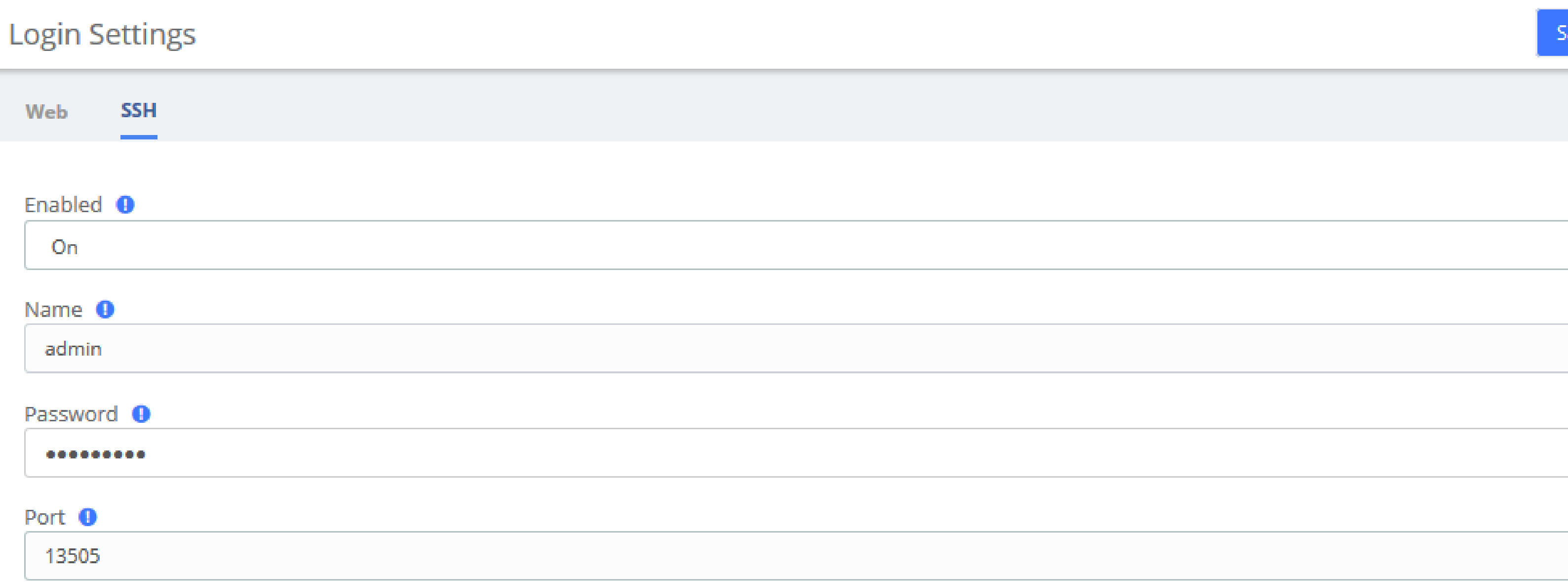
2.8.3 Login Settings

Navigate to **System** > **Maintenance** > **Login Settings** to setup the login mode and port.



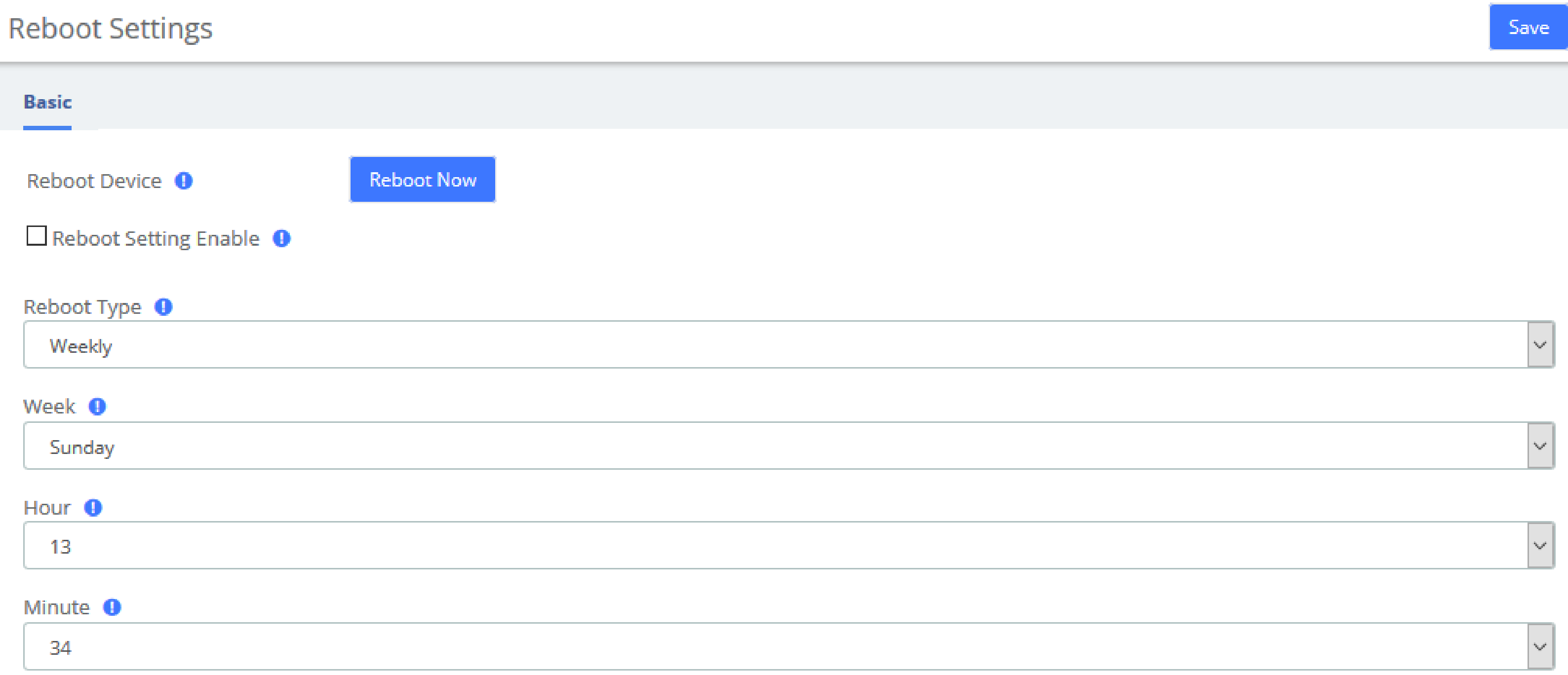
The SSH settings page requires **Developer Mode** to be enabled

After you turn on Developer Mode, you can log in and set up SSH. SSH default port is **13505**, select Enabled-On option, set Name and Password. Click Save.



2.8.4 Reboot Settings

SMB PBX system supports setting timed automatic restart. Navigate to **System** > **Maintenance** > **Reboot Settings**.

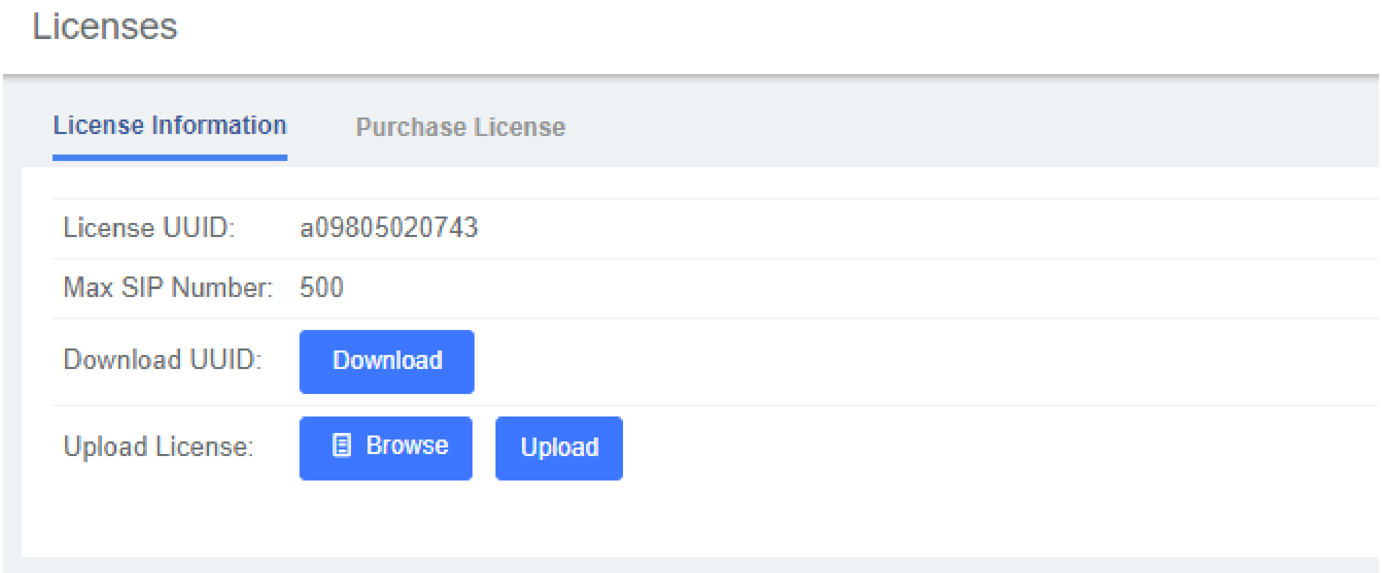


Reboot Settings Interface

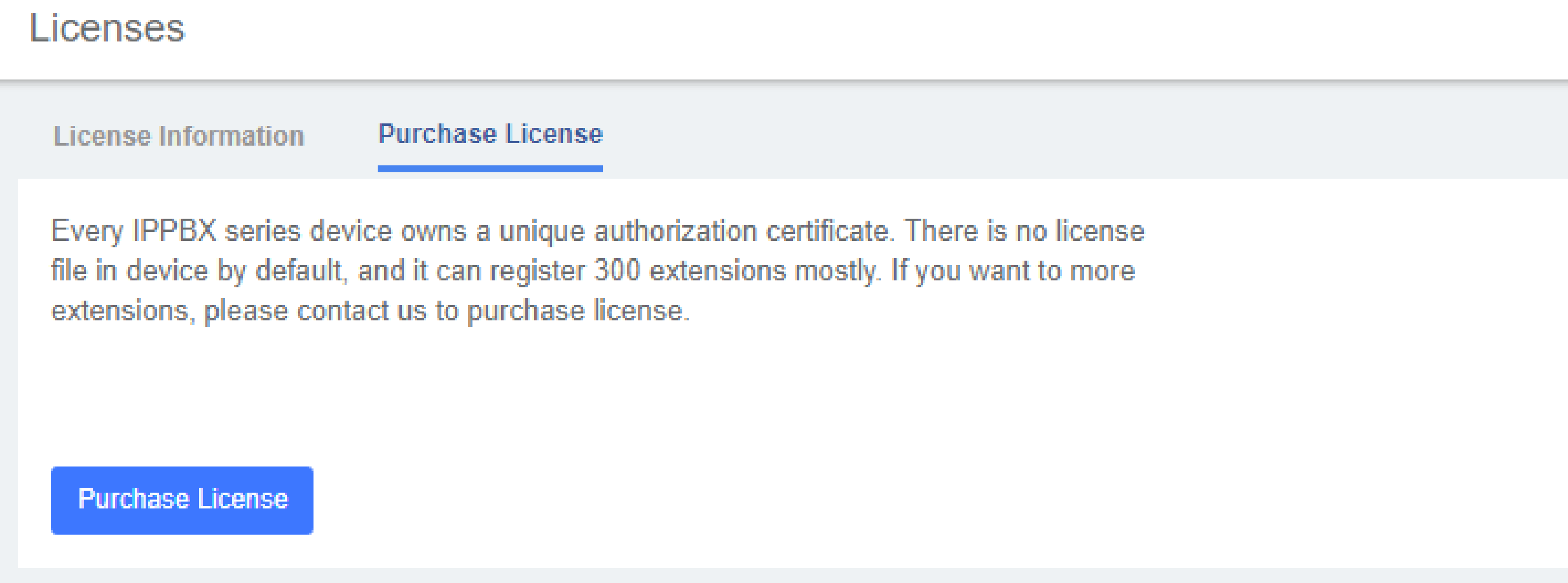
2.9 Licenses

There is no license file in device by default and it supports **300** extensions in maximum. Uploading license can make the device to support more extensions (for more info please contact XonTel).

Click Upload to upload the license.



Upload License Interface



Purchase License Interface

2.10 Event Center

SMB PBX system provides event monitoring and alert function, users can set events that need to be monitored and notification content, after adding notification contacts, the device will send reminders by sending emails or calling extensions, so that users can fully grasp the system dynamics.

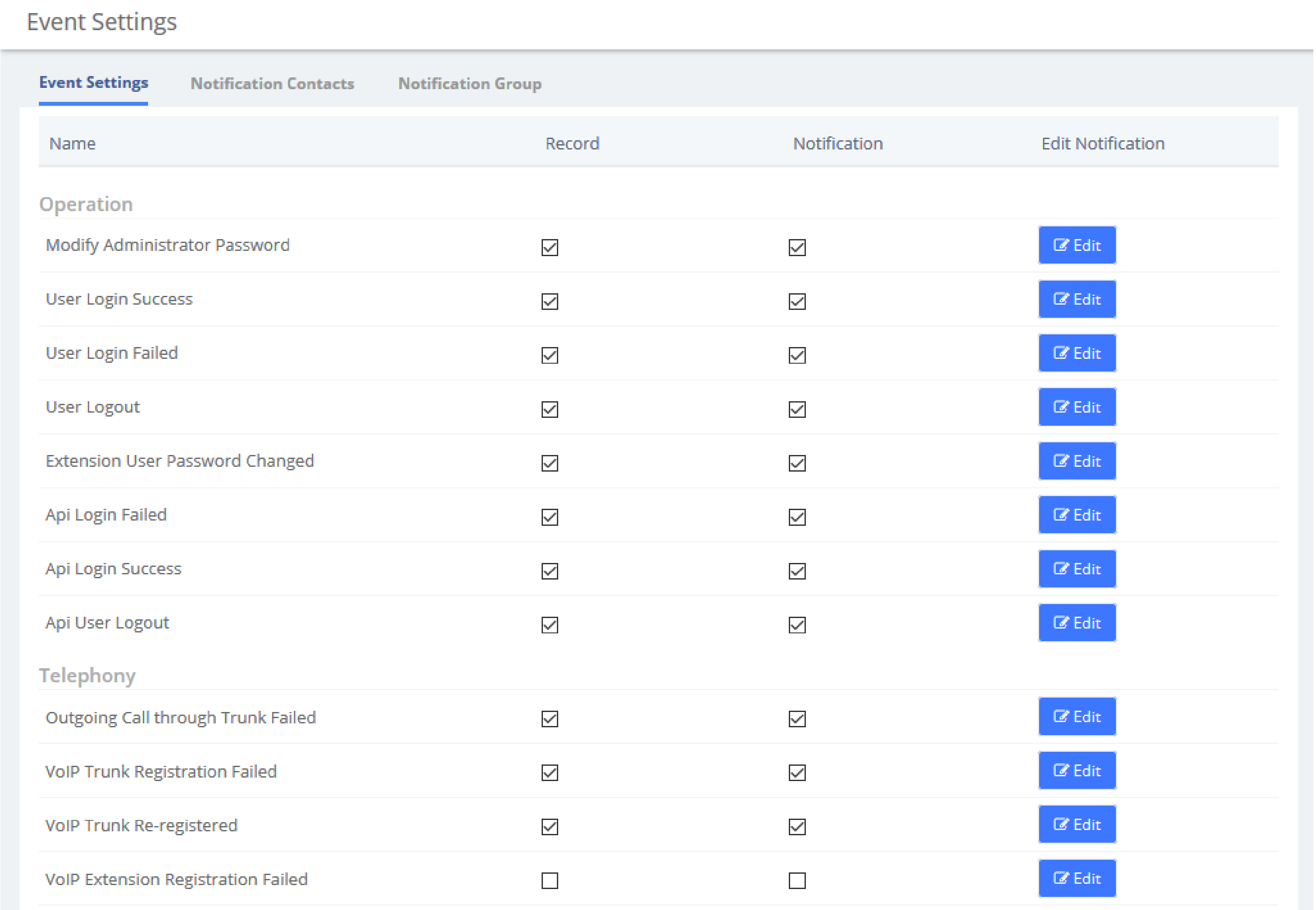
2.10.1 Event Settings

When the **Record** column checkbox is checked, the system will record the corresponding event in the Event Logs, or you can click the 

at the top to view it; when the checkbox of the **Notification** column is checked, you can set the notification by email or phone, but you need to add the contact information in advance.

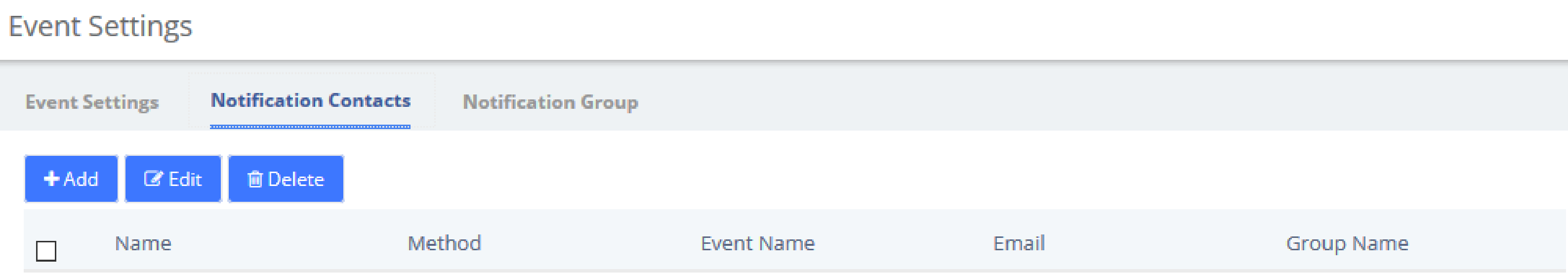
Click 

to edit the Notification Template and personalize the notification.



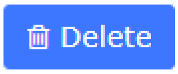
You can set up **Notification Contacts** to be notified by sending an email or calling when an event occurs. Click to **Add** contacts.

Note: For email notifications you must setup SMTP in the PBX

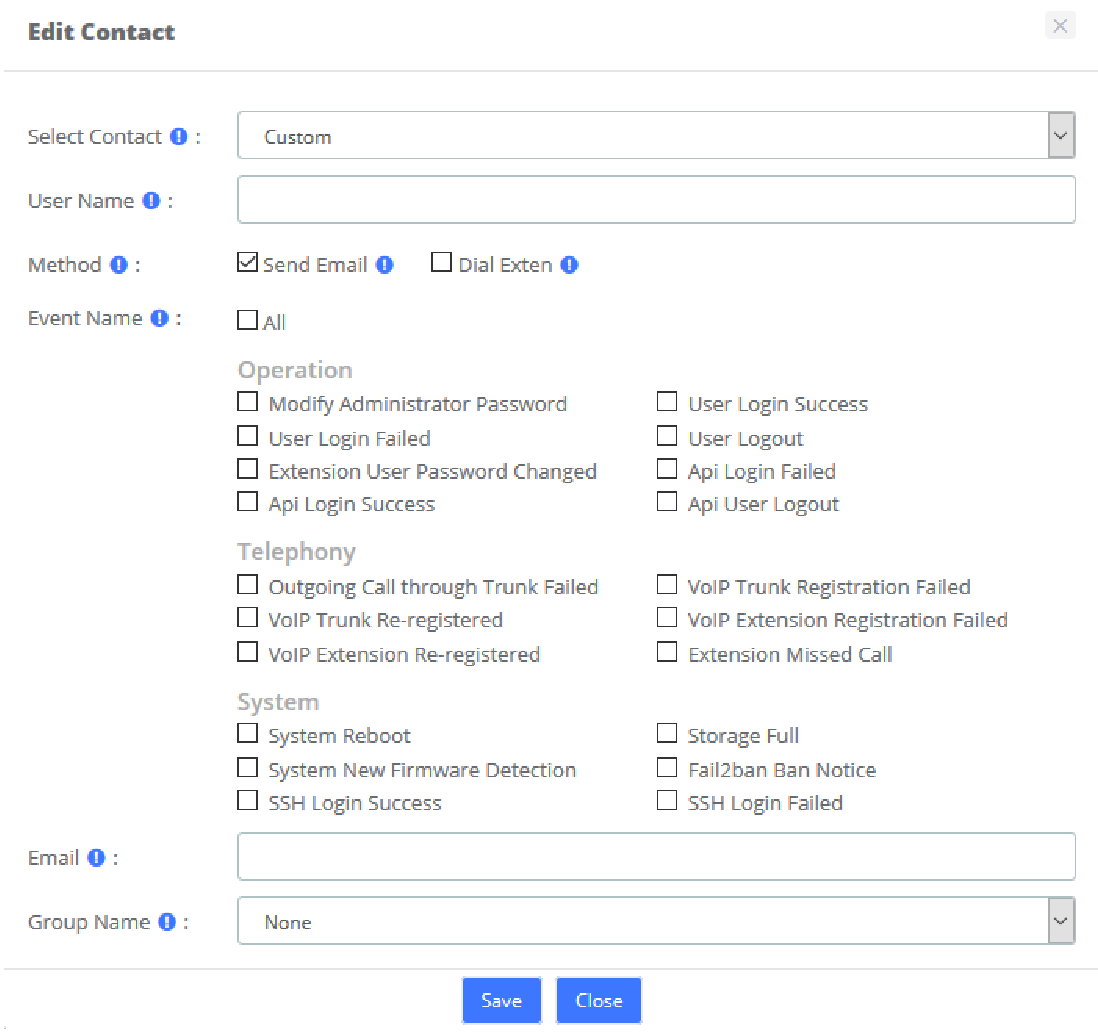


Notification Contacts

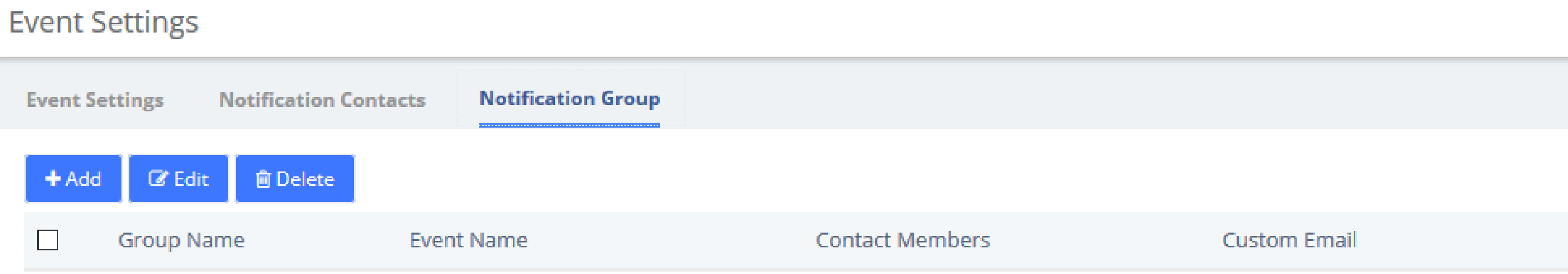
Once you are done, click 

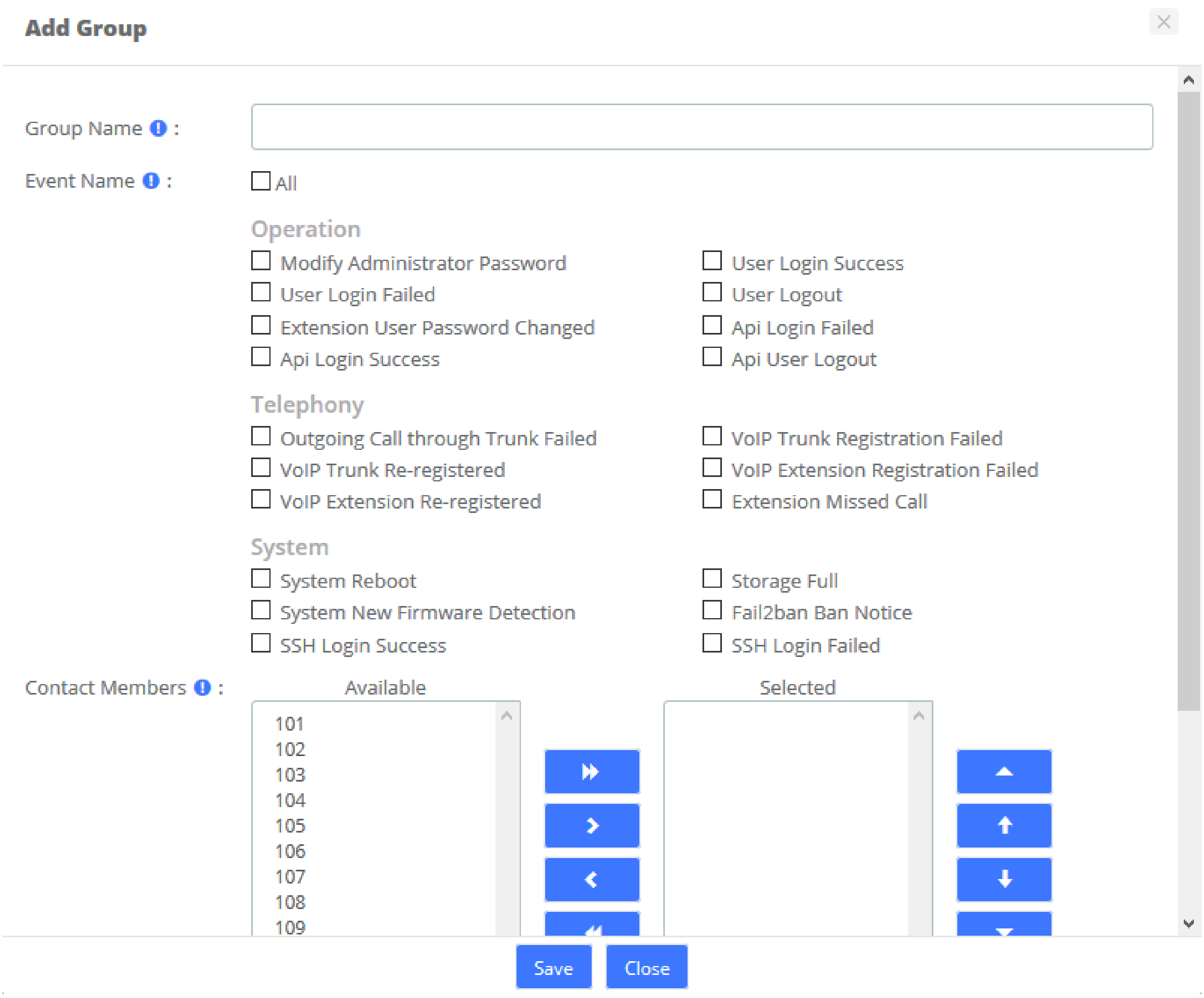
to edit the current contact and 

to delete the contact. Of course, it is also possible to select multiple contacts for bulk deletion.



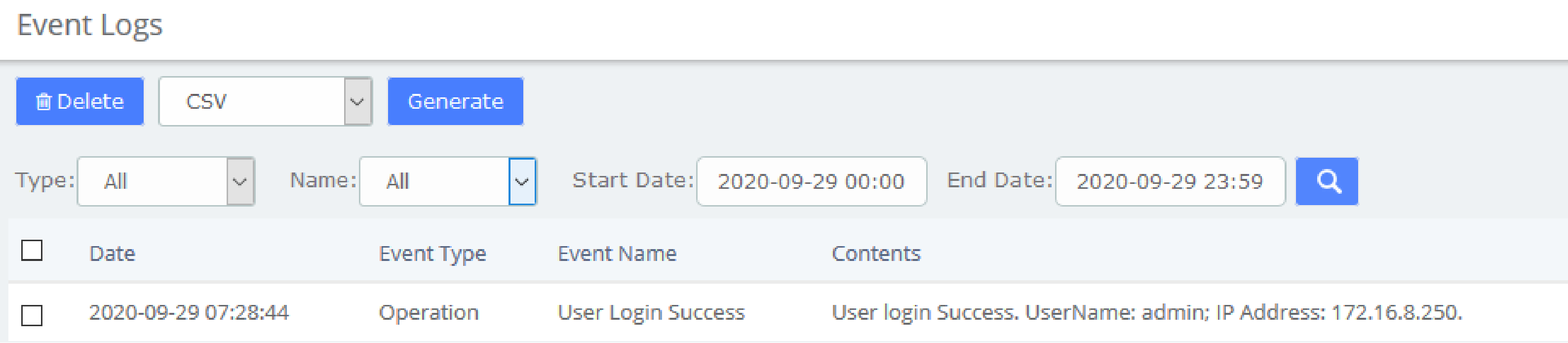
You can set up **Notification Group** to notify a group people by sending an email or calling when an event occurs. Click to **Add** a group.

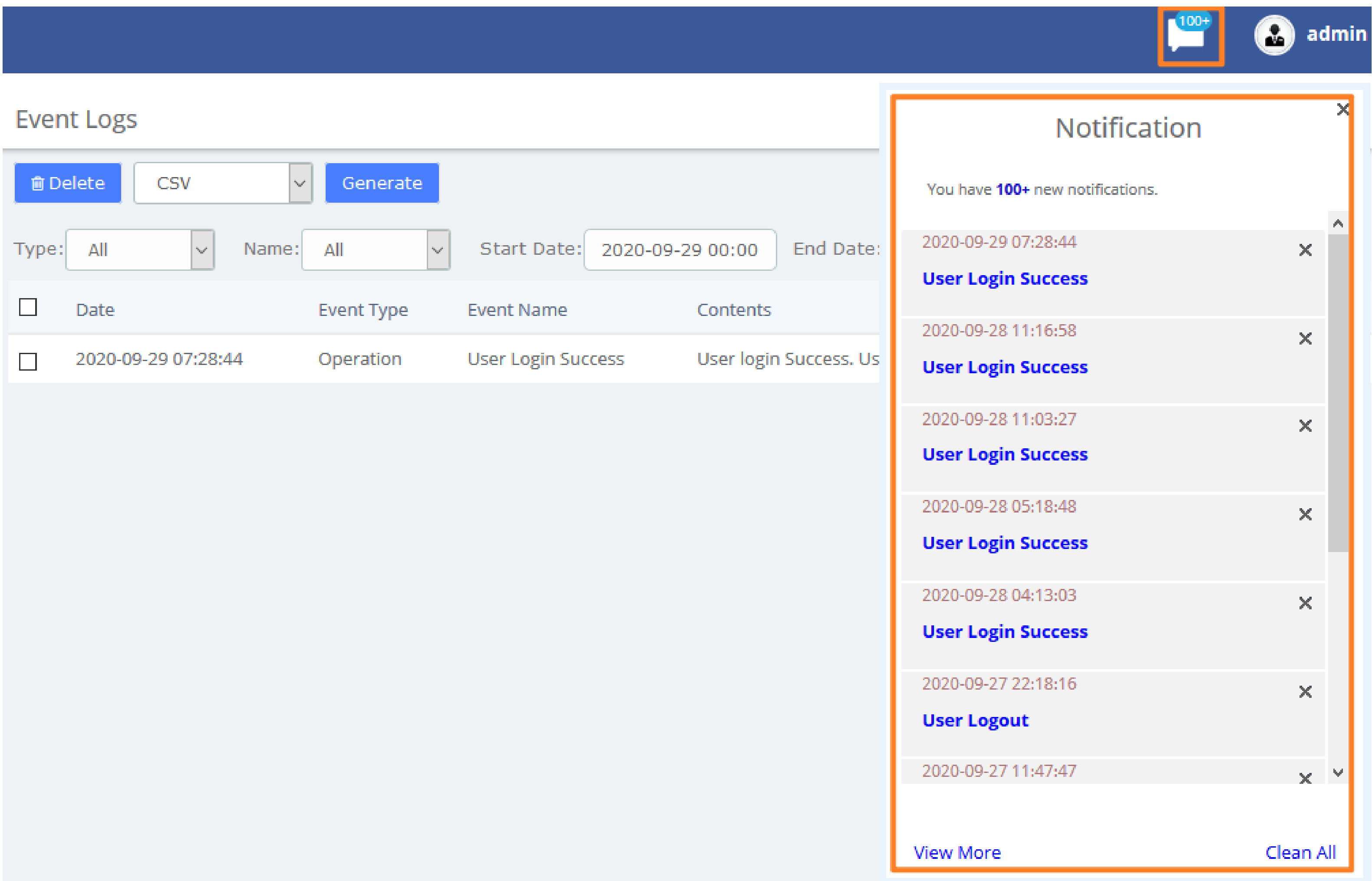




2.10.2 Event Logs

You can view logs related to monitored events in both the notification bar in the upper right corner and the **Event Center** > **Event Logs** page.



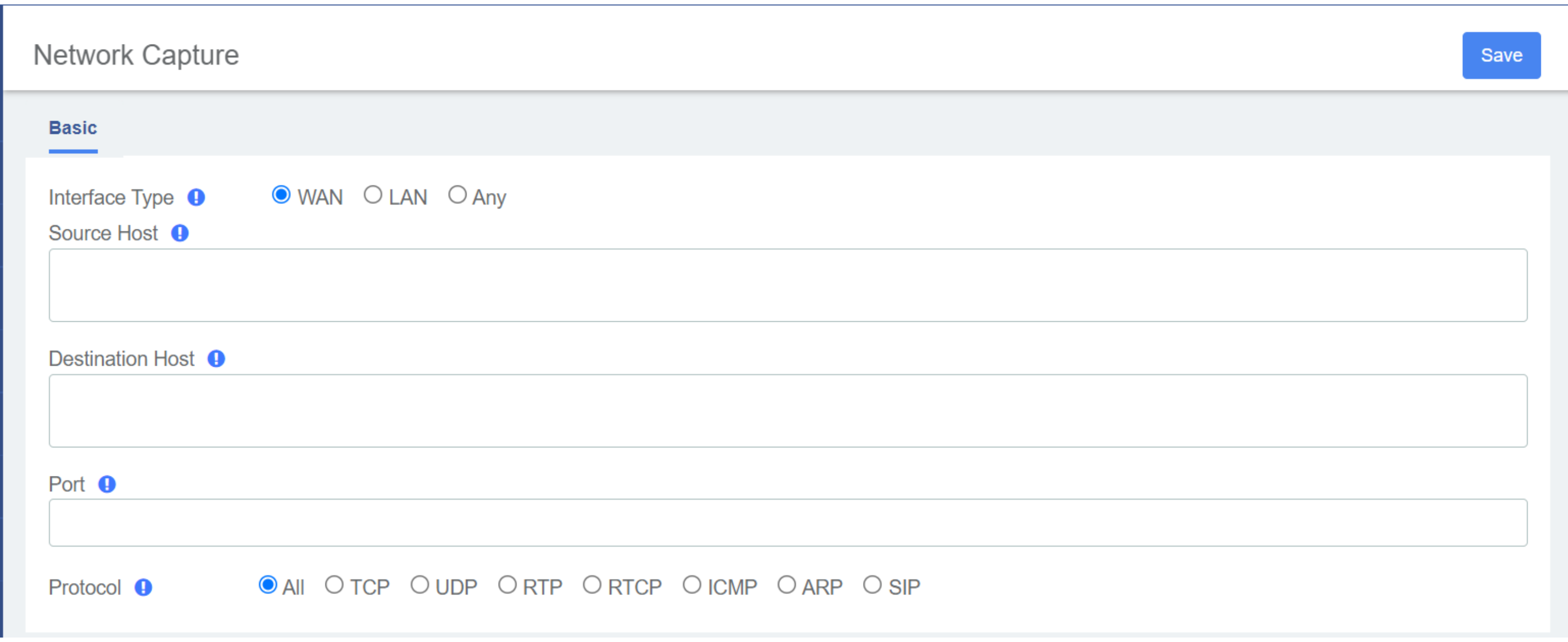


Event Logs

2.11 Tool Kit

2.11.1 Network Capture

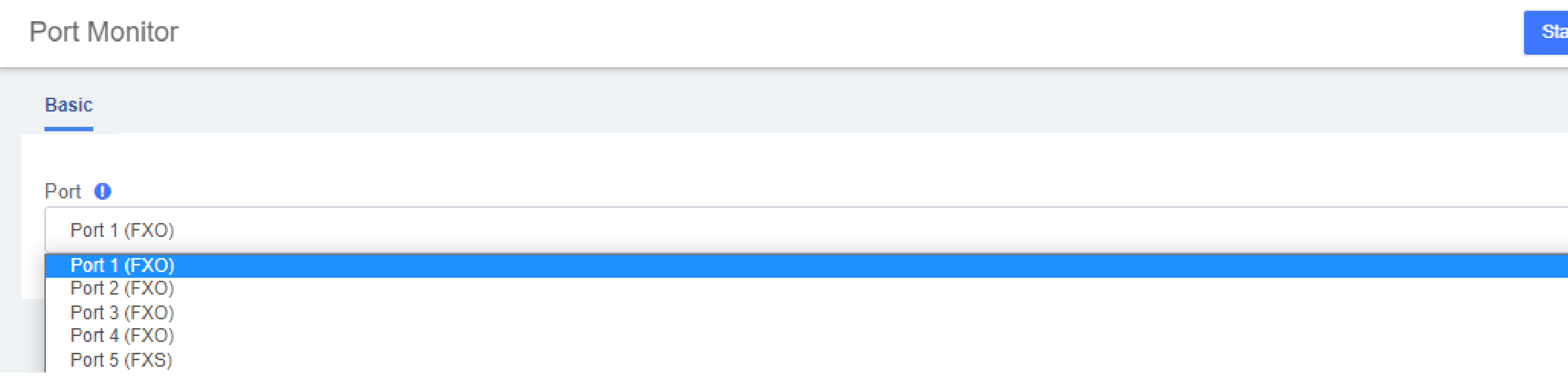
The XonTel SMB PBX provides network packet capture function for ease of user to analysis, capture and monitor the network status, RTP streams, protocol and so on.



Capture interface

2.11.2 Port Monitor

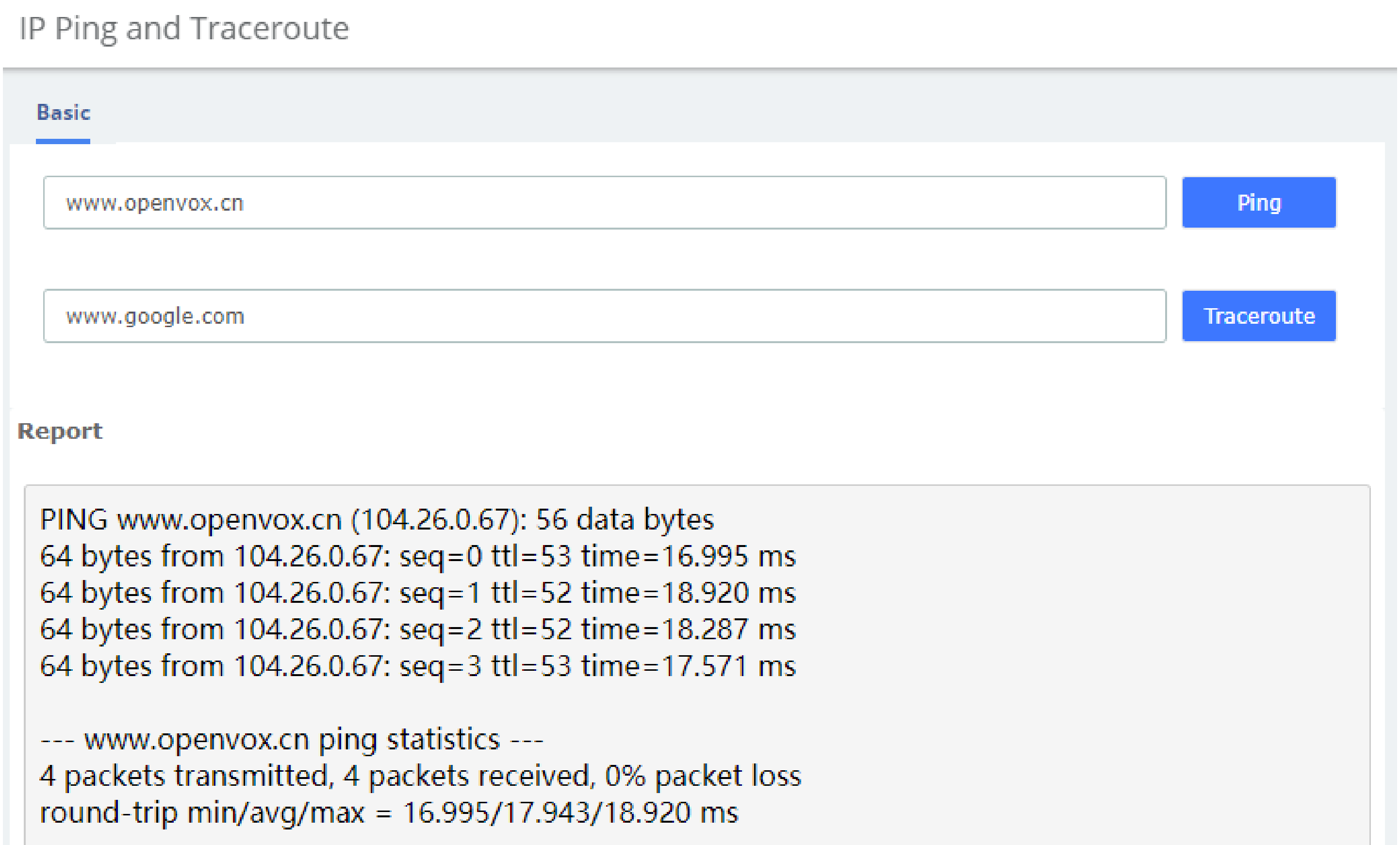
It also provides Port Monitor module for user to monitor and record the port communications.



Port Monitor interface

2.11.3 IP Ping and Traceroute

The IP Ping and Traceroute module assist user to check the network connectivity.



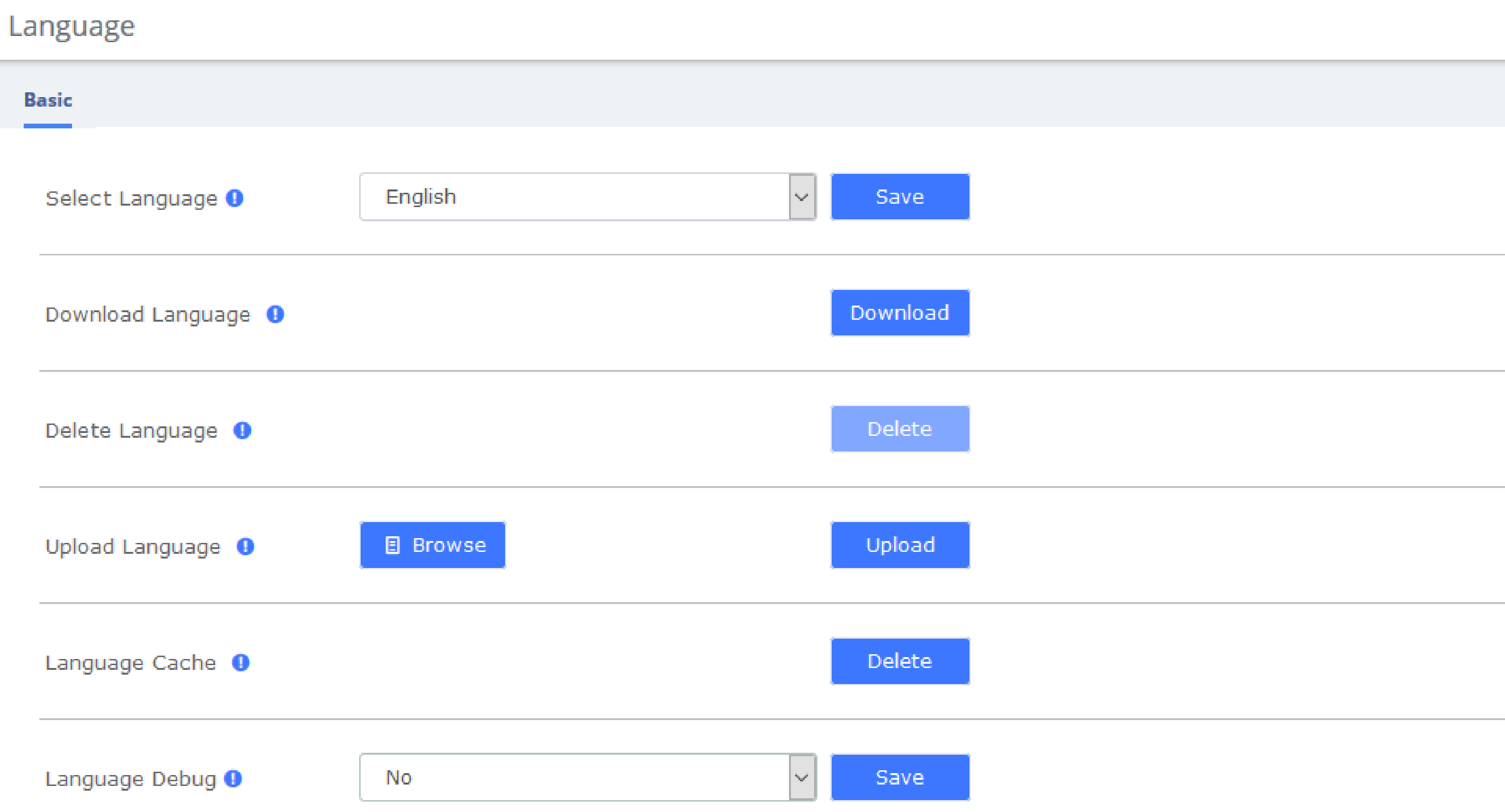
IP Ping and Traceroute interface

2.12 Preference

2.12.1 Language

Under the **Language** module in the **Preferences** menu, you can change the language of the SMB PBX system web interface. Select your desired language from the language list and click **Save**.

You can also download or upload languages you need.

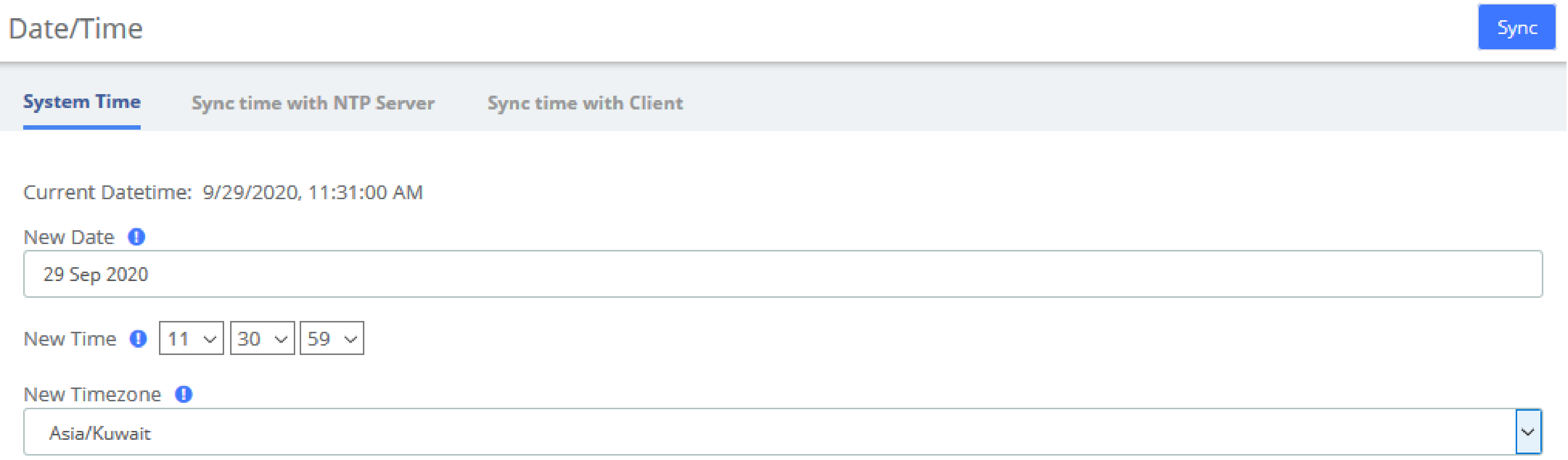


At the same time, the PBX system supports uploading language packs. You can click Download to download the current language pack, modify the language pack file based on it, then Upload and use the new language pack. Note that the language package is cached by default to ensure system smoothness.

When debugging a new language package, you can click Delete **Language Cache**, then select **Language Debug** Yes and save.

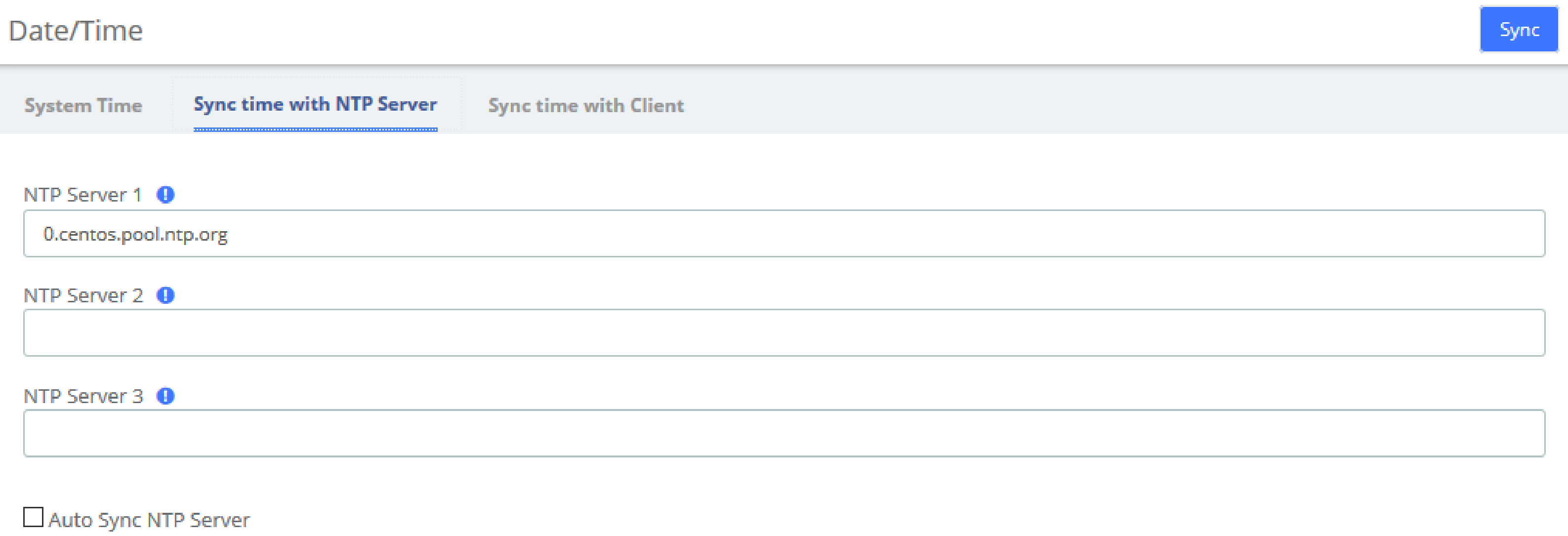
2.12.2 Date/Time

The option **Date/Time** of the Menu **Preferences** in PBX lets us configure the Date, Hour and Timezone for the SMB PBX Web Interface. Select the new date, hour and time zone and click on the **Apply changes** button.

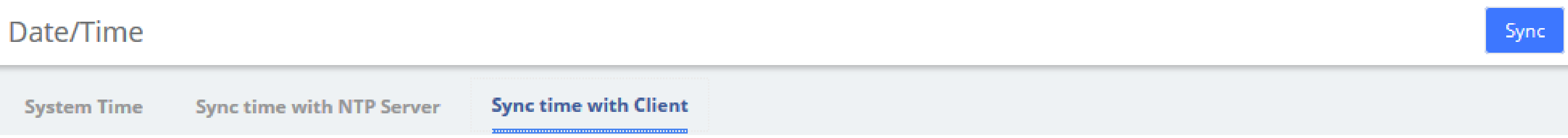


Date/Time Interface

Alternatively, system time can be synchronized automatically with the **NTP server/local client**.



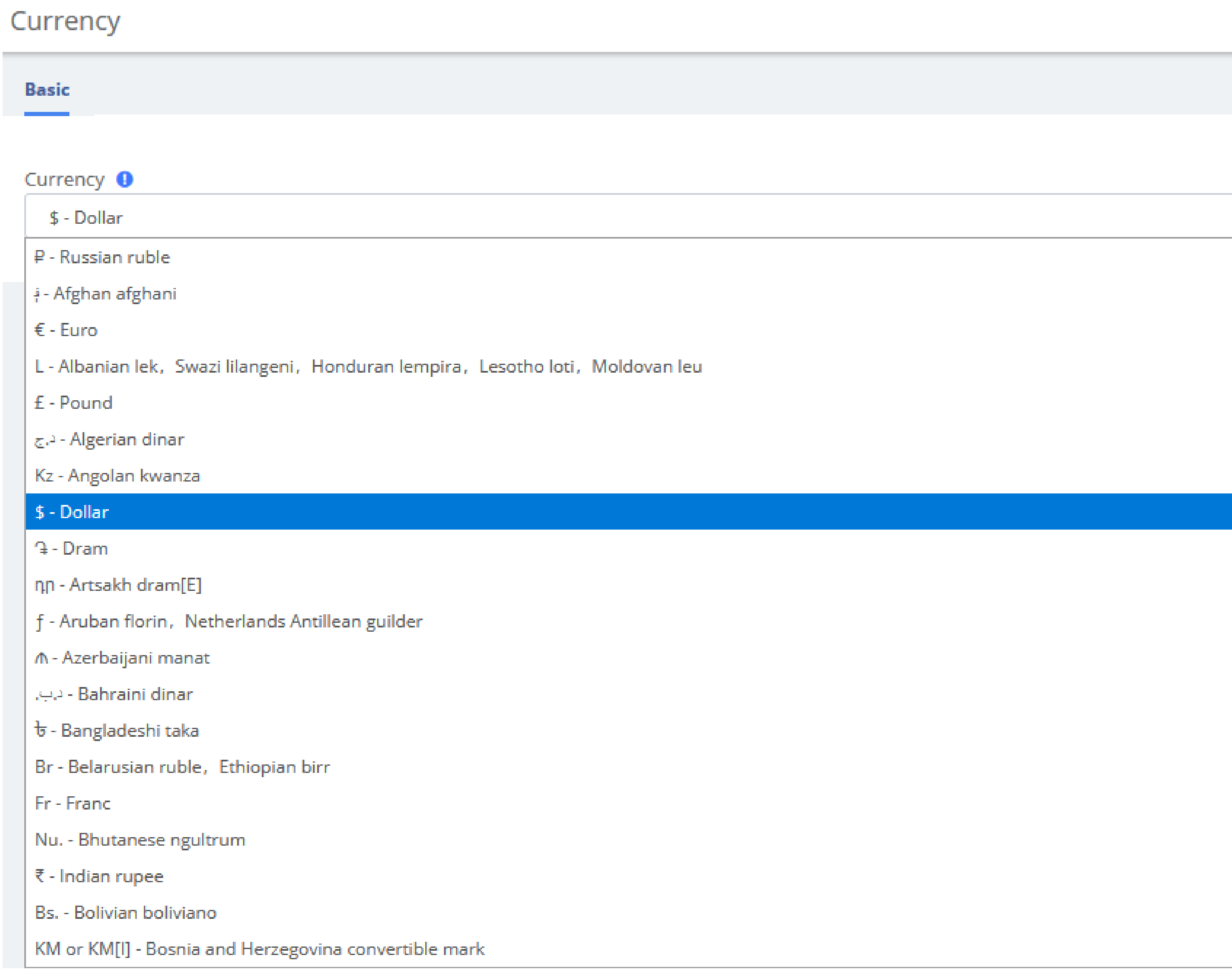
Sync time with NTP Server



Sync time with Client

2.12.3 Currency

Currency module of menu **Preferences** allows us change the currency for Reports in SMB PBX.

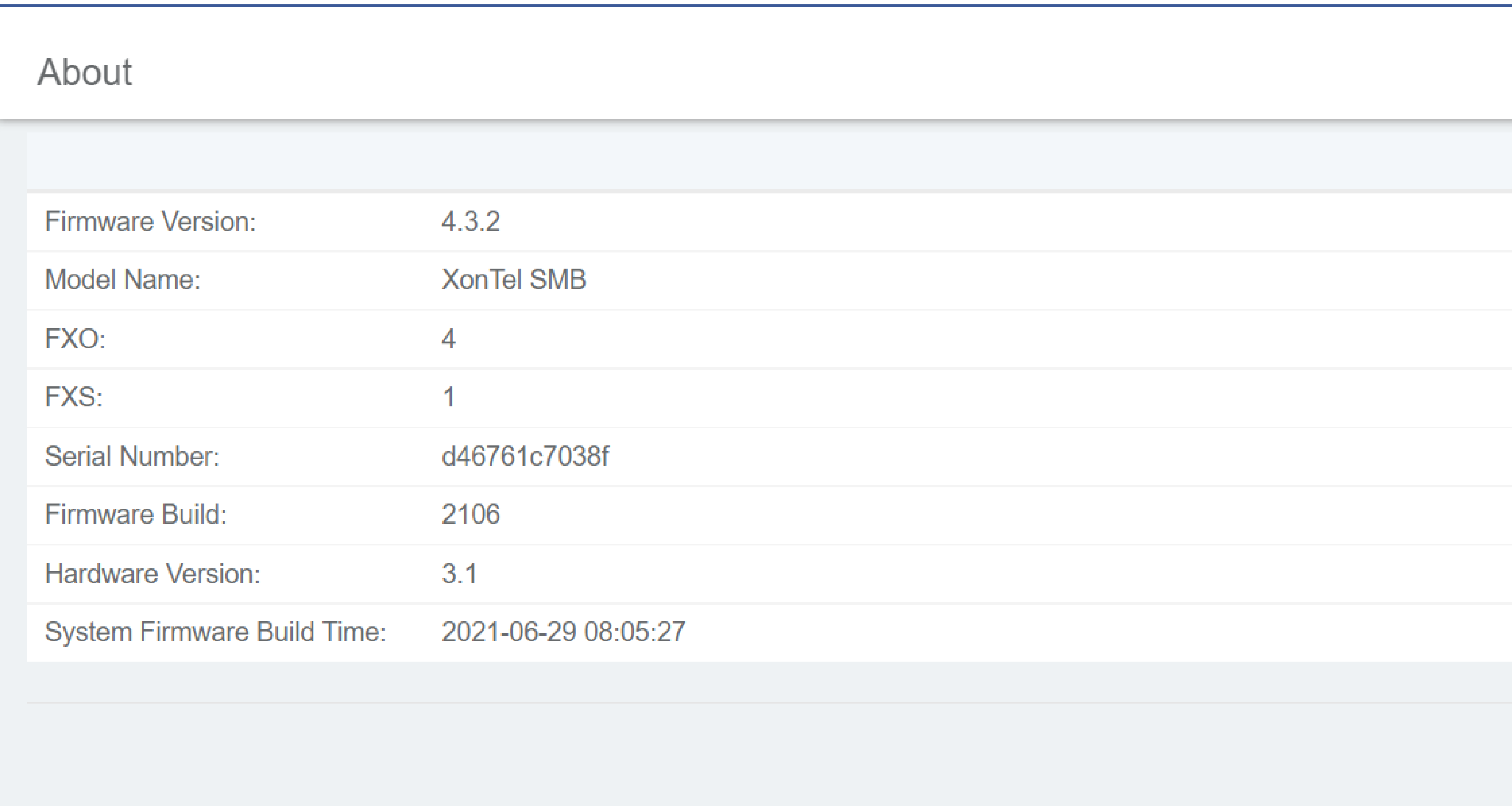


Currency Setting interface

Select a currency from the available options and click on the Save button.

2.12.4 About

Navigate to **System >About**, some basic information about the XonTel SMB PBX System is displayed, you can see the hardware version, model name, etc.



About

Firmware Version:	4.3.2
Model Name:	XonTel SMB
FXO:	4
FXS:	1
Serial Number:	d46761c7038f
Firmware Build:	2106
Hardware Version:	3.1
System Firmware Build Time:	2021-06-29 08:05:27

About information

3 PBX

The Menu **PBX** lets us configure extensions, trunks, routes, dial plan, queues, IVR and so on for SMB PBX.

In this menu, we can observe that we have different options for configuration.

3.1 Extensions

3.1.1 Extensions

The Extensions Module is used to set up each extension on your system. In the Extensions module, you will set up the extension number, the name of the extension, the password, voicemail settings for the extension, and other options.

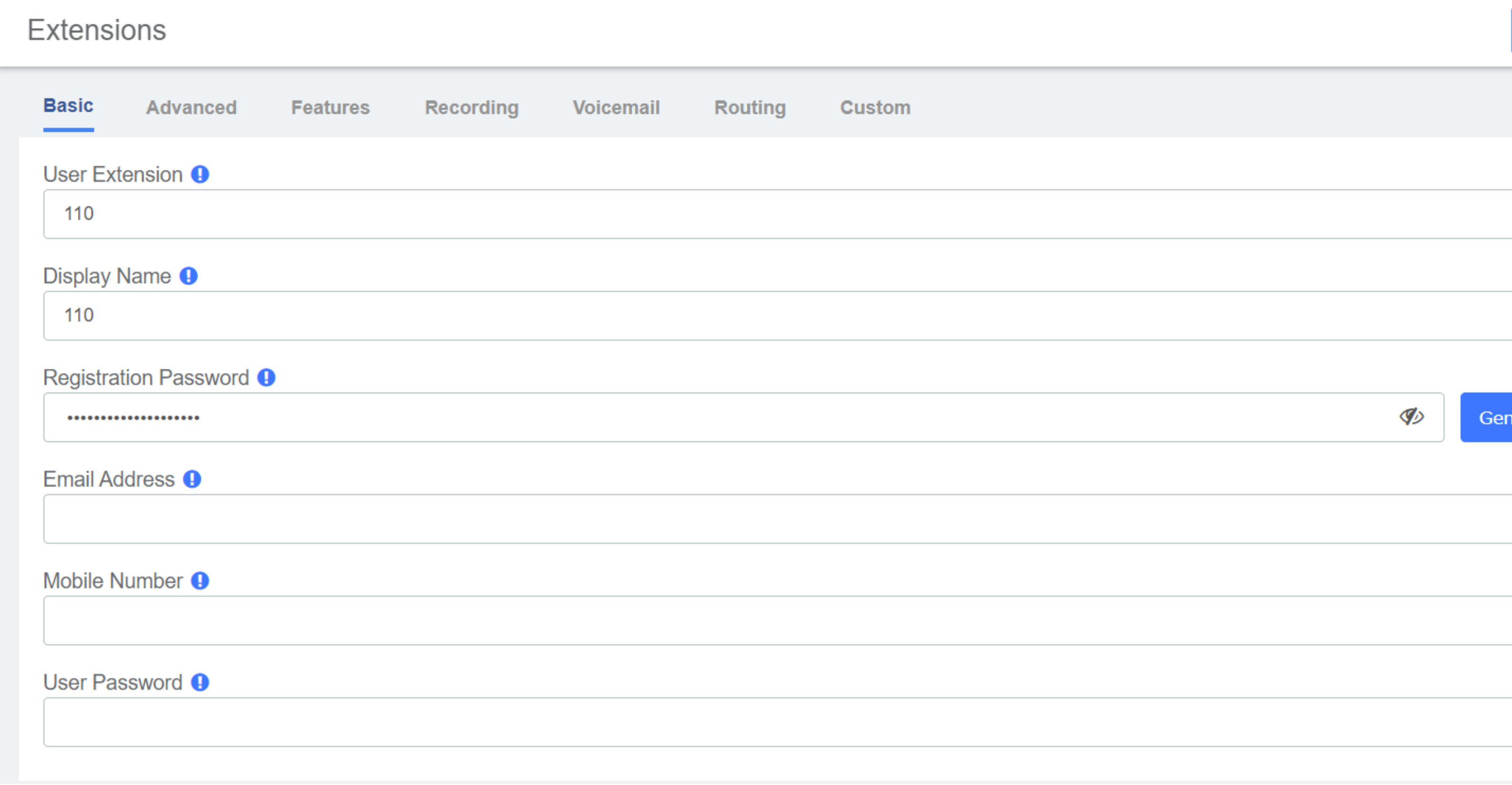
Normally, each physical phone will be assigned to one extension. If you have a phone that has more than one "line" button, you would normally make each line button register to the same extension number, and then use the line buttons to manage multiple calls to and from the same line. However, you could also create two or more extensions and assign each extension to a different line button.

Extensions

SIP Extension | + Add | + Add Bulk | Edit | Password | Export | Import | Delete

Name	Extension	Port	Type	Password
101	101	--	PJSIP	*******
102	102	--	PJSIP	*******
103	103	--	PJSIP	*******
104	104	--	PJSIP	*******
105	105	--	PJSIP	*******
106	106	--	PJSIP	*******
107	107	--	PJSIP	*******
108	108	--	PJSIP	*******
109	109	--	PJSIP	*******
110	110	--	PJSIP	*******
112	112	--	PJSIP	*******
200	200	Port 1	FXS	*******
201	201	Port 2	FXS	*******
Virtual_Fax_1301	1301	--	IAX2	*******

Click one of extensions number and edit it:



Item	**Description**
Basic	
User Extension	The extension number to dial to reach this user.
Display Name	The Caller ID name for calls from this user will be set to this name. only enter the name, not the number.
Registration Password	Password configured for the extension to register.
Email Address	The email can be used to email notification to the extension user.
Mobile Number	The extension contacts phone number.

User Password	Password configured for the extension to login web.
Advanced	
Account Code	Account code for the device
Max Contacts	Maximum number of endpoints that can associate with this device
Web Phone	Enable web phone will let user make and receive calls without installing any plugin in web browser.
Dtmfmode	The DTMF signaling mode used by this device, usually rfc2833 for most phone.
Audio Codecs	Codecs supported by the device, you can choose the codecs which you want.
Video Codecs	Video codecs supported by the device
Ring Timeout	Select the time in seconds.
Transport	This sets the allowed transport settings for this device and the default (Primary) transport for outgoing. The default transport is only used for outbound message until a registration takes place. During the peer registration the transport type may change to another supported type if the peer requests so. In most common cases, this does not have to be changed as most devices register in conjunction with the host=dynamic setting. If you are using TCP and/or TLS you need to make sure the general SIP Settings are configured for the system to operate in those modes and for TLS, proper certificates have been generated and configured. If you are using web sockets (such as Web RTC) then you must select an option that includes WS.
User Agent	When registering, SIP phones will be sending packets containing the user agent string. If the prefix of the user agent does not match the value defined here, the registration will fail.
Permitted IP/Subnet Mask	Permitted IP/Subnet Mask
Dictation Service	Allow the device to support dictation service.

Dictation Format	The format of dictation.
Language Code	Choose a different language for the user if he/she is not a native speaker than default system voice prompts.
CID Num Alias	The CID Number to use for internal calls, if different from the extension number. This is used to masquerade as a different user.
SIP Alias	If you want to support direct sip dialing of users internally or through anonymous sip calls, you can supply a friendly name that can be used in addition to the users extension to call them.
Features	
Outbound CID	Override the caller id when dialing out a trunk. Any setting here will override the common outbound caller id set in the trunk admin. Format: **"caller name"** <#######> Leave this filed blank to disable the outbound caller id feature for this user.
Asterisk Dial Options	Cryptic Asterisk Dial Options, check to customize for this extension or un-check to use system defaults set in Advanced Options. These will not apply to trunk options which are configured with the trunk.
Ring Time	Number of seconds to ring prior to going to voicemail. Default will use the value set in Advanced Settings. If no voicemail is configured this will be ignored.
Allow Being Monitored	Check this option to allow this user to be monitored.
Monitor Mode	Decide how you will monitor another extension.
Call Forward Ring Time	Number of seconds to ring during a Call Forward Busy or Call Forward Unavailable call prior to continuing to voicemail or specified destination. Setting to Always will not return, it will just continue to ring. Default will use the current Ring Time. If voicemail is disabled and there is not destination specified, it will be forced into Always mode.

Outbound Concurrency Limit	Maximum number of outbound simultaneous calls that an extension can make. This is also very useful as a Security Protection against a system that has been compromised. It will limit the number of simultaneous calls that can be made on the compromised extension.
Call Waiting	Set the initial/current Call Waiting state for this user's extension
Internal Auto Answer	When set to Intercom, calls to this extension/user from other internal users act as if they were intercom calls meaning they will be auto-answered if the endpoint supports this feature and the system is configured to operate in this mode. All the normal white list and black list settings will be honored if they are set. External calls will still ring as normal, as will certain other circumstances such as blind transfers and when a Follow Me is configured and enabled. If Disabled, the phone rings as a normal phone.
Call Screening	Call Screening requires external callers to say their name, which will be played back to the user and allow the user to accept or reject the call. Screening with memory only verifies a caller for their caller ID once. Screening without memory always required a caller to say their name. Either mode will always announce the caller based on the last introduction saved with that caller ID. If any user on the system uses the memory option, when that user is called, the caller will be required to re-introduce themselves and all users on the system will have that new introduction associated with the caller's Caller ID.
Pinless Dialing	Enabling Pinless Dialing will allow this extension to bypass any pin codes normally required on outbound calls.
Emergency CID	This caller id will always be set when dialing out an Outbound Route flagged ad Emergency. The Emergency CID overrides all other Caller ID settings.
Queue State Detection	If this extension is part of a Queue will attempt to use the user's extension state or device state information when determining if this queue member should be called. In some uncommon situations such as a Follow-Me with no physical device, or some virtual extension scenarios, the state information will indicate that this member is not available when they are. Setting this to 'Ignore-State' will make the Queue ignore all state information thus always trying to contact this member. Certain side effects can occur when this route is taken due to the nature of how Queues handle Local channels, such as subsequent transfers will continue to show the member as busy until the original call is terminated. In most cases, this SHOULD BE set to 'Use State'.
Recording	

On Demand Recording	Enable or disable the ability to do on demand (one-touch) recording. The overall calling policy rules still apply and if calls are already being recorded they cannot be paused.
Record Priority Policy	Call recording policy priority relative to other extensions when there is a conflict between an extension wanting recording and the other not wanting it. The higher of the two determines the policy, on a tie the global policy (caller or callee) determines the policy.
Voicemail	
Status	Enable or disable the voicemail function.
Voicemail Password	This is the password used to access the Voicemail system. This password can only contain numbers. A user can change the password you enter here after logging into the Voicemail system (*98) with a phone.
Pager Email Address	Page/mobile email address that short Voicemail notifications are sent to.
Email Attachment	Option to attach Voicemail to email.
Play CID	Read back caller's telephone number prior to playing the incoming message, and just after announcing the date and time the message was left.
Play Envelope	Envelope controls whether or not the Voicemail system will play the message envelope (date/time) before playing the voicemail message. This setting does not affect the operation of the envelope option in the advanced voicemail menu.
Delete Voicemail	If set to "yes" the message will be delete from the voicemail box (after having been emailed). Provides functionality that allows a user to receive their voicemail via email alone, rather than extension handset. CAUTION: must have attach voicemail to email set to yes otherwise your messages will be lost forever.
Send Voicemail	If set to 'yes', the voicemail will be sent by email.

VM Options	Separate options with pipe(|) Ie: review=yes|maxmessage=60
VM Context	This is the voicemail context which is normally set to default. Do not change unless you understand the implications.
Routing	
VmX Locater™	Enable/ disable the VmX locater feature for this user. When enabled all settings are controlled by the user in the user portal (ARI). Disabling will not delete any existing user settings but will disable access to the feature.
Use When	Menu options below are available during your personal voicemail greeting playback. Check both to use at all times.
Voicemail Instructions	Uncheck to play a deep after your personal voicemail greeting.
Press 0	Pressing 0 during your personal voicemail greeting goes to the operator. Uncheck to enter another destination here. This feature can be used while still disabling VmX to allow an alternative operator extension without requiring the VmX feature for the user.
Press 1	The remaining options can have internal extensions, ring groups, queues and external numbers that may be rung. It is often used to include your cell phone. You should run a test to make sure that the number is functional any time a change is made so you don't leave a caller stranded or receiving invalid number messages.
Press 2	Use any extensions, ring groups, queues or external numbers. Remember to re-record your personal voicemail greeting and include instructions. Run a test to make sure that the number is functional.
No Answer	Optional destination call is routed to when the call is not answered on an otherwise idle phone. If the phone is use and the call is simply ignored, then the busy destination will be used.
CID Prefix	Optional CID prefix to add before sending to this no answer destination.

Busy	Optional destination the call is route to when the phone is busy or the call is rejected the user. This destination is also used on an unanswered call if the phone is in use and the user choose not pickup the second call.
CID Prefix	Optional CID prefix to add before sending to this busy destination.
Not Reachable	Optional destination the call is routed when the phone is office, such as a softphone currently off or a phone unplugged.
CID Prefix	Optional CID prefix to add before sending to this not reachable destination.

The extension module allows you create extensions from a CSV file and download a CSV file with all the extensions that are currently configured in SMB PBX. This makes it easy the migration of data.

To download a CSV file with all the extensions created in SMB PBX, click on the Export button and save the file into your local hard drive.

To upload a CSV with the extensions you want to create, click on Import button, select the CSV file and click on "Upload CSV File" button.

Make sure the following indications are taken into account:

- Duplicated extensions are not allowed.
- The first line of the CSV file must contain the headers of the columns.
- The file must have at minimum four columns.
- This type of file can be created and opened with any text editor or spreadsheets such as Open Office Calc, Excel, etc.
- The separator of the columns is the comma.

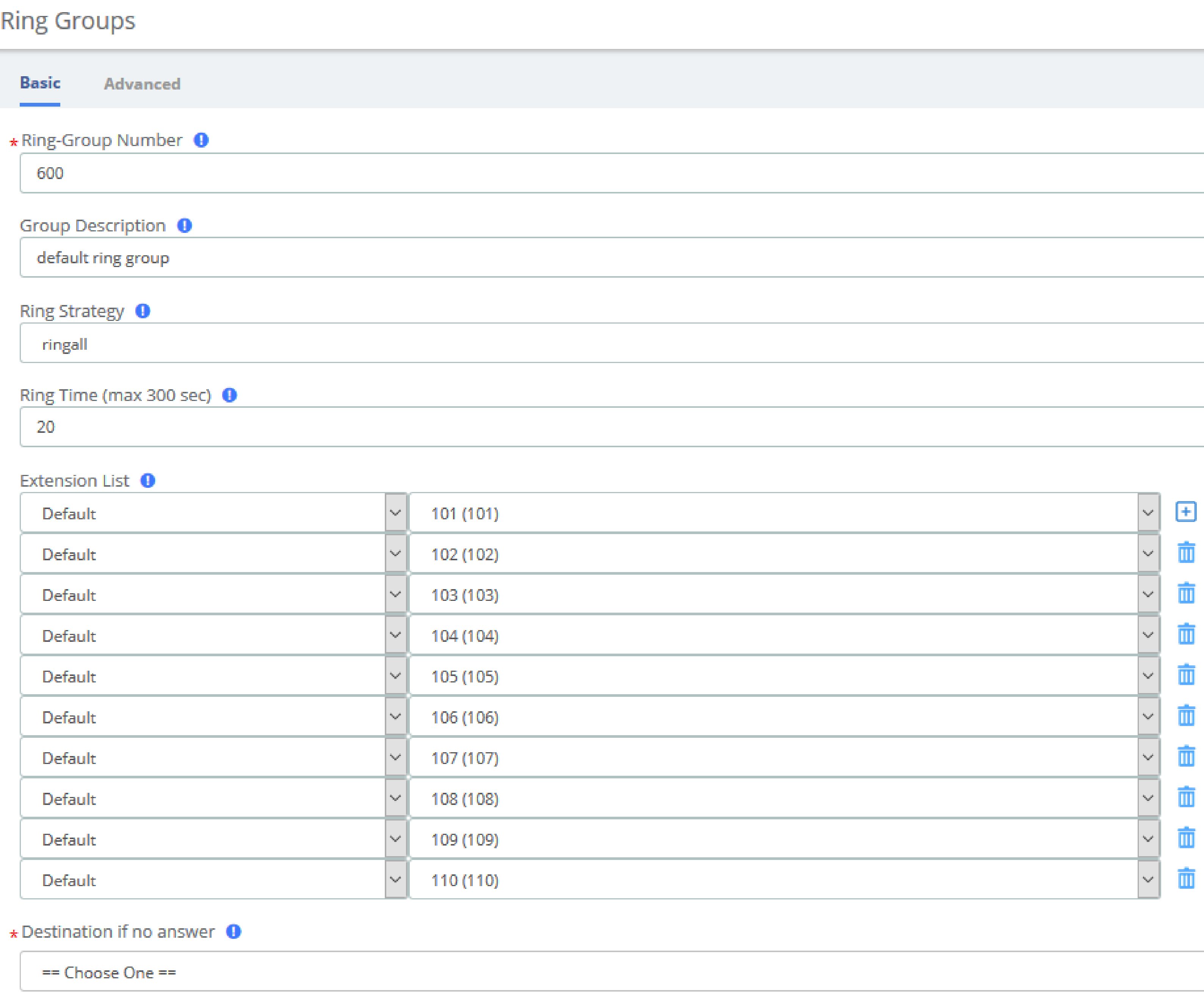
3.1.2 Ring Groups

A ring group is a group of extensions that will ring when there is an external incoming call. You can even put your Mobile Phone number in the ring group if you want to. For the mobile phone to work, you must have the appropriate route and trunk set up.

You may not want a ring group – it’s entirely up to you. If you don’t require a ring group, you may ignore this section.

When there is an incoming call to the ring group, the phones nominated in the selected group will ring. You may select different ring group for each of the incoming trunk or you may nominate the same group for all the trunks, in which case you will only need to define only one ring group.

The ring group screen is illustrated below:



Ring Groups

Basic | Advanced

Announcement

None

Play Music On Hold

Ring

CID Name Prefix

Alert Info

☐ Ignore CF Settings

☐ Enable Call Pickup

☐ Skip Busy Agent

☐ Confirm Calls

Remote Announce

Default

Too-Late Announce

Default

Mode

Default

Fixed CID Value

Item	Definition
Basic	
Ring-Group Number	The number users will dial to ring extensions in this ring group
Group Description	Provide a descriptive title for this Ring Group.
Ring Strategy	**Ringall :** Ring all available channels until one answers (default) **Hunt:** Take turns ringing each available extension **Memoryhunt:** Ring first extension in the list, then ring the 1 st and 2 nd extension, then ring 1st and 2nd and 3rd extension in the list…etc. ***-prim**: there mode act as described above. However, if the primary extension (first in list) is occupied, the other extensions will not be rung. If the primary is CF unconditional, then all will be rung **First available:** ring only the first available channel **Firstnotonphone:** ring only the first channel which is not off hook-ignored CW.
Ring Time (max 300 sec)	Time in seconds that the phones will ring. For all hunt style ring strategies, this is the time for each iteration of phone(s) that are rung.
Extension List	List extensions to ring, one per line, or use the Extension Quick Pick below to insert them here. You can include an extension on a remote system, or an external number by suffixing a number with a '#'. Ex:2448089# would dial 2448089 on the appropriate trunk (see outbound routing) Extension without a '#' will not ring a user's Follow-Me. To dial Follow-Me, Queues and other numbers that are not extensions, put a '#' at the end.
Destination if no answer	If there is no answer, the call will be sent to the destination.
Advanced	
Announcement	Message to be played to the caller before dialing this group. To add additional recordings please use the "System Recordings" MENU to the left.

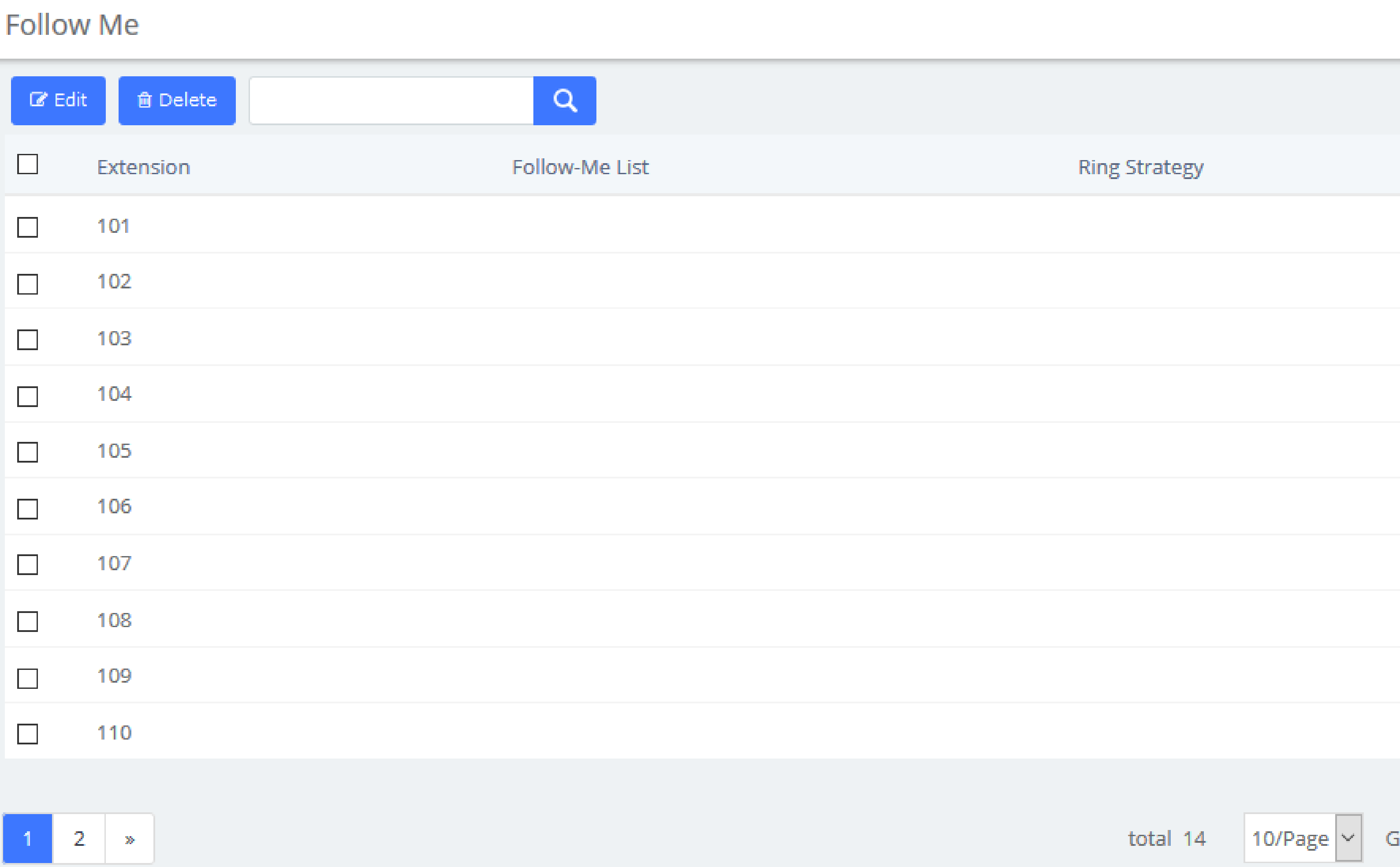
Play Music On Hold	If you select a music on hold class to play, instead of 'Ring', they will hear that instead of Ringing while they waiting for someone to pick up.
CID Name Prefix	You can optionally prefix the caller id name when ringing extensions in this group, ie: If you prefix with "Sales:", a call from John Doe would display as "Sales: John Doe" on the extensions that ring.
Alert Info	ALERT_INFO can be used for distinctive ring with SIP devices.
Ignore CF Settings	When checked, agents who attempt to Call Forward will be ignored, this applies to CF, CFU and CFB. Extensions entered with '#' at the end, for example to access the extension's Follow-Me, might not honor this setting.
Enable Call Pickup	Checking this will allow calls to the ring group to be picked up with the directed call pickup feature using the group number. When not checked, individual extensions that are part of the group can still be picked up by doing a directed call picked to the ringing extension, which works whether or not this is checked.
Skip Busy Agent	When checked, agents who are on an occupied phone will skipped as if the line were returning busy. This means that call waiting or multi-line phones will not be presented with the call and in the various hunt style ring strategies, the next agent will be attempted.
Confirm Calls	Enable this if you're calling external numbers that need confirmation-eg, a mobile phone may go to voicemail which will pick up the call. Enabling this requires the remote side push 1 on their phone before the call is put through. This feature only works with the ringall ring strategy.
Remote Announce	Message to be played to the person RECEIVING the call, if 'Confirm Calls' is enabled. To add additional recordings use the "System Recordings" MENU to the left
Too-Late Announce	Message to be played to the person RECEIVING the call, if the call has already been accepted before they push 1. To add additional recordings use the "System Recordings" MENU to the left
Mode	**Default**: Transmits the Callers CID if allowed by the trunk. **Fixed CID Value:** Always transmit the Fixed CID Value below. **Outside Calls Fixed CID Value**: Transmit the Fixed CID Value below on calls will continue to operate in default mode. **Use Dialed Number:** Transmit the number that was dialed as the CID for calls coming from outside. Internal extension to extension calls will continue to operate in default mode.

	There must be a DID on the inbound route for this. This will be BLOCKED on trunks that block foreign Caller ID **Force Dialed Number:** Transmit the number that was dialed as the CID for calls coming from outside. Internal extension to extension calls will be continue to operate in default mode. There must be a DID on the inbound route for this. This WILL be transmitted on trunks that block foreign Caller ID
Fixed CID Value	Fixed value to replace the CID with used with some of the modes above. Should be in a format of digits only with an option of E164 format using a leading "+".

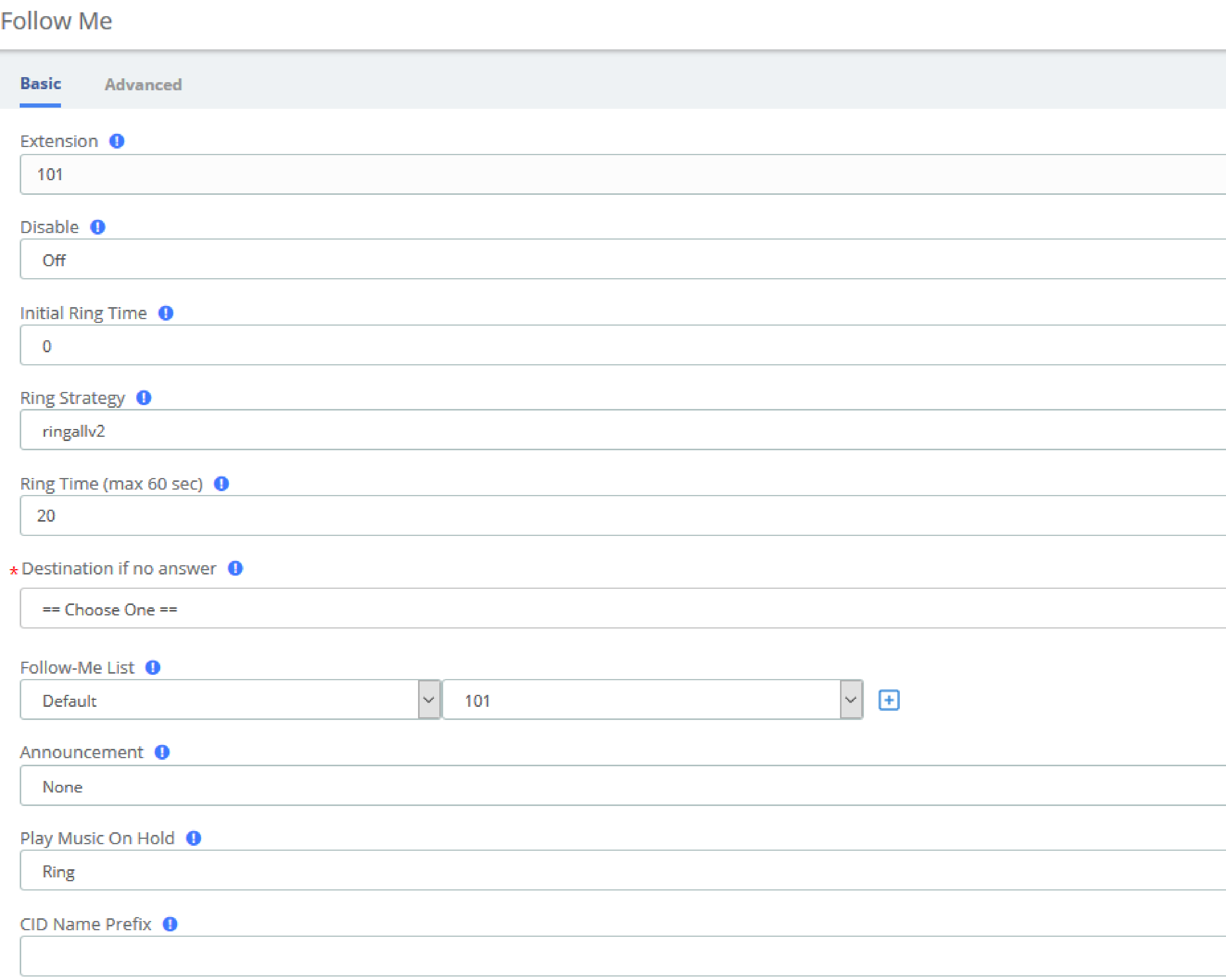
3.1.3 Follow Me

Follow Me (also known as **Find Me / Follow Me**) allows you to redirect a call that is placed to one of your extensions to another location. You can program the system to ring the extension alone for a certain period of time, then ring some other destination(s), such as a mobile phone or a related extension, and then go to the original extension's voicemail if the call is not answered. Follow Me can also be used to divert calls to another extension without ringing the primary extension.

Select the **PBX -> PBX Configuration -> Follow Me.**



Select the extensions that you want to define.



Item	**Definition**
	Basic
Extension	Edited extension
Disable	By default, (not checked) any call to this extension will go to this Follow-Me instead, including directory calls by name from IVRs. If checked, calls will go only to the extension.

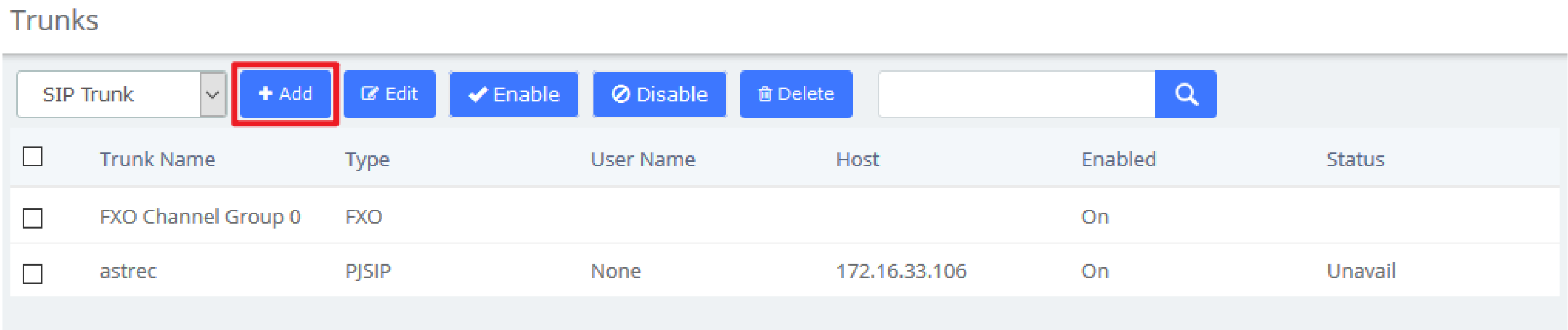
	However, destinations that specify FollowMe will come here. Checking this box is often used in conjunction with VmX Locater, where you want a call to ring the extension, and then only if the caller chooses to find you do you want it to come here.
Initial Ring Time	This is the number of seconds to ring the primary extension prior to proceeding to the follow-me list. The extension can also be included in the follow-me list. A 0 setting will bypass this
Ring Strategy	**Ringallv2**: ring Extension for duration set in Initial Ring Time, and then, while continuing call to extension, ring Follow-Me List for duration set in Ring Time. **Ringall:** ring Extension for duration set in Initial Ring Time, and then, terminate call to extension, ring Follow-Me List for duration set in Ring Time. **Hunt**: take turns ringing each available extension **Memoryhunt**: ring first extension in the list, then ring the 1st and 2nd extension, then ring 1st 2nd and 3rd extension in the list…. etc. ***-prim**: these mode act as described above. However, if the primary extension (first in the list) is occupied, the other extensions will not be rung. If the primary is DND, it won't be rung. If the primary is CF unconditional, then all will be rung **Firstavailable**: ring only the first available channel **Firstavailable**: ring only the first channel which is not off hook-ignore CW
Ring Time (max 60 sec)	Time in second that the phones will ring. For all hunt style ring strategies, this is the time for each iteration of phone(s) that are rung
Destination if no answer	Choose a destination when there is no answer.
Follow-Me List	List extensions to ring, one per line, or use the Extension Quick Pick below. You can include an extension on a remote system, or an external number by suffixing a number with a pound (#). Ex:2448089# would dial 2448089 on the appropriate trunk. see Outbound Routing.
Announcement	Message to be played to the caller before dialing this group. To add additional recordings please use the "System Recordings" MENU to the left.
Play Music On Hold	If you select a Music on Hold class to play, instead of 'Ring', they will hear that instead of Ringing while they are waiting for someone to pick up.
CID Name	You can optionally prefix the Caller ID name when ringing extensions in this group. Ie: if

Prefix	you prefix with “Sales:”, a call from John Doe would display as “Sales: John Doe” on the extensions that ring
Alert Info	You can optionally include an Alert Info which can create distinctive ring on SIP phones.
Advanced	
Confirm Calls	Enable this if you’re calling external numbers that need confirmation, eg, a mobile phone may go to voicemail which pick up the call. Enabling this require the remote side push 1 on their phone before the calls is put through. This feature only works with the ringall/ringall-prim ring strategy.
Remote Announce	Message to be played to the person RECEIVING the call, if ‘Confirm Calls” is enabled. To add additional recordings use the ‘System Recordings” MENU to the left
Too-Late Announce	Message to be played to the person RECEIVING the call, if the call has already been accepted before they push 1. To add additional recordings use the ‘System Recordings” MENU to the left
Mode	**Default**: Transmits the Caller CID if allowed by the trunk. **Fixed CID Value:** Always transmit the Fixed CID Value below. **Outside Calls Fixed CID Value**: Transmit the Fixed CID Value below on calls will continue to operate in default mode. **Use Dialed Number:** Transmit the number that was dialed as the CID for calls coming from outside. Internal extension to extension calls will continue to operate in default mode. There must be a DID on the inbound route for this. This will be BLOCKED on trunks that block foreign Caller ID **Force Dialed Number:** Transmit the number that was dialed as the CID for calls coming from outside. Internal extension to extension calls will be continue to operate in default mode. There must be a DID on the inbound route for this. This WILL be transmitted on trunks that block foreign Caller ID
Fixed CID Value	Fixed value to replace the CID with used with some of the modes above. Should be in a format of digits only with an option of E164 format using a leading “+”.

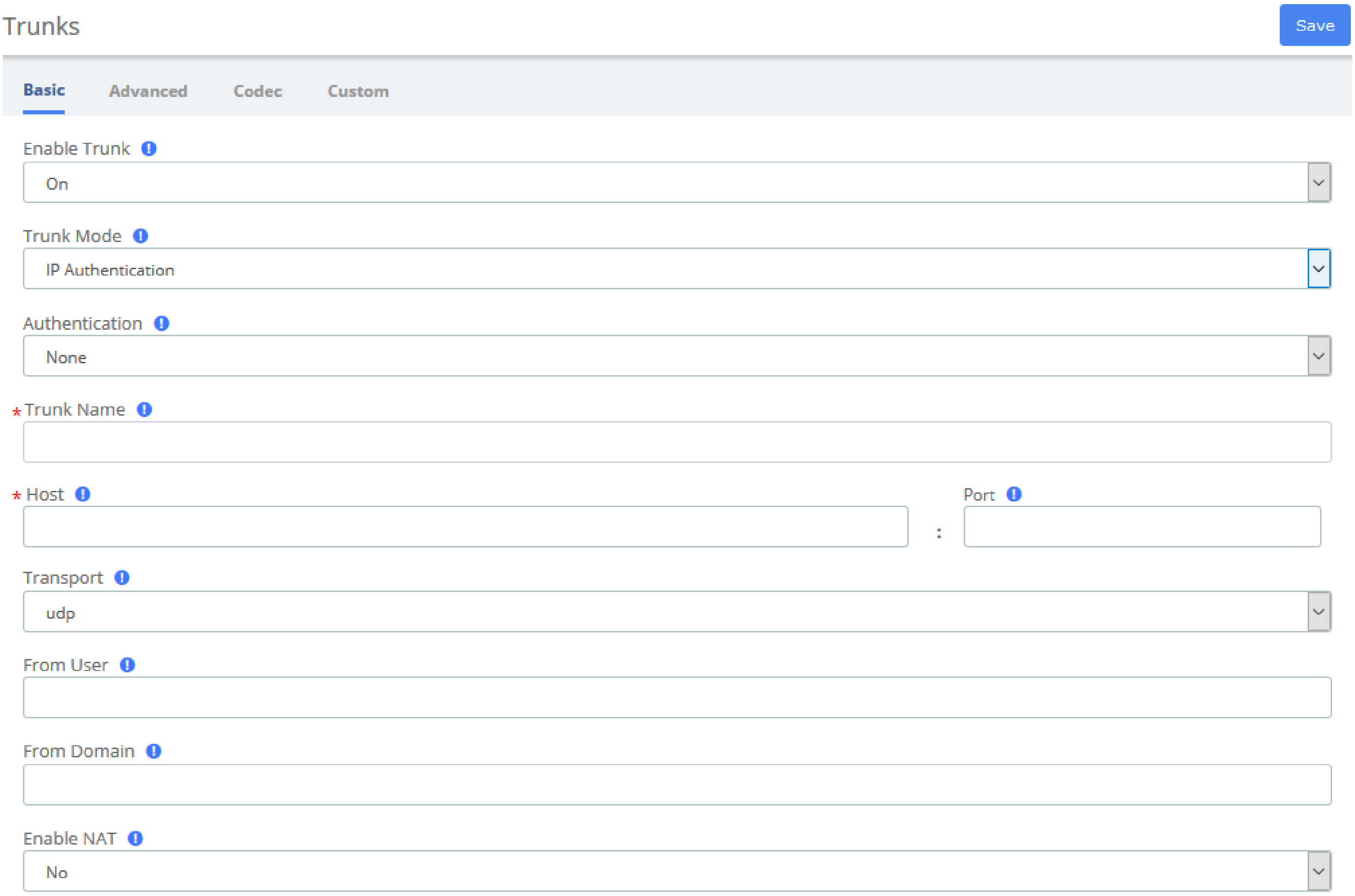
3.2 Trunks

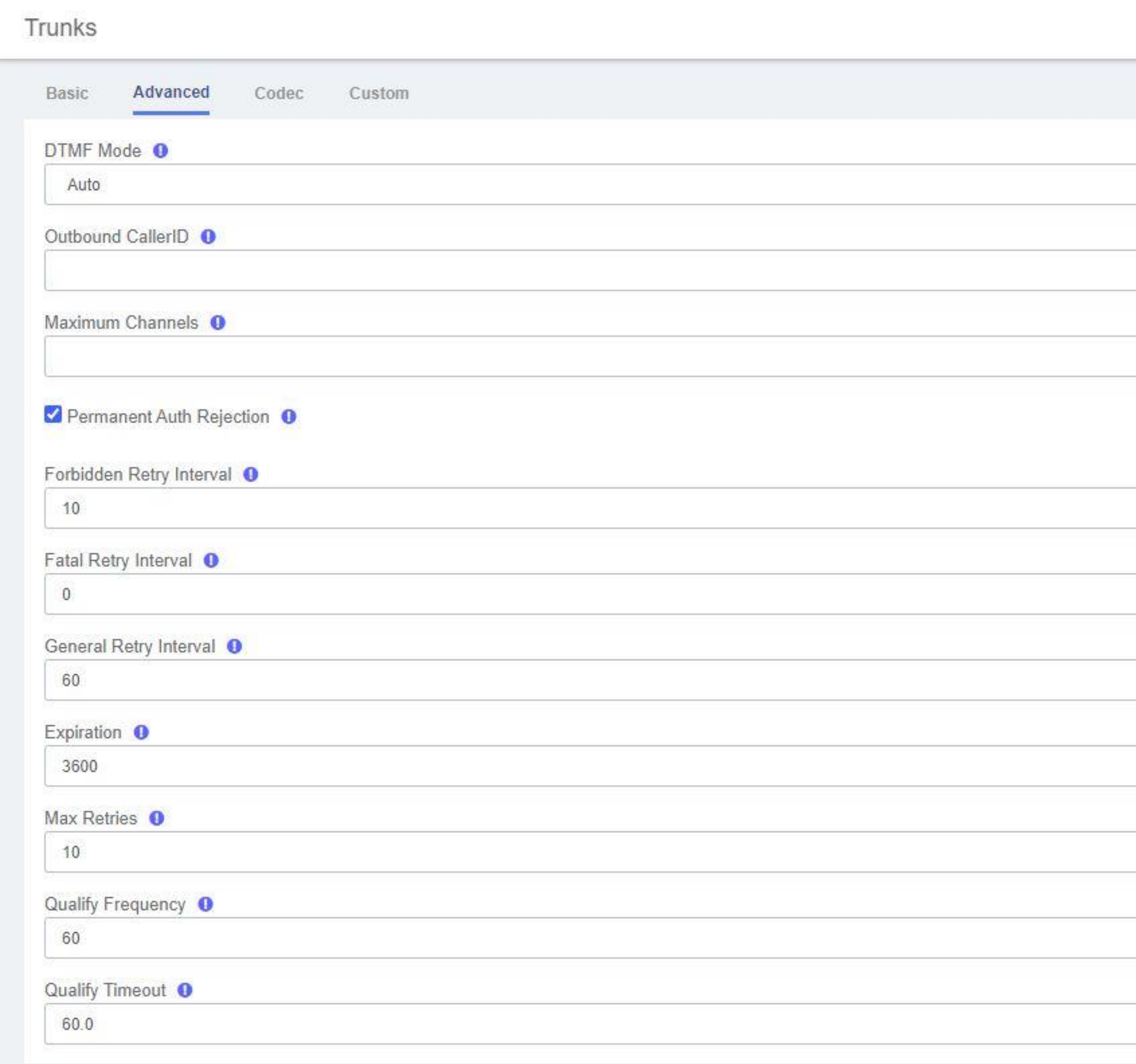
The **Trunks Module** is used to connect your Asterisk system to another VoIP system or VoIP device so that you can send calls out to and receive calls in from that system/device. You can create connections with Internet Telephone Service Providers ("ITSPs"), with other Asterisk systems, with commercial VOIP phone systems, with FXO Gateways (a device that connects an ordinary telephone line with a VoIP phone system using a network connection), and with FXO cards (cards that are installed in your computer and allow you to connect a standard telephone line).

If you don't have a Trunk set-up, you can still make calls, but only to other extensions on your same phone system.



A. Add SIP Trunk




Trunks
Basic
Advanced
Codec
Custom
DTMF Mode
Auto
Outbound CallerID
Maximum Channels
Permanent Auth Rejection
Forbidden Retry Interval
10
Fatal Retry Interval
0
General Retry Interval
60
Expiration
3600
Max Retries
10
Qualify Frequency
60
Qualify Timeout
60.0

Contact User

AOR Contact

Support Path

No

Support T.38 UDPTL

No

T.38 UDPTL Error Correction

None

T.38 UDPTL NAT

No

Fax Detect

No

Inband Progress

No

Direct Media Method

Invite

Trust Connected Line

Yes

Send Connected Line

Yes

Connected Line Method

Invite

Direct Media

No

RTP Symmetric

No

Rewrite Contact

No

Get CID From

Default

Get DID From

Default

Remote Party ID

None

P Asserted Identity

None

Diversion

None

P Preferred Identity

None

Asterisk Trunk Dial Options

* Context

from-pstn

Continue if Busy

On

Kuwait
Tel.: 1880005 Fax: 22413877
KSA
Tel.: 920007622 Fax: 011-4700403

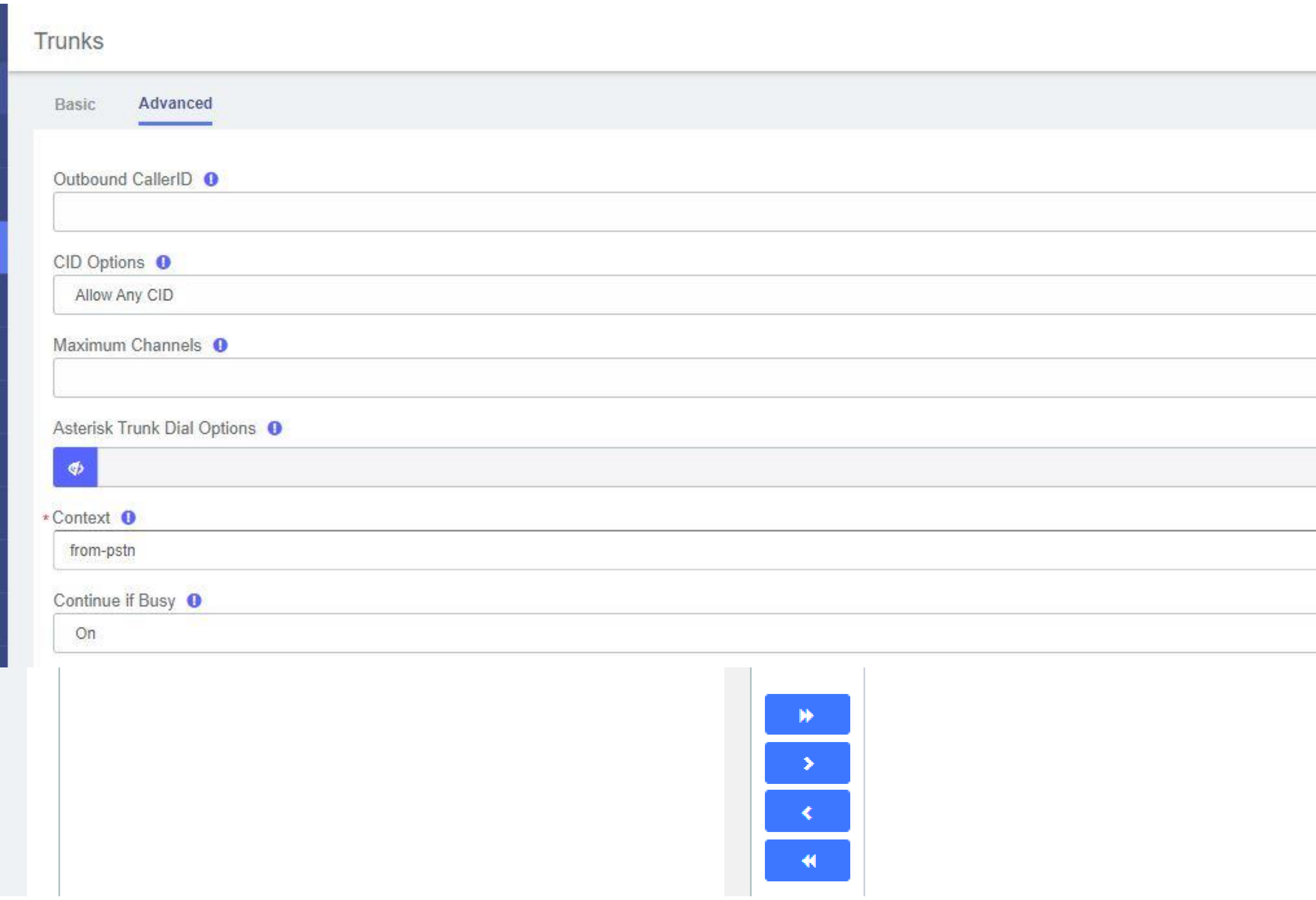
Item	Definition
Basic	
Enable Trunk	Check this to disable this trunk in all routes where it is used.
Trunk Mode	Authentication mode of this trunk.
Authentication	Usually, this will be set to "Outbound", which authenticates calls going out, and allows unauthenticated calls in from the other server. If you select "None", all calls from or to the specified SIP Server are unauthenticated.
Trunk Name	Descriptive Name for this trunk.
Host	Host settings for this device, almost always dynamic for endpoint.
Transport	Transports which the device supports.
From user	Rewrite the caller id
From Domain	Example: proxy.provider.domain
Enable NAT	Check this to enable or disable NAT
Codec	Allow specified codecs, the available codecs are on the left options bar and the selected on the right.
Advanced	
DTMF Mode	Types of DTMF.
Outbound CallerID	Caller ID for calls placed out on this trunk Format: <#######>. You can also use the format: “hidden” <#######> to hide the Caller ID sent out over Digital lines if supported (SIP/IAX).
Maximum Channels	Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. To count inbound calls against this maximum, use the auto-generated context: as the inbound trunk's context. (see extensions_additional.conf) Leave blank to specify no maximum.
Permanent Auth Rejection	Determines whether failed authentication challenges are treated as permanent.
Forbidden Retry	How long to wait before retry when receiving a 403 Forbidden response.

Interval	
Fatal Retry Interval	How long to wait before retry when receiving a fatal response.
General Retry Interval	The interval between two registered request packets.
Expiration	Expiration time for registrations in seconds.
Max Retries	The times asterisk will attempt to register before give up.
Qualify Frequency	Interval between two qualifies.
Qualify Timeout	Timeout of qualify
Contact User	Contact user to use in request.
AOR Contact	Permanent contacts assigned to AoR.
Support Path	When the button is enabled, registering request of outbound will advertise support for path header.
Support T.38 UDPTL	Allow the device to support T.38 UDPTL
T.38 UDPTL Error Correction	T.38 UDPTL error correction method
T.38 UDPTL NAT	Whether NAT support is enabled on UDPTL sessions
Fax Detect	When a CNG is detected, the session will be sent to the fax extension.
Inband Progress	Determine whether chan_sip indicates ringing using inbound progress.
Direct Media Method	Method for building direct media between endpoints.
Trust Connected Line	Accept Connected Line updates from this endpoint.
Send Connected Line	Send Connected Line updates to this endpoint
Connected Line Method	Method used when updating connected line information.
Direct Media	Determines whether media may flow directly between endpoints.
RTP Symmetric	Enforce that RTP must be symmetric.

Rewrite Contact	Allow contact header to be rewritten
Get CID From	Get CID from
Get DID From	Get DID from
Remote Party ID	This configuration requires professional knowledge of SIP protocol; as incorrect configuration may cause calling issues. Configure the value in the "Remote Party ID" header field of SIP INVITE packet which will be sent when making a direct call. • None: do not send this parameter with the SIP INVITE packet. • Default: the same as the value in "From" header field when making a direct call. • Trunk Username: the username you configured for the trunk. • Extension Number: the extension number (e.g. 1000). • From User: the "From User" value you configured for the trunk in the "Basic" settings
P Asserted Identify	This configuration requires professional knowledge of SIP protocol; as incorrect configuration may cause calling issues. Configure the value in the "Remote Party ID" header field of SIP INVITE packet which will be sent when making a direct call. • None: do not send this parameter with the SIP INVITE packet. • Default: the same as the value in "From" header field when making a direct call. • Trunk Username: the username you configured for the trunk. • Extension Number: the extension number (e.g. 1000). • From User: the "From User" value you configured for the trunk in the "Basic" settings
Diversion	This configuration requires professional knowledge of SIP protocol; as incorrect configuration may cause calling issues. Configure the value in the "Remote Party ID" header field of SIP INVITE packet which will be sent when making a direct call. • None: do not send this parameter with the SIP INVITE packet. • Default: the same as the value in "From" header field when making a direct call. • Trunk Username: the username you configured for the trunk. • Extension Number: the extension number (e.g. 1000). From User: the "From User" value you configured for the trunk in the "Basic" settings
P Preferred Identifiy	This configuration requires professional knowledge of SIP protocol; as incorrect configuration may cause calling issues. Configure the value in the "Remote Party ID" header field of SIP INVITE packet which will be sent when making a direct call. • None: do not send this parameter with the SIP INVITE packet.

	• Default: the same as the value in "From" header field when making a direct call. • Trunk Username: the username you configured for the trunk. • Extension Number: the extension number (e.g. 1000). From User: the "From User" value you configured for the trunk in the "Basic" settings
Asterisk Trunk Dial Options	Asterisk Dial command options to be used when calling out this trunk. To override the Advanced Settings default, check the box and then provide the required options for this trunk
Context	(Experts Only) Set the context that calls will originate from. Leaving this as from-internal unless you know what you're doing.
Continue if Busy	Normally the next trunk is only tried upon a trunk being 'Congested' in some form, or unavailable. Checking this box will force a failed call to always continue to the next configured trunk or destination even when the channel reports BUSY or INVALID NUMBER.

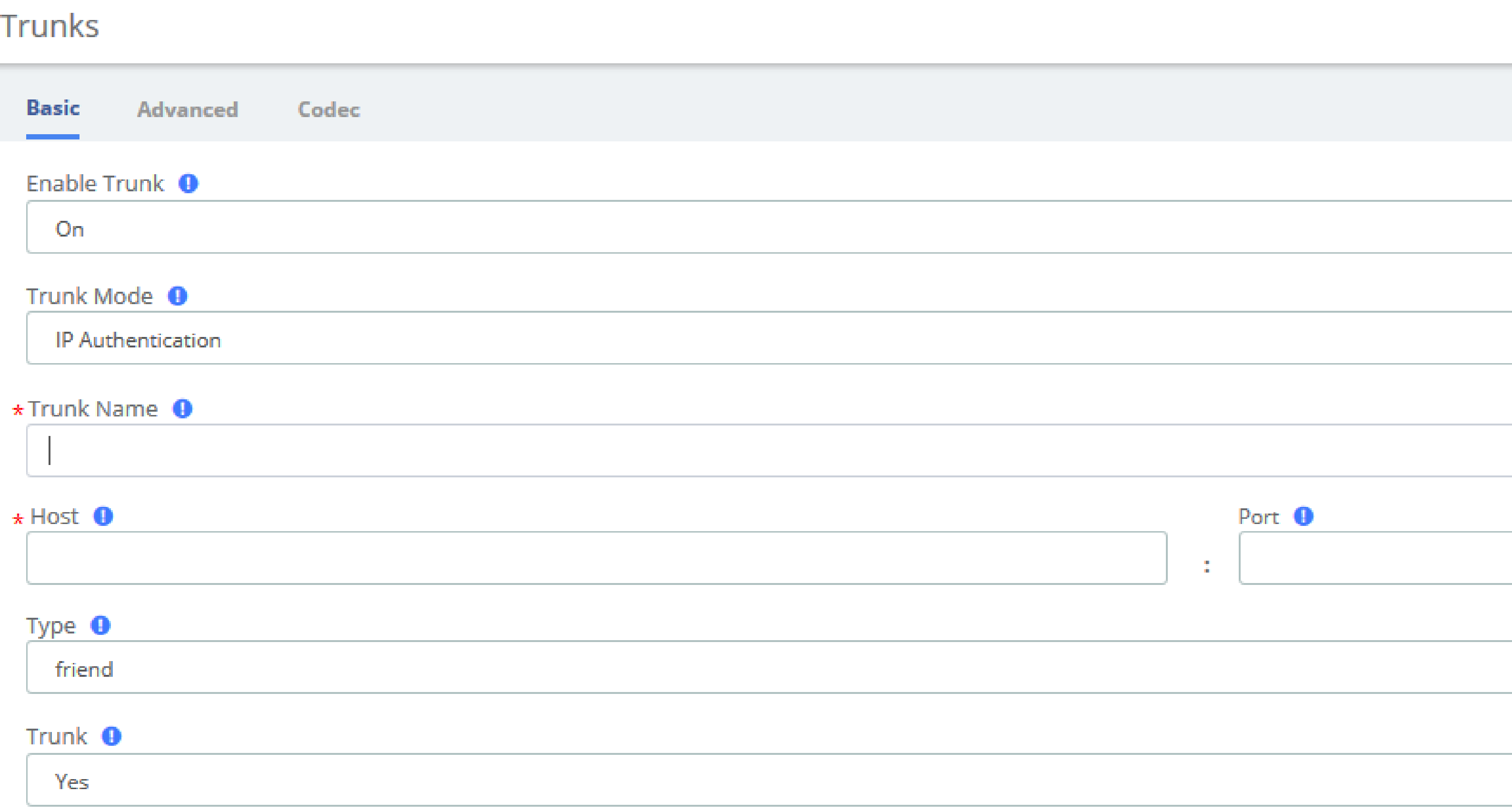
B. Add FXO Trunk



Item	Definition
Basic	
Enable Trunk	Check this to disable this trunk in all routes where it is used.
Trunk Name	Descriptive Name for this trunk.
Group ID	FXO channels are referenced either by a group number or channel number (which is defined in chan_dahdi.conf).
Policy	Used to make FXO trunks decisions, help determine the ringing order among multiple

	members of group
Member of Groups	Adding FXO ports into trunk groups allow automatic selection of the selected idle port for outgoing calls.
Advanced	
Outbound CallerID	Caller ID for calls placed out on this trunk Format: <#######>. You can also use the format: “hidden” <#######> to hide the Caller ID sent out over Digital lines if supported (SIP/IAX).
CID Options	Determines what CIDs will be allowed out this trunk. IMPORTANT: EMERGENCY CIDs defined on an extension/device will ALWAYS be used if this trunk is part of an EMERGENCY Route regardless of these settings. **Allow Any CID:** all CIDs including foreign CIDS from forwarded external calls will be transmitted. **Block Foreign CIDs:** blocks any CID that is the result of a forwarded call from off the system. CIDs defined for extensions/users are transmitted. **Remove CNAM:** this will remove CNAM from any CID sent out this trunk **Force Trunk CID:** Always use the CID defined for this trunk except if part of any EMERGENCY Route with an EMERGENCY CID defined for the extension/device. Intra-Company Routes will always transmit an extension’s internal number and name.
Maximum Channels	Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. Inbound calls are not counted against the maximum. Leave blank to specify no maximum.
Asterisk Trunk Dial Options	Asterisk Dial command options to be used when calling out this trunk. To override the Advanced Settings default, check the box and then provide the required options for this trunk
Context	(Experts Only) Set the context that calls will originate from. Leaving this as from-internal unless you know what you’re doing.
Continue if Busy	Normally the next trunk is only tried upon a trunk being ‘Congested’ in some form, or unavailable. Checking this box will force a failed call to always continue to the next configured trunk or destination even when the channel reports BUSY or INVALID NUMBER.

C. Add IAX Trunk



Trunks

Basic | Advanced | Codec

Outbound CallerID

CID Options

Allow Any CID

Maximum Channels

Outbound Dial Prefix

Qualifyfreq OK

60000

Qualifyfreq Not OK

10000

Qualify

Yes

Asterisk Trunk Dial Options

* Context

from-pstn

Continue if Busy

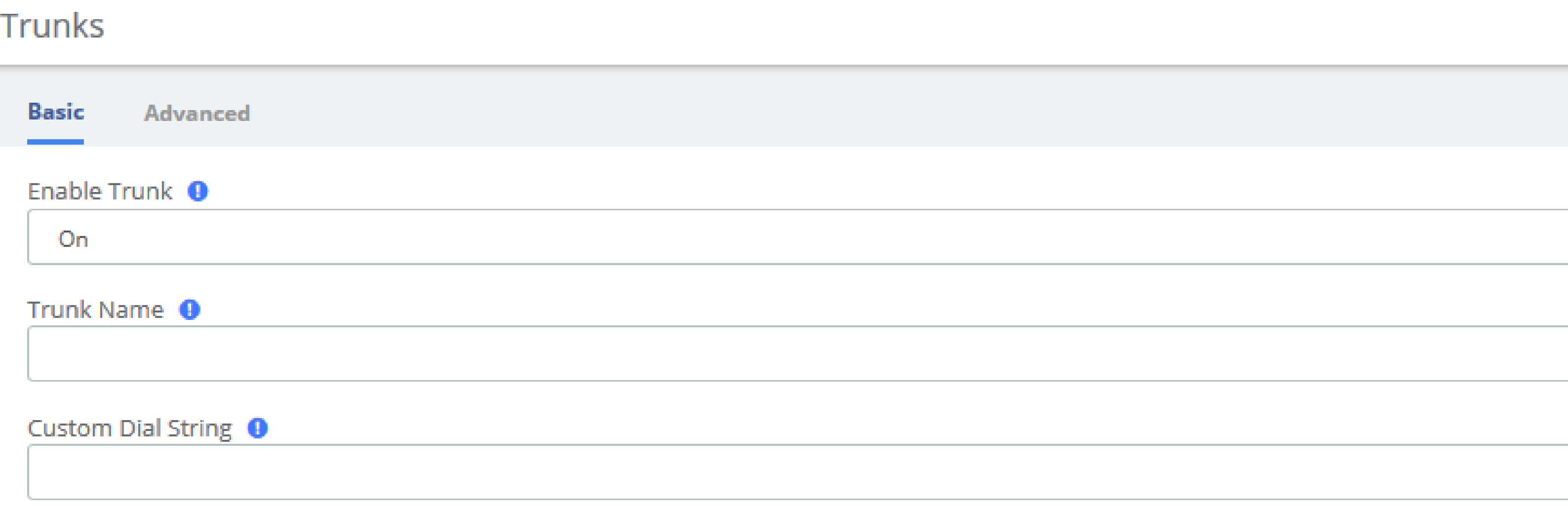
On

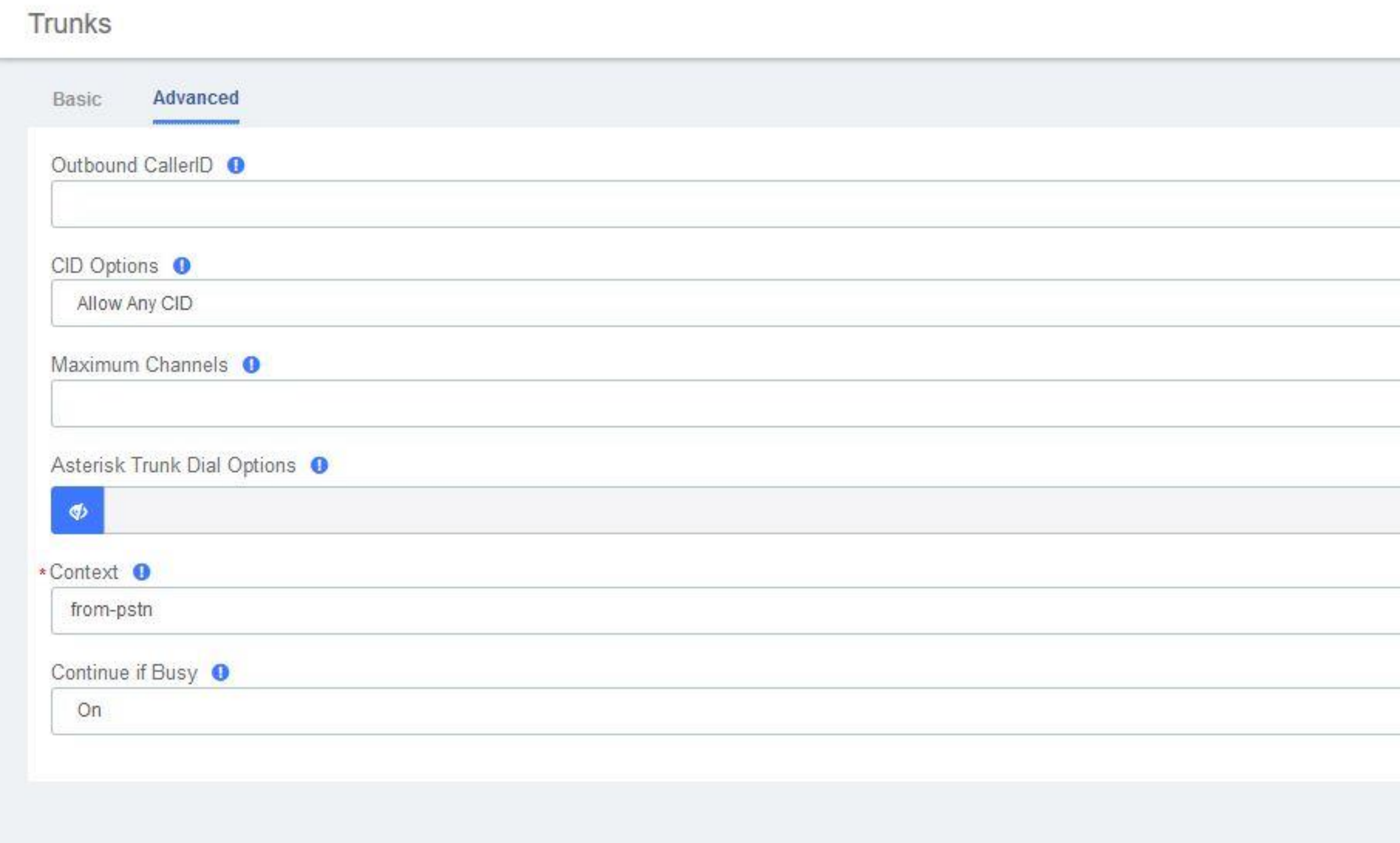
XonTel
Kuwait
Tel.: 1880005 Fax: 22413877
KSA
Tel.: 920007622 Fax: 011-4700403

Item	Definition
Basic	
Enable Trunk	Check this to disable this trunk in all routes where it is used.
Trunk Mode	Authentication mode of this trunk.
Trunk Name	Descriptive Name for this trunk
Host	Host settings for this device, almost always dynamic for endpoint.
Type	Asterisk connection type. There are three type you can choose, friend, peer and user.usually friend for endpoint.
Trunk	Use IAX2 trunk with this host.
Advanced	
Outbound CallerID	Caller ID for calls placed out on this trunk Format: <#######>. You can also use the format: “hidden” <#######> to hide the Caller ID sent out over Digital lines if supported (SIP/IAX).
CID Options	Determines what CIDs will be allowed out this trunk. IMPORTANT: EMERGENCY CIDs defined on an extension/device will ALWAYS be used if this trunk is part of an EMERGENCY Route regardless of these settings. **Allow Any CID:** all CIDs including foreign CIDS from forwarded external calls will be transmitted. **Block Foreign CIDs:** blocks any CID that is the result of a forwarded call from off the system. CIDs defined for extensions/users are transmitted. **Remove CNAM:** this will remove CNAM from any CID sent out this trunk **Force Trunk CID:** Always use the CID defined for this trunk except if part of any EMERGENCY Route with an EMERGENCY CID defined for the extension/device. Intra-Company Routes will always transmit an extension’s internal number and name.
Maximum Channels	Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. To count inbound calls against this maximum, use auto-generated context: from-trunk-[trunkname] as the inbound trunk’s context. (see extesions_additional.conf) Leave blank to specify no maximum.
Outbound Dial	The outbound dialing prefix is used to prefix a dialing string to all outbound calls

Prefix	placed on this trunk. For example, if this trunk is behind another PBX or is a Centrex line, then you would put 9 here to access an outbound line. Another common use is to prefix calls with 'w' on a POTS line that need time to obtain dial tone to avoid eating digits. Most users should leave this option blank.
Qualifyfreq OK	Frequency in milliseconds to send qualify messages to the endpoint.
Qualifyfreq Not OK	Frequency in milliseconds to send qualify messages to the endpoint.
Qualify	Setting to yes (equivalent to 2000 msec) will send an OPTIONS packet to the endpoint periodically (default every minute). Used to monitor the health of the endpoint. If delays are longer then the quality time, the endpoint will be taken offline and considered unreachable. Can be set to a value which is the msec threshold. Setting to no will turn this off. Can also be helpful to keep NAT pinholes open.
Asterisk Trunk Dial Options	Asterisk Dial command options to be used when calling out this trunk. To override the Advanced Settings default, check the box and then provide the required options for this trunk.
Context	(Experts Only) Set the context that calls will originate from. Leaving this as from-internal unless you know what you're doing.
Continue if Busy	Normally the next trunk is only tried upon a trunk being 'Congested' in some form, or unavailable. Checking this box will force a failed call to always continue to the next configured trunk or destination even when the channel reports BUSY or INVALID NUMBER.
Codec	
Audio Codecs	You can choose specific audio codecs here
Video Codecs	You can choose specific video codecs here

D. **Add Custom Trunk**





Item	Definition
Basic	
Enable Trunk	Check this to disable this trunk in all routes where it is used.
Trunk Name	Descriptive Name for this trunk
Custom Dial String	Define the custom Dial String. Include the token $OUTNUM$ wherever the number to dial should go. examples: CAPI/XXXXXXXX/$OUTNUM$ H323/$OUTNUM$@XX.XX.XX.XX OH323/$OUTNUM$@XX.XX.XX.XX:XXXX vpb/1-1/$OUTNUM$
Advanced	
Outbound CallerID	Caller ID for calls placed out on this trunk Format: <#######>. You can also use the format: “hidden” <#######> to hide the Caller ID sent out over Digital lines if supported (SIP/IAX).
CID Options	Determines what CIDs will be allowed out this trunk. IMPORTANT: EMERGENCY CIDs defined on an extension/device will ALWAYS be used if this trunk is part of an EMERGENCY Route regardless of these settings. **Allow Any CID:** all CIDs including foreign CIDS from forwarded external calls will be transmitted. **Block Foreign CIDs:** blocks any CID that is the result of a forwarded call from off the system. CIDs defined for extensions/users are transmitted. **Remove CNAM:** this will remove CNAM from any CID sent out this trunk **Force Trunk CID:** Always use the CID defined for this trunk except if part of any EMERGENCY Route with an EMERGENCY CID defined for the extension/device. Intra-Company Routes will always transmit an extension’s internal number and name.
Maximum Channels	Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. Inbound calls are not counted against the maximum.

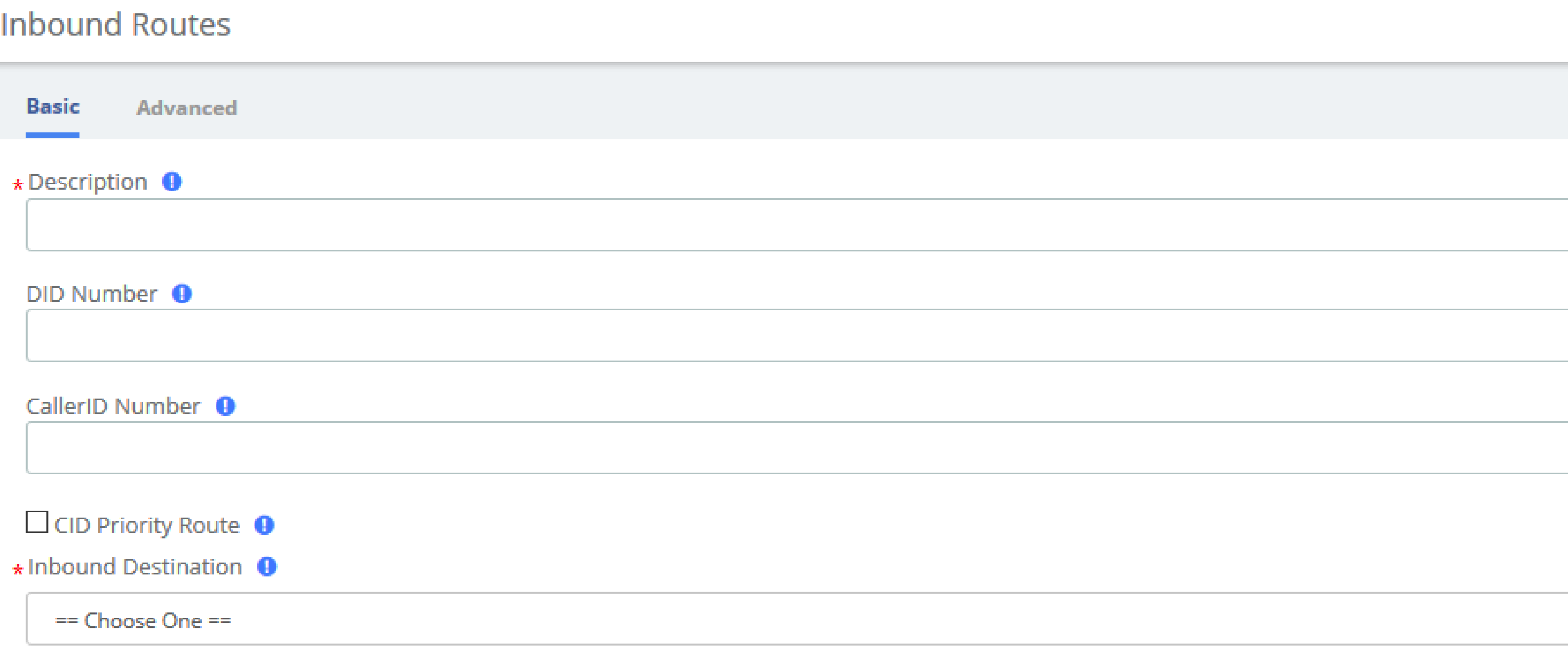
	Leave blank to specify no maximum.
Asterisk Trunk Dial Options	Asterisk Dial command options to be used when calling out this trunk. To override the Advanced Settings default, check the box and then provide the required options for this trunk
Context	(Experts Only) Set the context that calls will originate from. Leaving this as from-internal unless you know what you're doing.
Continue if Busy	Normally the next trunk is only tried upon a trunk being 'Congested' in some form, or unavailable. Checking this box will force a failed call to always continue to the next configured trunk or destination even when the channel reports BUSY or INVALID NUMBER.

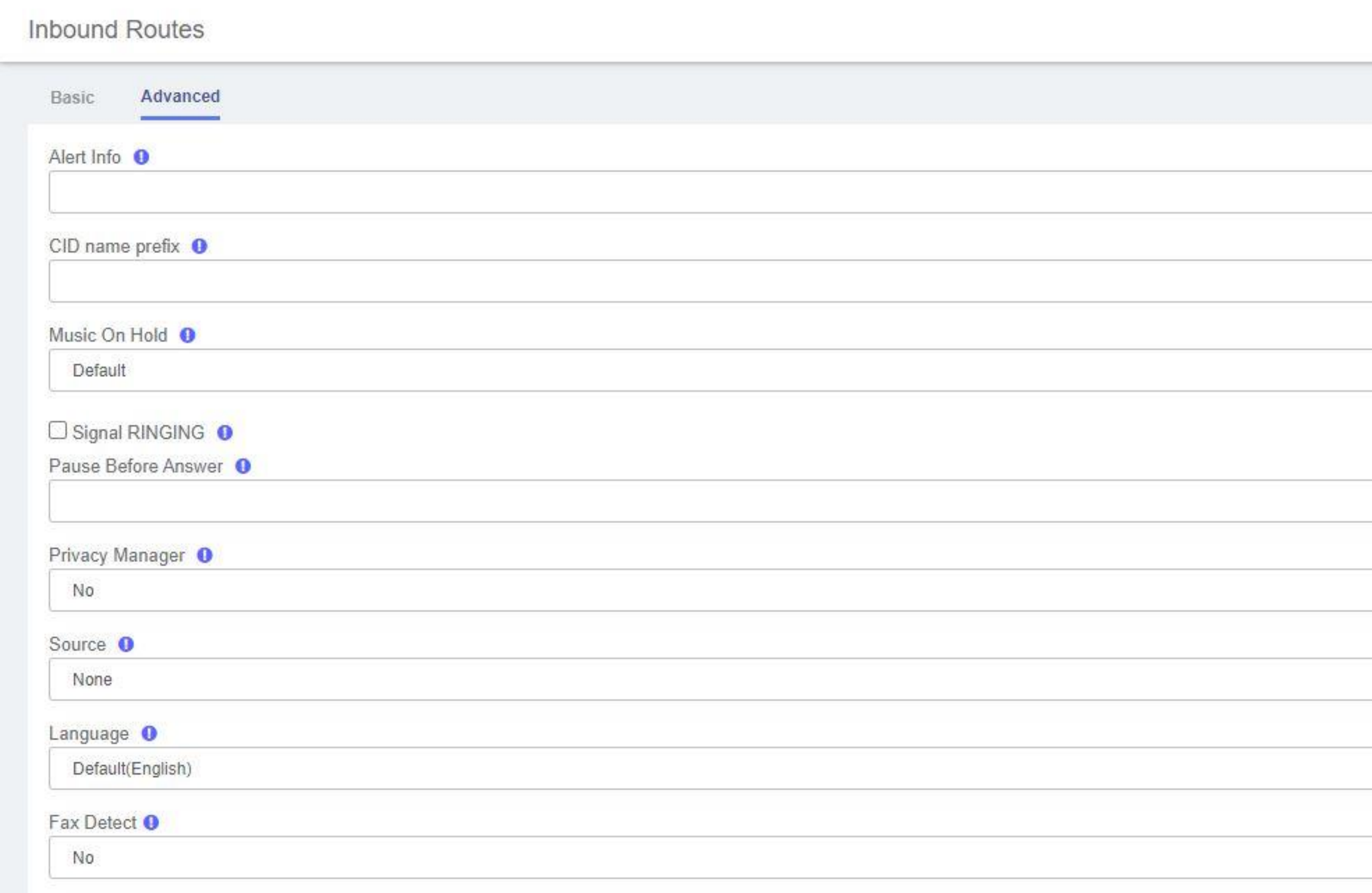
3.3 Call Control

3.3.1 Inbound Routes

When a call comes into your system from the outside, it will usually arrive along with information about the telephone number that was dialed (also known as the "DID") and the Caller ID of the person who called.

The Inbound Routes module is used to tell your system what to do with calls that come into your system on any trunk that has the "context=from-trunk" parameter in the PEER details.





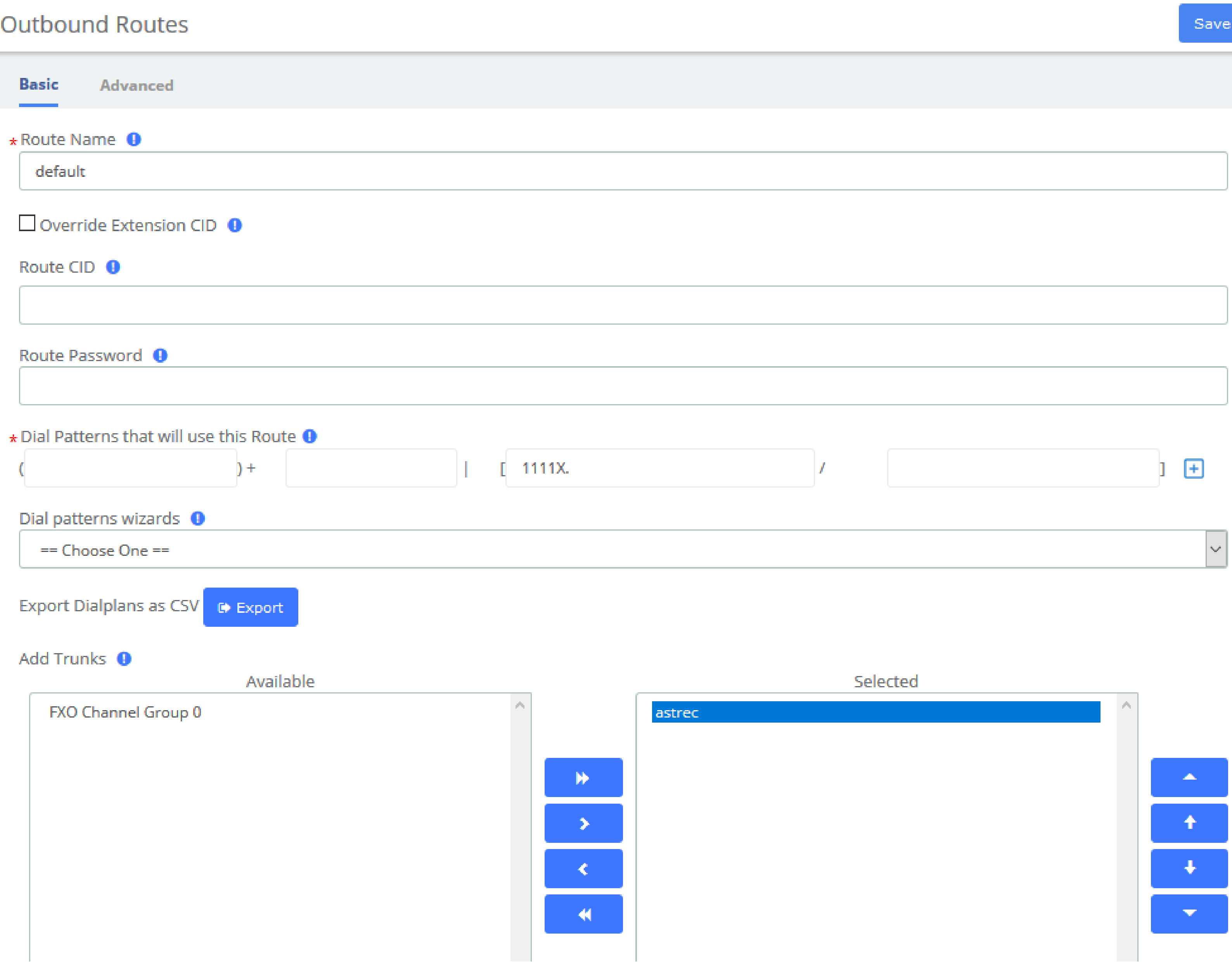
Item	Definition
Basic	
Description	Provide a meaningful description of what this incoming route is
DID Number	Define the expected DID Number if your trunk passes DID on incoming calls. Leaving this blank to match calls with any or no DID info. You can also use a pattern match (eg_2[345]X) to match a range of numbers.
CallerID Number	Define the Caller ID Number to be matched on incoming calls. Leave this field blank to match any or no CID info. In addition to standard dial sequences, you can also put Private, Blocked, Unknown, Restricted, Anonymous and Unavailable in order to catch these special cases if the Telco transmits them.

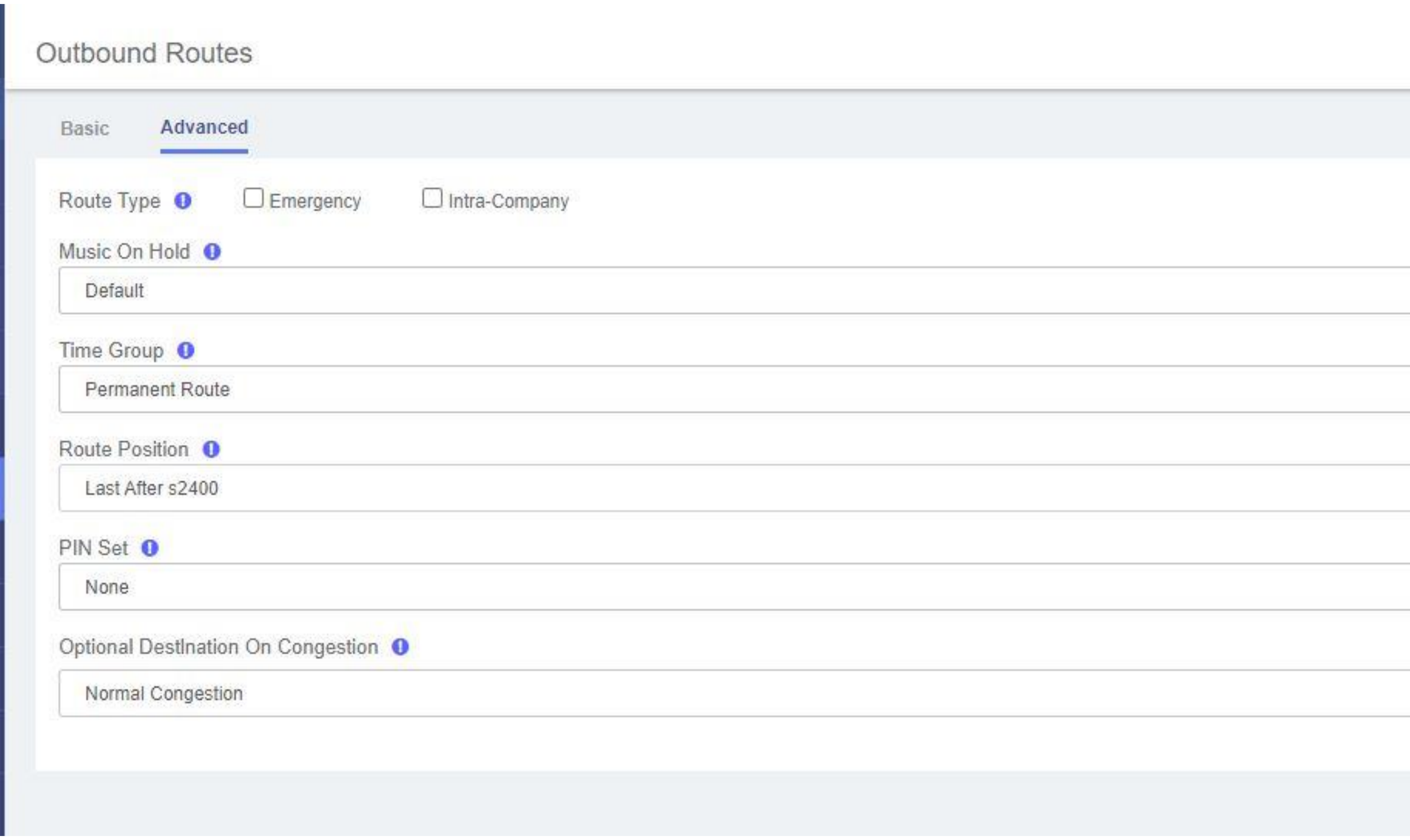
CID Priority Route	This effects CID ONLY routes where no DID is specified. If checked, calls with this CID will routed to this route, even if there is a route to the DID that was called. Normal behavior is for the DID route to take the calls. If there is a specific DID/CID route for this CID, that route will still take the call when that DID is called.
Inbound Destination	Indicates extension, Ring Group, Voicemail or other destination to which the call is supposed to be directed when the outside callers have called specified DID Number
Advanced	
Alert Info	ALERT_INFO can be used for distinctive ring with SIP devices.
CID name prefix	You can optionally prefix the Caller ID name. ie: If you prefix with “Sales:”, a call from john Doe would display as “Sales: John Doe” on the extension that ring
Music On Hold	Set the MoH class that will be used for calls that come in on this route. For example, choose a type appropriate for routes coming in from a country which may have announcements in their language.
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.
Pause Before Answer	An optional delay to wait before processing this route. Setting this value will delay the channel from answering the call. This may be handy if external fax equipment or security systems are installed in parallel and you would like them to be able to seize the line.
Privacy Manager	If no Caller ID has been received, Privacy Manager will ask the caller to enter their phone number. If an user/extension has Call Screening enabled, the incoming caller will be prompted to say their name when the call reaches the user/extension.
Source	Source can be added in Caller Name Lookup Sources section.
Language	Allows you to set the language for this DID.
Fax Detect	Attempt to detect faxes on this DID. • No: No attempts are made to auto-determine the call type; all calls sent to destination below. Use this option if this DID is used exclusively for voice OR fax. • Yes: try to auto determine the type of call; route to the fax destination if call is a fax, otherwise send to regular destination. Use this option if you receive both voice and fax calls on this line.

3.3.2 Outbound Routes

The Outbound Routes Module is used to tell your Asterisk system which numbers your phones are permitted to call and which Trunk to send the calls to.

Generally, an Asterisk system will have a Restricted route which designates certain numbers that can never be dialed (such as 900 and 976 numbers), an Emergency route to use for routing 110 calls, and a route for ordinary calls. A phone system might also have special routes for interoffice calls, international calls, and other special circumstances





Item	Definition
Basic	
Route Name	Name of this route. Should be used to describe what type of calls this route matches (for example, 'local' or 'long distance').
Route CID	Optional Route CID to be used for this route. If set, this will override all CIDS specified except: ● extension/device EMERGENCY CIDs if this route is checked as an EMERGENCY Route ● trunk CID if trunk is set to force it's CID ● Forwarded call CIDs (CF, Follow Me, Ring Groups, etc.) ● Extension/User CIDs if checked
Route	Optional: A route can prompt users for a password before allowing calls to progress. This is

Password	useful for restricting calls to international destinations or 1-900 numbers. A numerical password, or the path to an Authenticate password file can be used. Leave this field blank to not prompt for password.
Dial Patterns that will use this Route	A Dial Pattern is a unique set of digits that will select this route and send the call to the designated trunks. If a dialed pattern matches this route, no subsequent routes will be tried. If Time Groups are enabled, subsequent routes will be checked for matches outside of the designated time(s). **Rules:** **X** matches any digit from 0-9 **Z** matches any digit from 1-9 **N** matches any digit from 2-9 **[1237-9]** matches any digit in the brackets (example: 1,2,3,7,8,9). wildcard, matches one or more dialed digits **Prepend:** Digits to prepend to a successful match. If the dialed number matches the patterns specified by the subsequent columns, then this will be prepended before sending to the trunks. **Prefix:** Prefix to remove on a successful match. The dialed number is compared to this and the subsequent columns for a match. Upon a match, this prefix is removed from the dialed number before sending it to the trunks. **Match pattern:** The dialed number will be compared against the prefix + this match pattern. Upon a match, the match pattern portion of the dialed number will be sent to the trunks. **CallerID:** If Caller ID is supplied, the dialed number will only match the prefix + match pattern if the Caller ID being transmitted matches this. When extensions make outbound calls, the Caller ID will be their extension number and not their Outbound CID. The above special matching sequences can be used for Caller ID matching similar to other number matches.
Dial patterns wizards	These options provide a quick way to add outbound dialing rules. Follow the prompts for each. **Lookup local prefixes** This looks up your local number on ww.localcallingguide.com (NA-only), and sets up so you can dial either 7, 10 or 11 digits (5551234, 6135551234, 16135551234) to access this route. **Upload from CSV** Upload patterns from a CSV file replacing existing entries. If there are no headers, then the file must have 4 columns of patterns in the same order as in the GUI. You can also supply headers: **prepend, prefix, match pattern** and **caller id** in the first

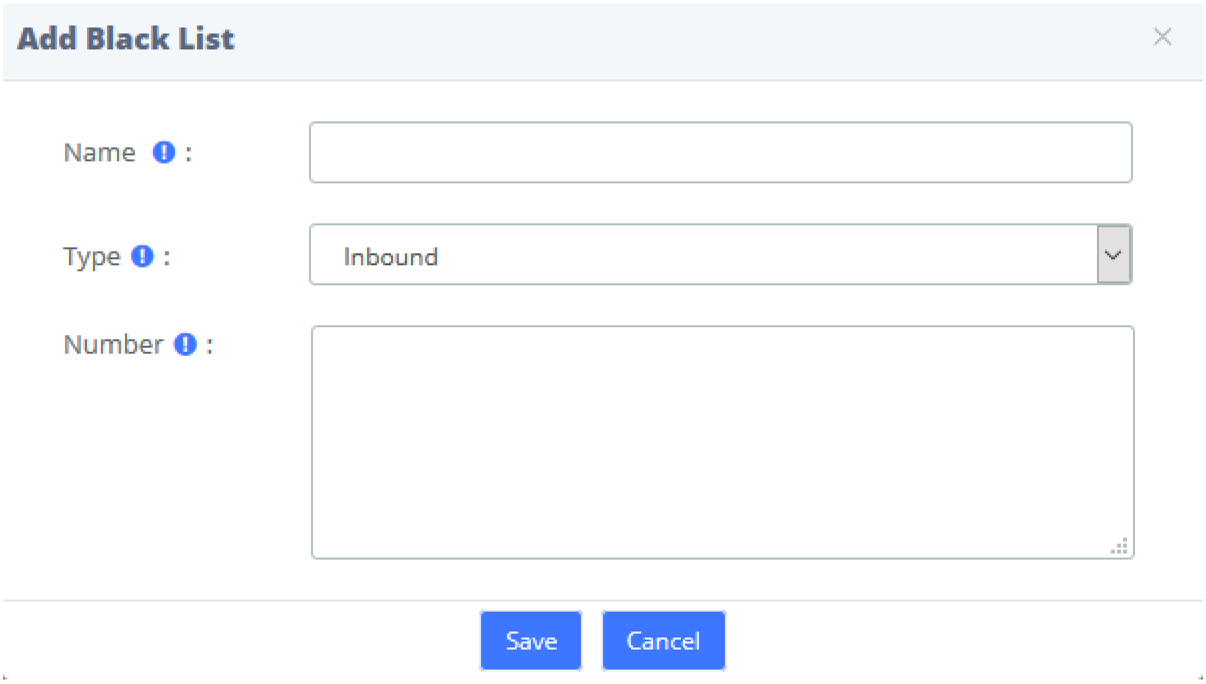
	row. If there are less than 4 recognized headers then the remaining columns will be blank.
Add Trunks	Trunks used by this outbound route, the available trunks are on the left options bar and the selected on the right.
Advanced	
Route Type	Optional: Selecting Emergency will enforce the use of a device
Music On Hold	You can choose which music category to use. For example, choose a type appropriate for a destination country which may have announcements in the appropriate language.
Time Group	If this route should only be available during certain times, then Select a Time Group created under Time Groups. The route will be ignored outside of times specified in that Time Group. If left as default of Permanent Route then it will always be available.
Route Position	Where to insert this route or relocate it relative to the other routes.
PIN Set	Optional: Select a PIN set to use. If using this option, leave the Route Password field blank.
Optional Destination on Congestion	If all the trunks fail because of Asterisk 'CONGESTION' dial status you can optionally go to a destination such as a unique recorded message or anywhere else. This destination will not be engaged if the trunk is reporting busy, invalid numbers or anything else that would imply the trunk was able to make an 'intelligent' choice about the number that was dialed. The 'Normal Congestion' behavior is to play the 'ALL Circuits Busy' recording or other options configured in the route Congestion Messages module when installed.

3.3.3 Call Restrictions

Black List

The blacklist module is used to add a phone number to a blacklist or remove a phone number from a blacklist. You can also choose to blacklist any blocked or unknown calls.

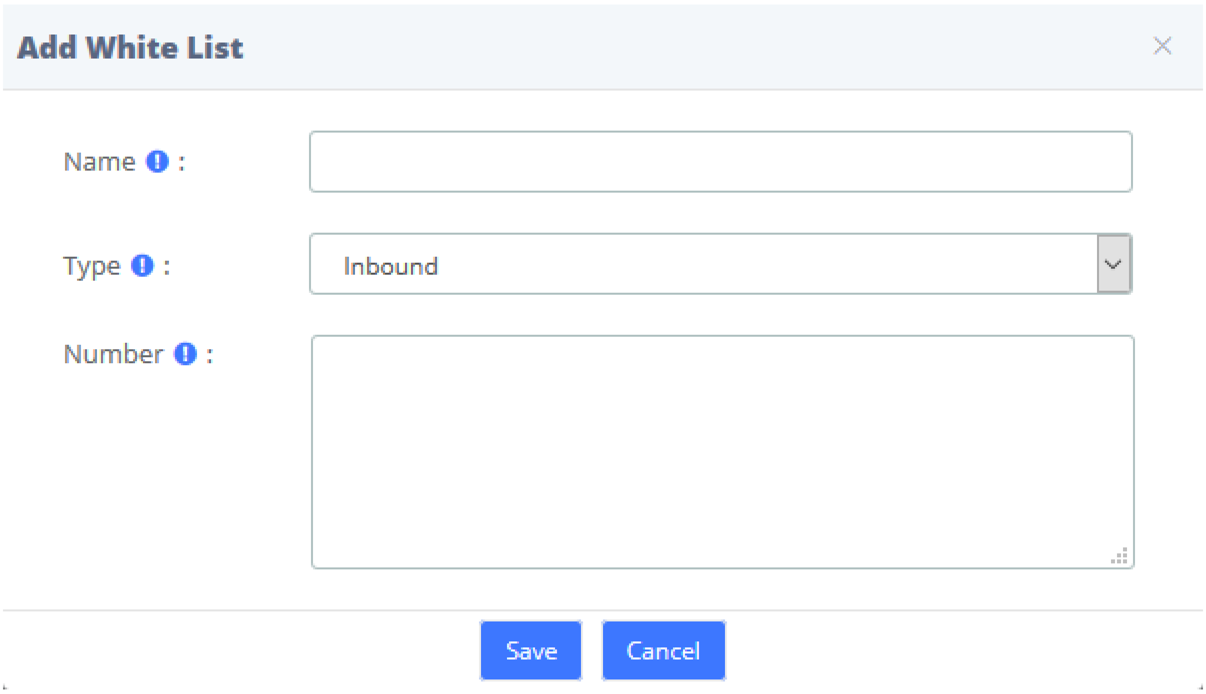
When a number is blacklisted, any calls with that number in the Caller ID field received by the system will be routed to the disconnected record.



Blacklist interface

Item	**Definition**
Name	Name of this blacklist rule.
Type	Which type the rule applies to, including Inbound/Outbound/Both When you choose type Outbound, you can select the extensions which will be blacklisted.
Number	Enter the number you want to block; you can input sets of digits that match the Dial Pattern Rules.

White List

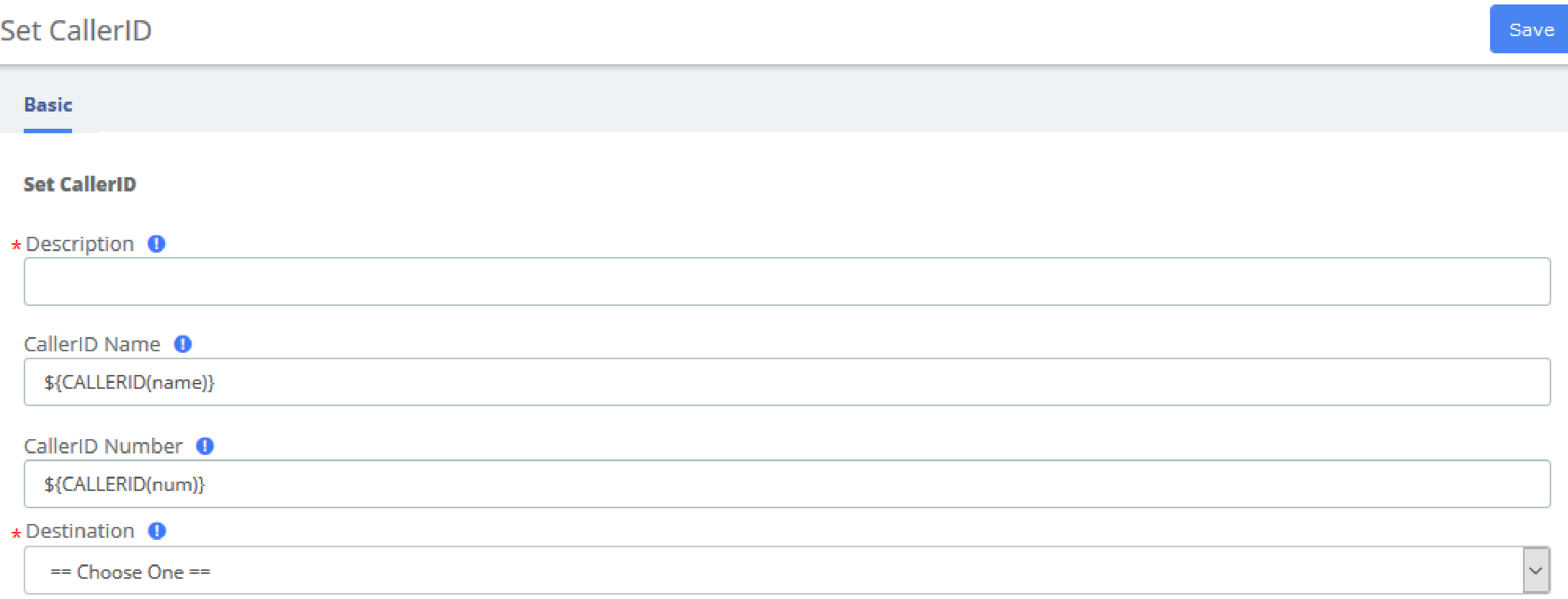


Whitelist interface

Item	Definition
Name	Name of this whitelist rule.
Type	Which type the rule applies to, including Inbound/Outbound/Both
Number	Enter the number you want to add into whitelist, you can input sets of digits that match the Dial Pattern Rules.

3.3.4 Set CallerID

You can change name and number of incoming call display.



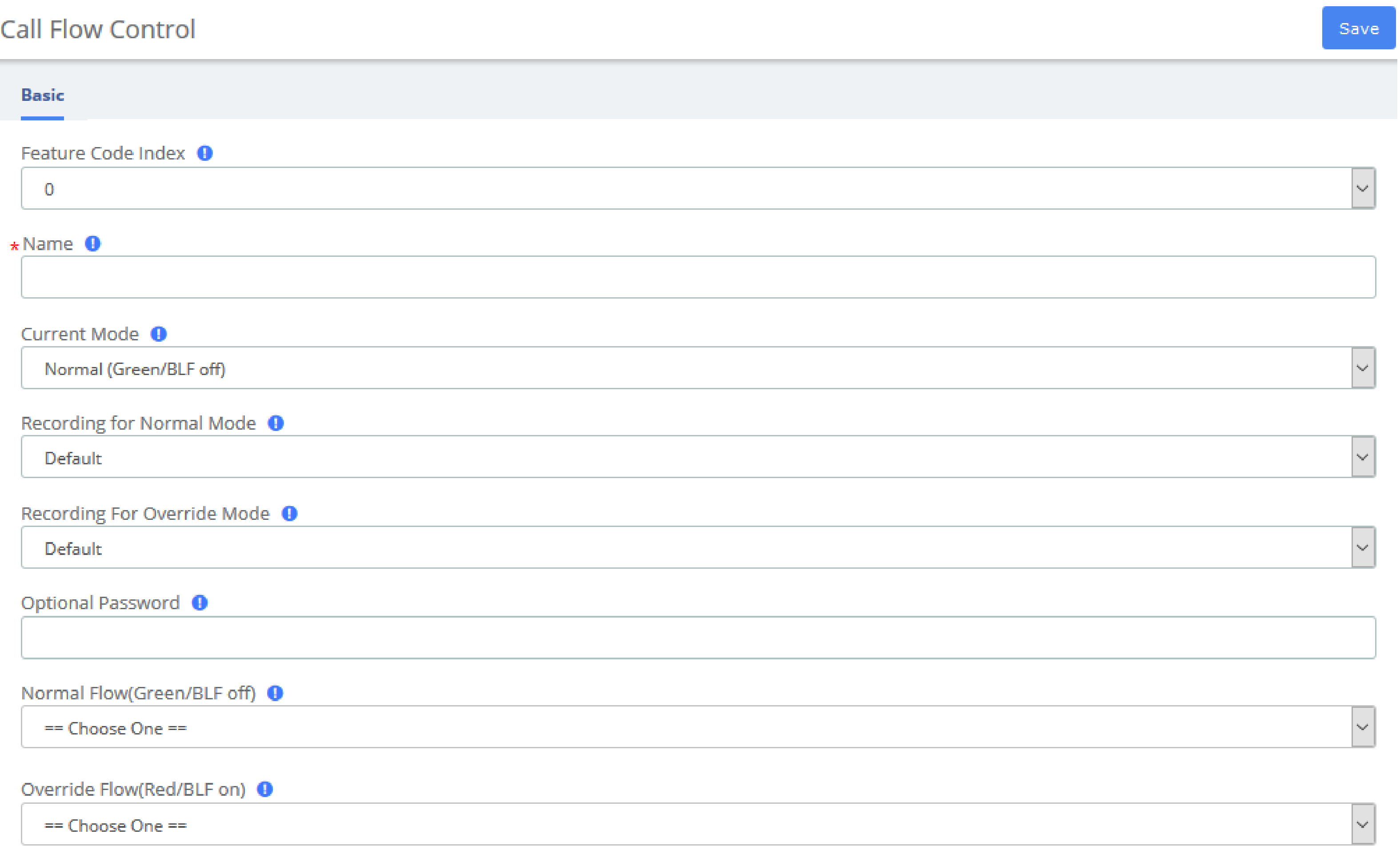
Set CallerID interface

Item	Definition
Description	Provide a title for it
CallerID Name	The caller ID name will be changed to it.
CallerID Number	The caller ID number will be changed to it.
Destination	Destination the call will be sent to after CID has been processed.

3.3.5 Call Flow Control

The Call Flow Control module is used to create a single destination that can act as a switch that can be toggled by anyone who has access to a local phone. It is commonly used to allow phone system users to manually switch between "Daytime Mode" and "Nighttime Mode."

Call Flow Control should not be confused with Time Conditions. While both of these modules relate to call flow, Call Flow Control is designed to be a **manual** switch, while a Time Condition is designed to be a **scheduled, automatic** switch.

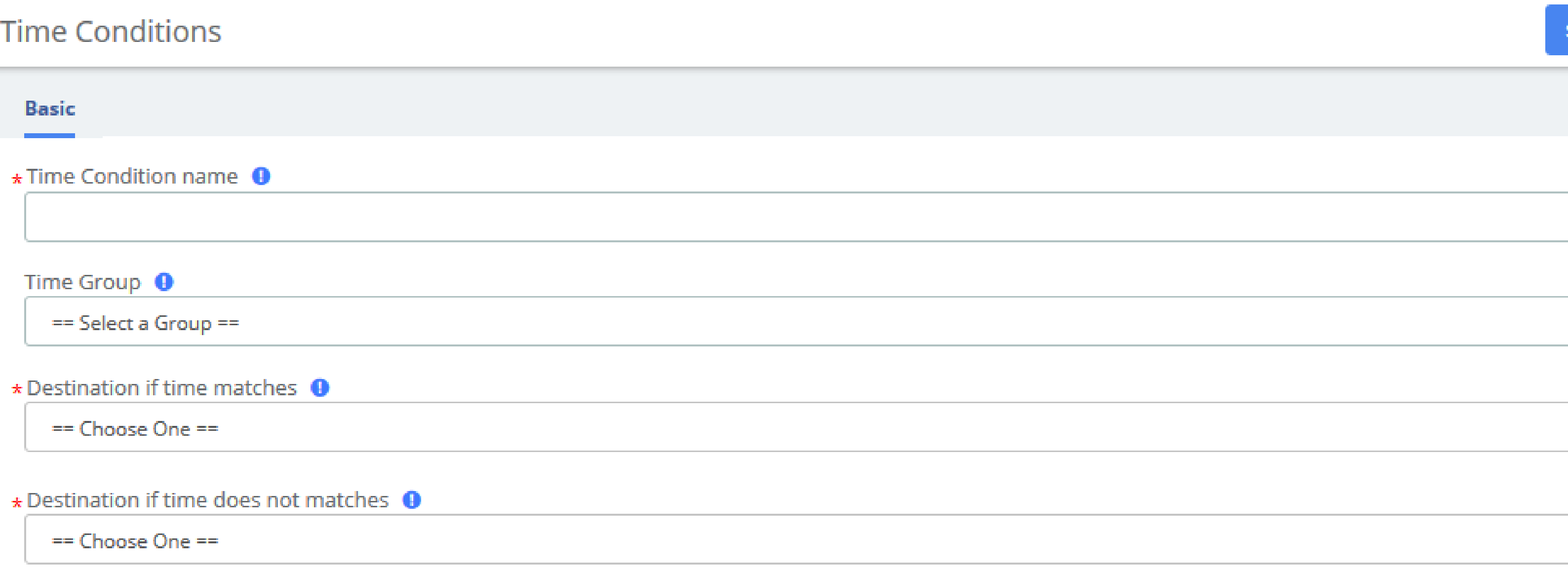


Item	**Definition**
Feature Code Index	There are a total of 10 Feature code objects,0-9, each can control a call flow and be toggled using the call flow toggled feature code plus the index
Name	Description for this Call Flow Toggle Control
Current Mode	This will change the current state for this Call Flow Toggle Control, or set the initial state when creating a new one.
Recording for Normal Mode	Message to be played in normal mode (Green/BLF off) To add additional recordings use the “System Recordings” MENU to the left
Recording for Override Mode	Message to be played in override mode (Green/BLF off) To add additional recordings use the “System Recordings” MENU to the left

Optional Password	You can optionally include a password to authenticate before toggling the call flow. If left blank anyone can use the feature code and it will be un-protected
Normal Flow (Green/BLF off)	Destination to use when set to Normal Flow (Green/BLF off) mode
Override Flow (Red/BLF on)	Destination to use when set to Override Flow (Red/BLF off) mode

3.3.6 Time Conditions

You can create various time conditions and use these time conditions in conjunction with your Inbound Route to individualize each of the incoming trunk's behavior.

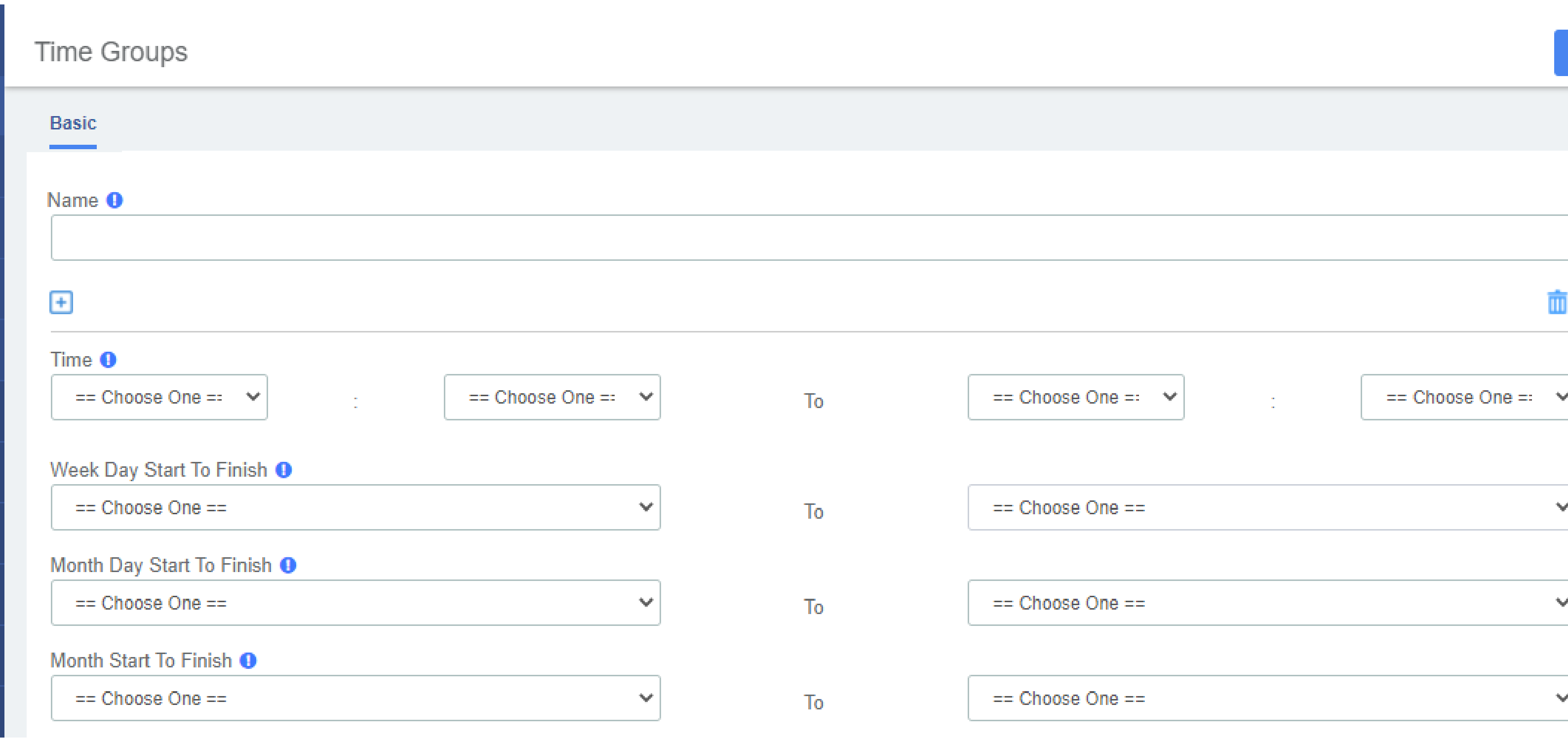


Item	Definition
Time Condition name	Give this Time Condition a brief name to help you identify it.
Time Group	Select a time group created under Time Groups. Matching times will be sent to matching destination. If no group is selected, call will always go to no-match destination.
Destination if time matches	The destination the call will be sent to when the time matches.
Destination if time does not match	The destination the call will be sent to when the time doesn't match.

3.3.7 Time Groups

The Time Groups Module is used to define periods of time that can then be selected in the Time Conditions module or Outbound Routes module.

For example, you might create a Time Group called "Lunch" that might start at 12:00 pm and end at 1:00 pm. You could then create a Time Condition that would use the Lunch Time Group to send calls to voicemail during lunch, and to a ring group at other times.

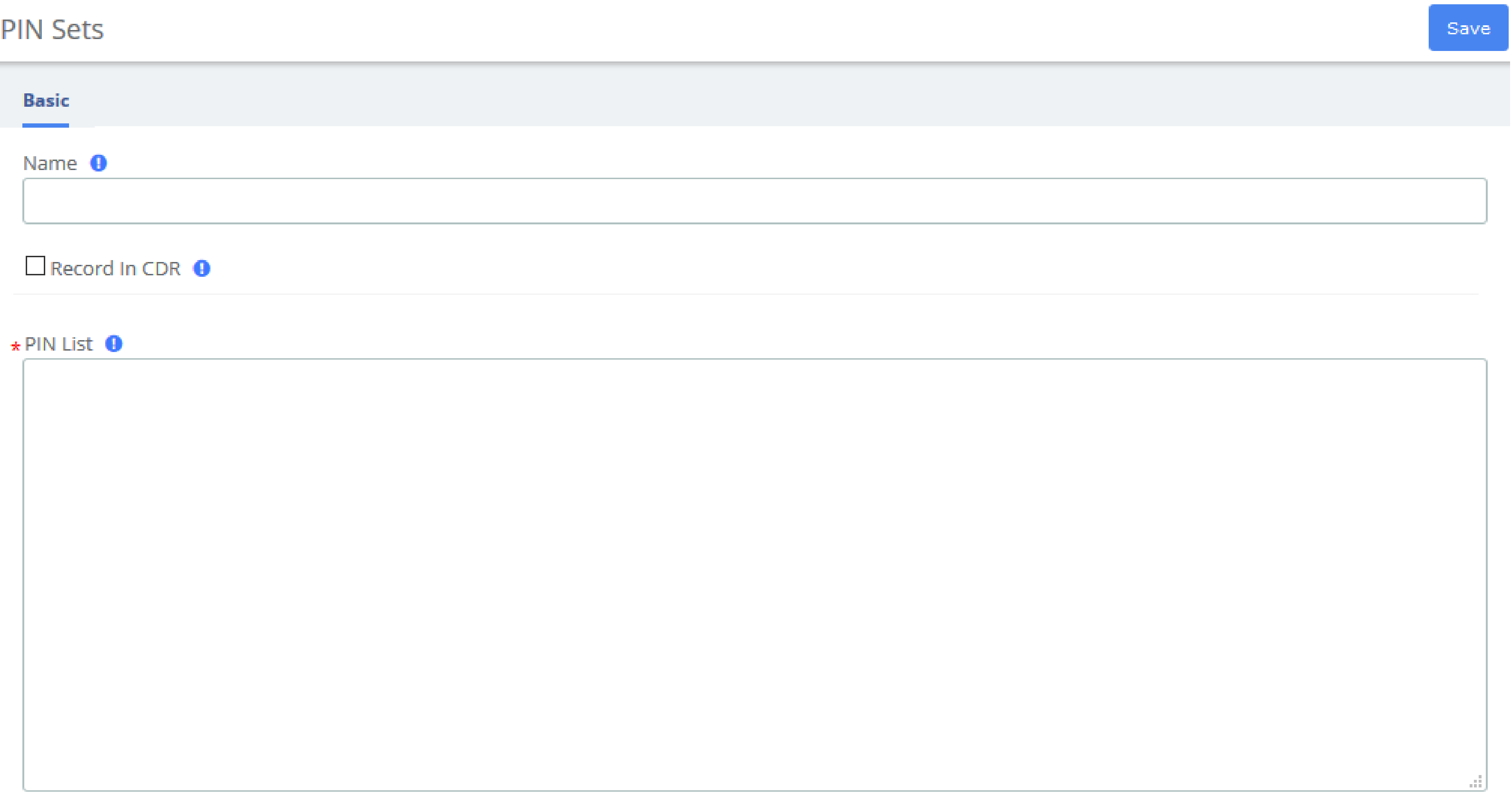


Item	Definition
Name	It will display as the name of time group
Time	Choose a time interval.
Week Day Start To Finish	Start and end times of one week.
Month Day Start To Finish	Start and end times of one month.
Month Start To Finish	Start and end times of one year.

3.3.8 PIN Sets

SMB PBX allows you to require callers to dial a password before an outbound call will go through. You can require a password on all calls, or only on calls to certain numbers.

The PIN Sets Module allows you to create define groups and then assign a list of passwords to each group. You can then restrict certain calls to certain groups by going to the Outbound Routes Module and limiting the route to a certain PIN Set group. Each Outbound Route can be limited to just one PIN Set group. So, if you want to allow more than one PIN Set group to make a certain type of call, just create a duplicate Outbound Route and assign the second Outbound Route to a different PIN Set Group.



PIN Sets Interface

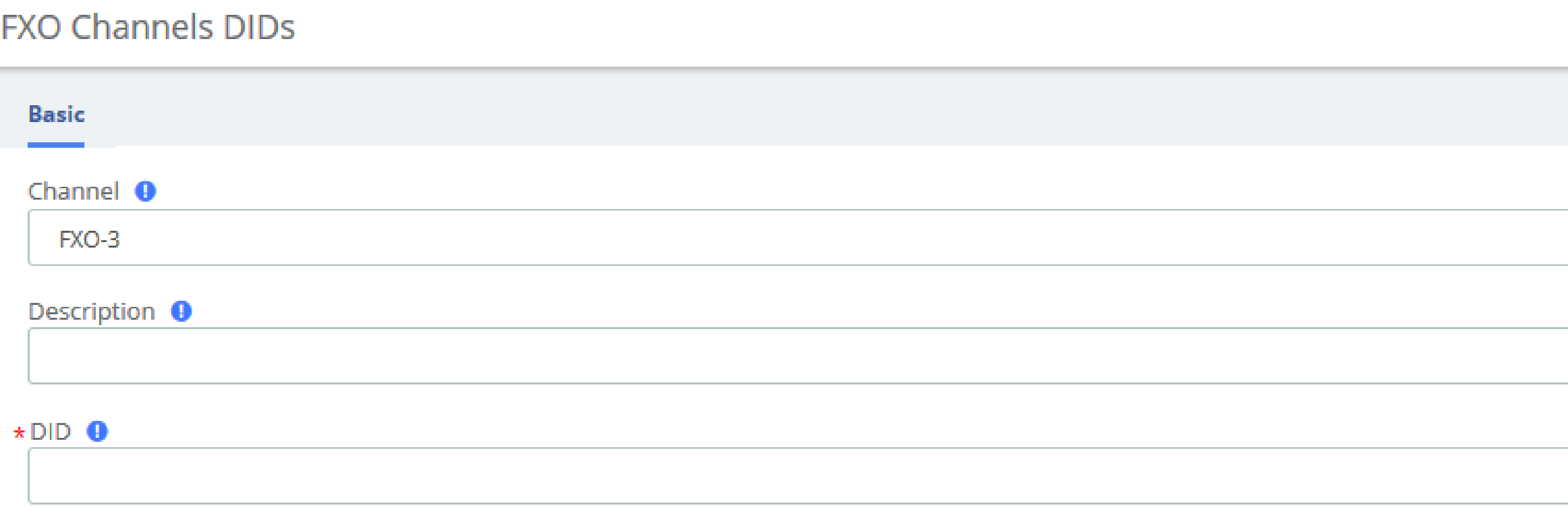
Item	Definition
Name	Name of the pin sets.
Record In CDR	Select this box if you would like to record the PIN in the call detail records when used.
PIN List	Enter a list of one more PINs. One PIN per line.

3.3.9 FXO Channels DIDs

The FXO Channel DIDs module allows you to assign a DID or phone number to specific analog channels.

Unlike SIP or PRI trunks, analog lines do not send a DID or dialed number to the PBX. Since the PBX routes all inbound calls based on the DID or number dialed, we need to map each analog port or channel to a fake number so we can match that number to an Inbound Route number and route your calls.

Each channel can be mapped to the same phone number if you want all calls on the analog lines to go to the same destination. This would be a common scenario if you have multiple POTS lines that are on a hunt group from your provider.



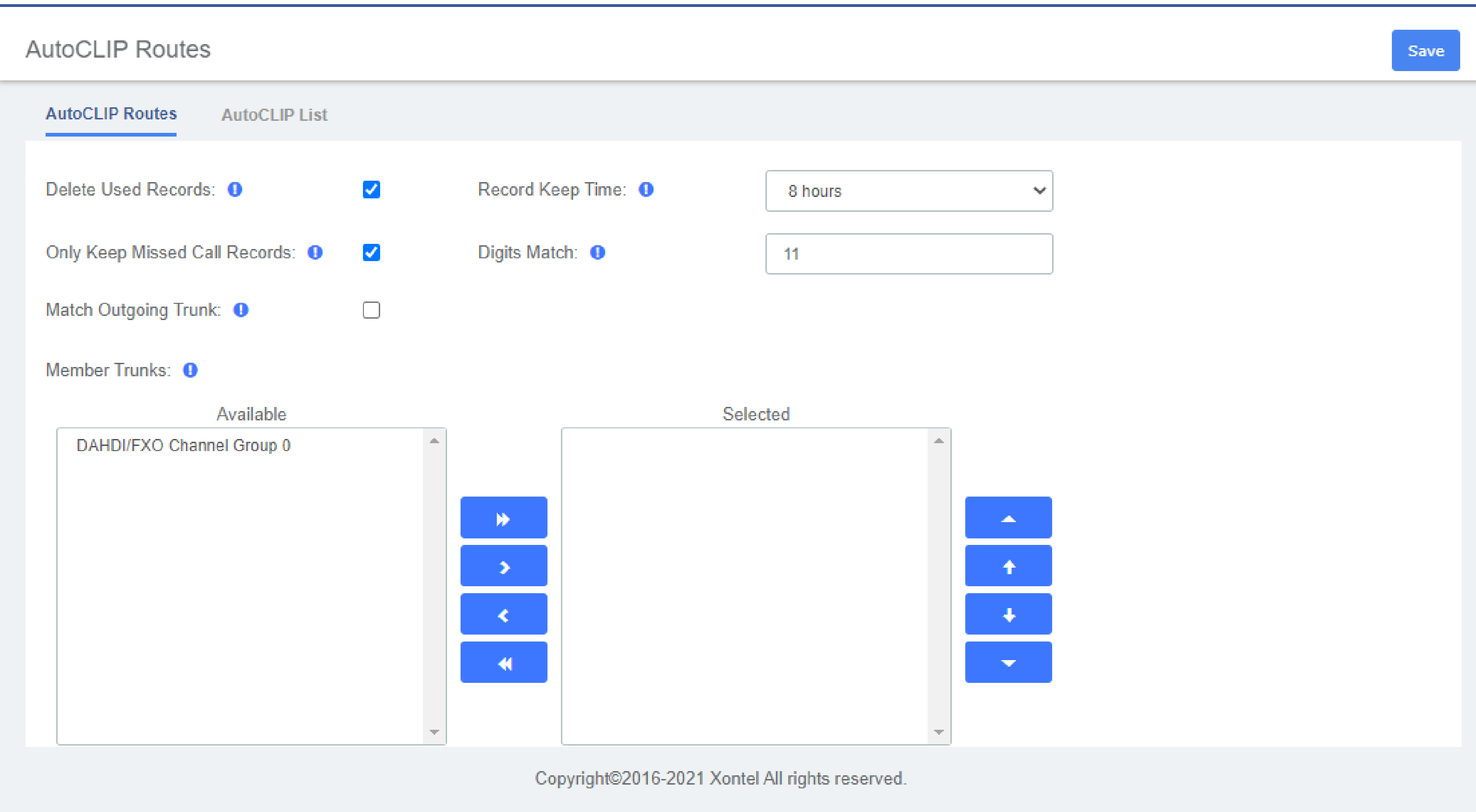
Add FXO Channel DID interface

Item	Definition
Channel	The FXO Channel number to map to a DID
Description	A useful description this channel
DID	The DID that this channel represents. The incoming call on this channel will be treated as if it came in with this DID and can be managed with Inbound Routing on DIDs

3.3.10 AutoCLIP Route

Generally, in the enterprise's telephony system, incoming calls are routed to IVR, ring groups, queues, and so on instead of specific extensions. AutoCLIP can redirect calls to the extension of the original caller instead of the automated attendant or the default ring group.

You may encounter situations that when you use an internal extension to call a client or colleague and they don't answer the call in time. By the time he/she dials back, the IP telephony system directs him/her to the default inbound routing destination such as IVR, making it difficult for callers who are not in touch to find you. AutoCLIP deals with this by ignoring the routing destination and redirecting this call to the original extension (your IP Phone line) according to stored records of outgoing calls in the AutoCLIP route table. This feature will retain many opportunities and possibilities for customers.

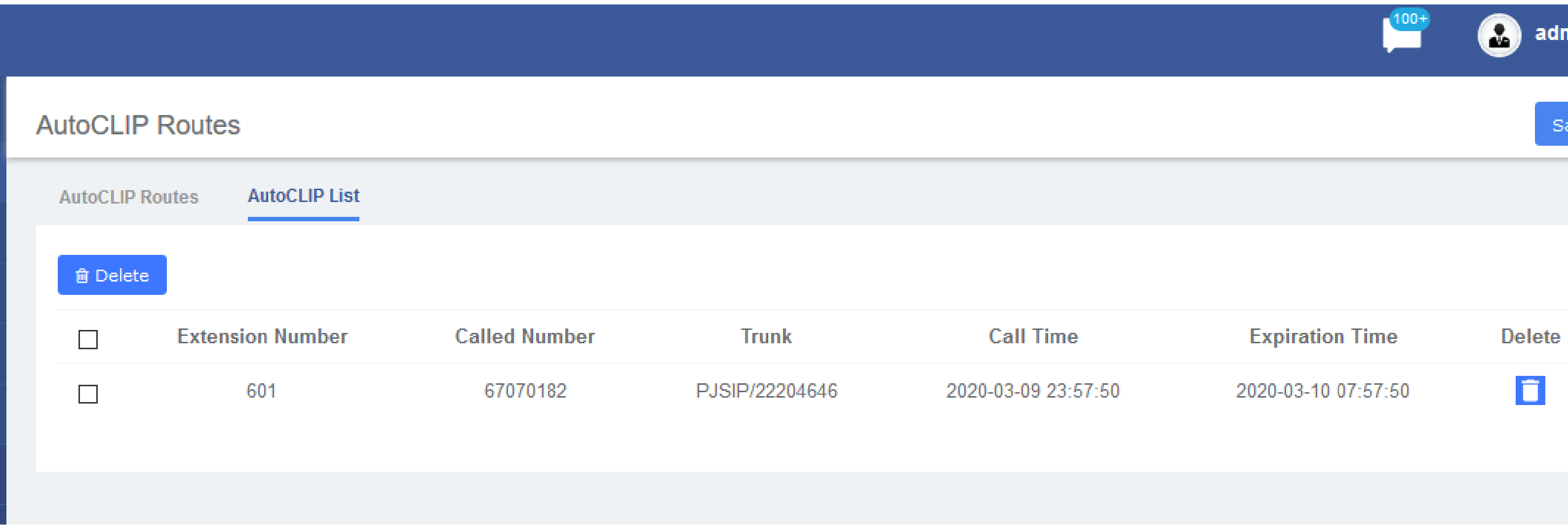


Add AutoCLIP Route interface

Item	**Definition**
Delete Used Records	If enabled, when an AutoCLIP record is matched, it will be automatically deleted afterwards.
Record Keep Time	This sets how long each record will be kept in the AutoCLIP List.

Only Keep Missed Call Records	If enabled, the system will only keep records of calls that are not answered by the called party in the AutoCLIP list. Note: PSTN line will keep records of all calls whether this option is enabled or disabled.
Digits Match	Define how many digits from the last digit of the incoming phone number will be used to match the AutoCLIP record. If the number has fewer digits than the value defined here, all digits will be matched.
Match Outgoing Trunk	If enabled, only the incoming call that came to the PBX through the same trunk which made the call will be match against the AutoCLIP List.
Member Trunks	This defines AutoCLIP Route will apply to which trunk and which trunk's record will be kept in the AutoCLIP list. If no trunk's selected, AutoCLIP will stop working.

AutoCLIP Route example

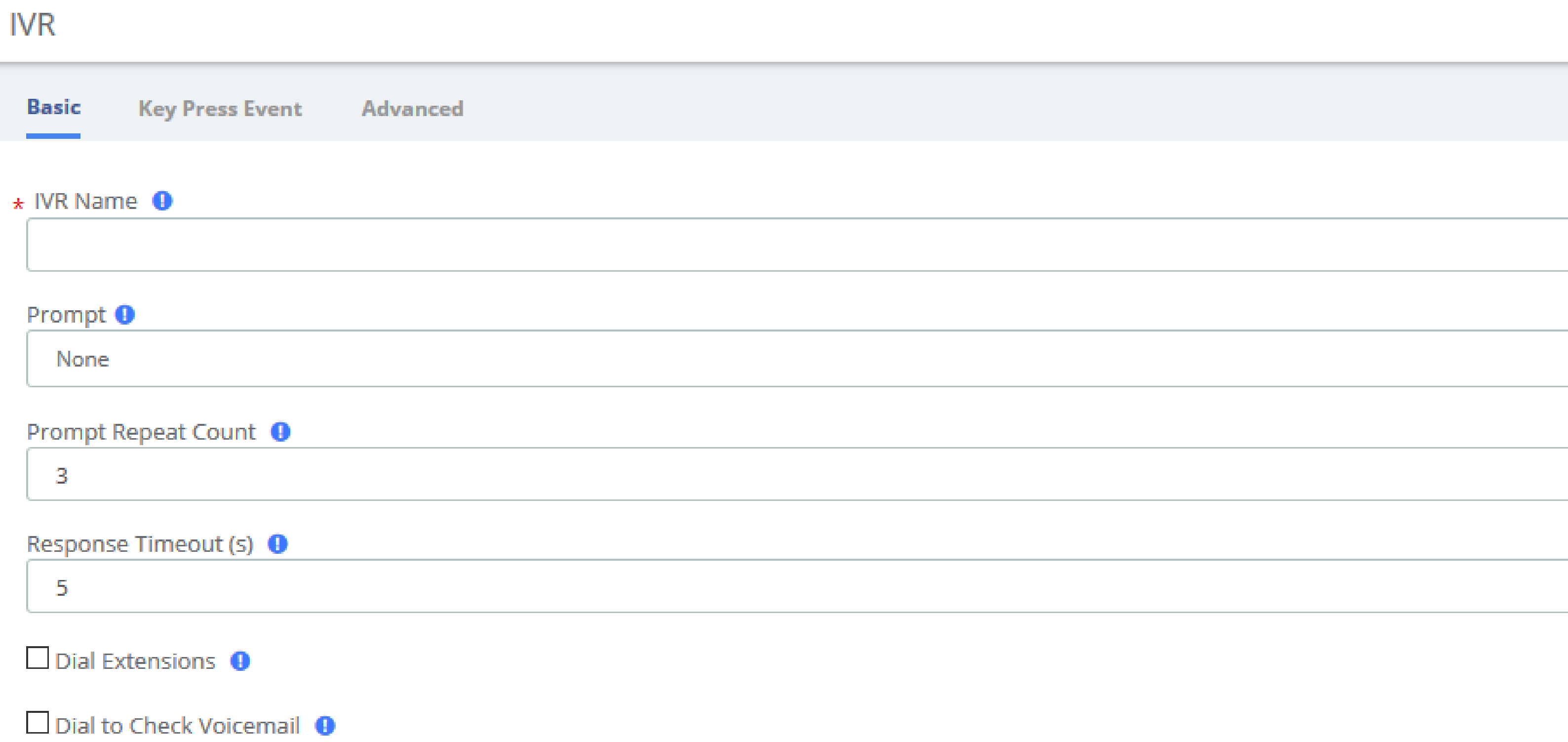


1. Extension user **601** makes a call to number **67070182**, the called party doesn't answer the call.
2. The number **67070182** calls back to PBX.
3. As we set in **Digits Match** in this example it will be "**8**", so the call will match against the Autoclip records, and the call will be forwarded directly to the extension user **601**.
4. The record will be deleted automatically in Autoclip and next time when the number **67070182** calls the PBX, no Autoclip route will be matched and the call will go to the inbound route destination.

3.4 Call Features

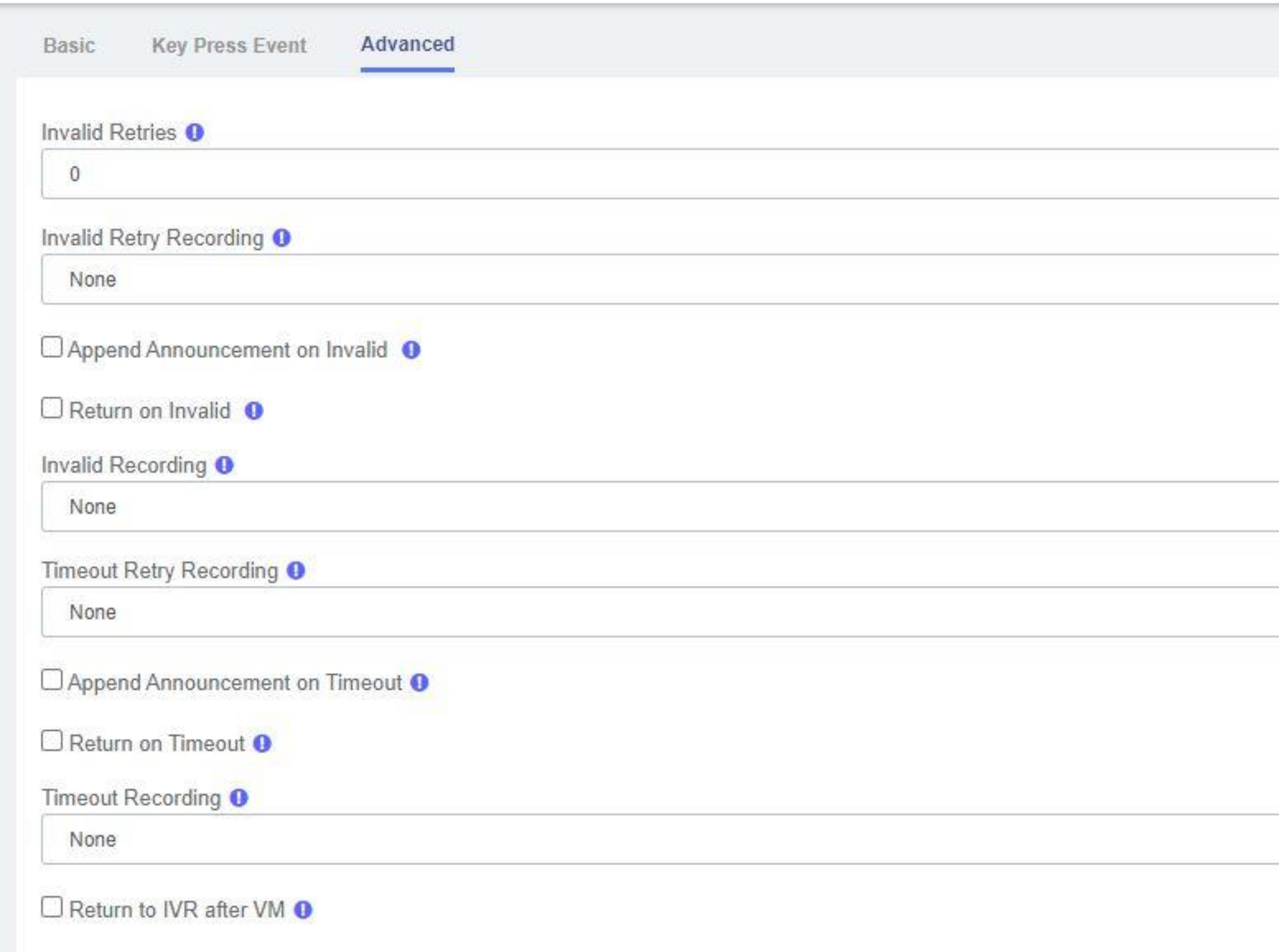
3.4.1 IVR

The IVR module allows you to create one or more IVRs ("Interactive Voice Response" systems or Auto Attendants). You can then route calls to the IVR and play a recording to prompt callers what options to enter, such as “press 1 for sales and press 2 for the company directory.” An IVR can also route calls to another IVR, or in other words, a sub-menu. As a general rule, you never want more than five or six options in a single IVR, or it will become too confusing to navigate. It is better to only include a few options at a single menu level, and route callers to a sub-menu for more choices.



Basic | **Key Press Event** | Advanced

Press 0:	Select an Option
Press 1:	Select an Option
Press 2:	Select an Option
Press 3:	Select an Option
Press 4:	Select an Option
Press 5:	Select an Option
Press 6:	Select an Option
Press 7:	Select an Option
Press 8:	Select an Option
Press 9:	Select an Option
Press #:	Select an Option
Press *:	Select an Option
Timeout :	Select an Option
Invalid :	Select an Option
digits pressed	Select an Option



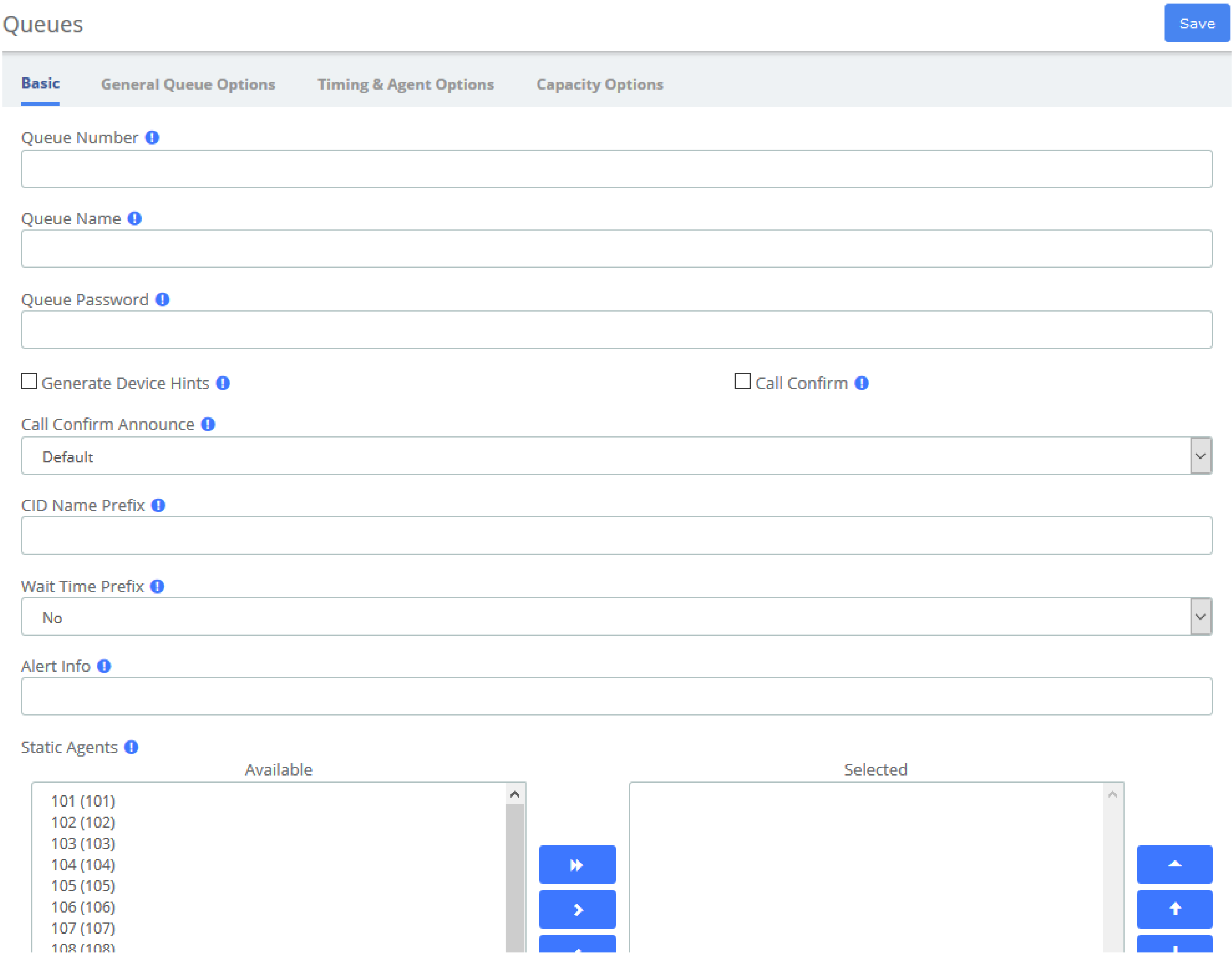
Item	Definition
Basic	
IVR Name	Name of this IVR
Prompt	The prompt will be played when a call reaches the IVR.
Prompt Repeat Count	The number of times that the prompt will be played.
Response Timeout (s)	The number of seconds to wait for a digit input after prompt.
Dial Extensions	Allow the caller to dial extension directly.

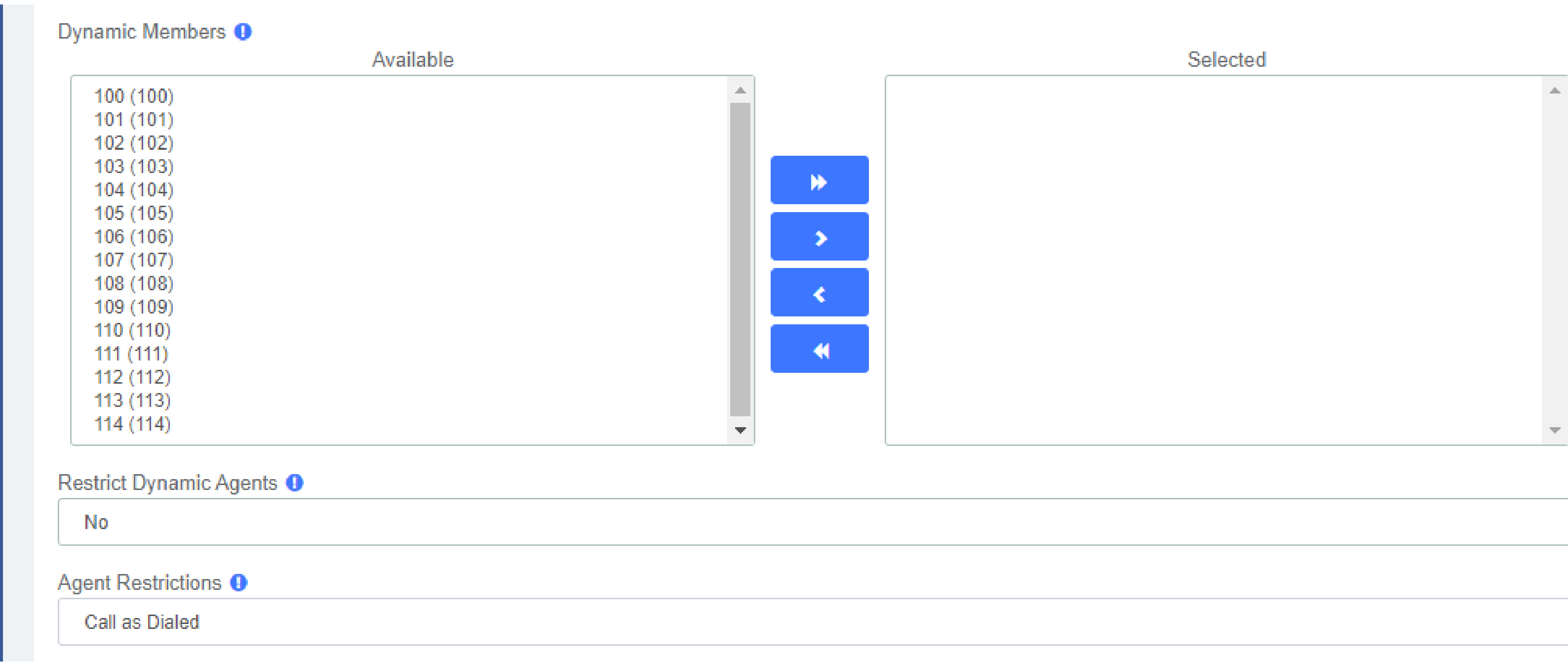
Dial to Check Voicemail	If enabled, the caller will be allowed to dial '*97' to check voicemail.
Key Press Event	
Select the destination for each key pressing: digits 0-9, "#", "*", Timeout and Invalid. When the callers press the corresponding key, the call will be routed to extension, voicemail, ring Group, IVR, queues, etc...	
Advanced	
Invalid Retries	Number of time to retry when receiving an invalid/unmatched response from the caller
Invalid Retry Recording	Prompt to be played when an invalid/unmatched response is received, before prompt the caller to try again
Append Announcement on Invalid	After playing Invalid Retry Recording the system will replay mail IVR Announcement
Return on Invalid	Check this box to have this option return to a parent IVR if it was called from a parent IVR. If not, it will go to the chosen destination. The return path will be to any IVR that was in the call path prior to this IVR which could lead to strange result if there was an IVR called in the call path but not immediately before this.
Invalid Recording	Prompt to be played before sending the caller to an alternate destination due to the caller pressing 0 or receiving the maximum amount of invalid/unmatched responses (as determined by Invalid Retries)
Timeout Retry Recording	Prompt to be played when a timeout occurs, before prompting the caller to try again
Append Announcement on Timeout	After playing the Timeout Retry Recording the system will replay the main IVR Announcement.
Return on Timeout	Check this box to have this option return to a parent IVR if it was called from a parent IVR. If not, it will go to the chosen destination. The return path will be to any IVR that was in the call path prior to this IVR which could lead to strange result if there was an IVR called in the call path but not immediately before this

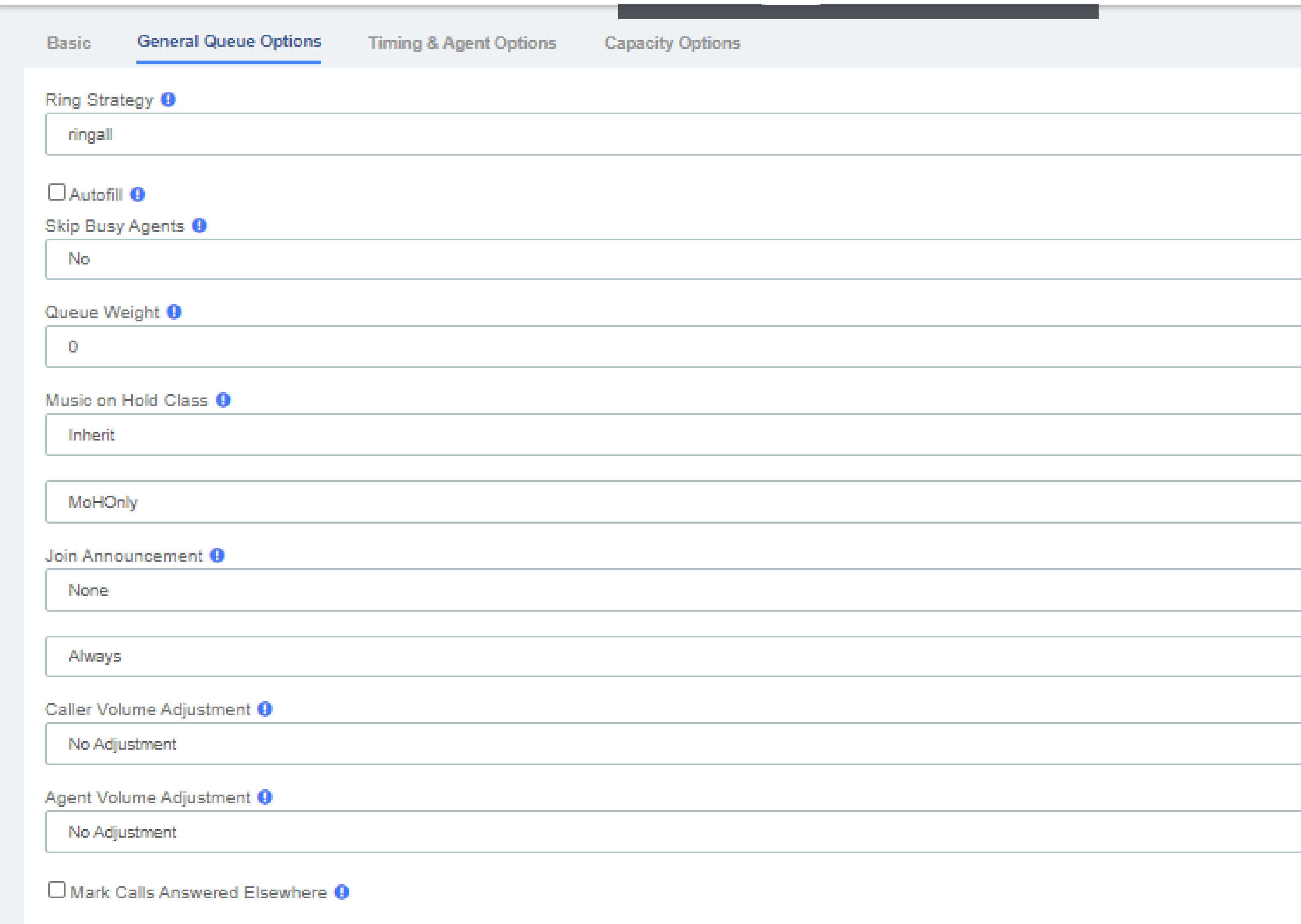
Timeout Recording	Prompt to be played before sending the caller to an alternate destination due to the caller pressing 0 or receiving the maximum amount of invalid/unmatched responses (as determined by Invalid Retries)
Return to IVR after VM	If checked, upon exiting voicemail a caller will be returned to this IVR if they got a user voicemail

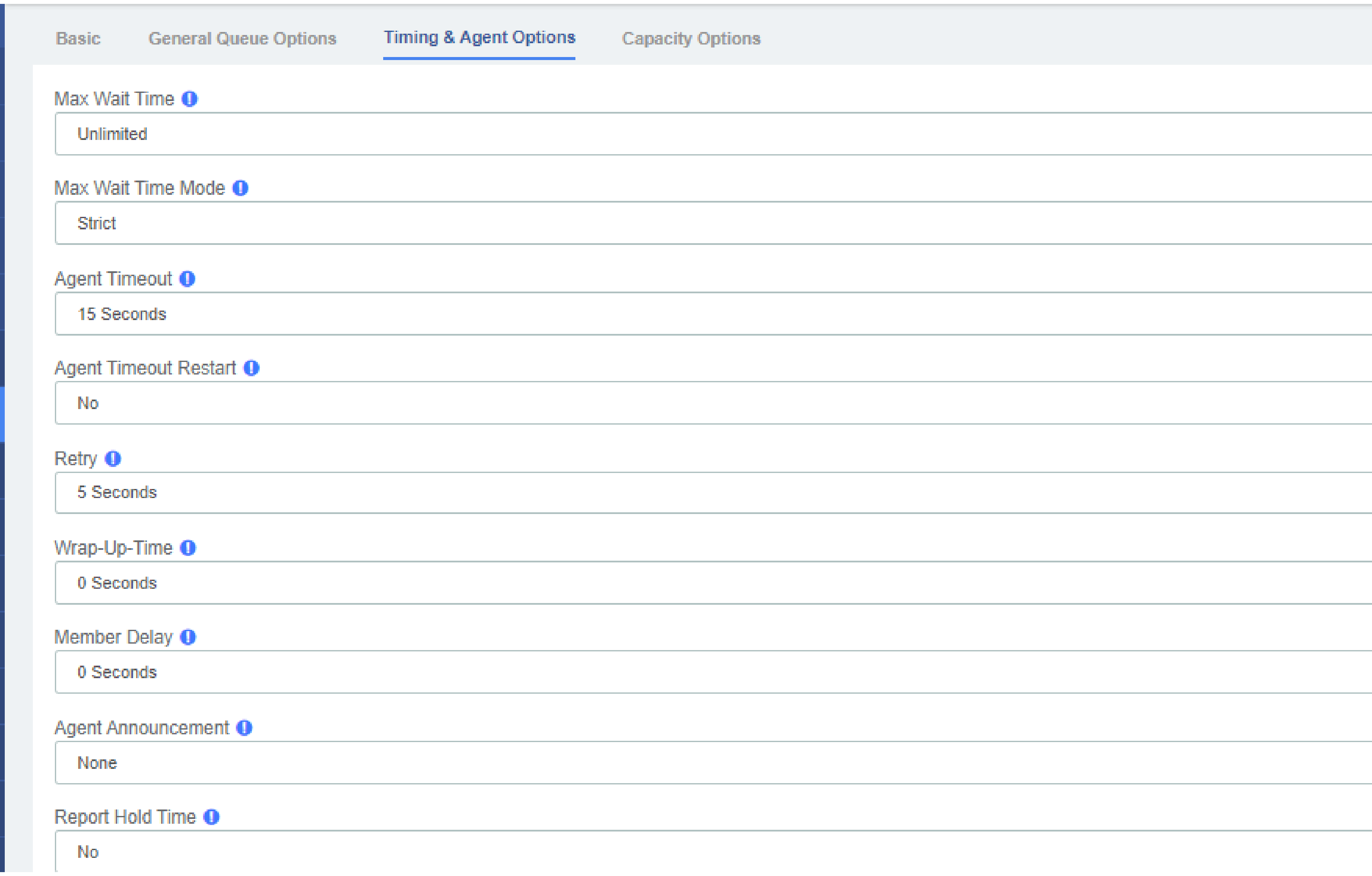
3.4.2 Queues

The Queues module is a more advanced version of the Ring Groups module. Like the Ring Groups module, the Queues module is used to create an extension number that your users can dial in order to ring multiple extensions at the same time. It also creates a destination to which you can send calls that will ring those multiple extensions.

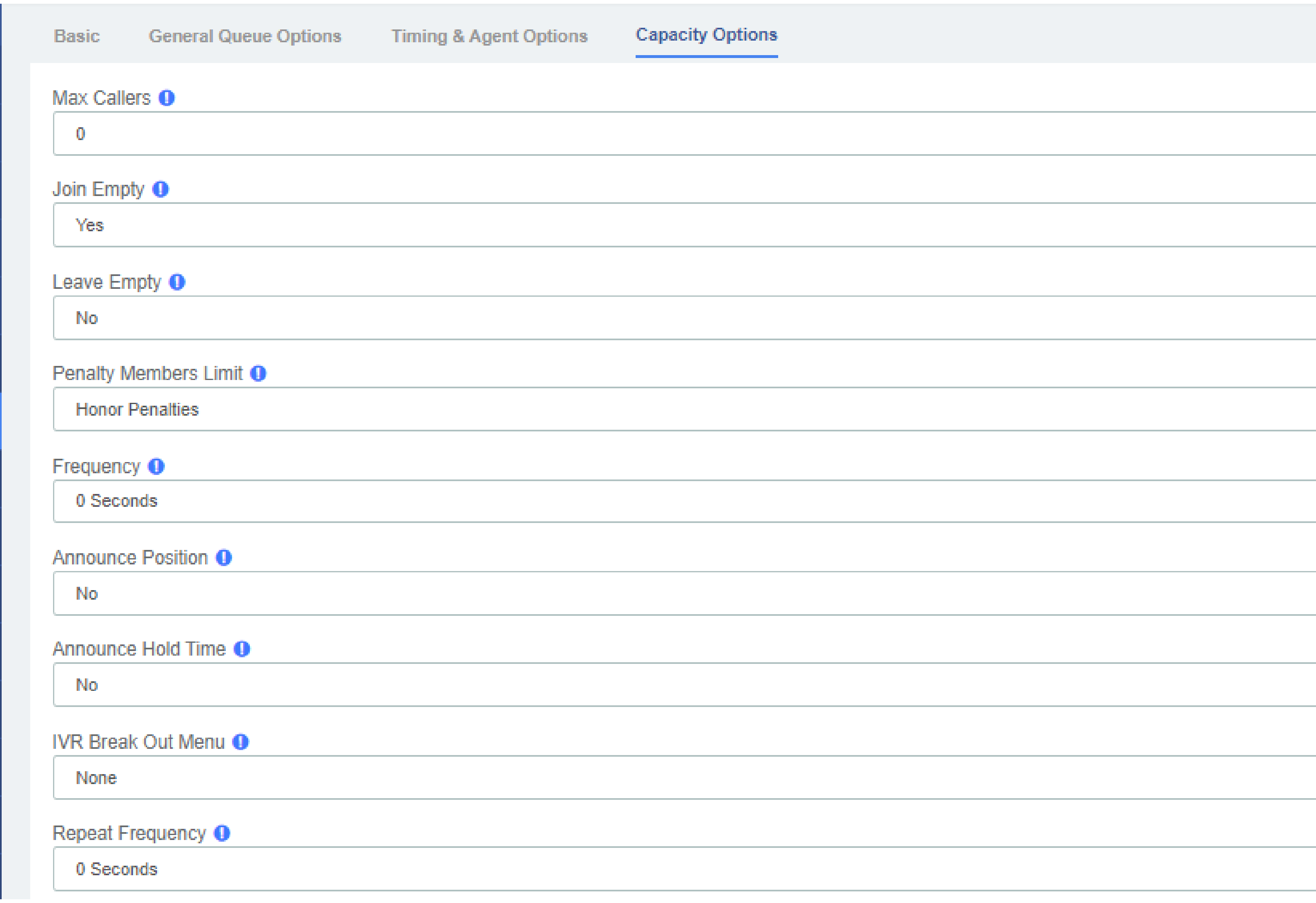


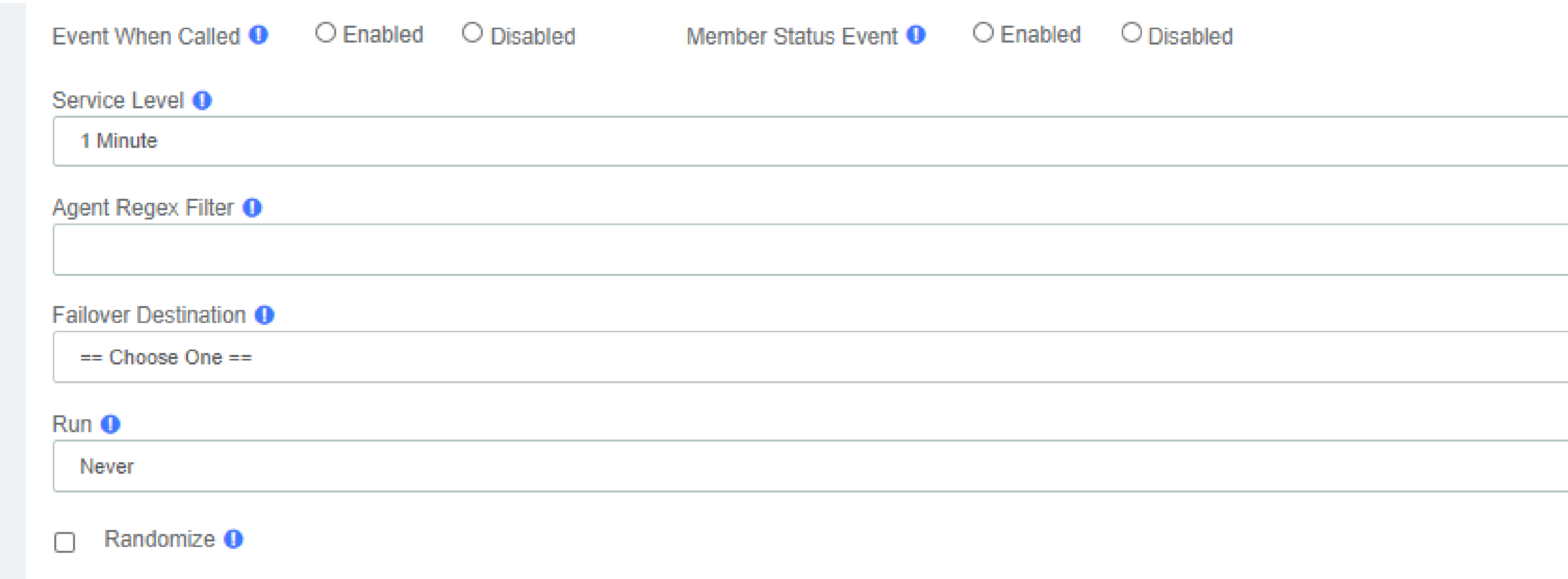

Dynamic Members
Available
Selected
100 (100)
101 (101)
102 (102)
103 (103)
104 (104)
105 (105)
106 (106)
107 (107)
108 (108)
109 (109)
110 (110)
111 (111)
112 (112)
113 (113)
114 (114)
Restrict Dynamic Agents
No
Agent Restrictions
Call as Dialed


Basic
General Queue Options
Timing & Agent Options
Capacity Options
Ring Strategy
ringall
Autofill
Skip Busy Agents
No
Queue Weight
0
Music on Hold Class
Inherit
MoHOnly
Join Announcement
None
Always
Caller Volume Adjustment
No Adjustment
Agent Volume Adjustment
No Adjustment
Mark Calls Answered Elsewhere


Basic
General Queue Options
Timing & Agent Options
Capacity Options
Max Wait Time
Unlimited
Max Wait Time Mode
Strict
Agent Timeout
15 Seconds
Agent Timeout Restart
No
Retry
5 Seconds
Wrap-Up-Time
0 Seconds
Member Delay
0 Seconds
Agent Announcement
None
Report Hold Time
No


Auto Pause
No
Auto Pause on Busy
No
Auto Pause on Unavailable
No
Auto Pause Delay
0


Basic
General Queue Options
Timing & Agent Options
Capacity Options
Max Callers
0
Join Empty
Yes
Leave Empty
No
Penalty Members Limit
Honor Penalties
Frequency
0 Seconds
Announce Position
No
Announce Hold Time
No
IVR Break Out Menu
None
Repeat Frequency
0 Seconds


Event When Called
Enabled
Disabled
Member Status Event
Enabled
Disabled
Service Level
1 Minute
Agent Regex Filter
Failover Destination
== Choose One ==
Run
Never
Randomize

Item	Definition
Basic	
Queue Number	Use this number to dial into the queue, or transfer callers to this number to put them into the queue. Agents will dial this queue number plus* to log the queue, and this queue number plus** to log out the queue. For example, if the queue number is 123: 123*=log in 123**=log out
Queue Name	Give the queue a brief name to help you identify it.
Queue Password	You can require agents to enter a password before they can log in to this queue. This setting is optional. The password is only used when logging in with the legacy queue no* code. When using the toggle codes, you must use the Restrict Dynamic Agents option in conjunction with the Dynamic Members list to control access.
Generate Device Hints	If checked, individual hints and dial plan will be generated for each SIP and IAX2 device that could be part of this queue. These are used in conjunction with programmable BLF status as to the current state, the format of this hints is *45ddd*qqq Where *45 is the currently define toggle feature code, ddd is the device number (typically the same as the extension number) and qqq is this queue's number
Call Confirm	If checked, any queue member that is actually an outside telephone number, or any extension Follow-Me or call forwarding that are pursued and leave the PBX will be forced into Call Confirmation mode where the member must acknowledge the call before it is answered and delivered.
Call Confirm Announce	Announcement played to the Queue Member announcing the Queue call and requesting confirmation prior to answering. If set to default, the standard call confirmation default message will be played unless the number is reached through a Follow-Me and this is an alternate message provided in the Follow-Me. This message will override any other message specified. To add additional recordings please use the "System Recordings" MENU.

CID Name Prefix	You can optionally prefix the Caller ID name of callers to the queue. ie: If you prefix with "Sales:", a call from John Doe would display as "Sales: John Doe" on the extensions that ring.
Wait Time Prefix	When set to Yes, the CID Name will be prefix with the total wait time in the queue so the answering agent is aware how long they have waited. It will be rounded to the nearest minute, in the form of Mnn: where nn is the number of minutes. If the call is subsequently transferred, the wait time will reflect the time since it first entered the queue or reset if the call is transferred to another queue with this feature set.
Alert Info	ALERT_INFO can be used for distinctive ring with SIP device.
Static Agents	Static agents are extensions that are assumed to always be on the queue. Static agents do not need to 'log in' to the queue, and cannot 'log out' of the queue. List extensions to ring, one per line. You can include an extension on a remote system, or an external number (Outbound Routing must contain a valid route for external numbers). You can put a "," after the agent followed by a penalty value, see Asterisk documentation concerning penalties. An advanced mode has been added which allows you to prefix an agent number with S, X, Z, D or A. This will force the agent number to be dialed as an Asterisk device of type SIP, IAX2, ZAP, DAHDI or Agent respectively. This mode is for advanced users and can cause known issues in PBX as you are by-passing the normal dial plan. If your 'Agent Restrictions' are not set to 'Extension Only' you will have problems with subsequent transfers to voicemail and other issues may also exist. (Channel Agent is deprecated starting with Asterisk 1.4 and gone in 1.6+.)
Dynamic Members	Dynamic Members are extensions or callback numbers that can log in and out of the queue. When a member logs in to a queue, their penalty in the queue will be as specified here. Extensions included here will NOT automatically be logged in to the queue.
Restrict Dynamic Agents	Restrict dynamic queue member logins to only those listed in the Dynamic Members list above. When set to Yes, members not listed will be DENIED ACCESS to the queue.
Agent Restrictions	When set to 'Call as Dialed' the queue will call an extension just as if the queue were another user. Any Follow-Me or Call Forward states active on the extension will result in the queue call following these call paths. This behavior has been the standard queue behavior on past PBX versions.

	When set to 'No Follow-Me or Call Forward', all agents that are extensions on the system will be limited to ring their extensions only. Follow-Me and Call Forward settings will be ignored. Any other agent will be called as dialed. This behavior is similar to how extensions are dialed in ringgroups When set to 'Extensions Only' the queue will dial Extensions as described for 'No Follow –Me or Call Forward'. Any other number entered for an agent that is NOT a valid extension will be ignored. No error checking is provided when entering a static agent or when logging on as a dynamic agent, the call will simply be blocked when the queue tries to call it. For dynamic agents, see the 'Agent Regex filter' to provide some validation.
General Queue Options	
Ring Strategy	**Ringall**: ring all available agents until one answers (default) **Leastrecent:** ring agent which was least recently called by this queue **Fewestcalls:** ring the agent with fewest completed calls from this queue **Random:** ring random agent **Rrmemory:** round robin with memory, remember where we left off last ring pass **Rrordered:** same as rrmemory, except the queue member where order from config file is preserved **Linear:** rings agents in the order specified, for dynamic agents in the order they logged in **Wrandom:** random using the member's penalty as a weighting factor, see asterisk documentation for specifics.
Autofill	Starting with Asterisk 1.4, if this is checked, and multiple agents are available, Asterisk will send one call to each waiting agent (depending on the ring strategy). Otherwise, it will hold all calls while it tries to find an agent for the top call in the queue making other calls wait. This was the behavior in Asterisk 1.2 and has no effect in 1.2. See Asterisk documentation for more details of this feature.
Skip Busy Agents	When set to 'Yes' agents who are on an occupied phone will be skipped as if the line were returning busy. This means that Call Waiting or multi-line phones will not be presented with the call and in the various hunt style ring strategies, the next agent will be attempted. When set to 'Yes + (ringinuse=no)' the queue configuration flag 'ringinuse=no' is set for this queue in addition to the phone's device status being monitored. This results in the queue tracking remote agents (agents who are a remote PSTN phone, called through Follow-Me, and other means) as well as PBX connected agents, so the queue

	will not attempt to send another call if they are already on a call from any queue. When set to 'Queue calls only (ringinuse=no)' the queue configuration flag 'ringinuse=no' is set for this queue also but the device status of locally connected agents is not monitored. The behavior is to limit an agent belonging to one or more queues to a single queue call. If they are occupied from other calls, such as outbound calls they initiated, the queue will consider them available and ring them since the device state is not monitored with this option. WARNING: When using the settings that set the 'ringinuse=no' flag, there is a NEGATIVE side effect. An agent who transfers a queue call will remain unavailable by any queue until that call is terminated as the call still appears as 'inuse' to the queue UNLESS 'Agent Restrictions' is set to 'Extensions Only'.
Queue Weight	Gives queue 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. Choose “inherit” if you want the MoH class to be what is currently selected, such as by the inbound route. MoH Only will play music until the agent answers. Agent Ringing will play MoH until an agent’s phone is presented with the call and is ringing. If they don’t answer MoH will return. Ring only makes callers hear a ringing tone instead of MoH ignoring any MoH class selected as well as any configured periodic announcements. This music is defined in the “Music on Hold” Menu.
Join Announcement	Announcement played to callers prior to joining the queue. This can be skipped if there are agents ready to answer a call (meaning they still may be wrapping up from a previous call) or when they are free to answer the call right now. To add additional recordings please use the “System Recordings” MENU.
Caller Volume Adjustment	Adjust the recording volume of the caller.
Agent Volume Adjustment	Adjust the recording volume of the queue member (Agent).
Mark calls answered elsewhere	Enabling this option, all calls are marked as ‘answered elsewhere’ when cancelled. The effect is that missed queue calls are *not* shown on the phone(if the phone support it)
Timing & Agent Options	
Max Wait Time	The maximum number of seconds a caller can wait in a queue before being pulled out.(0 for unlimited).

Max Wait Time Mode	Asterisk timeout priority. In ‘Strict’ mode, when the ‘Max Wait Time’ of a caller is hit, they will be pulled out of the queue immediately. In ‘Loose’ mode, if a queue stops ringing with this call, then we will wait until the queue stops ringing this queue number or otherwise the call is rejected by the queue member before taking the caller out of the queue. This means that the ‘Max Wait Time’ could be as long as ‘Max Wait Time’+’Agent Timeout’ combined.
Agent Timeout	The number of seconds an agent’s phone can ring before we consider it a timeout. Unlimited or other timeout values may still be limited by system ring time or individual extension defaults.
Agent Timeout Restart	If timeout restart is set to yes, then the time out for an agent to answer is reset if a BUSY or CONGESTION is received. This can be useful if agents are able to cancel a call with reject or similar
Retry	The number of seconds we wait before trying all the phones again. Choosing “No Retry” will exit the queue and go to the fail-over destination as soon as the first attempted agent time-out, additional agents will not be attempted.
Wrap-Up-Time	After a successful call, how many seconds to wait before sending a potentially free agent another call (default is 0, or no delay) If using Asterisk 1.6+, you can also set the ‘Honor Wrap-up Time Across Queues setting (Asterisk: shared_lastcall) on the Advanced Settings page so that this is honored across queues for members logged on to multiple queues.
Member Delay	If you wish to have a delay before the member is connected to the caller (or before the member hears any announcement messages), set this to the number of seconds to delay.
Agent Announcement	Announcement played to the Agent prior to bridging in the caller. Example:” the Following call is from the Sales Queue” or “This call is from the Technical Support Queue”. To add additional recordings please use the “System Recordings” MENU. Compound recordings composed of 2 or more sound files are not displayed as options since this feature cannot accept such recordings.
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.
Auto Pause	Auto Pause an agent in this queue (or all queues they are a member of) if they don’t answer a call. Specific behavior can be modified by the Auto Pause Delay as well Auto Pause Busy/Unavailable settings if supported on this version of Asterisk.

Auto Pause on Busy	When set to Yes agents devices that report busy upon a call attempt will be considered as a missed call and auto paused immediately or after the auto pause delay if configured
Auto Pause on Unavailable	When set to Yes agents devices that report congestion upon a call attempt will be considered as a missed call and paused immediately or after that auto pause delay if configured
Auto Pause Delay	This setting will delay the auto pause of an agent by auto pause delay seconds from when it last took a call. For example, if this were set to 120 seconds, and a new call is presented to the agent 90 seconds after they last took a call, will not be auto paused if they don't answer the call. If presented with a call 120 seconds or later after answering the last calls, this will have no effect.
Capacity Options	
Max Callers	Maximum number of people waiting in the queue (0 for unlimited)
Join Empty	Determines if new callers will be admitted to the Queue, if not, the failover destination will be immediately pursued. The options include: ● **Yes** Always allows the caller to join the Queue. ● **Strict** Same as Yes but stricter. Simply speaking, if no agent could answer the phone then don't admit them. If agents are infused or ringing someone else, caller will still be admitted. ● **Ultra Strict** Same as Strict plus a queue member must be able to answer the phone 'now' to let them in. simply speaking, any 'available' agents that could answer but are currently on the phone or ringing on behalf of another caller will be considered unavailable. ● **No** Callers will not be admitted if all agents are paused, show an invalid status for their device, or have penalty values less than QUEUE_MAX_PENALTY (not currently set in dial plan). ● **Loose** Same as No except Callers will be admitted if there are paused agents who could become available.
Leave Empty	Determines if callers should be exited prematurely from the queue in situations where it appears no one is currently available to take the call. The options include: ● **Yes** Callers will exit if all agents are paused, show an invalid state for their device or have penalty values less than QUEUE_MAX_PENALTY (not currently set in dial plan) ● **Strict** Same as Yes but stricter. Simply speaking, if no agent could answer the

	phone then have them leave the queue. If agents are in use or ringing someone else, caller will still be held. ● **Ultra Strict** Same as Strict plus a queue member must be able to answer the phone 'now' to let them remain. simply speaking, any 'available' agents that could answer but are currently on the phone or ringing on behalf of another caller will be considered unavailable. ● **Loose** Same as No except Callers will remain in the queue, if there are paused agents who could become available. ● **No** never have a caller leave the Queue until the Max Wait Time has expired.
Penalty Members Limit	Asterisk: penalty members limit. A limit can be set to disregard penalty settings, allowing all members to be tried, when the queue has too fewer members. No penalty will be weight in if there are only X or fewer queue members.
Frequency	How often to announce queue position and estimated hold time (0 to Dis able Announcements).
Announce Position	Announce position of caller in the queue
Announce Hold Time	Should we include estimated hold time in position announcements? Either yes, no, or only once; hold time will not be announced if <1 minute.
IVR Break Out Menu	You can optionally present an existing IVR as a 'break out' menu. This IVR must only contain single-digit 'dialed options'. The recording set for the IVR will be played at intervals specified in 'Repeat Frequency', below.
Repeat Frequency	How often to announce a voice menu to the caller (0 disable Announcements)
Event When Called	When this option is set to YES, the following manager events will be generated: AgentCalled, AgentDump, AgentConnect and AgentComplete.
Member Status Event	When set to YES, the following manager event will be generated: QueueMemberStatus.
Service Level	Used for service level statistics (calls answered within service level time frame)
Agent Regex Filter	Provides an optional regex expression that will be applied against the agent callback number. If the callback number does not pass the regex filter then it will be treated as invalid. This can be used to restrict agents to extensions within a range, not allow callbacks to include keys like *, or any other use that may be appropriate. An example input might be:

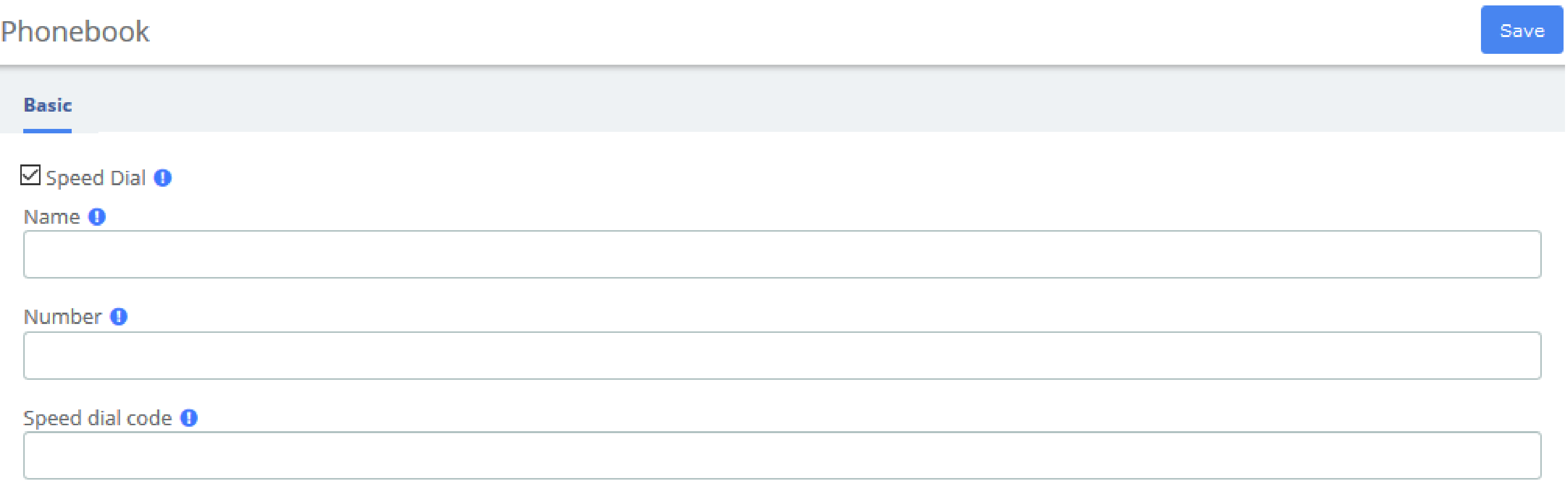
	^([2-4][0-9]{3})$ This would restrict agents to extensions 2000-4999. Or ^([0-9]+)$ would allow any number of any length, but restrict the * key. WARNING: make sure you understand what you are doing or otherwise leave this blank!
Failover Destination	Set the failover destination for the queue.
Run	Select how often to reset queue stats. The following schedule will be followed for all but custom: Hourly Run once an hour, beginning of hour Daily Run once a day, at midnight Weekly Run once a week, midnight on Sun Monthly Run once a month, midnight, first of month Annually Run once a year, midnight, Jan.1 Reboot Run at startup of the server OP of the cron deamon (i.e. after every service cron restart) If Randomize is selected, a similar frequency will be followed, only the exact times will be randomized (avoiding peak business hours, when possible). Please note: randomized schedules will be rescheduled (randomly) every time ANY backup is saved.

3.4.3 Phonebook

With the Phonebook module, we can have a centralized list of numbers that can be accessed by the users. Each number of this list has a special code in order to dial it quicker than by dialing the number itself.

Navigate to **PBX > Call Control > Phonebook**, add a speed dial number by using the following information.

The next screenshot shows this configuration:



Item	Definition
Speed Dial	This option must be checked
Name	Name of the speed dial
Number	Destination external number
Speed dial code	A number to associate this code to the external number to dial

To dial this speed dial number, we dial *088, where *0 is to access the speed dial system's feature and 88 is the speed dial code we entered.

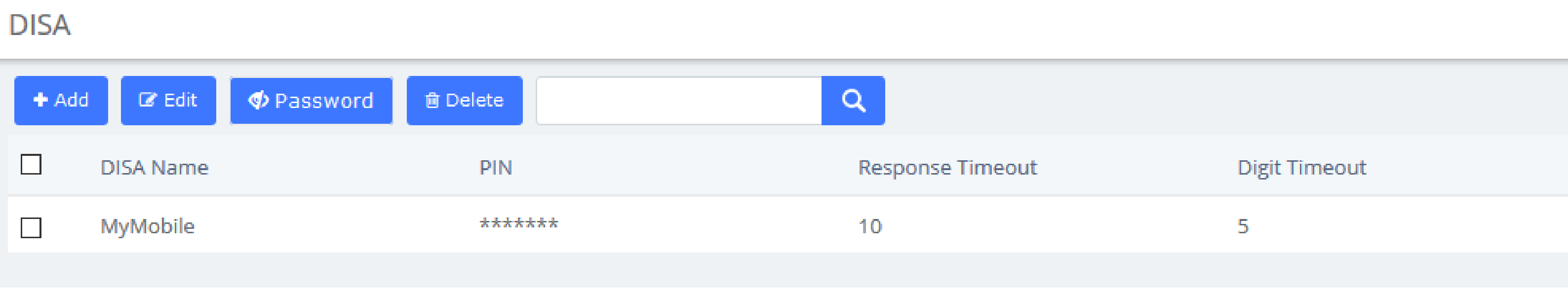
Some actions that we can perform on the speed dial administration web page are as follows:

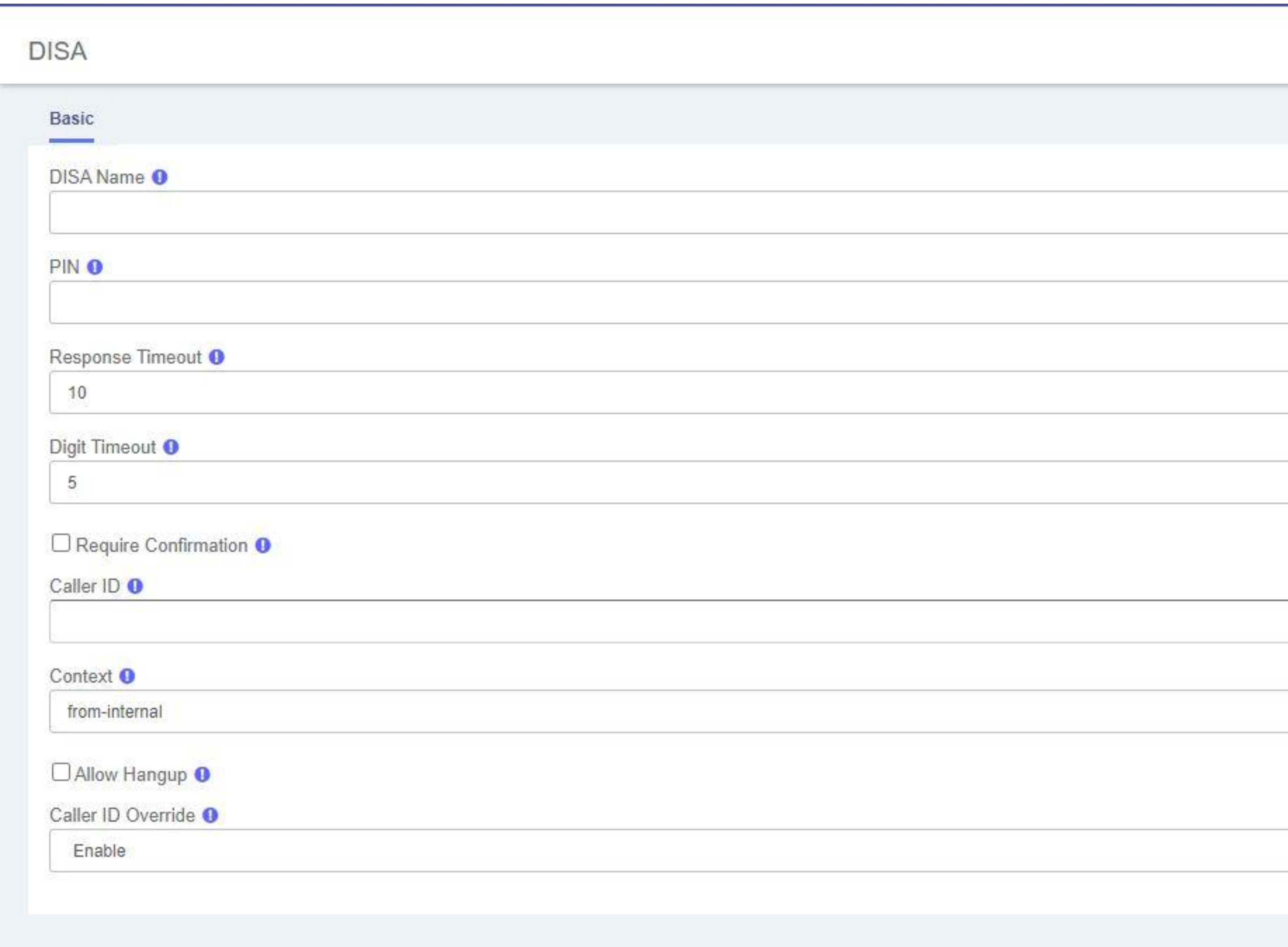
- Export in CSV: If we click on this link, we can download the current speed dial list.
- Import from CSV: We can upload a CSV file with the format: “Name”; Number; Speed dial

Navigate to **PBX > Settings > Functions Code,** switch the Speed dial prefix to Enabled.

3.4.4 DISA

DISA (Direct Inward System Access) allows you to dial in from outside to the Asterisk switch (PBX) to obtain an "internal" system dial tone. You can place calls from it as if they were placed from within.





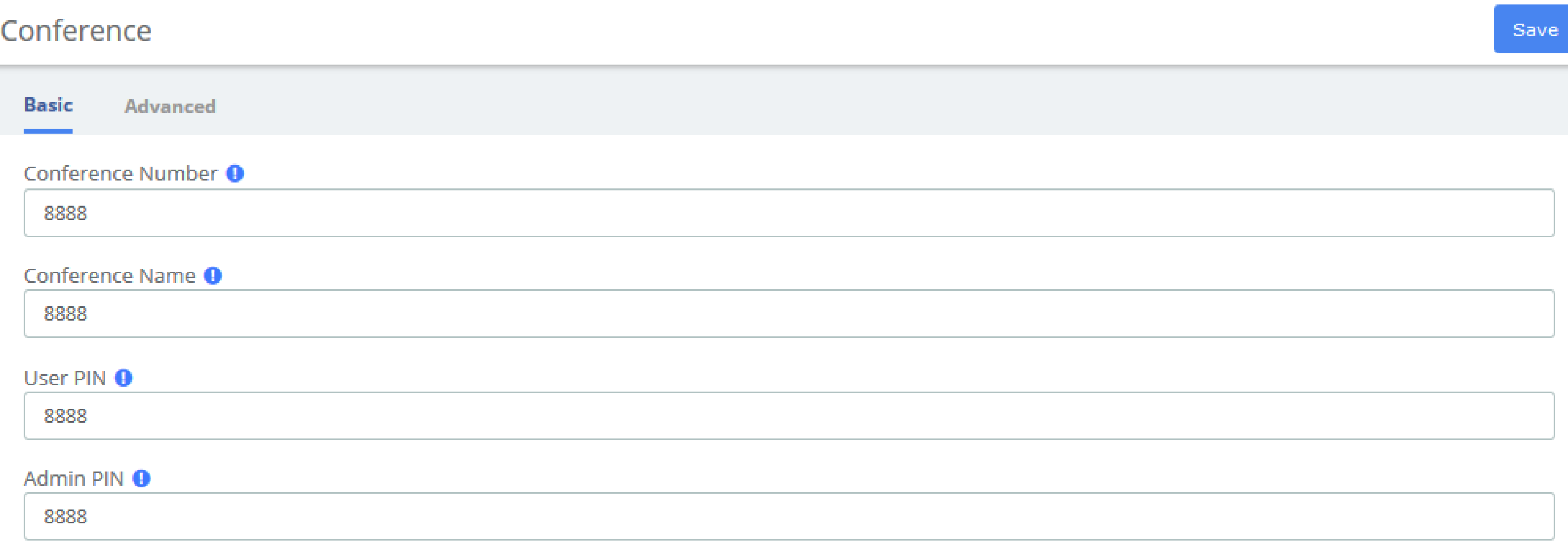
Item	Definition
DISA name	Give this DISA a brief name to help you identify it.
PIN	The user will be prompted for this number. If you wish to have multiple PIN's, separate them with commas.
Response Timeout	The maximum amount of time it will before hanging up if the user has dialed an incomplete or invalid number. Default of 10 seconds.
Digit Timeout	The maximum amount of time permitted between digits when the user is typing in an extension. Default of 5.
Require Confirmation	Require Confirmation before prompting for password. Used when your PSTN connection appears to answer the call immediately.
Caller ID	(Optional) When using this DISA, the users CallerID will be set to this. Format is "User Name" <5551234>
Context	(Experts Only) Set the context that calls will originate from. Leaving this as from-internal unless you know what you're doing.
Allow Hangup	Allow the current call to be disconnected and dial tone presented for a new call by pressing the Handup feature code: ** while in a call.
Caller ID Override	Determine if we keep the Caller ID being presented or if we override it. Default is Enable.

After configuring DISA when you choose the DISA option to call a number, you will be greeted with "Please enter your password followed by the pound key" and after entering your password, you will then get a dial tone. You may start dialing the telephone number.

3.4.5 Conference

The Conference option is used to create a single extension number that your users can dial so that they can talk to each other in a conference call. It also creates a destination to which you can send calls so that they can participate in the conference call.

For example, you could create a Conference that will allow your local phones to dial 800, and then enter into a conference call.



Item	**Definition**
	Basic
Conference Number	Use this number to dial into the conference.
Conference Name	Give this conference a brief name to help you identify it.
User PIN	You can require callers to enter a password before they can enter this conference. This setting is optional. If either PIN is entered, the user will be prompted to enter a PIN.
Admin PIN	Enter a PIN number for the admin user. This setting is optional unless the ‘leader wait’ option is in use, then this PIN will identify the

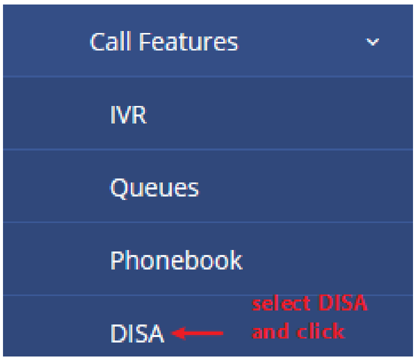
	leader.
Advanced	
Join Message	Message to be played to the caller before joining the conference. To add additional recordings use the "System Recordings" MENU to the left
Leader Wait	Wait until the conference leader (admin user) arrives before starting the conference
Talker Optimization	Turn on talker optimization. With talker optimization, Asterisk treats talkers who are not speaking as being muted, meaning that no encoding is done on transmission and that received audio that is not registered as talking is omitted, causing no buildup in background noise.
Talker Detection	Sets talker detection. Asterisk will send events on the Manager Interface identifying the channel that is talking. The talker will also be identified on the output of the meet me list CLT command.
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
Music on Hold	Enable Music on Hold when the conference has single caller
Music on Hold Class	Music (or Commercial) played to the caller while they wait line for the conference to start. Choose "inherit" if you want the MoH class to be what is currently selected, such as by the inbound route. This music is defined in the "Music on Hold" to the left.
Allow Menu	Present Menu (user or admin) when '*' is received ('send' to menu).
Record Conference	Record the conference call
Maximum Participants	Maximum Number of users allowed to join this conference.
Mute on Join	Mute everyone when they initially join the conference. Please note that if you do not have 'Leader Wait' set to yes you must have 'Allow Menu' set to Yes to unmute yourself.

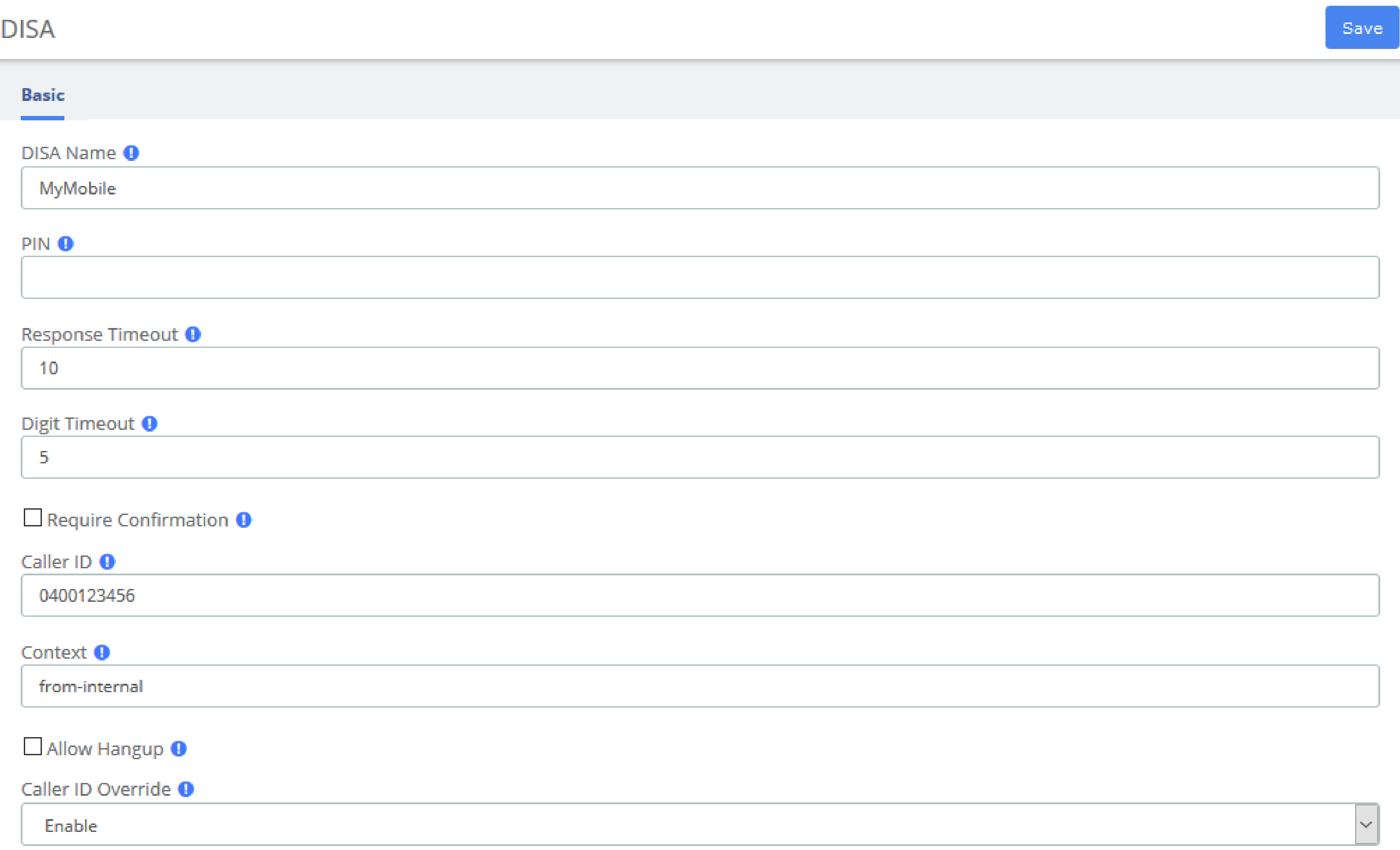
3.4.6 Callback

Callback is where you make a call to your IP-PBX and when reached you will be disconnected, but it does not end there. Your PBX will in turn call your mobile and reconnect you relieving you of the cost of the lengthy Mobile phone call that you will otherwise be up for.

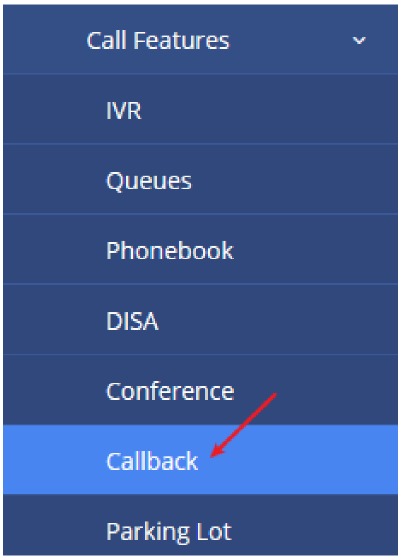
Let's take this step by step.

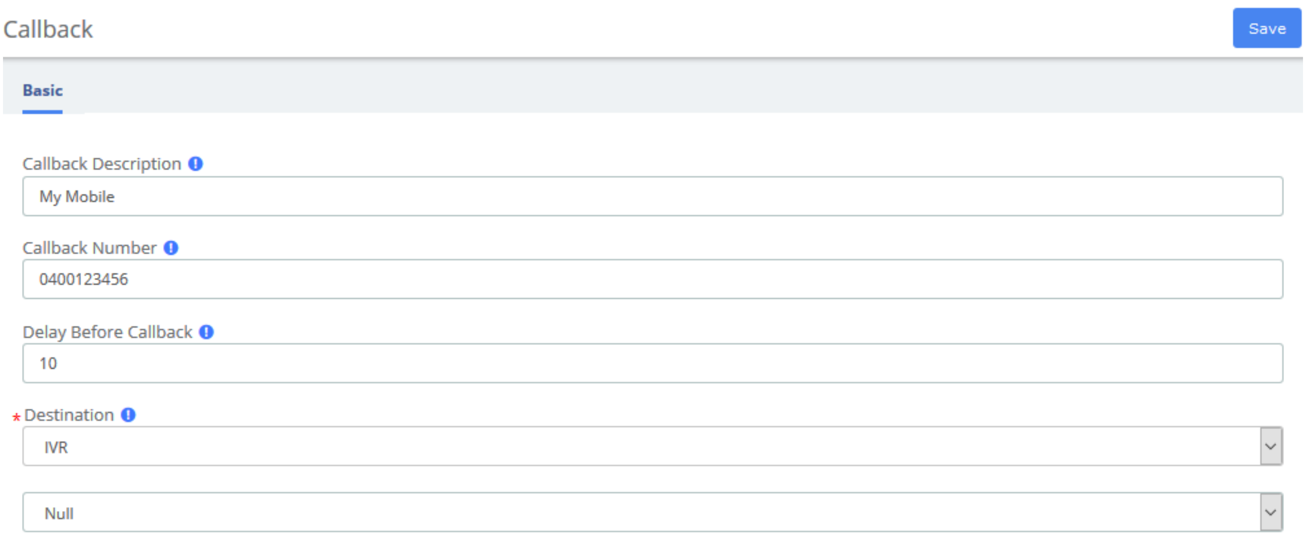
1. **Setup DISA**





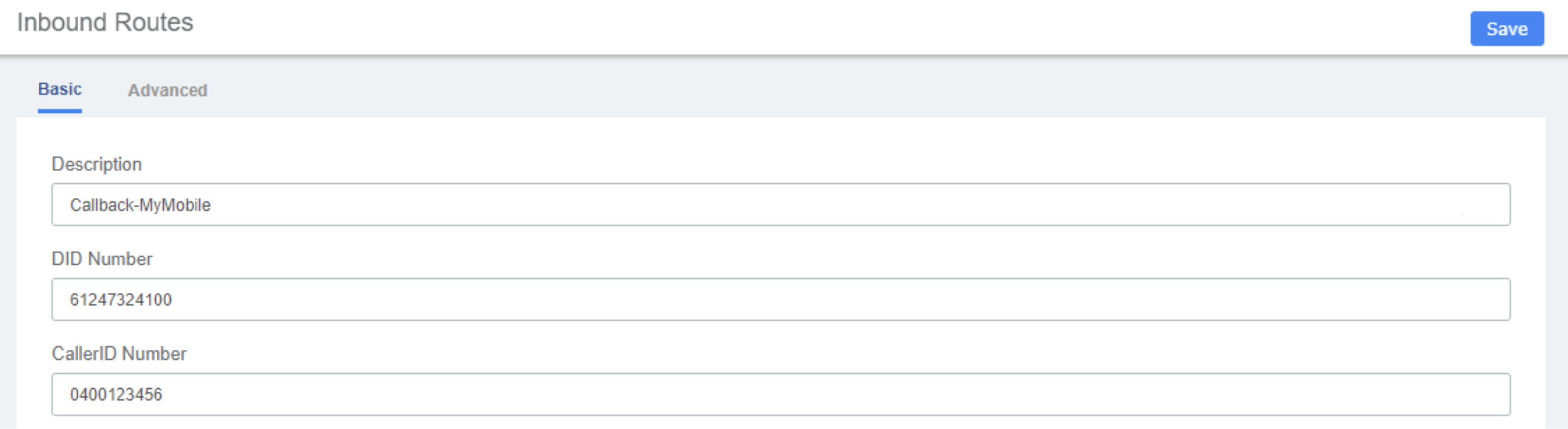
2. Setup Callback





Item	Definition
Callback Description	Enter a description for this callback
Callback Number	Optional: Enter the number to dial for the callback. Leave this blank to just dial the incoming Caller ID Number.
Delay Before Callback	Optional: Enter the number of seconds the system should wait before calling back.
Destination	Destination of a callback.

3. Inbound Routes



Click Save button then Click on the red circle at the top & follow on screen prompts

Now enable send caller ID on your mobile and call your DID number. When connected you will get one beep and then followed by silence. Hang up your mobile and wait for approximately10 seconds and your mobile will ring.

When you answer your mobile, you will hear your IVR playing with the various options. One of the silent options in my IVR is DISA. If I need to make an external call using my PBX. If I know the option and select it, I will then get DISA where I can make an external call at no cost to my Mobile.

3.4.7 Parking Lot

This module is used to configure Parking Lot(s) in Asterisk.

Simply transfer the call to said parking lot extension. Asterisk will then read back the parking lot number the call has been placed in. To retrieve the call simply dial that number back.

Parking Lot

Basic Advanced

Parking Lot Extension
*6

Parking Lot Name
Default Lot

Parking Lot Starting Position
80

Number of Slots
4
(80-83)

Parking Timeout (seconds)
45

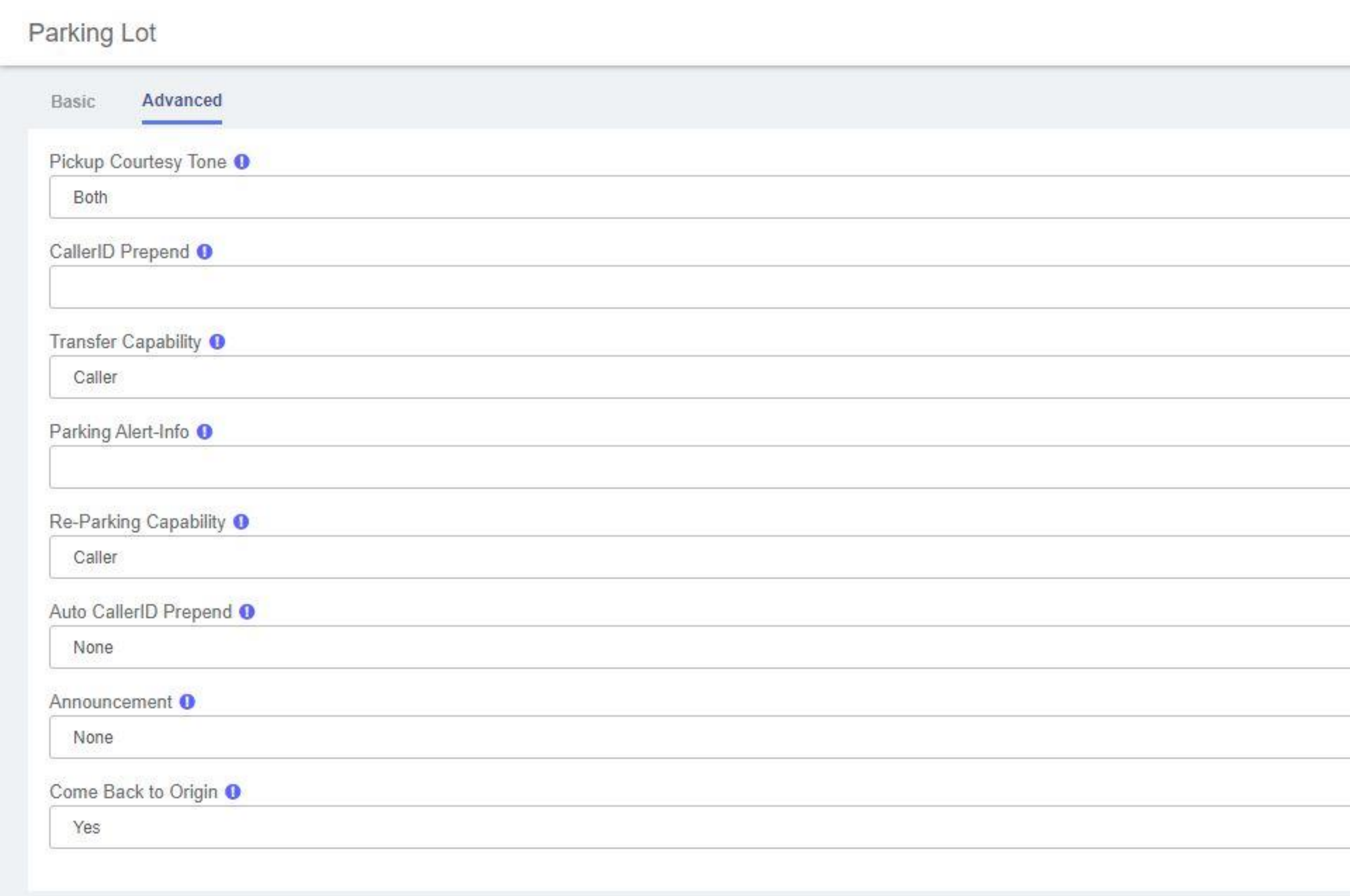
Parked Music Class
Default

BLF Capabilities
Enable

Find Slot
First

Destination
Terminate Call

Hangup

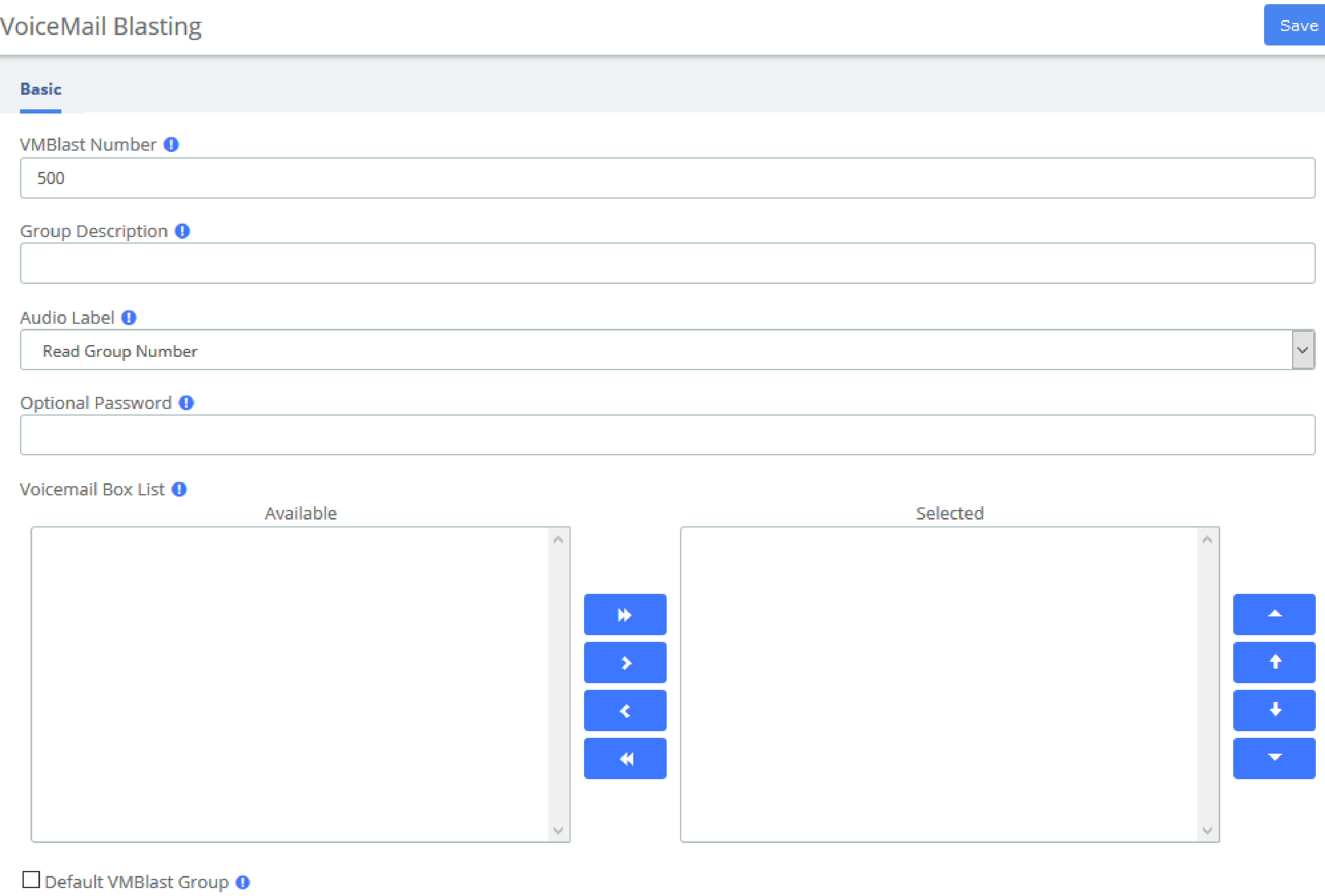


Item	Definition
Basic	
Parking Lot Extension	This is the extension where you will transfer a call to park it
Parking Lot Name	Name of the parking Lot.
Parking Lot Starting Position	The starting position of the parking lot.
Number of Slots	The total number of parking lot spaces to configure.
Parking Timeout (seconds)	The timeout period in seconds that a parked call will attempt to ring back the original parker if not answered.
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
BLF Capabilities	Enable this to have Asterisk “hints” generated to use with BLF buttons.
Find Slot	If you want the parking lot to seek the next sequential parking slot relative to the last parked call instead of seeking the first available slot.
Destination	Destination of Parking Lot.
Advanced	
Pickup Courtesy Tone	Whom to play the courtesy tone to when a parked call is retrieved.
CallerID Prepend	String to prepend to the current Caller ID associated with the parked call prior to sending back to the Originator or the Alternate Destination.
Transfer Capability	Parked call transfers. Enables or disables DTMF based transfers when picking up a parked call.
Parking Alert-Info	Alert-Info to add to the call prior to sending back to the Originator or to the Alternate Destination.
Re-Parking Capability	Parked call reparking. Enables or disables DTMF based parking when picking up a parked call
Auto CallerID Prepend	These options will be appended after Caller ID Prepend if set.
Announcement	Optional message to be played to the call prior to send back to the originator.
Come Back to Origin	Where to send a parked call that has timed out. If set to yes then the parked call will be sent back to the originating device that sent the call to this parking lot.
*2nn:	Attended Transfer call into Park lot nnn (It will announce the slot back to you)
nn:	Park Yourself into Parking lot nnn (Announcing your parked slot to you)

3.4.8 Voicemail Blasting

Voicemail blasting lets you send a voicemail message to multiple users at the same time. The Voicemail Blasting module is used to create a group of users and assign a number to the group. A user can dial this number to leave a voicemail message for the group. All members of the group will receive the message in their voicemail boxes.



Item	Definition
VMBlast Number	The number users will dial to voicemail boxes in this VMBlast group
Group Description	Provide a descriptive title for this VMBlast Group.

Audio Label	Paly this message to the caller so they can confirm they have dialed the proper voice mail group number, or have the system simply read the group number.
Optional Password	You can optionally include a password to authenticate before providing access to this group voicemail list.
Voicemail Box List	Select voice mail boxes to add to this group. Use Ctrl key to select multiple.
Default VMBlast Group	Each PBX system cam have a single Default VOICEMAIL Blast Group. If specified, extensions can be automatically added (or removed) from this default group in the Extensions (or Users) tab. Making this group the default will uncheck the option from the current default group if specified.

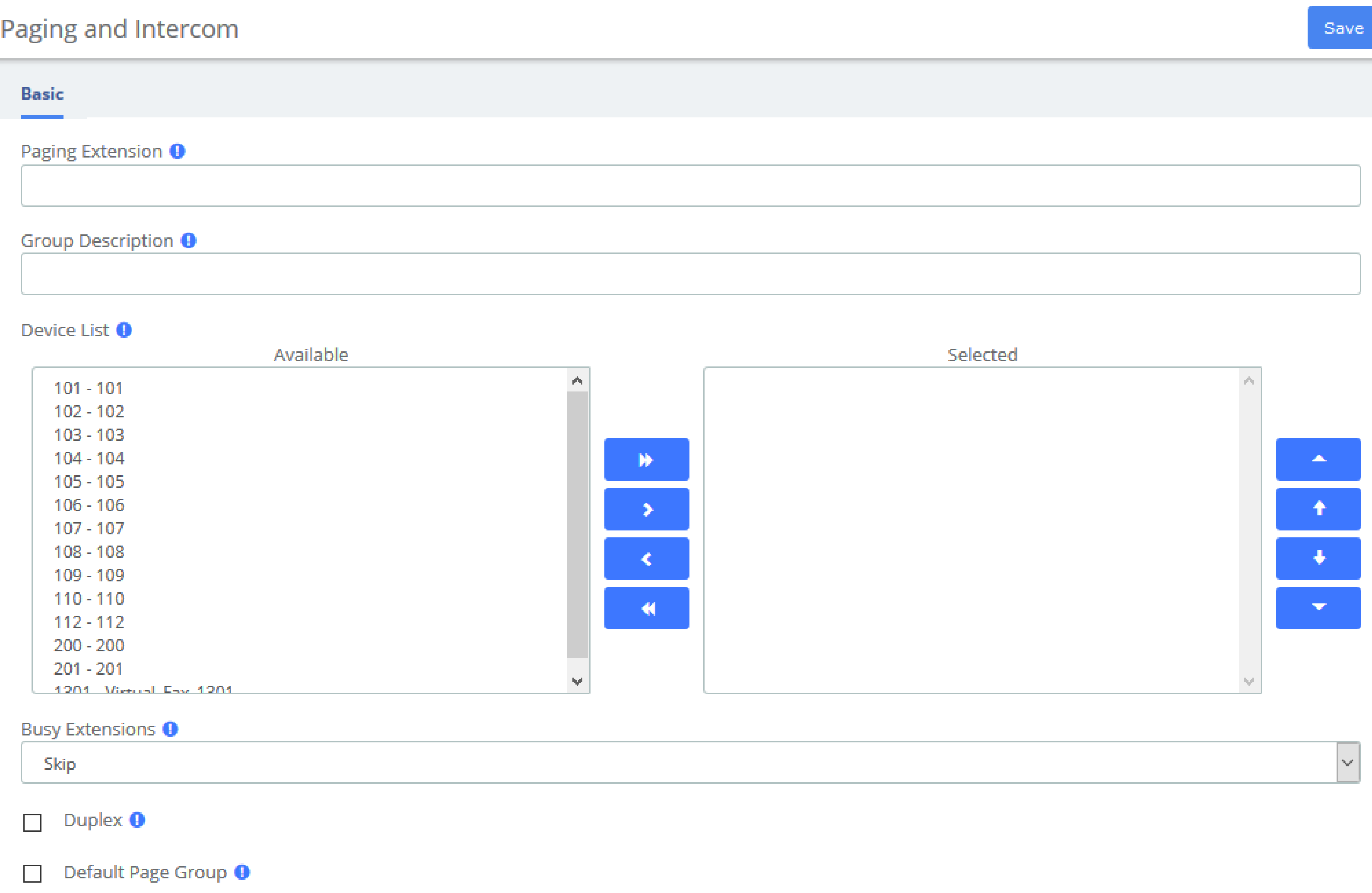
3.4.9 Paging and Intercom

The Paging and Intercom module is used to set up an extension number that your users can dial in order to place an intercom call to multiple phones on your system at the same time.

For example, in a small office, you might set up a page group with extension number "100."   When 100 is dialed by a local user, all of the phones in the office would go off-hook, and you could speak to everyone at every extension at the same time. Alternatively, you could set up page groups with different extension numbers for each department in the office, i.e. 100 for sales, 110 for service, and so on.

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. Intercom must be enabled on a handset before it will allow incoming calls. It is possible to restrict incoming intercom calls to specific extensions only, or to allow intercom calls from all extensions but explicitly deny from specific extensions.

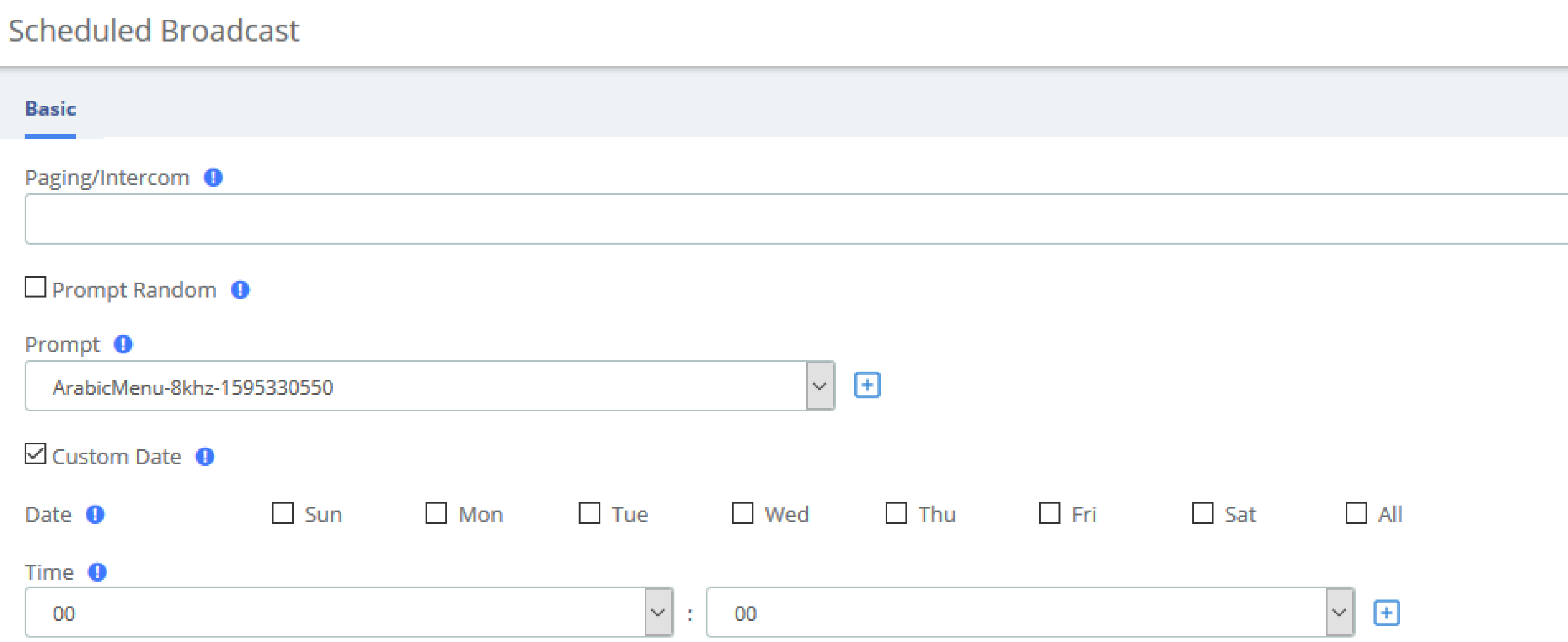
This module should work with Aastra, Grandstream, Linksys/Sipura, Mitel, Polycom, SNOM, and possibly other SIP phones (not ATAs). Any phone that is always set to auto-answer should also work (such as the console extension if configured). Intercom mode is currently disabled; it can be enabled in the Feature Codes Panel.



Item	Definition
Paging Extension	The number users will dial to page this group.
Group Description	Provide a descriptive title for this VMBlast Group.
Device List	Choose extensions.
Busy Extensions	Skip will not page any busy extension. All other extensions will be paged as normal Force will not check if the device is in use before paging it. This means conversations can be interrupted by a page (depending on how the device handles it).
Duplex	Paging is typically one way for announcements only Checking this will make the paging duplex, allowing all phones in the paging group to be able to talk and be heard by all.

3.4.10 Scheduled Broadcast

You can broadcast some audio by setting up the scheduled broadcast feature to inform the group.

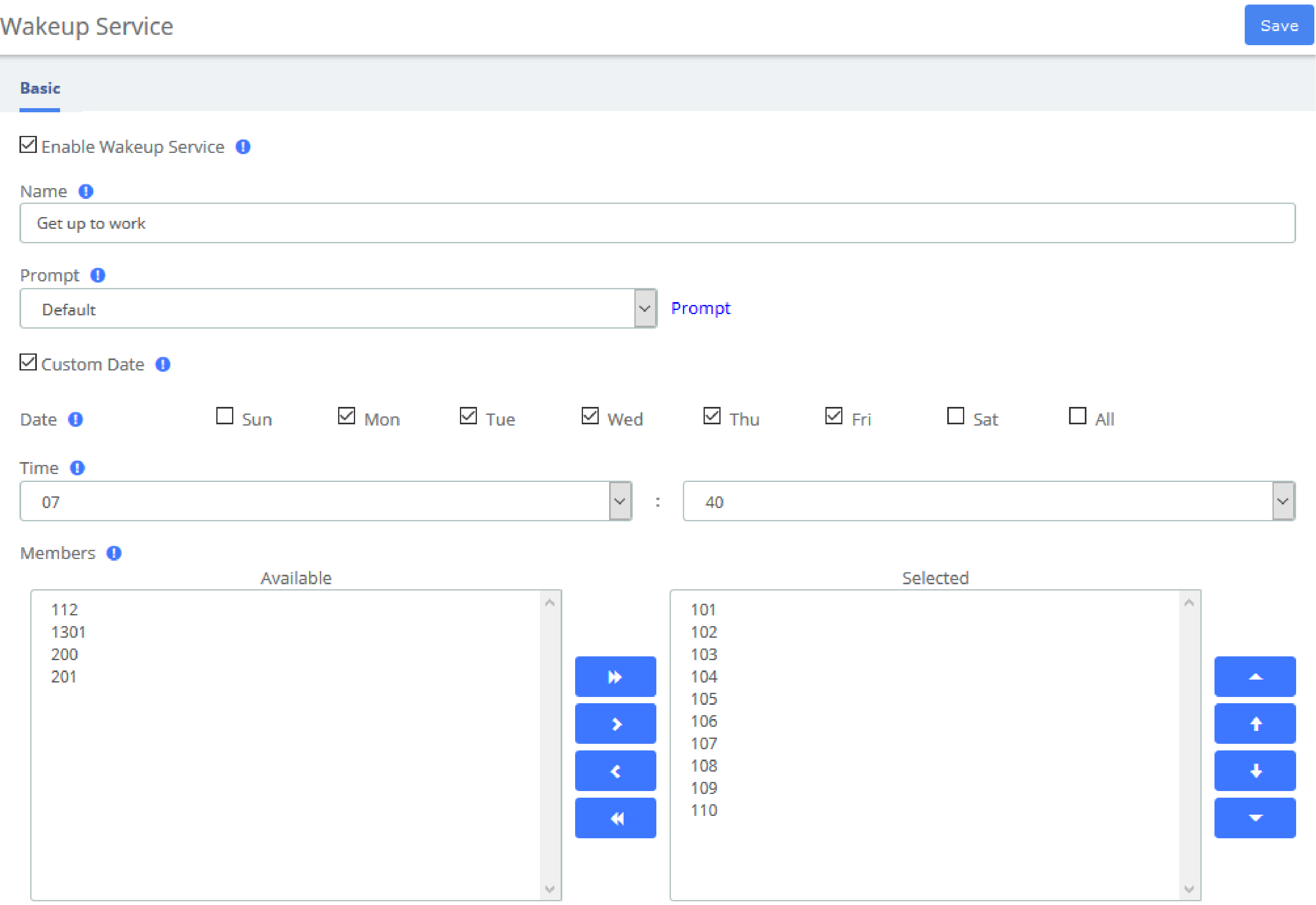


Item	Definition
Paging/Intercom	Select the desired paging group or intercom group.
Prompt Random	If enabled, the prompt should be played randomly.
Prompt	Select the desired prompt.
Custom Date	You can select a time to play the scheduled broadcast on a special day.

3.4.11 Wakeup Service

User can enable the Wakeup service and set the time and date, members, and receive the call reminder after the time. The wake-up service will ring for 30 seconds every 30 seconds for the duration of the wake-up service.

In the following example, the wakeup service is set up for extension 101-110 from Monday to Friday on 7:40AM.

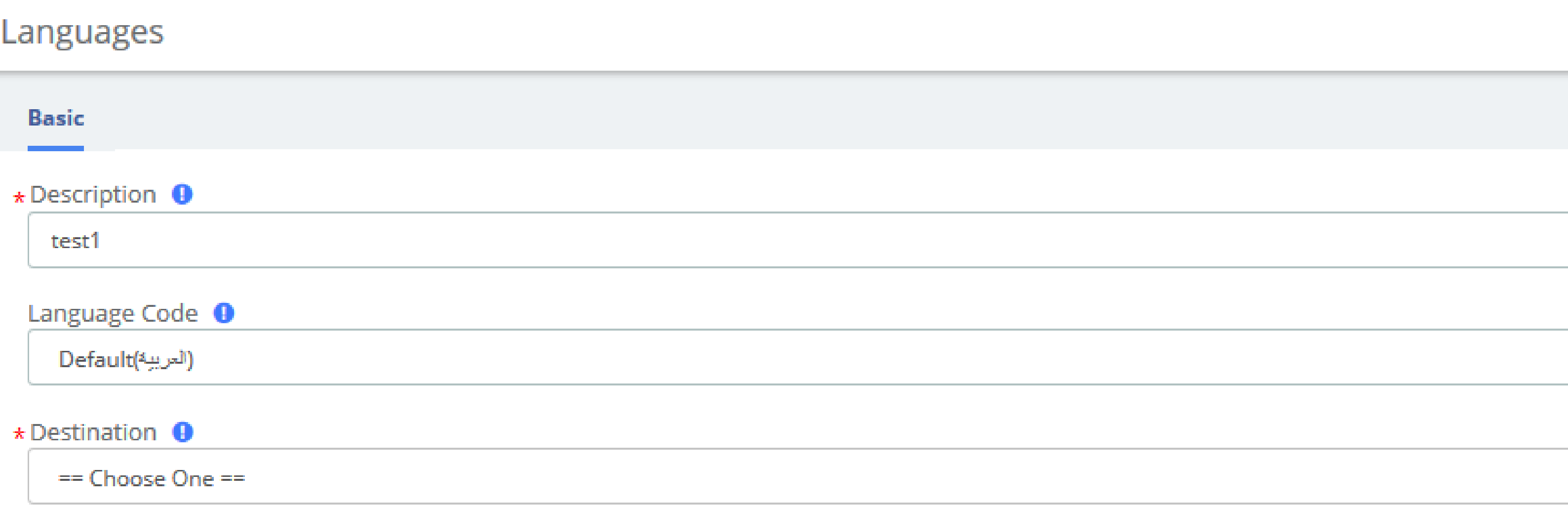


Wakeup Service Interface

3.5 Voice Prompts

3.5.1 Languages

The Languages module is used to allow calls to be routed to localized or alternate language recordings.



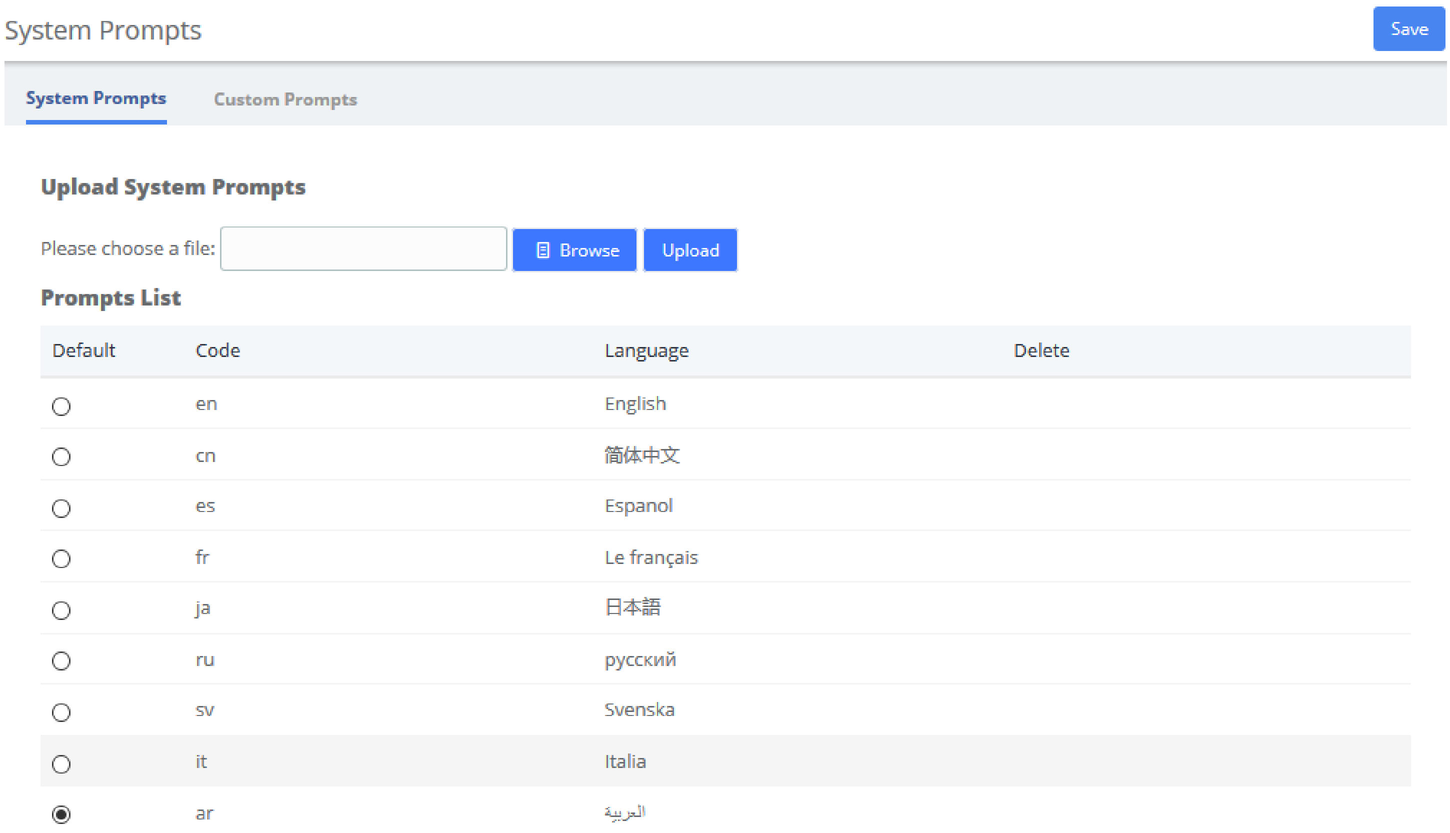
Languages allow you to change the language of the call flow and then continue on to the desired destination. For example, you may have an IVR option that says "For French Press 5 now". You would then create a French language instance and point its destination at a French IVR. The language of the call's channel will now be in French. This will result in French sounds being chosen if installed.

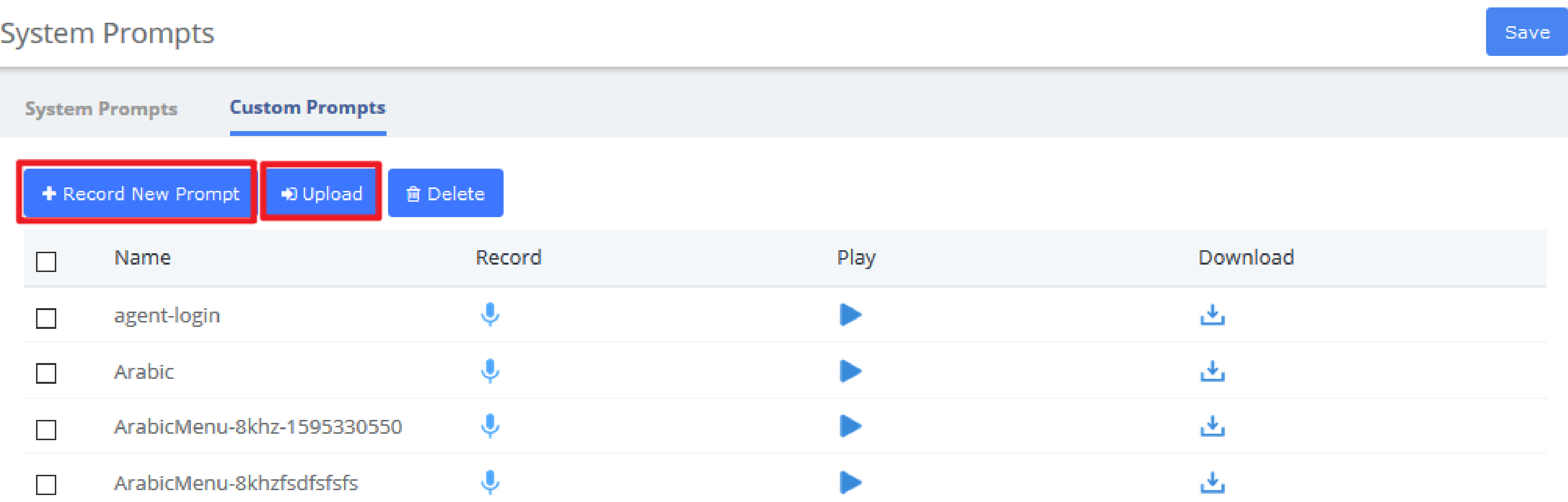
Item	**Definition**
Description	The descriptive name of this language instance. For example, “French Main IVR”
Language Code	The Asterisk language code you want to change to. For example, “fr” for French.
Destination	Indicates extension, Ring Group, Voicemail or other destination to which the call is supposed to be directed when the outside callers have called specified

3.5.2 System Prompts

The System Prompts module is used to record or upload messages that can be played back to callers in other modules. It can also be used to make pre-installed Asterisk recordings available for use in other modules.

For example, you might create a recording called "Main Menu" and then play that message in an IVR before a caller is asked to select. Or, you might record a recording called "Holiday Message" and then use that message in an Announcement. You would then route incoming calls to the Announcement or IVR using the Inbound Routes Module.

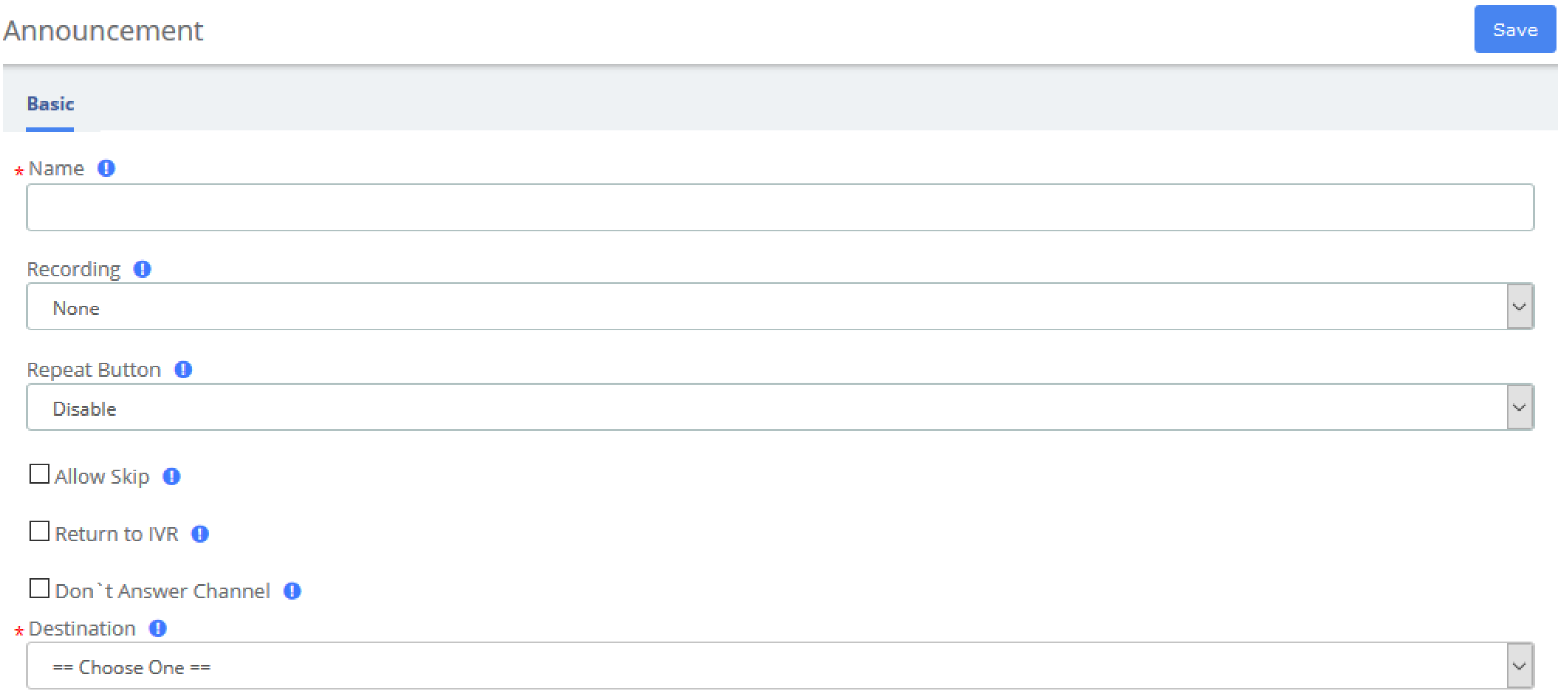




3.5.3 Announcement

The Announcements Module is used to create a destination that will play an informational message to a caller. After the message is played, the call will proceed to another destination.

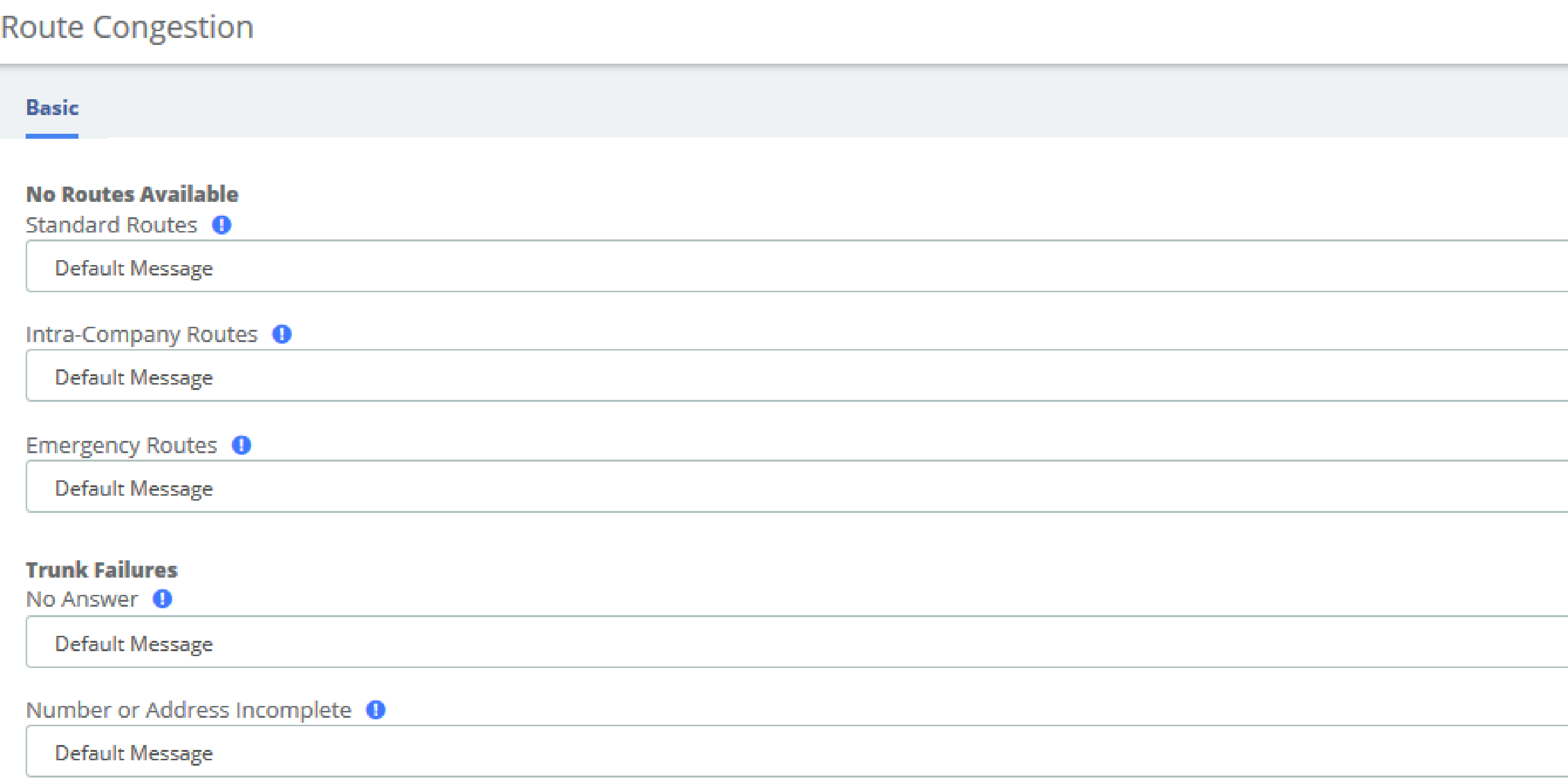
For example, you might create an Announcement that plays the address, fax number, and the web-site of your business. A caller could reach that message by pressing the number 2 from the company's main menu. After hearing the message, the call might be routed back to the company's main menu and allowed to make another selection.



Item	**Definition**
Name	The name of this announcement
Recording	Message to be played. To add additional recordings, use the “System Recordings” MENU to the left
Repeat	Key to press that will allow for the message to be replayed. If you choose this option there will be a short delay inserted after the message. If a longer delay is needed it should be incorporated into the recording.
Allow Skip	If the caller is allowed to press a key to skip the message
Return to	If the announcement came from an IVR and this box is checked, the destination below will

IVR	be ignored and instead it will be return to the calling IVR. Otherwise, the destination below will be taken. Don't check if not using in this mode. The IVR return location will be to the last IVR in the call chain that was called so be careful to only check when needed. For example, if an IVR directs a call to another destination which eventually calls this announcement and this box is checked, it will return to that IVR which may not be the expected behavior.
Don't Answer Channel	Check this to keep the channel from explicitly being answered. When checked, the message will be played and if the channel supports that. When not checked, the channel is answered followed by a 1 second delay. When using an announcement from an IVR or other sources that have already answered the channel, that 1 second delay may not be desired.
Destination	Indicates extension, Ring Group, Voicemail or other destination to which the call is supposed to be directed when the outside callers have called specified

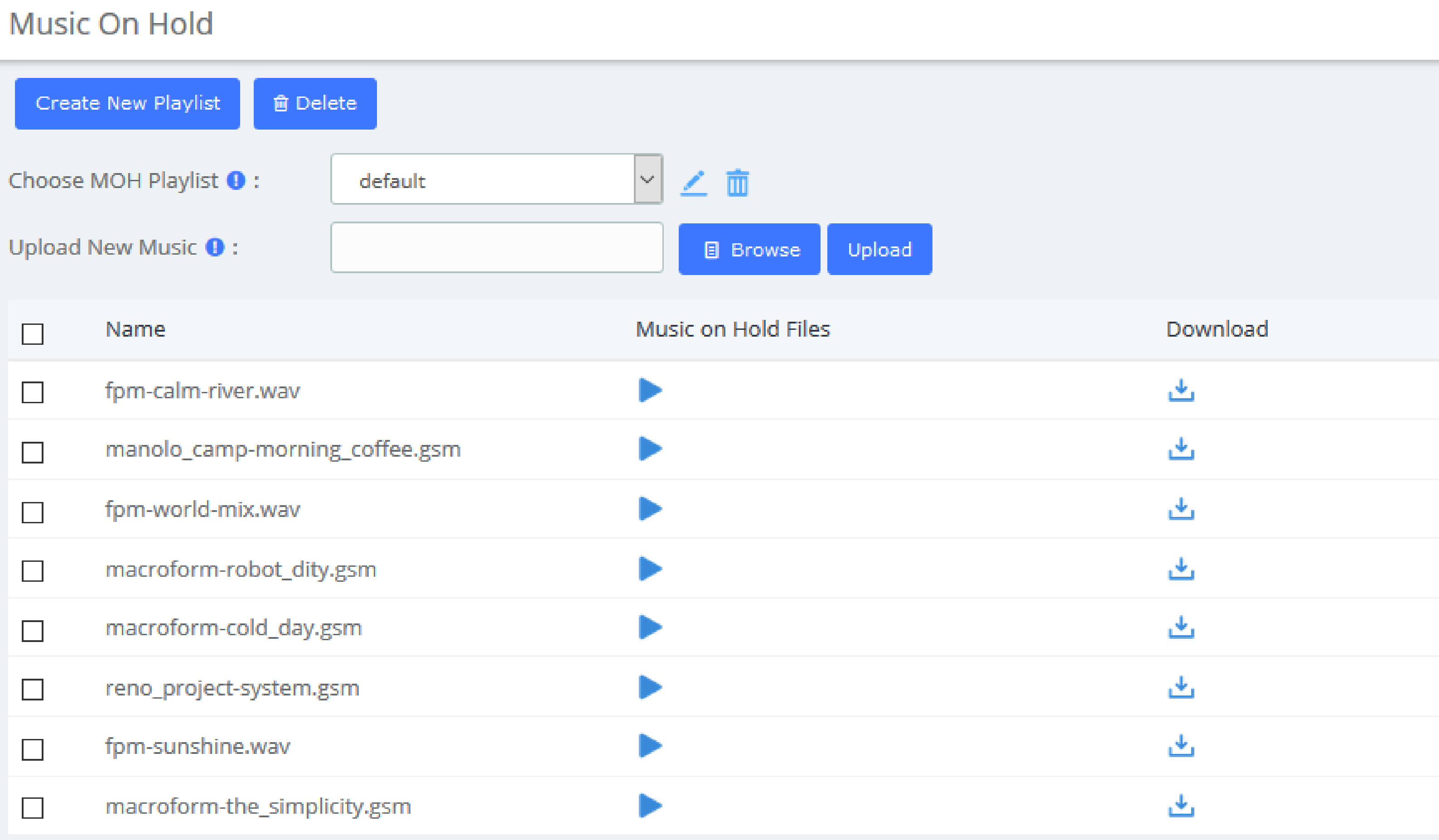
3.5.4 Route Congestion



Item	Definition
No Routes Available	
Standard Routes	Message or tone to be played if no trunks are available.
Intra-Company Routes	Message or tone to be played if no trunks are available. Used on routes marked as intra-company only.
Emergency Routes	Message or tone to be played if no trunks are available. Used on all emergency routes. Consider a message instructing caller to find an alternative means of calling emergency services such as a cell phone or alarm system panel.
Trunk Failures	
No Answer	Message or tone to be played if there was no answer. Default message is: “The number is not answering." Hang-up cause is 18 or 19
Number or Address Incomplete	Message or tone to be played if trunk reports Number or Address Incomplete. Usually this means that the number you have dialed is to short. Default message is: “The number you have dialed is not in service. Please check the number and try again. “Hang-up cause is 28

3.5.5 Music On Hold

Music on hold is the business practice of playing recorded music to fill the silence that would be heard by callers who have been placed on hold. The PBX has a default music on hold playlists. You also can add the music on hold playlists and upload music files to the PBX.



Music on Hold Interface

You can add a custom music on hold playlist and upload your audio files to the PBX.

1. Add a Custom music on hold Playlist.

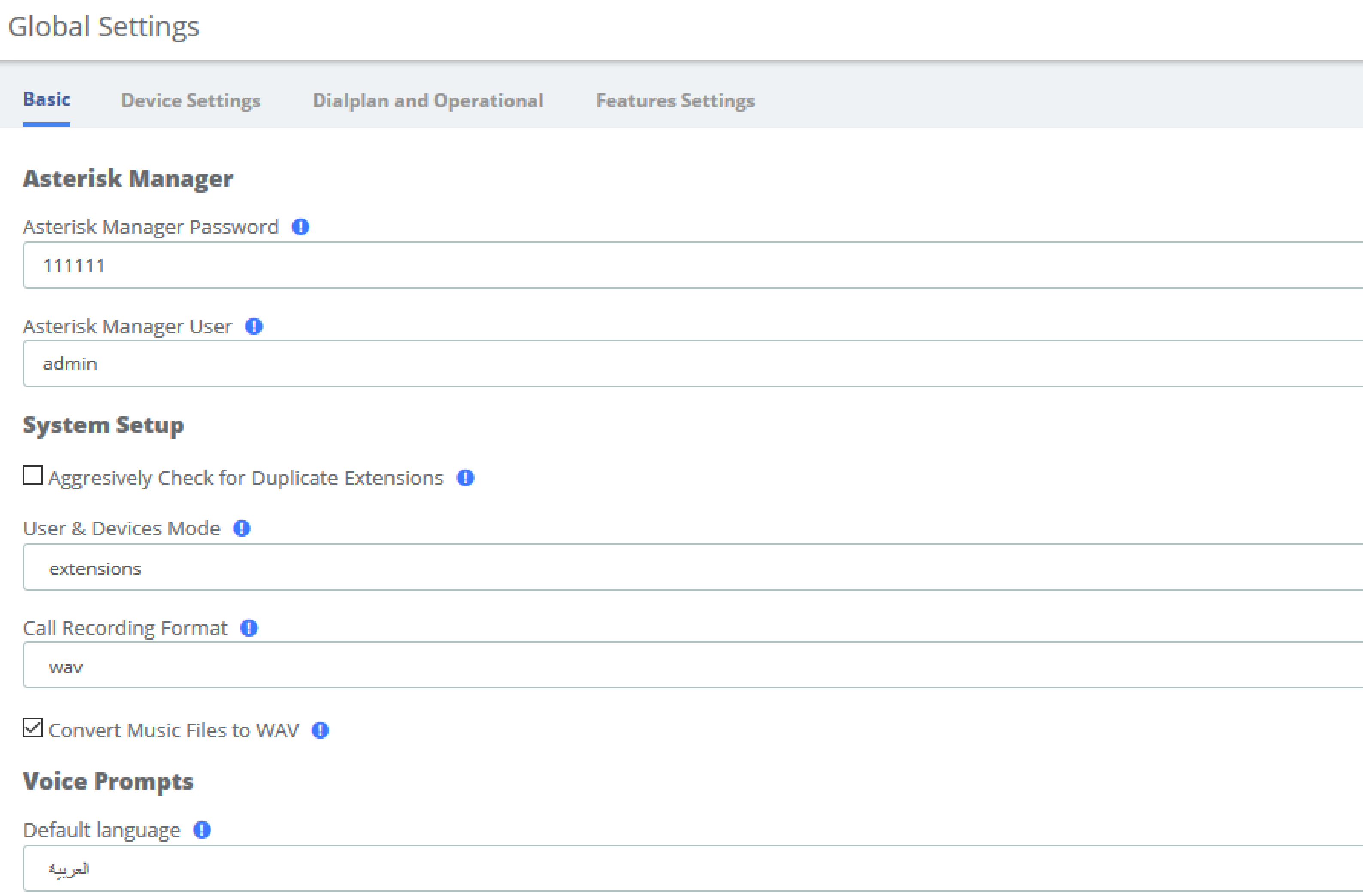
   Go to **PBX > Voice Prompts > Music on Hold** page, click Create New Playlist. On the configuration page, set the playlist name and the playlist order, click Save.

2. Upload the audio file.

   Click **Browse** to choose an audio file from your local PC, and then click **Upload**

3.6 Settings

3.6.1 Global Settings



Options	Definition
Asterisk Manager	
Asterisk Manager Password	Password for accessing the Asterisk Manager Interface (AMI), this will be automatically updated in manager.conf.
Asterisk Manager User	Username for accessing the Asterisk Manager Interface (AMI), this will be automatically updated in manager.conf.

System Setup	
Aggressively Check for Duplicate Extensions	Aggressively Check for Duplicate Extensions
User & Devices Mode	Sets the extension behavior in XonTel SMB device. If set to extensions, Devices and Users are administered together as a unified Extension, and appear on a single page. If set to deviceanduser, Devices and Users will be administered separately.
Call Recording Format	Format to save recorded calls for most call recording unless specified differently in specific applications.
Convert Music Files to WAV	When set to false, the MP3 files can be loaded and WAV files converted to MP3 in the MoH module. The default behavior of true assumes you have mpg123 loaded as well as sox and will convert MP3 files to WAV. This is highly recommended as MP3 files heavily tax the system and can cause instability on a busy phone system
Voice Prompts	
Default language	The default language for Voice Prompts.

Global Settings

Basic | Device Settings | Dialplan and Operational | Features Settings

Device Settings

- [ ] Show all Device Setting on Add
- [x] Require Strong Secrets
- [ ] Remove mailbox Setting when no Voicemail

SIP canrenivite (directmedia)

no

SIP trustrpid

yes

SIP sendrpid

pai

SIP nat

no

SIP encryption

no

SIP qualifyfreq

60

SIP and IAX qualify

yes

SIP and IAX allow

SIP and IAX disallow

SIP and DAHDi callgroup

SIP and DAHDi pickupgroup

Option	Definition
Show all Device Setting on Add	When adding a new extension/device, setting this to true will show most available device settings that are displayed when you edit the same extension/device. Otherwise, just a few basic settings are displayed.
Require Strong Secrets	Requires a strong secret on SIP and IAX devices requiring at least two numeric and non-numeric characters and 6 or more characters. This can be disabled if using devices that cannot meet these needs, or you prefer to put other constraints including more rigid constraints that this rule actually considers weak when it may not be.
Remove mailbox Setting when no Voicemail	If you enable this option, any fixed device associated with a user that has no voicemail configured will have the "mailbox=" setting removed in the generated technology configuration file such as sip_additional.conf. This will not affect the value in the GUI.
SIP canrenivite (directmedia)	Default setting for SIP canreinvite (same as directmedia). See Asterisk documentation for details.
SIP trustrpid	Default setting for SIP trustrpid. See Asterisk documentation for details.
SIP sendrpid	Default setting for SIP sendrpid. A value of 'yes' is equivalent to 'rpid' and will send the 'Remote-Party-ID' header. A value of 'pai' is only valid starting with Asterisk 1.8 and will send the 'P-Asserted-Identity' header. See Asterisk documentation for details.
SIP nat	Default setting for SIP nat. A 'yes' will attempt to handle nat, also works for local (uses the network ports and address instead of the reported ports), 'no' follows the protocol, 'never' tries to block it, no RFC3581, 'route' ignores the rport information. See Asterisk documentation for details
SIP encryption	Default setting for SIP encryption. Whether to offer SRTP encrypted media (and only SRTP encrypted media) on outgoing calls to a peer. Calls will fail with HANGUPCAUSE=58 if the peer does not support SRTP. See Asterisk documentation for details.
SIP qualifyfreq	Default setting for SIP qualifyfreq. Only valid for Asterisk 1.6 and above. Frequency that 'qualify' OPTIONS messages will be sent to the device. Can help to keep NAT holes open but not dependable for remote client firewalls. See Asterisk documentation for details.
SIP and IAX qualify	Default setting for SIP and IAX qualify. Whether to send periodic OPTIONS messages (for SIP) or otherwise monitor the channel, and at what point to consider the channel unavailable. A value of 'yes' is equivalent to 2000, time in msec. Can help

	to keep NAT holes open with SIP but not dependable for remote client firewalls. See Asterisk documentation for details.
SIP and IAX allow	Default setting for SIP and IAX allow (for codecs). Codecs to allow in addition to those set in general settings unless explicitly 'disallowed' for the device. Values van be separated with '&' e.g. 'ulaw&g729&g729' where the preference order is preserved. See Asterisk documentation for details.
SIP and IAX disallow	Default setting for SIP and IAX disallow (for codecs). Codecs to disallow, can help to reset from the general settings by setting a value of 'all' and then specifically including allowed codecs with the 'allow' directive. Values van be separated with '&' e.g. 'g729&g722'. See Asterisk documentation for details.
SIP and DAHDi callgroup	Default setting for SIP, DAHDi (and Zap) callgroup. Callgroup(s) that the device is part of, can be one or more callgroups, e.g. '1,3-5' would be in groups 1,3,4,5. See Asterisk documentation for details.
SIP and DAHDi pickupgroup	Default setting for SIP, DAHDi (and Zap) pickupgroup. Pickupgroups(s) that the device can pickup calls from, can be one or more groups, e.g. '1,3-5' would be in groups 1,3,4,5. Device does not have to be in a group to be able to pickup calls from that group. See Asterisk documentation for details.

Global Settings

Basic | Device Settings | Dialplan and Operational | Features Settings

Dialplan and Operational

☐ Block CNAM on External Trunks

Call Forward Ringtimer Default

0

Call Recording Policy

caller

Conference Room App

app_confbridge

☐ CW Enabled by Default

☐ Disable -custom Context Includes

Ditech VQA Inbound Setting

7

Global Settings

Ditech VQA Outbound Setting

7

☐ Enable Custom Device States

Extension Concurrency Limit

0

Extension Dial Timeout

45

☐ Feature Codes Beep Only

☐ Force All Internal Auto Answer

☐ Generate Diversion Headers

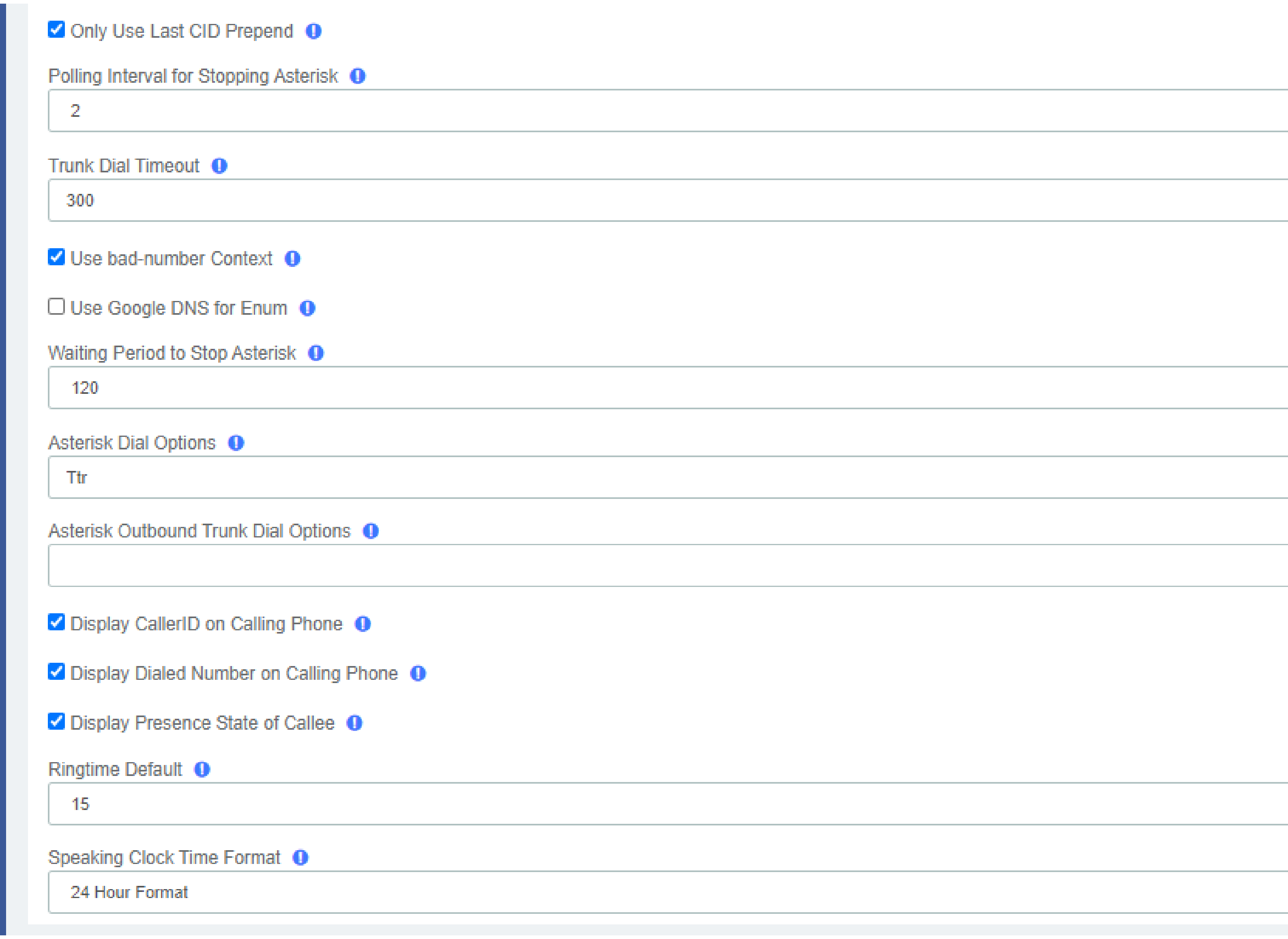
Internal Auto Answer Default

disabled

NoOp Traces in Dialplan

0

☑ Occupied Lines CW Busy



Option	Definition
Block CNAM on External Trunks	Some carriers will reject a call if a CallerID Name (CNAM) is presented. This occurs in several areas when configuring CID on the PBX using the format of 'CNAM' \<CNUM\>. To remove the CNAM part of CID on all external trunks, set this value to true. This WILL NOT remove CNAM when a trunk is called from an Intra-Company route. This can be done on each individual trunk in addition to globally if there are trunks where it is desirable to keep CNAM information, though most carriers ignore CNAM.
Call Forward Ringtimer Default	This is the default time in seconds to try and connect a call that has been call forwarded by the server side CF, CFU and CFB options. (If your phones use client side CF such as SIP redirects, this will not have any affect) If set to the default of 0, it will use the

	standard ring timer. If set to -1 it will ring the forwarded number with no limit which is consistent with the behavior of some existing PBX systems. If set to any other value, it will ring for that duration before diverting the call to the users voicemail if they have one. This can be overridden for each extension.
Call Recording Policy	Call Recording Policy used to resove the winner in a conflict between two extensions when one wants a call recorded and the other does not, if both their priorities are also the same.
Conference Room App	The asterisk application to use for conferencing. If only one is compiled into asterisk, PBX will auto detect and change this value if set wrong. The app_confbridge application is considered.
CW Enabled by Default	Enable call waiting by default when an extension is created (Default is yes). Set to no to if you do not want phones to be commissioned with call waiting already enabled. The user would then be required to dial the CW feature code (*70 default) to enable their phone. Most installations should leave this alone. It allows multi-line phones to receive multiple calls on their line appearances.
Disable -custom Context Includes	Normally PBX auto-generates a custom context that may be usable for adding custom dialplan to modify the normal behavior of PBX. It takes a good understanding of how Asterisk processes these includes to use this and in many of the cases, there is no useful application. All includes will result in a WARNING in the Asterisk log if there is no context found to include though it results in no errors. If you know that you want the includes, you can set this to true. If you comment it out PBX will revert to legacy behavior and include the contexts.
Ditech VQA Inbound Setting	If Ditech's VQA, Voice Quality application is installed, this setting will be used for all inbound calls. For more information 'core show application VQA' at the Asterisk CLI will show the different settings.
Ditech VQA Outbound Setting	If Ditech's VQA, Voice Quality application is installed, this setting will be used for all outbound calls. For more information 'core show application VQA' at the Asterisk CLI will show the different settings.
Enable Custom Device States	If this is set, it assumes that you are running Asterisk 1.4 or higher and want to take advantage of the func_devstate.c backport available from Asterisk 1.6. This allows custom hints to be created to support BLF for server side feature codes such as daynight, followme, etc..
Extension Concurrency Limit	Default maximum number of outbound simultaneous calls that an extension can make. This is also very useful as a Security Protection against a system that has been compromised. It will limit the number of simultaneous calls that can be made on the compromised extension. This default is used when an extension is created. A default of

	0 means no limit.
Extension Dial Timeout	How many seconds to try a call on your extension before giving up. This should normally be a very long time and is usually only changed if you have some sort of problematic extension. This is the Asterisk Dial Command timeout parameter.
Feature Codes Beep Only	When set to true, a beep is played instead of confirmation message when activating/de-activating: CallForward, CallWaiting, DayNight, DoNotDisturb and FindMeFollow.
Force All Internal Auto Answer	Force all extensions to operate in the Internal Auto Answer mode regardless of their individual settings. See 'Internal Auto Answer Default' for more information.
Generate Diversion Headers	If this value is set to true, then calls going out your outbound routes that originate from outside your PBX and were subsequently forwarded through a call forward, ring group, follow-me or other means, will have a SIP diversion header added to the call with the original incoming DID assuming there is a DID available. This is useful with some carriers that may require this under certain circumstances.
Internal Auto Answer Default	Default setting for new extensions. When set to Intercom, calls to new extensions/users from other internal users act as if they were intercom calls meaning they will be auto-answered if the endpoint supports this feature and the system is configured to operate in this mode. All the normal white list and black list settings will be honored if they are set. External calls will still ring as normal, as will certain other circumstances such as blind transfers and when a Follow Me is configured and enabled. If Disabled, the phone rings as a normal phone.
NoOp Traces in Dialplan	Some modules will generate lots of NoOp() commands proceeded by a 'TRACE'(trace_level) that can be used during development or while trying to trace call flows. These NoOp() commands serve no other purpose so if you do not want to see excessive NoOp()s in your dialplan you can set this to 0. The higher the number the more detailed level of trace NoOp()s will be generated.
Occupied Lines CW Busy	For extensions that have CW enabled, report unanswered CW calls as busy (resulting in busy voicemail greeting). If set to no, unanswered CW calls simply report as no-answer.
Only Use Last CID Prepend	Some modules allow the CNAM to be prepended. If a previous prepend was done, the default behavior is to remove the previous prepend and only use the most recent one. Setting this to false will turn that off allowing all prepends to be 'starcked' in front of one another.
Polling Interval for Stopping Asterisk	When Asterisk is stopped or restarted with the 'amportal stop/restart' commands, it does a graceful stop waiting for active channels to hangup. This sets the polling interval to

	check if Asterisk is shutdown and update the countdown timer.
Trunk Dial Timeout	How many seconds to try a call on your trunks before giving up. This should normally be a very long time and is usually only changed if you have some sort of problematic trunks. This is the Asterisk Dial Command timeout parameter.
Use bad-number Context	Generate the bad-number context which traps any bogus number or feature code and plays a message to the effect. If you use the Early Dial feature on some Grandstream phones, you will disable this option.
Use Google DNS for Enum	Setting this flag will generate the required global variable so that enumlookup.agi will use Google DNS 8.8.8.8 when performing an ENUM lookup. Not all DNS deals with NAPTR record, but Google does. There is a drawback to this as Google tracks every lookup. If you are not comfortable with this, do not enable this setting. Please read Google FAQ about this: <b>http://code.google.com/speed/public-dns/faq.html#privacy.
Waiting Period to Stop Asterisk	When Asterisk is stopped or restarted with the 'amportal stop/restart' commands, it does a graceful stop waiting for active channels to hangup. This sets the maximum time in seconds to wait prior to force stopping Asterisk.
Asterisk Dial Options	Options to be passed to the Asterisk Dial Command when making internal calls or for calls ringing internal phones. The options are documented in Asterisk documentation, a subset of which are described here. The default options T and t allow the calling and called users to transfer a call with ##. The r option allows Asterisk to generate ringing back to the calling phones which is needed by some phones and sometimes needed in complex dialplan features that may otherwise result in silence to the caller.
Asterisk Outbound Trunk Dial Options	Options to be passed to the Asterisk Dial Command when making outbound calls on your trunks when not part of an Intra-Company Route. The options are documented in Asterisk documentation, a subset of which are described here. The default options T and t allow the calling and called users to transfer a call with ##. It is HIGHLY DISCOURAGED to use the r option here as this will prevent early media from being delivered from the PSTN and can result in the inability to interact with some external IVRs.
Display CallerID on Calling Phone	When you enable this option and when CONNECTEDLINE() capabilities are configured and supported by your handset, the CID value being transmitted on this call will be updated on your handset in the CNAM field prepended with CID: so you know what is being presented to the caller if the outbound trunk supports and honors setting the transmitted CID.
Display Dialed Number on Calling Phone	When you enable this option and when CONNECTEDLINE() capabilities are configured and supported by your handset, the number actually dialled will be updated on your handset in the CNUM field. This allows you to see the final manipulation of

	your number after outbound route and trunk dial manipulation rules have been applied. For example, if you have configured 7 digit dialing on a North America dialplan, the ultimate 10 or 11 digit transmission will be displayed back. Any 'Outbound Dial Prefixes' configured at the trunk level will NOT be shown as these are foten analog line pauses (w) or other characters that distort the CNUM field on updates.
Display Presence State of Callee	When set to true and when CONNECTEDLINE() capabilities are configured and supported by your handset, the name displayed will include the presence state of the callee.
Ringtime Default	Default number of seconds to ring phones before sending callers to voicemail or other extension destinations. This can be set per extension/user. Phones with no voicemail or other destination options will ring indefinitely.
Speaking Clock Time Format	Time format to use with the Speaking Clock.

Global Settings

Basic | Device Settings | Dialplan and Operational | Features Settings

Follow Me Module

- [ ] Create Follow Me at Extension Creation Time
- [x] Disable Follow Me Upon Creation

Default Follow Me Ring Time

20

Default Follow Me Initial Ring Time

7

Default Follow Me Ring Strategy

ringallv2-prim

Queues Module

- [ ] Set Agent Name in CDR dstchannel
- [x] Use MixMonitor for Recordings
- [x] Hide Queue No Answer Option
- [ ] Asterisk Queues Patch 15168 Installed
- [ ] Agent Called Events Default
- [x] Generate queuenum*/** Login/off Codes
- [ ] Memeber Status Event Default

Ring Group Module

- [ ] Display Extension Ring Group Members

Time Condition Module

- [x] Enable Maintenance Polling

Maintenance Polling Interval

60

Voicemail Module

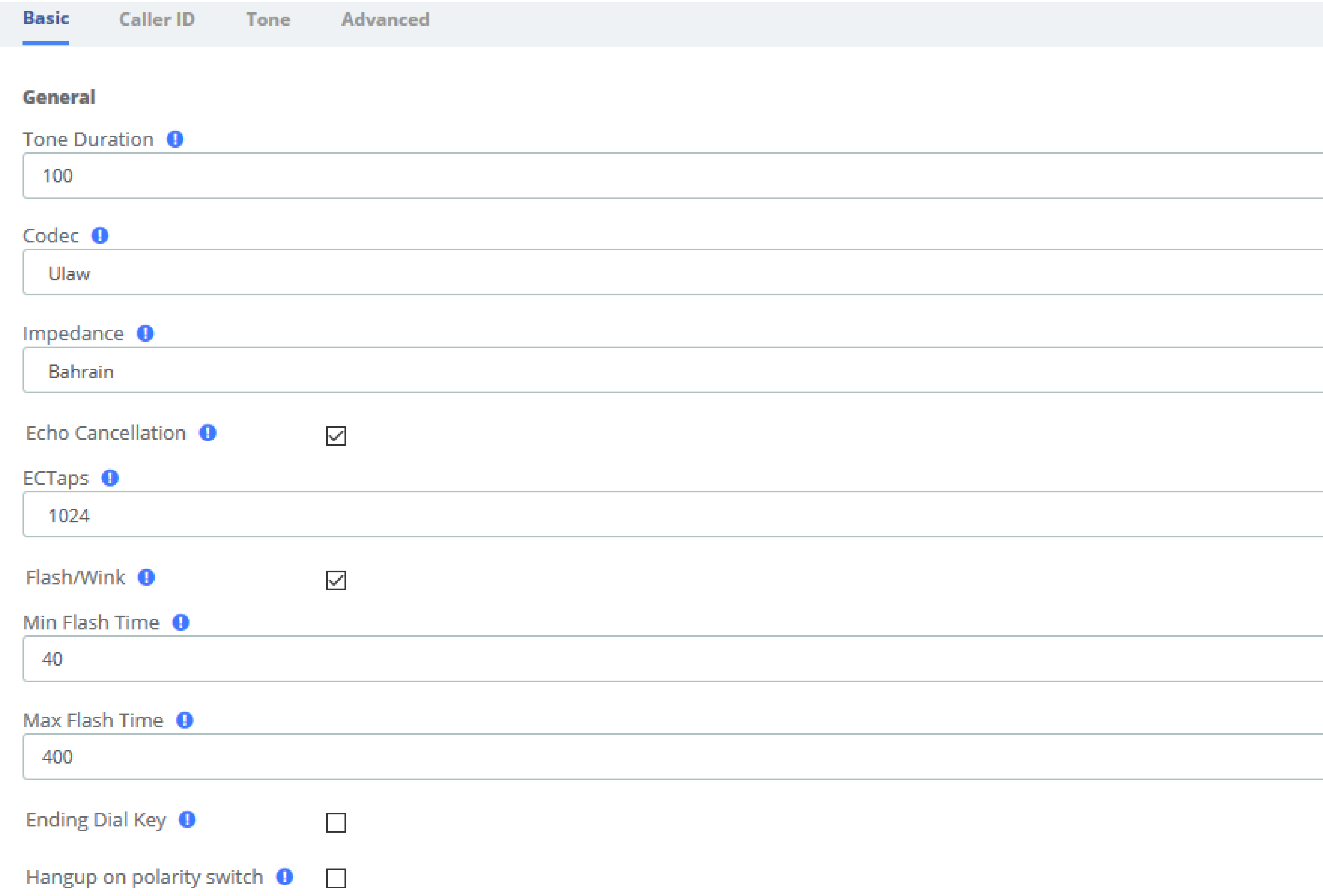
- [x] Create Voicemail Hints
- [ ] Provide IMAP Voicemail Fields

Option	Definition
Follow Me Module	
Create Follow Me at Extension Creation Time	When creating a new user or extension, setting this to true will automatically create a new Follow Me for that user using the default settings listed below.
Disable Follow Me Upon Creation	This is the default value for the Follow Me.
Default Follow Me Ring Time	The default Ring Time for a Follow Me set upon creation and used if auto-created with a new extension.
Default Follow Me Initial Ring Time	The default Ring Strategy selected for a Follow Me set upon creation and used if auto-created with an extension.
Queue Module	
Set Agent Name in CDR dstchannel	Queues: updatecdr, only valid with Asterisk 1.6+. This option is implemented to mimic chan_agents behavior of populating CDR dstchannel field of a call with an agent name, which is set if available at the login time with AddQueueMember membername parameter, or with static members.
Use MixMonitor for Recordings	Queues: monitor-type = MixMonitor. Setting true will use the MixMonitor application instead of Monitor so the concept of 'joining/mixing' the in/out files now goes away when this is enabled.
Hide Queue No Answer Option	It is possible for a queue to NOT Answer a call and still enter callers to the queue. The normal behavior is that all allers are answered before entering the queue. If the call is not answered, it is possible that some early media delivery would still allow callers to hear recordings, MoH, etc. but this can be inconsistent and vary. Because of the volatility of this option, it is not displayed by default. If a queue is set to not answer, the setting will be displayed for that queue regardless of this setting.
Asterisk Queues Patch 15168 Installed	Setting this flag will generate the required dialplan to integrate with the following Asterisk patch: https://issues.asterisk.org/view.php?id=15168. This setting is obsolete on Asterisk 1.8+ systems where the hint state is now standard and always used. This asterisk patch is only available on Asterisk 1.4, trying to use this setting on Asterisk 1.6 will break some queue behavior and should be avoided.
Agent Called Events Default	Default state for AMI emit events related to an agent's call. This setting will only affect the default for NEW queues, it won't change existing queues or enfore the

	option on in new ones.
Generate queuenum*/** Login/off Codes	Queue login and out codes were historically queunum* and queunum**. These have been largely replaced by the *45 queue toggle codes. The legacy codes are required to login or out a third party user that is not the extension dialing. These can be removed from the system by disabling this option.
Memeber Status Event Default	Default state for AMI to emit the QueueMemberStatus event. This setting will only affect the default for NEW queues, it won't change existing queues or enforce the option on in new ones.
Ring Group Module	
Display Extension Ring Group Members	When set to true extensions that belong to one or more Ring Groups will have a Ring Group section and link back to each group they are a member of.
Time Condition Module	
Enable Maintenance Polling	If set to false, this will override the execution of the Time Conditons maintenace task launched by call files. If all the feature codes for time conditions are disabled, the maintenance task will not be launched anyhow. Setting this to false would be fairly un-common. You may want to set this temporarily if debugging a system to avoid the periodic dialplan running through the CLI that the maintenance task launches and can be distracting.
Maintenance Polling Interval	The polling interval in seconds used by the Time Conditions manintenace task, launched by an Asterisk call file used to update Time Conditions override states as well as keep custom device state hint values up-to-date when being used with BLF. A shorter interval will assure that BLF keys states are accurate. The interval should be less than the shortest configured span between two time condition states, so that a manual overide during such a period is properly reset when the new period starts.
Voicemail Module	
Create Voicemail Hint	Setting this flag with generate the required dialplan to integrate with res_mwi_blf which is included with the Official PBX Distro. It allows users to subscribe to other voicemail box and be notified via BLF of changes.
Provide IMAP Voicemail Fields	Installations that have configured Voicemail with IMAP should set this to true so that the IMAP username and password fields are provided in the Voicemail setup screen for extensions. If an extension already has these fields populated, they will be displayed even if you disable this option.

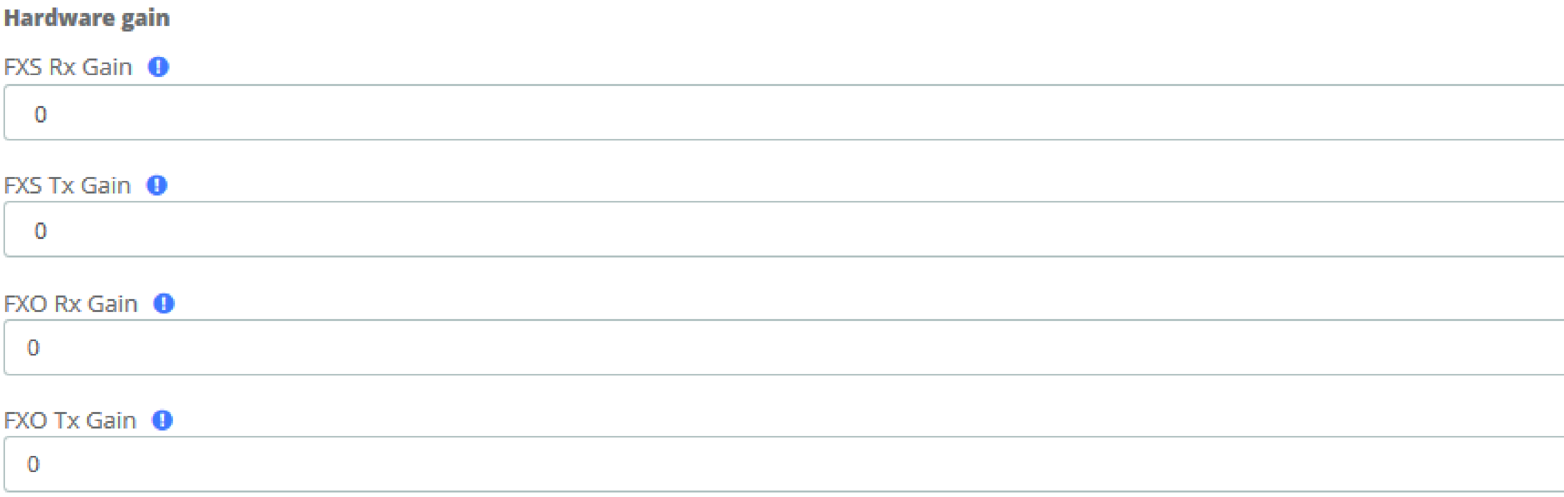
3.6.2 Analog Settings

Basic setting



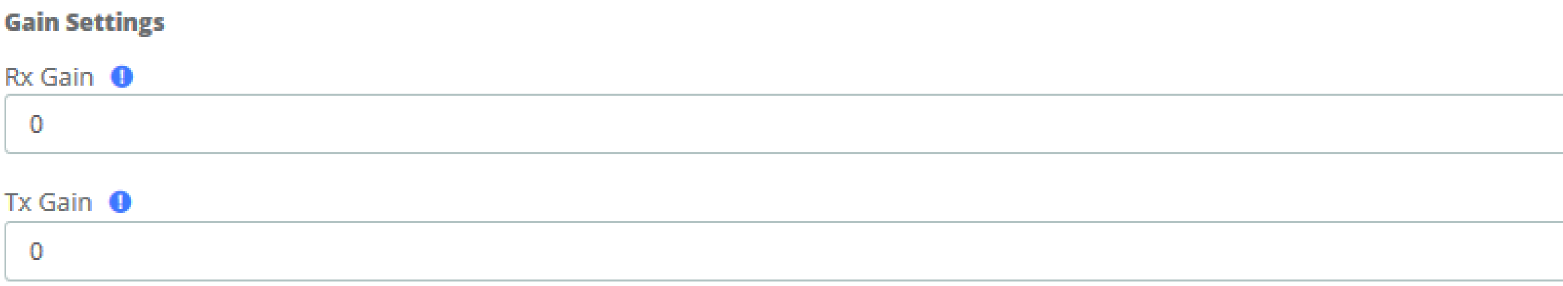
Options	Definition
Tone duration	How long generated tones (DTMF and MF) will be played on the channel. (in milliseconds)
Codec	Set the global encoding: ulaw, alaw.
Impedance	Configuration for impedance.
Echo Cancellation	Enable/Disable

ECTaps	128/256/512/1024
Flash/Wink	Turn on/off Flash/wink.
Min flash time	Min flash time. (in milliseconds).
Max flash time	Max flash time. (in milliseconds).
Ending Dial Key	Turn on/off Ending Dial Key (#).
Hang up on polarity switch	Turn on/off Hang-up on polarity switch



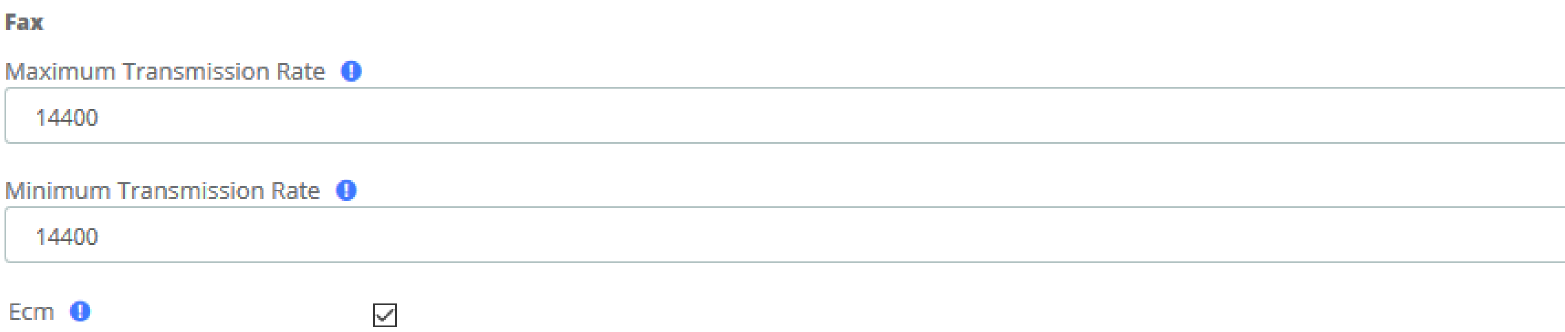
Analog Settings/Basic/Hardware gain

Options	**Definition**
FXO Rx gain	Set the FXO port Rx gain. Range: from -150 to 120.
FXO Tx gain	Set the FXO port Tx gain. Range: from -150 to 120.
FXS Rx gain	Set the FXS port Rx gain. Range: -35, 0 or 35.
FXS Tx gain	Set the FXS port Tx gain. Range: -35, 0 or 35.



Analog Settings/Basic/Gain Settings

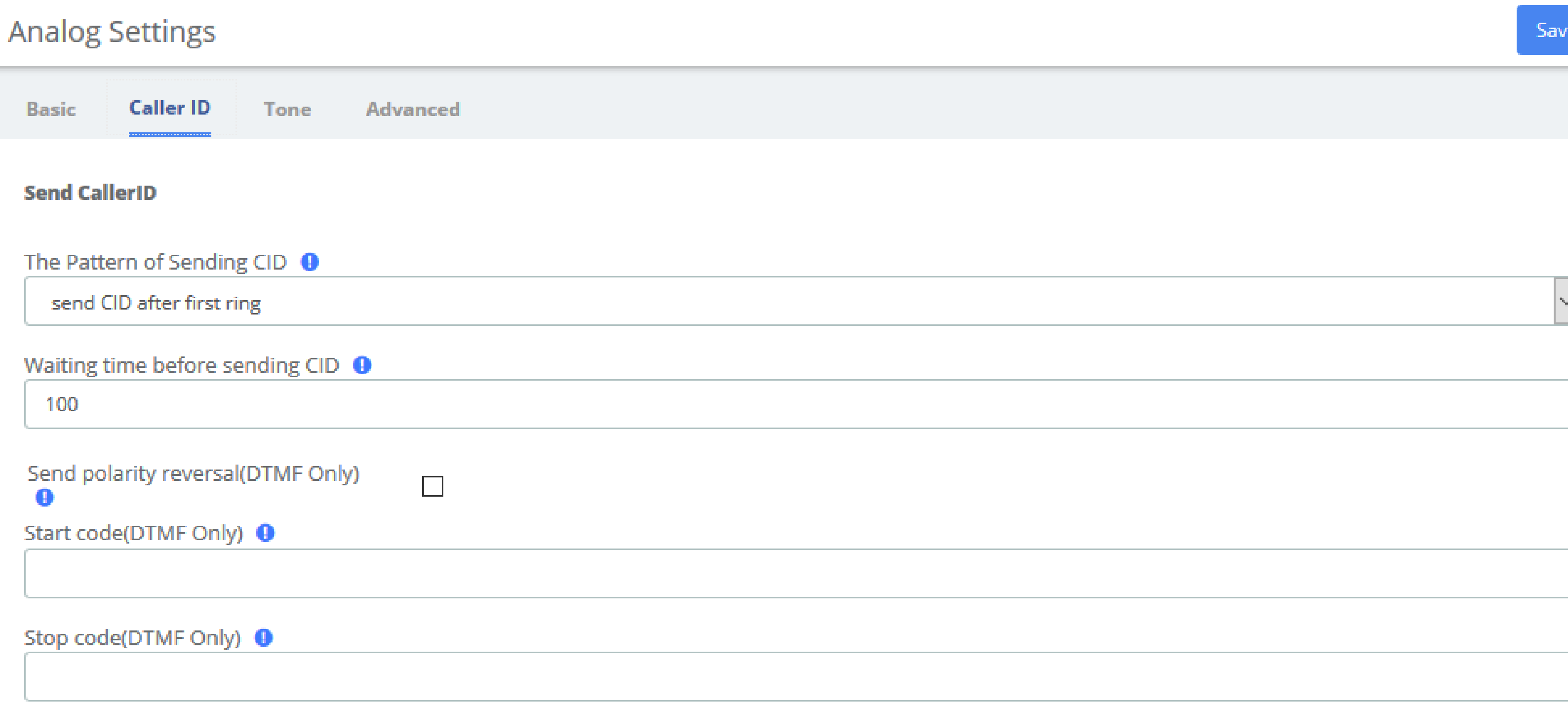
Options	Definition
Rx gain	Gain for the rx (receive) channel. Default: 0.0
Tx gain	Gain for the tx (transmit) channel. Default: 0.0



Analog Settings/Basic/Fax

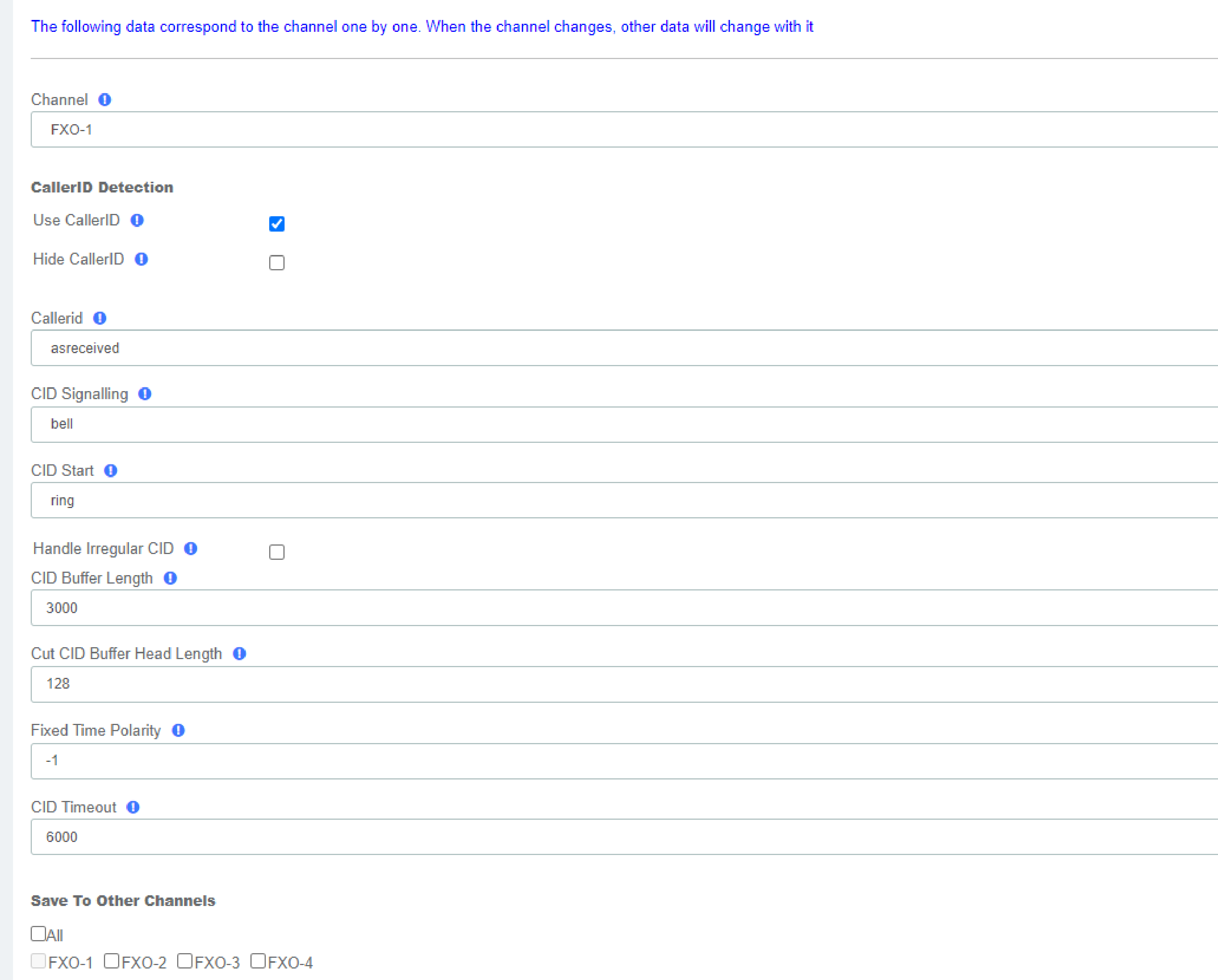
Options	Definition
Maximum Transmission Rate	Set the maximum transmission rate
Minimum Transmission Rate	Set the minimum transmission rate
Ecm	Enable/disable T.30 ECM (error correction mode) by default.

Caller ID setting



Analog Settings/Caller ID/Send Caller ID

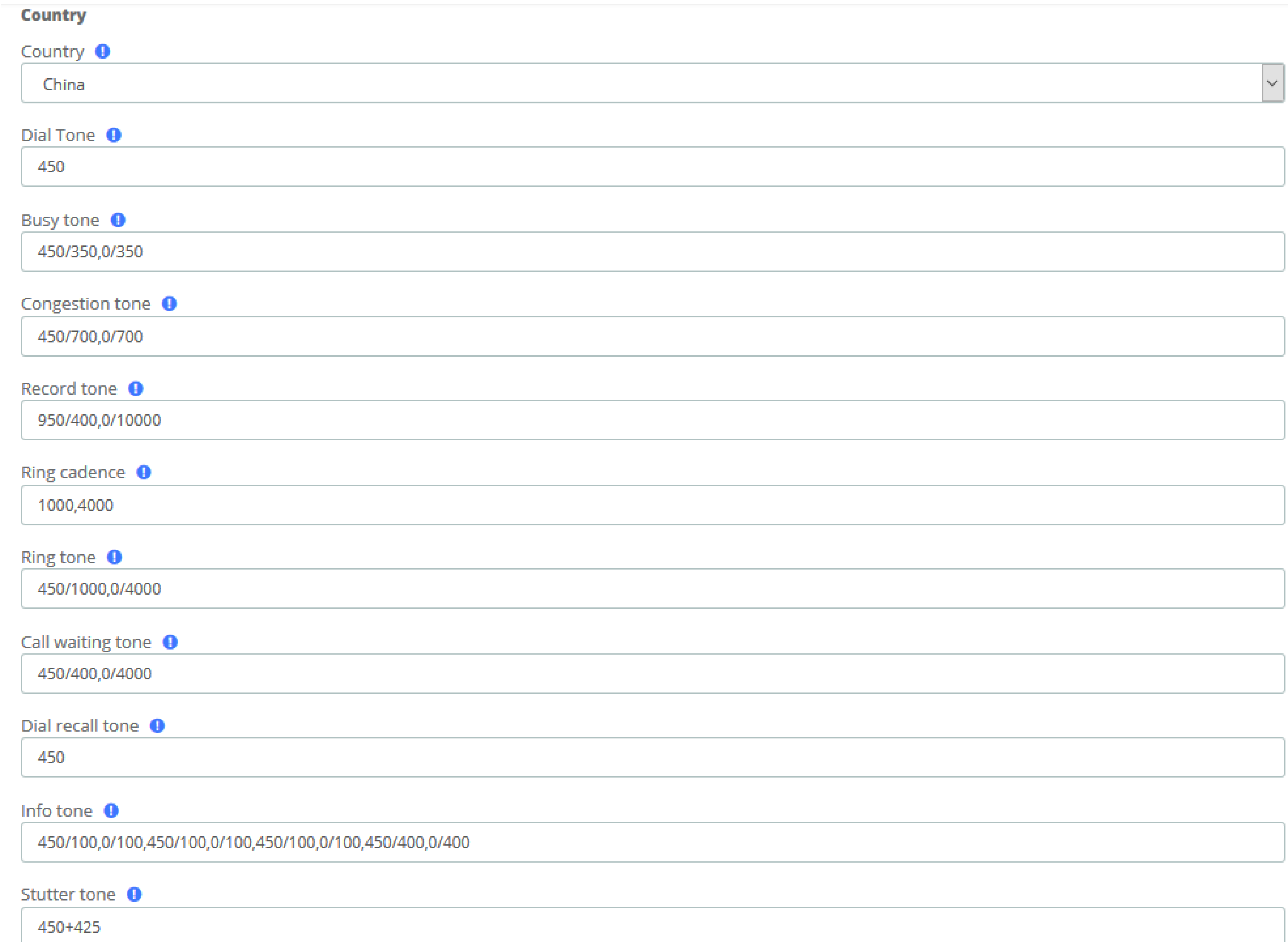
Option	Description
The pattern of sending CID	Some countries (UK) have ring tones with different ring tones(ring-ring), which means the caller ID needs to be set later on, and not just after the first ring, as per the default (1).
Waiting time before sending CID	How long we will be waiting before sending the CID on the channel. (in milliseconds).
Sending polarity reversal (DTMF Only)	Send polarity reversal before sending the CID on the channel.
Start code (DTMF Only)	Start code.
Stop code (DTMF Only)	Stop code.



Analog Settings/Caller ID/CallerID Detection

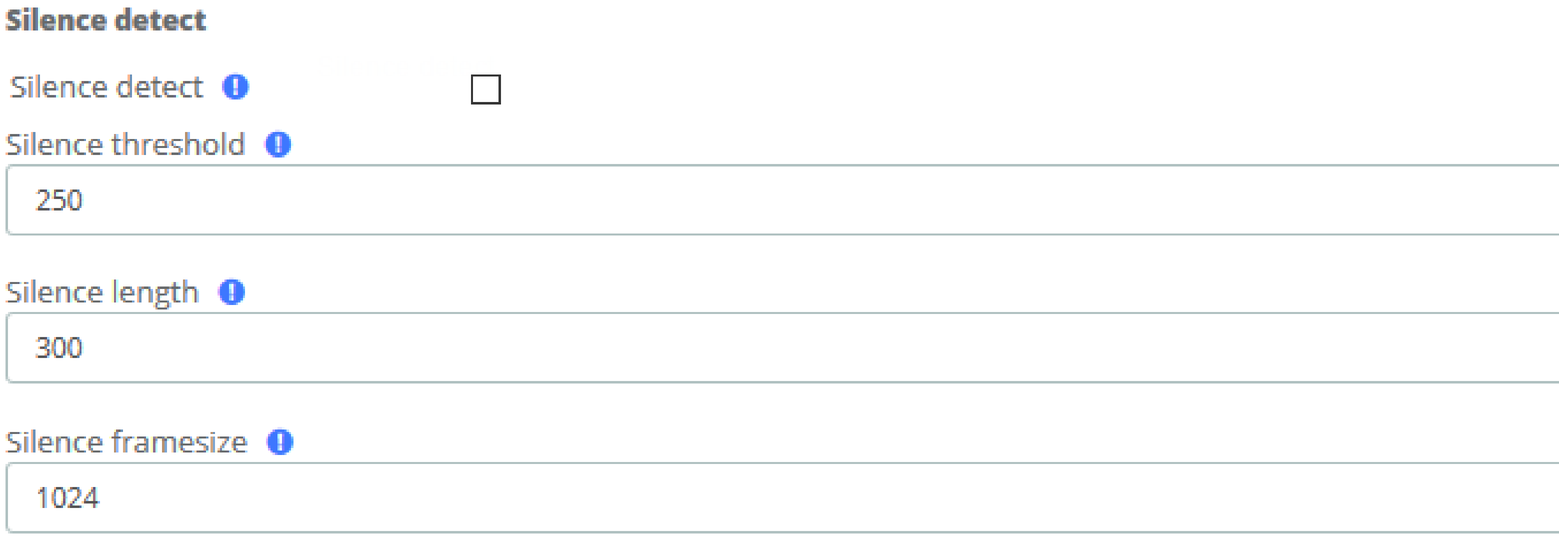
Options	**Definition**
Channel	Choose the desired FXO port for caller ID settings configuration
Use CallerID	Turn on/off CallerID detect function
Hide CallerID	Turn on/off CallerID detect function
Callerid	Caller ID can be set to "as received" or a specific number if you want to override it.

	Note that "as received" only applies to trunk interfaces.
CID Signalling	Type of caller ID signaling in use bell = bell202 as used in US (default) v23 = v23 as used in the UK dtmf = DTMF as used in Denmark, Sweden and Netherlands
CID Start	What signals the start of caller ID ring=a ring signals the start (default) polarity=polarity reversal signals the start



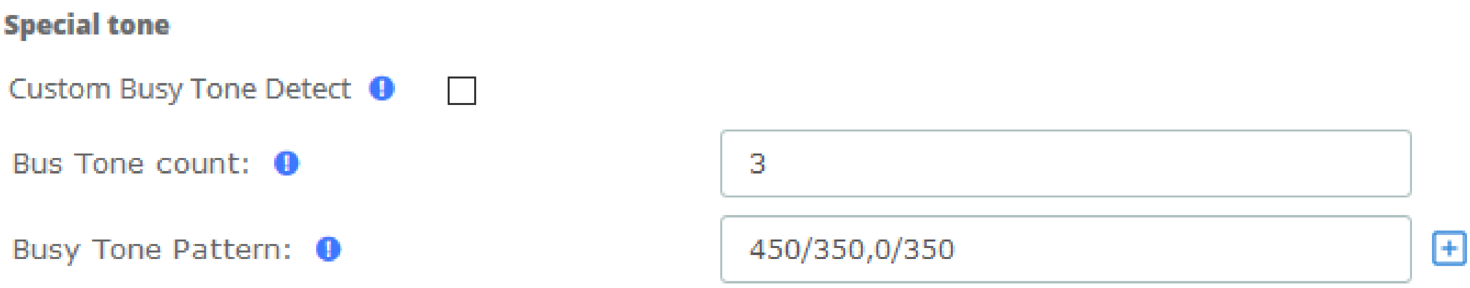
Analog Settings/Tone/Country

Options	Definition
Country	Configuration for location specific tone indications.
Dial tone	Set of tones to be played when one picks up the hook.
Busy tone	Set of tones played when the receiving end is busy.
Congestion tone	Set of tones played when there is some congestion.
Record tone	Set of tones played when call recording is in progress.
Ring cadence	List of durations the physical bell rings.
Ring tone	Set of tones to be played when the receiving end is ringing.
Call waiting tone	Set of tones played when there is a call waiting in the background.
Dial recall tone	Many phone systems play a recall dial tone after hook flash.
Info tone	Set of tones played with special information messages (e.g., number is out of service.)



Analog Settings/Tone/Silence detect

Options		Definition
Silence detect		Turn on/off silence detect function
Silence threshold		What we consider silence: the lower, the more sensitive, eg:250 is 250ms. Range: 100 to 500(100 to 500ms), default: 250
Silence length		How many silence threshold of silence before hanging up(eg: 16 is 250ms*16=4s). Range: 2 to 1020 (200ms to 512s), default: 80(20s)
Silence framesize	Rx threshold	Range: -20 dBm0 to -40 dBm0, default: 20(-20 dBm0), all values are understood to be negative.
	Tx threshold	Range: -20 dBm0 to -40 dBm0, default: 20(-20 dBm0), all values are understood to be negative.



Analog Settings/Tone/Special tone

Options	Definition
Custom Busy Tone detect	Turn on/off busy detect function
Busy Tone count	How many busy tones to wait for before hanging up. The default is 3, but it might be safer to set to 6 or even 8.
Busy Tone pattern	Set the busy detect country

COMMON

Chan Check Intv

1

Debug

0

DSR Intv

1000

Ext Reset Delay

300

Ex tReset Len

200

Io Op INTV

1

TDM URB Num

16

FXS

Boost Ringer ☐

Dial Debounce

64

Fast Ringer ☐

Fxs Honor Mode ☐

Loop Current

20

Low Power ☐

Ms Per ChkFxs

8

Reverse Polarity ☐

Robust ☑

FXO

Voltage Alarm

1000

Voltage Debounce

64

Voltage Thresh

3

Fast Pickup ☐

Fast Ring Offhook

0

FW Ring Detect ☐ FXO Full Scale ☐

FXO Voltage

0

Ms Per ChkFxo

4

Polarity Debounce

64

Ring Debounce

64

Ring Off Count

0

Ring On Count

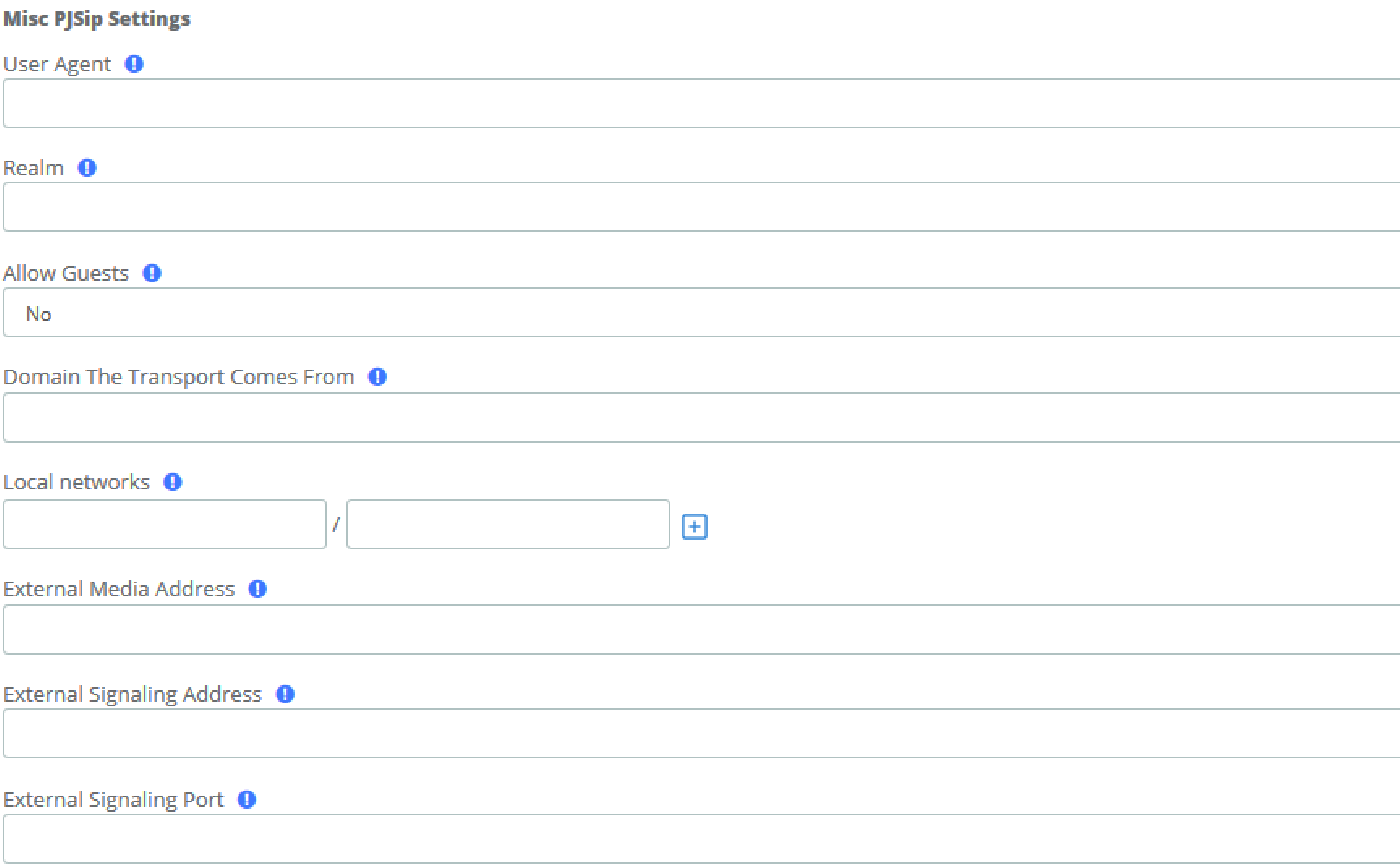
4

Timing Only

0

Two Way Charge Flag ☑

3.6.3 SIP Settings



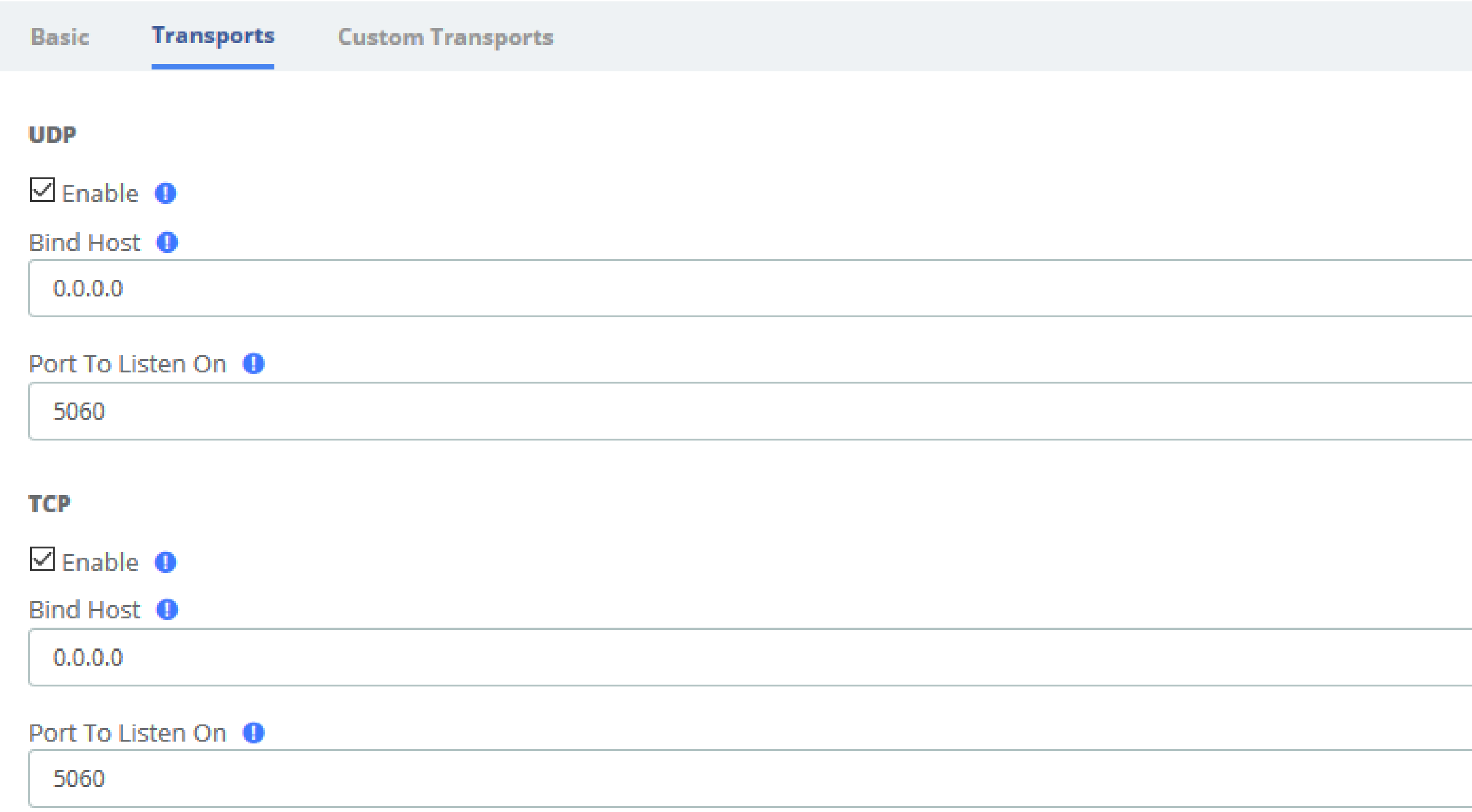
Options	Definition
User Agent	Value used in User-Agent header for SIP requests and Server header for SIP responses.
Realm	When generates a challenge, the digest realm will be set to this value if there is no better option (such as auth/realm) to be used.
Allow Guests	When set Asterisk will allow Guest SIP calls and send them to the Default SIP context. Turning this off will keep anonymous SIP calls from entering the system. Doing such will also stop 'Allow Anonymous Inbound SIP Calls' from functioning. Allowing guest calls but rejecting the Anonymous SIP calls below will enable you to see the call attempts and debug incoming calls that may be misconfigured and appearing as guests.

Domain The Transport Comes From	Typically used with SIP calling. Example user@domain, where domain is the value that would be entered here
Local networks	Local network settings in the form of IP/CIDR or IP/Netmask. For networks with more than 1 LAN subnets, use the Add Local Network Field button for more fields. Blank fields will be ignored.
External Media Address	External IP address to use in RTP handling.
External Signaling Address	External address for SIP signaling.
External Signaling Port	External port for SIP signaling.

SIP Settings/Basic/NAT STUN Settings

Options	**Definition**
Enable	If enabled, the Nat Stun will be enabled.
STUN Server	Address of the STUN server to query. Custom valid form:[(hostname | IP-address)]

STUN Port	The port defaults to the standard STUN port (3478).
Refresh Time	Number of seconds between STUN refreshes, Default is 30.



TLS

☑ Enable

Bind Host

0.0.0.0

Port To Listen On

5061

Certificate Manager

default

SSL Method

default

Verify Client

No

Verify Server

No

WS

☑ Enable

Bind Host

0.0.0.0

WSS

☑ Enable

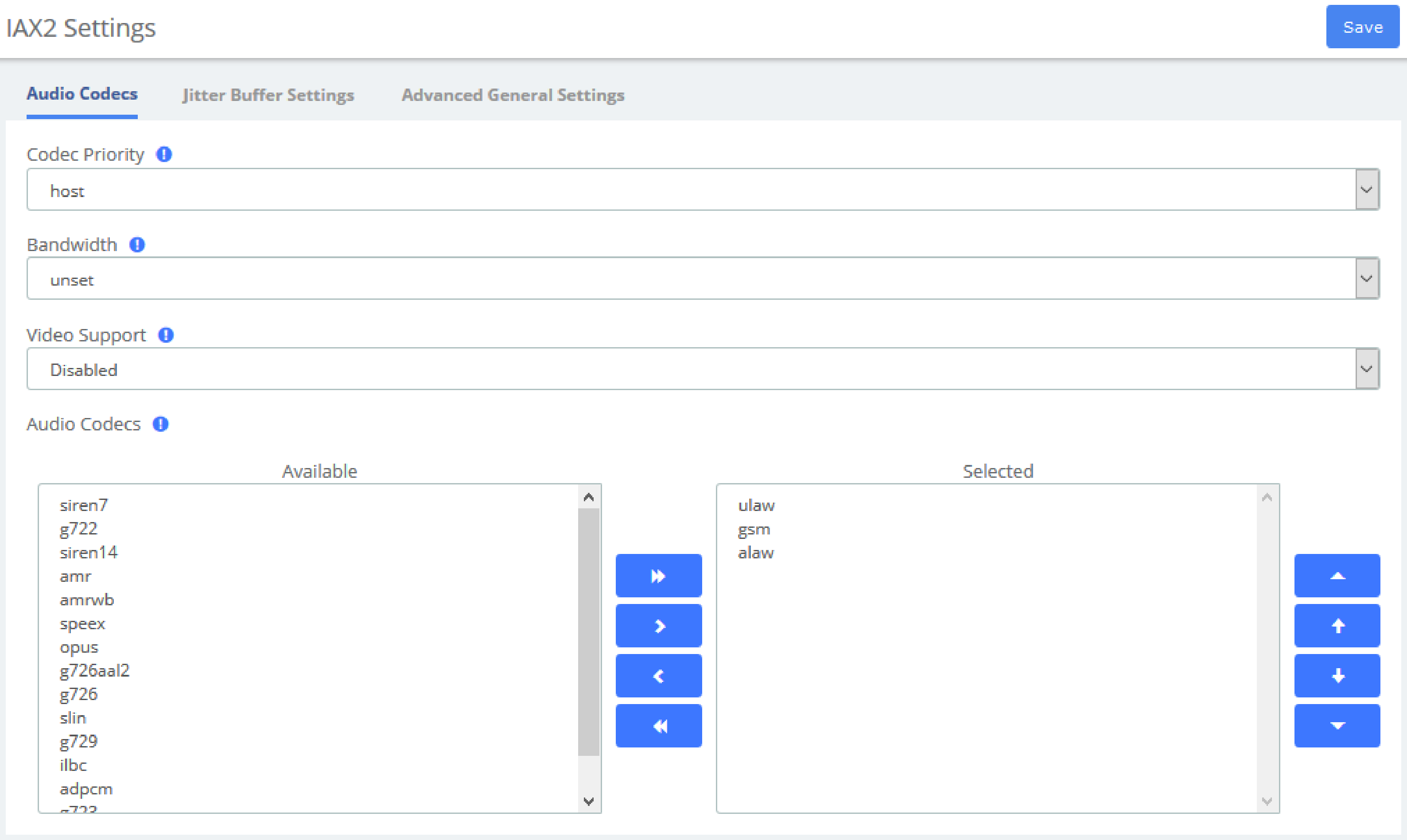
Bind Host

0.0.0.0

Options	Definition
UDP	
Enable	Use 0.0.0.0 - All
Bind Host	You can customize the UDP bind host, the default is 0.0.0.0
Port To Listen On	The port that this transport should listen on

TCP	
Bind Host	You can customize the TCP bind host, the default is 0.0.0.0
Port To Listen On	The port that this transport should listen on
TLS	
Bind Host	You can customize the TLS bind host, the default is 0.0.0.0
Port To Listen On	The port that this transport should listen on
Certificate Manager	Select a certificate to use for the TLS transport. These are configured in the module Certificate Manager1
SSL Method	Method of SSL transport (TLS ONLY). The default is currently TLSv1, but may change with future releases.1
Verify Client	Require verification of server certificate (TLS ONLY).
Verify Server	Require verification of server certificate (TLS ONLY).
WS	
Bind Host	You can customize the WS bind host, the default is 0.0.0.0
WSS	
Bind Host	You can customize the WSS bind host, the default is 0.0.0.0

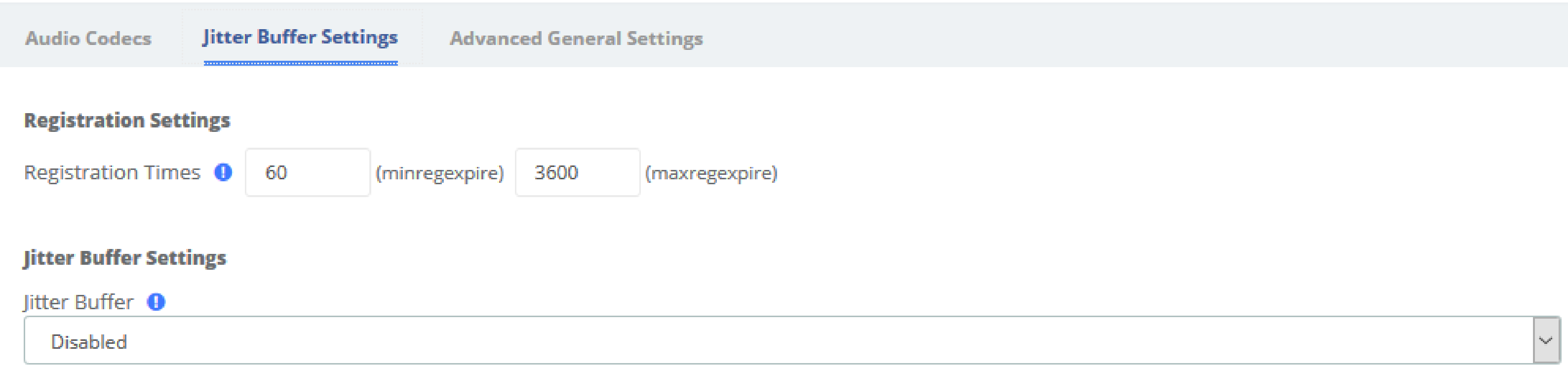
3.6.4 IAX2 Settings



IAX2 Settings/Audio Codecs

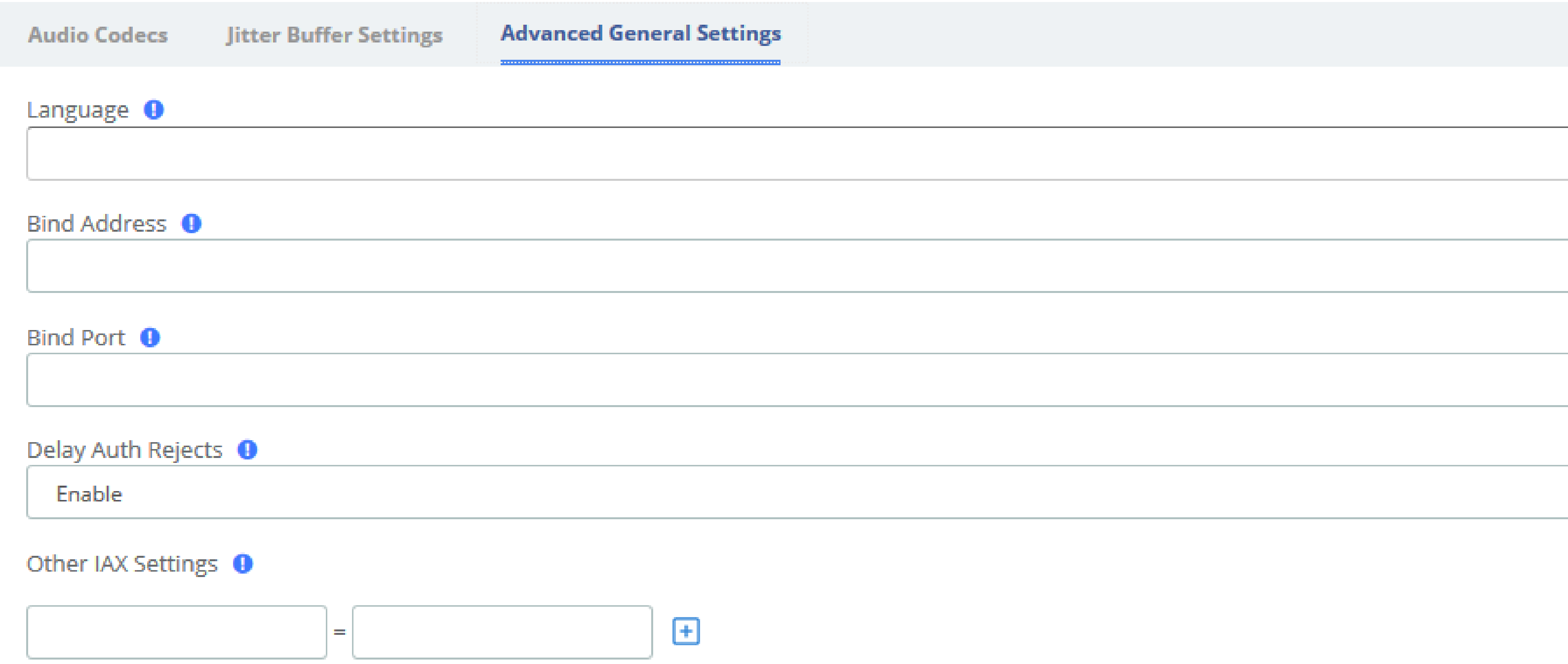
Options	**Definition**
Codec Priority	Asterisk: code priority. Controls the codec negotiation of an inbound IAX call. This option is inherited to all user entities. It can also be defined in each user entity separately which will override the setting here. The valid values are: host - Consider the host's preferred order ahead of the caller's. Caller - consider callers host's. disabled disable consideration codec preference altogether. (this is original behavior before preferences were added). regonly same as disabled, only do not capabilities if requested format available call will be accepted available.
Bandwidth	Asterisk: bandwidth. Specify bandwidth of low, medium, or high to control which codecs are used in general.

Video Support	Check to enable and then choose allowed codecs. If you clear each codec and then add them one at a time, submitting with each addition, they will be added in order which will affect the codec priority.
Audio Codecs	Check the desired codecs, all others will be disabled unless explicitly enabled in a device or trunks configuration. Drag to re-order.



IAX2 Settings/Jitter Buffer Settings

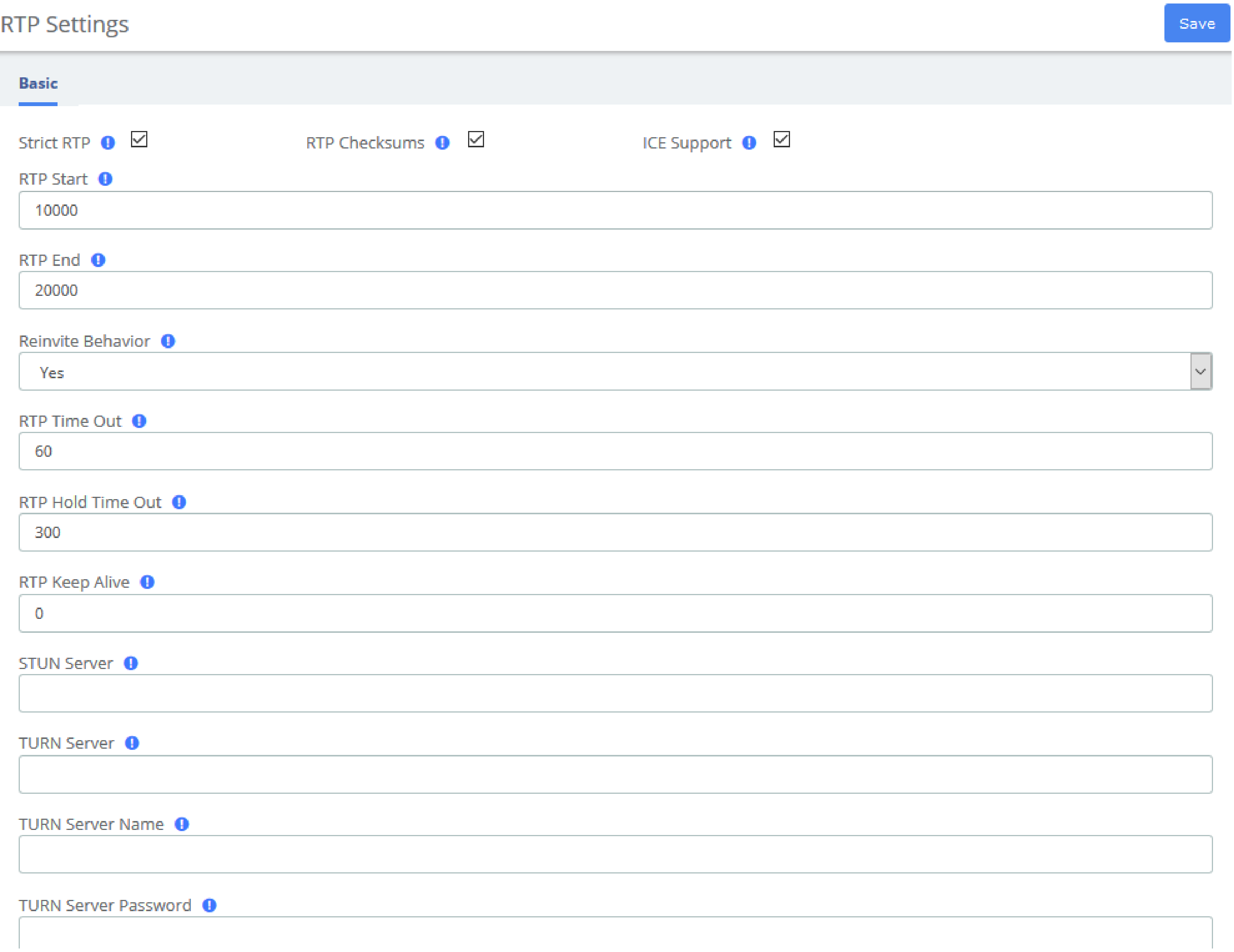
Options	Definition
Registration Settings	
Registration Times	Asterisk: minregexpire, maxregexpire. Minimum and maximum length of time that IAX peers can request as a registration expiration interval (in seconds).
Jitter Buffer Settings	
Jitter Buffer	Asterisk: jitter buffer. You can adjust several parameters relating to the jitter buffer. The jitter buffer’ s function is to compensate for varying network delay. the jitter buffer works incoming audio - outbound will be dejittered by at other end.



IAX2 Settings/Advanced General Settings

Options	Definition
Language	Default Language for a channel, Asterisk: language
Bind Address	Asterisk: bind address. The IP address to bind to and listen for calls on the Bind Port. If set to 0.0.0.0 Asterisk will listen on all addresses. To bind to multiple IP addresses or ports, use the Other iax settings' fields where you can put settings such as:bindaddr='192.168.10.100:4555.' it is recommended to leave this blank.
Bind Port	Asterisk: bind port. Local incoming UDP Port that Asterisk will bind to and listen for IAX messages. The IAX standard is 4569 and in most cases this is what you want. It is recommended to leave this blank.
Delay Auth Rejects	Asterisk: delay reject. For increased security against brute force password attacks enable this which will delay the sending of authentication reject for REGREQ or AUTHREP if there is a password.
Other IAX Settings	You may set any other IAX settings not present here that are allowed to be configured in the General section of iax.conf. There will be no error checking against these settings so check them carefully. They should be entered as: [setting] = [value]in the boxes below. Click the Add Field box to add additional fields. Blank boxes will be deleted when submitted.

3.6.5 RTP Settings

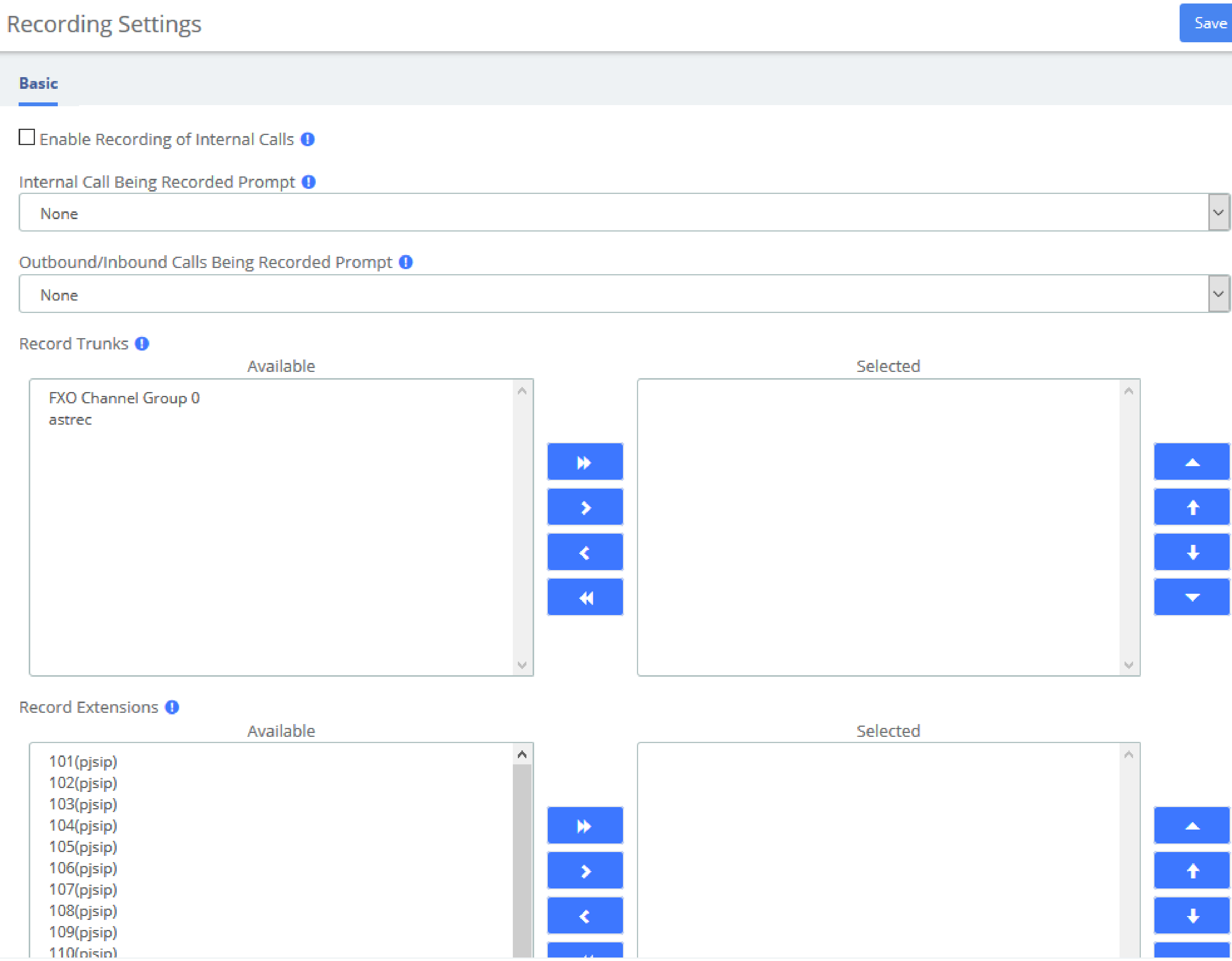


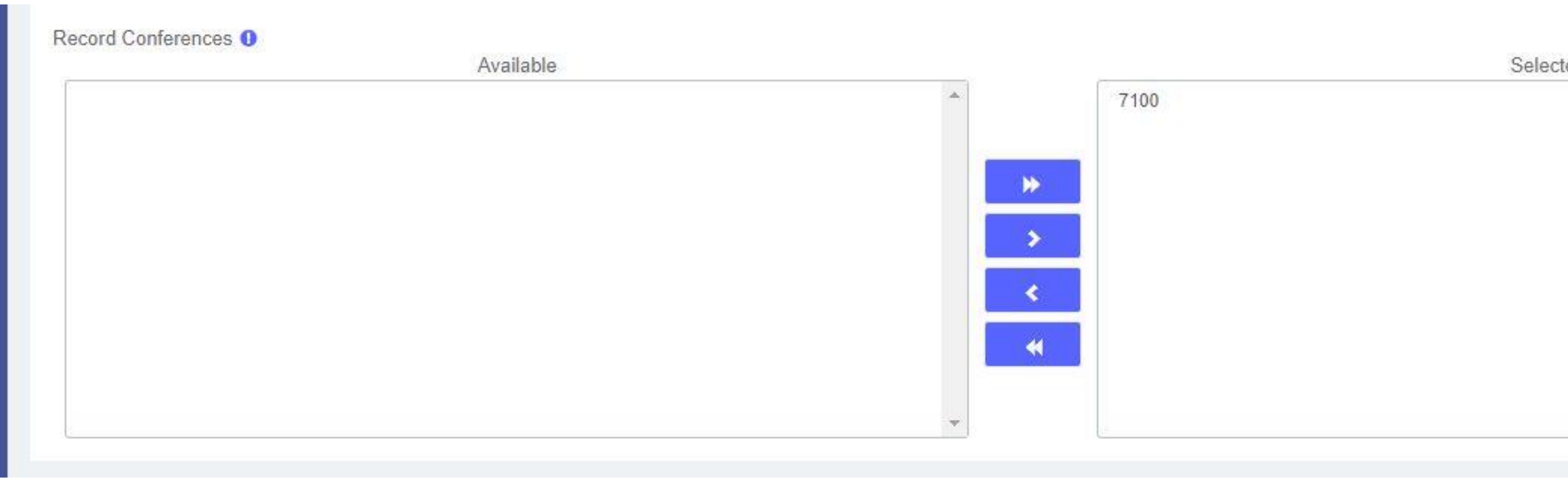
RTP Settings

Options	Definition
Strict RTP	Enable strict RTP protection. This will drop RTP packets that do not come from the source of the RTP stream. This option is disabled by default.
RTP Checksums	Whether to enable or disable UDP checksums on RTP traffic

ICE Support	Whether to enable ICE support. Defaults to no. ICE (Interactive Connectivity Establishment) is a protocol for network address Translator (NAT) traversal for UDP-based multimedia sessions established with the offer/answer model. This option is commonly enabled in Web RTC setups
RTP Start	Start of range of port numbers to be used for RTP. Defaults is 10000.
RTP End	End of range of port numbers to be used for RTP. Defaults is 20000.
Reinvite Behavior	yes: standard reinvites; (yes = update + nonat) no: never; nonat: An additional option is to allow media path redirection (reinvite) but only when the peer where the media is being sent is known to not be behind a NAT (as the RTP core can determine it based on the apparent IP address the media arrives from; update: use UPDATE for media path redirection, instead of INVITE.
RTP Time Out	The call is terminated when there is no RTP or RTCP activity on the audio channel for a period of time (that is, the set timeout period). This is to be able to hang up the call in case of network interruption (not on hold)
RTP Hold Time Out	If there is no RTP or RTCP activity on the audio channel for a period of time (that is, the set hold timeout period), the call will be terminated (in hold state). This value must be greater than the timeout period.
RTP Keep Alive	Send Keep alive in RTP stream to keep NAT open (default is off)
STUN Server	Configure the STUN server address. STUN is a Client/Server protocol and also a Request/Response protocol. It is used here to check the connectivity between two terminals, like a way to maintain NAT binding entries Keep-alive agreement.
TURN Server	Configure the TURN server address, STUN can handle most of the NAT problems. TURN is an enhanced version of the STUN protocol, dedicated to dealing with symmetric NAT problems.
TURN Server Name	Configure the TURN Server name
TURN Server Password	Configure the TURN Server password

3.6.6 Recording Settings





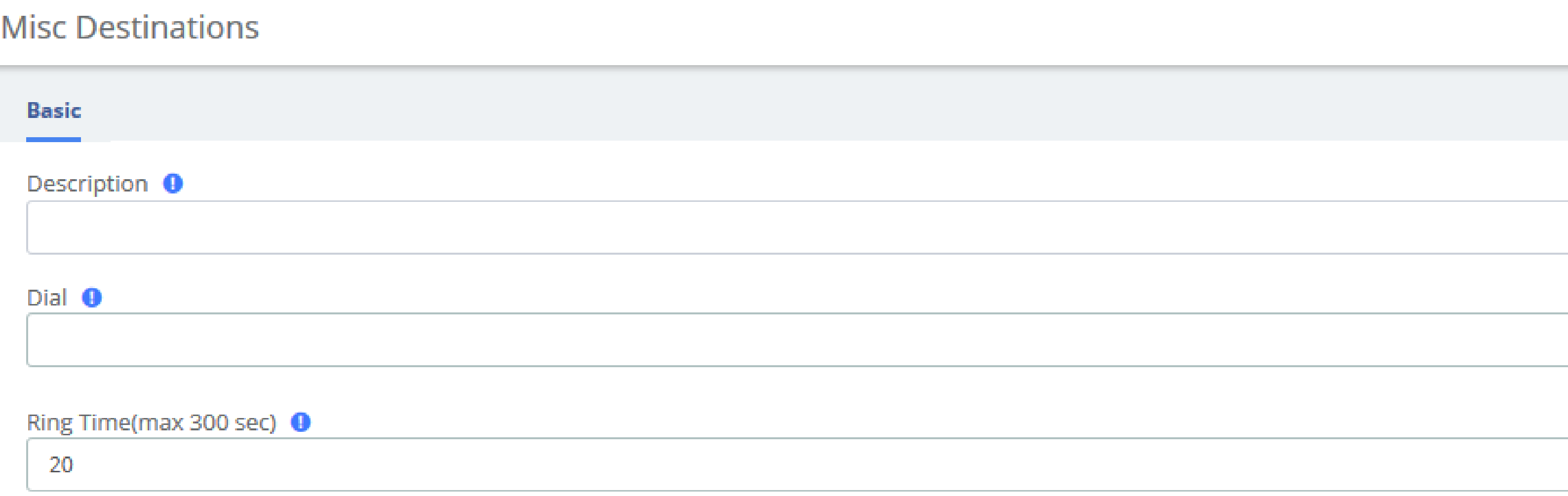
Options	Definition
Enable Recording of Internal Calls	Check this option, and all internal calls made by the selected extensions will be recorded automatically.
Internal Call Being Recorded Prompt	If the internal call has enabled call recording, this prompt will notify the called party that the call is being recorded.
Outbound/Inbound Calls Being Recorded Prompt	If the external call (outbound/inbound) has enabled call recording, this prompt will notify the external party that the call is being recorded.
Record Trunks	When a call reaches the selected trunk, it will be recorded.
Record Extensions	The selected extensions will be recorded.
Record Conferences	The selected conferences will be recorded.

3.6.7 Misc Destinations

The Misc Destinations Module is used to create a miscellaneous destination to which you can route calls from another module.

For example, you might create a Misc destination called "My Mobile Phone" that dials your mobile telephone number. Then, you could set up an IVR so that if a caller presses 9, they would be routed to "Misc Destinations: My Mobile Phone."

Misc Destinations are for adding destinations that can be used by other modules, generally used to route incoming calls. If you want to create feature codes that can be dialed by internal users and go to various destinations, please see the Misc Applications module. If you need access to a Feature Code, such as *98 to dial voicemail or a Time Condition toggle, these destinations are now provided as Feature Code Admin destinations. For upgrade compatibility, if you previously had configured such a destination, it will still work but the Feature Code short cuts select list is no longer provided.

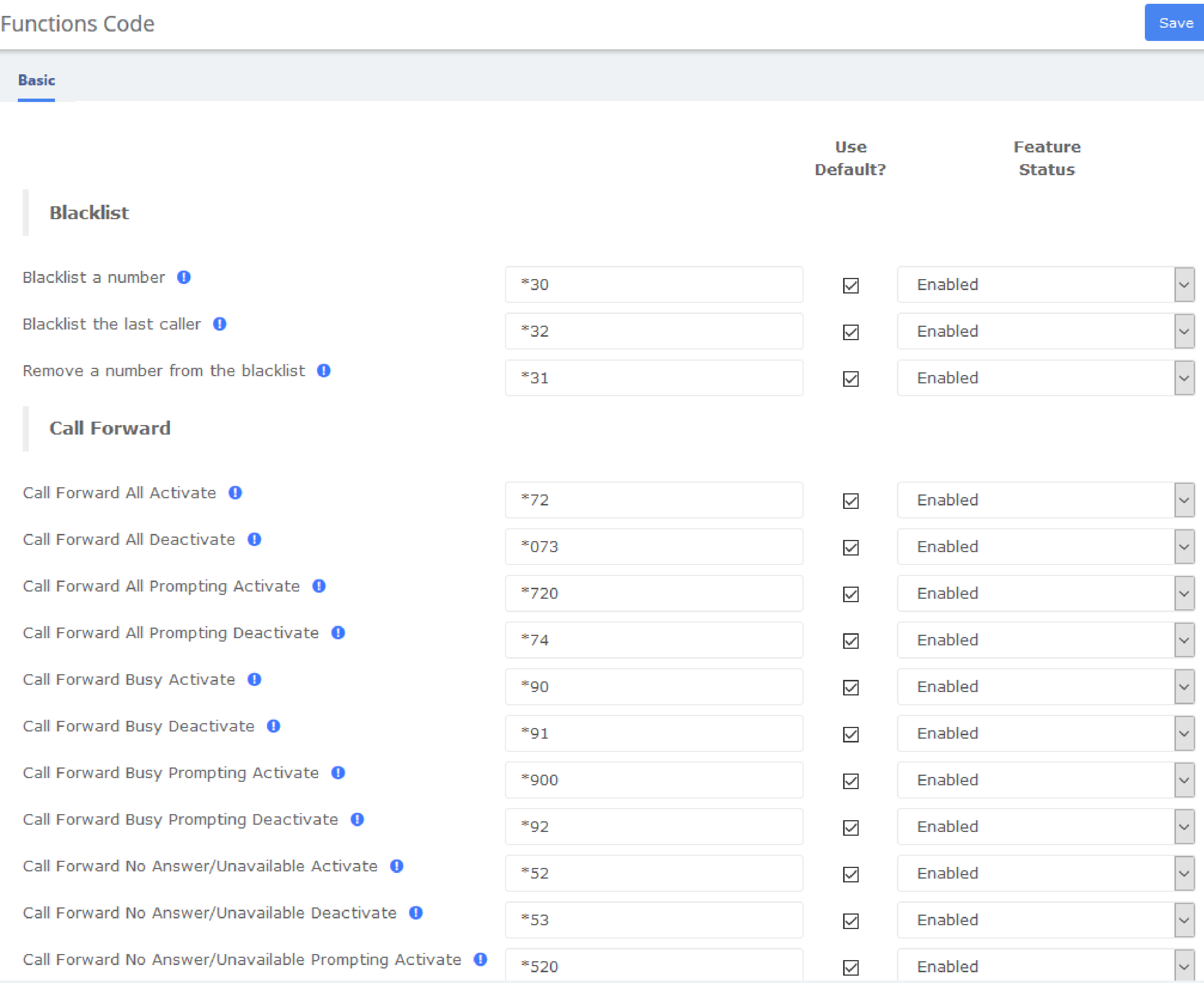


Item	Definition
Description	Give this Misc Destination a brief name to help you identify it.
Dial	Enter the number this destination will simulate dialing, exactly as you would dial it from an internal phone. When you route a call to this destination, it will be as if the caller dialed this number from an internal phone.

3.6.8 Functions Code

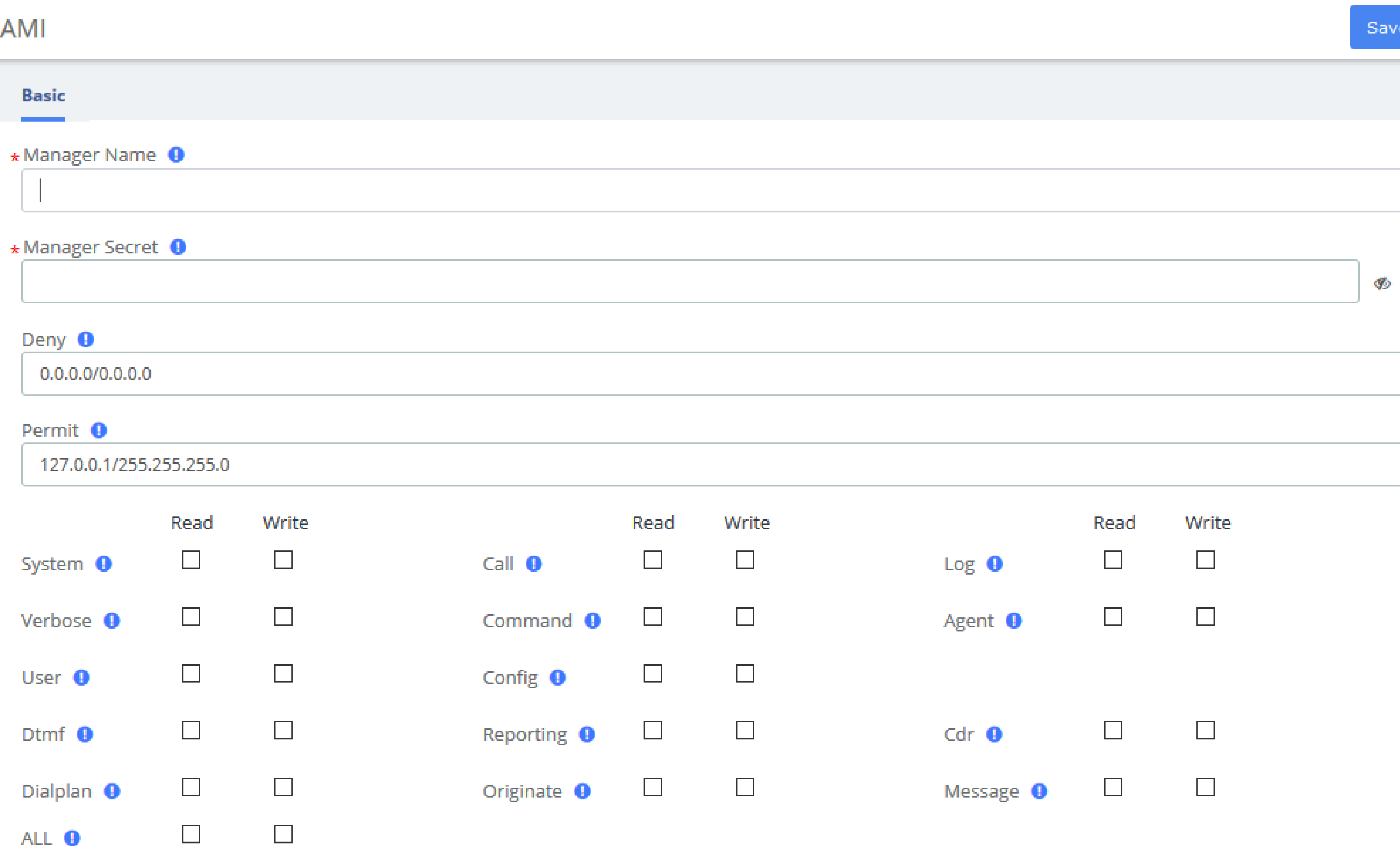
The Feature Codes Module is used to enable and disable certain features available in your PBX and Asterisk, and to set the codes that local users will dial on their phones to use that particular feature.

For example, the Feature Codes Module can be used to set the code that a user will dial to activate or deactivate Call Forwarding. It can also be used to set a Code that can be used to enter into an Echo Test, to hear your extension number, or to hear the time of day.



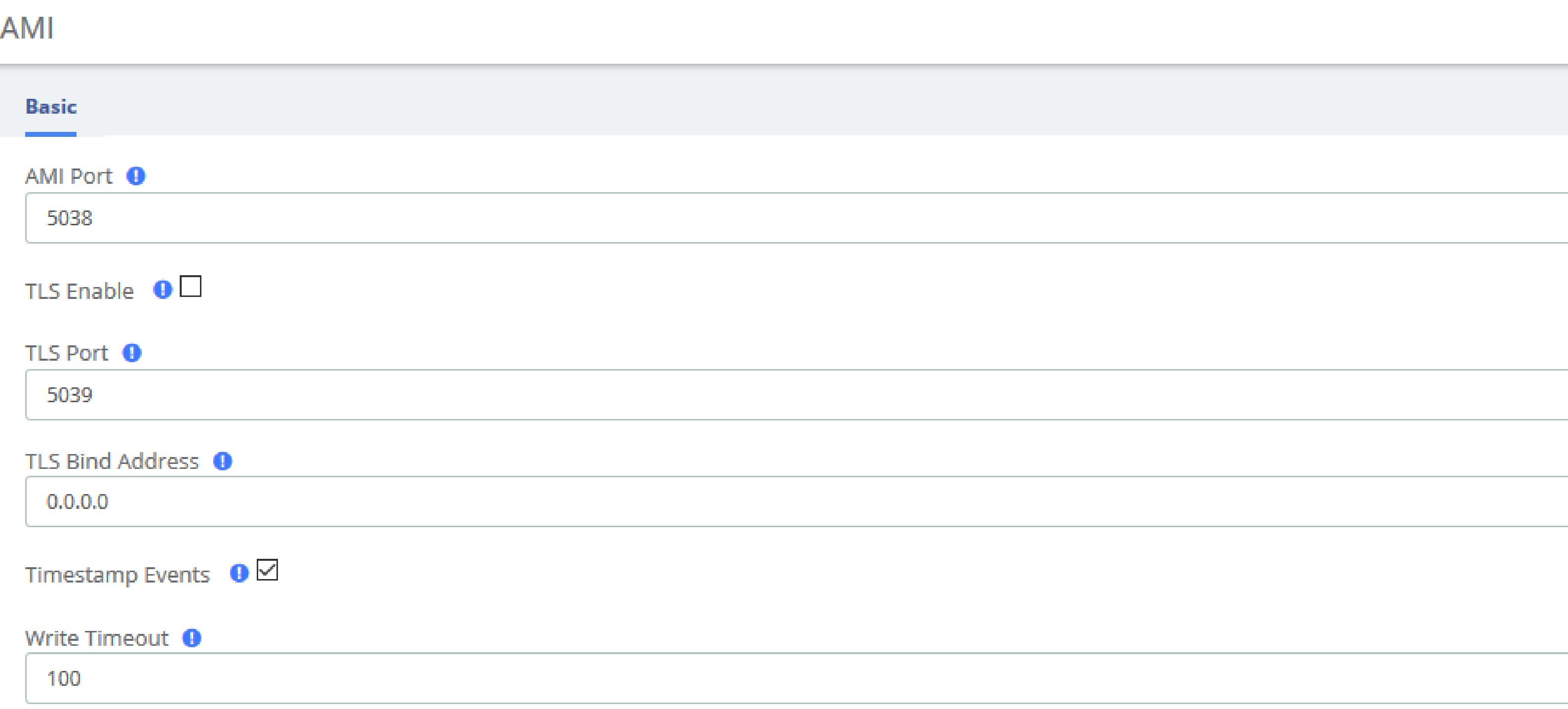
Feature code admin interface

3.6.9 AMI



Manager User interface

Item	Definition
Manager name	Name of the manager without space.
Manager secret	Password for the manager.
Deny	If you want to deny many hosts or networks, use & char as separator. Example: 192.168.1.0/255.255.255.0&10.0.0.0/255.0.0.0
Permit	If you want to permit many hosts or networks, use & char as separator. Look at deny example.



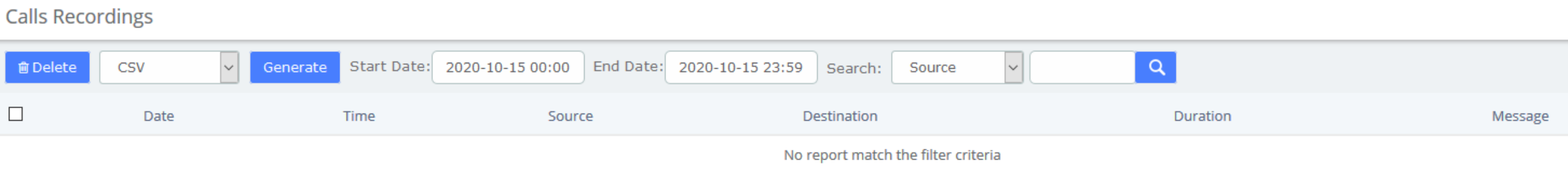
AMI Setting interface

Item	Definition
AMI Port	Sets the port number to listen on for AMI connections. Port numbers cannot be less than 1024. The default is 7777.
TLS Enable	Enables listening for AMI connections using TLS. The default is no.
TLS Port	Sets the port to listen on for TLS connections to the AMI. The default is 5039.
TLS Bind Address	Sets the address to listen on for TLS-based AMI connections. The default is to listen on all addresses (0.0.0.0).
Timestamp Events	Add a Unix epoch timestamp to events (but not action responses.
Write Timeout	Write Timeout

3.7 Recording

3.7.1 Call Recordings

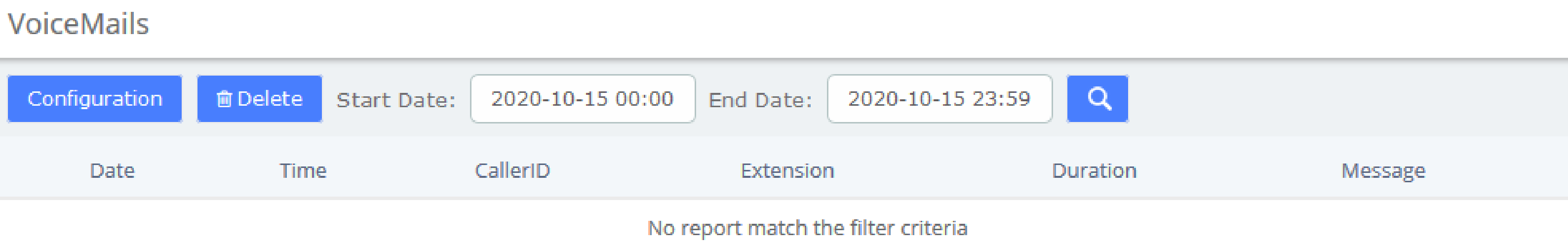
The option "Calls Recordings" of the Menu "Recordings" in SMB PBX lets us view a list with details of all recorded calls for the extension associated to the connected user. The administrator account can see all the recordings.



Calls Recordings interface

3.7.2 VoiceMails

The option "Voicemail" of the Menu “Recordings” in SMB PBX lets us view a list with details of the voicemails for the extension of the logged user.



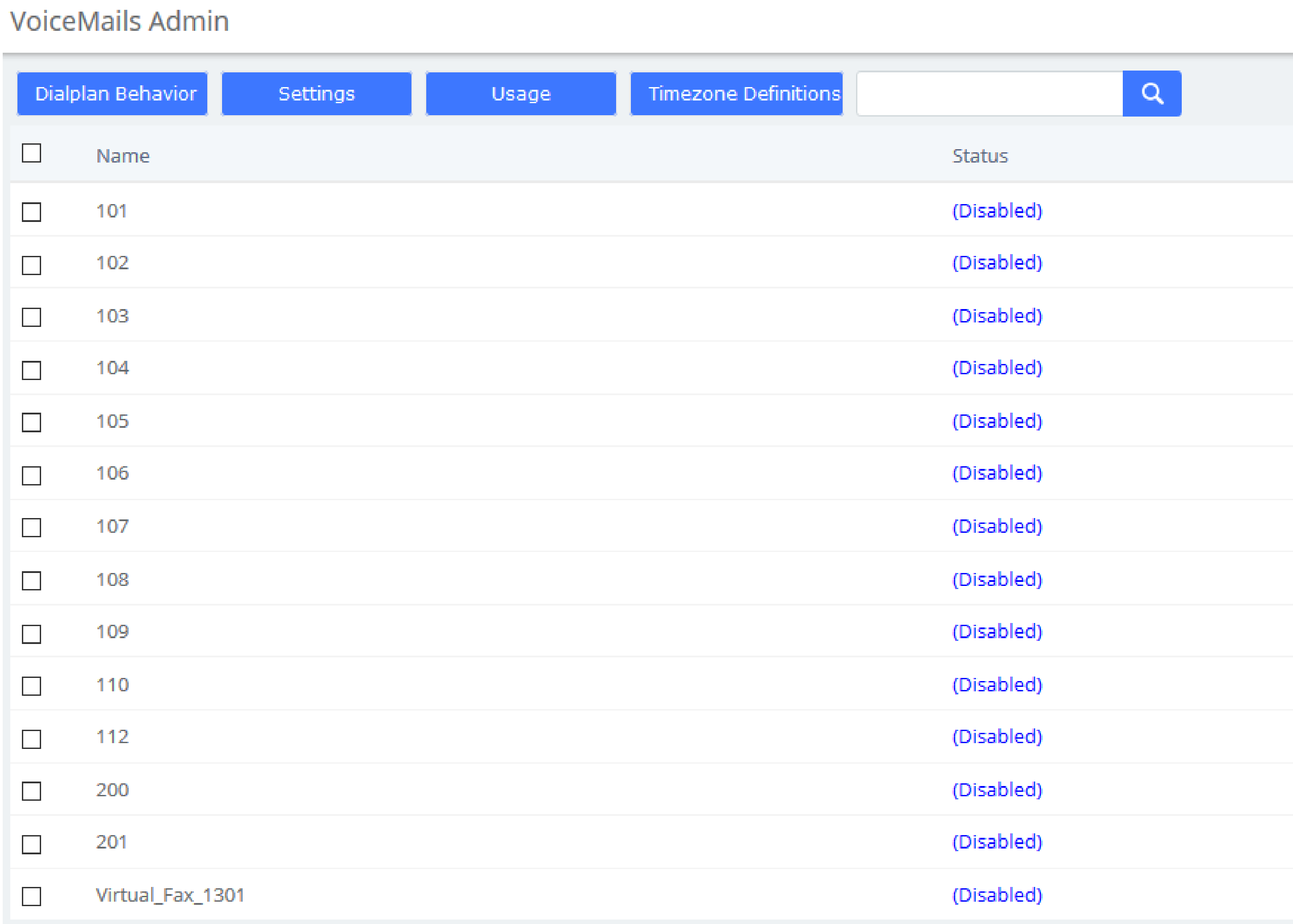
The report will change depending on the values of the filter:

Parameter	**Description**
Start Date	Start date for the selection of voicemails.
End Date	End date for the selection of voicemails.

To delete a voicemail, just select the voicemail from the list and click on "**Delete**" button.

3.7.3 VoiceMails Admin

The option "**Voicemail Admin**" of the Menu "**Recordings**" lets us view or modify some voicemail configuration.

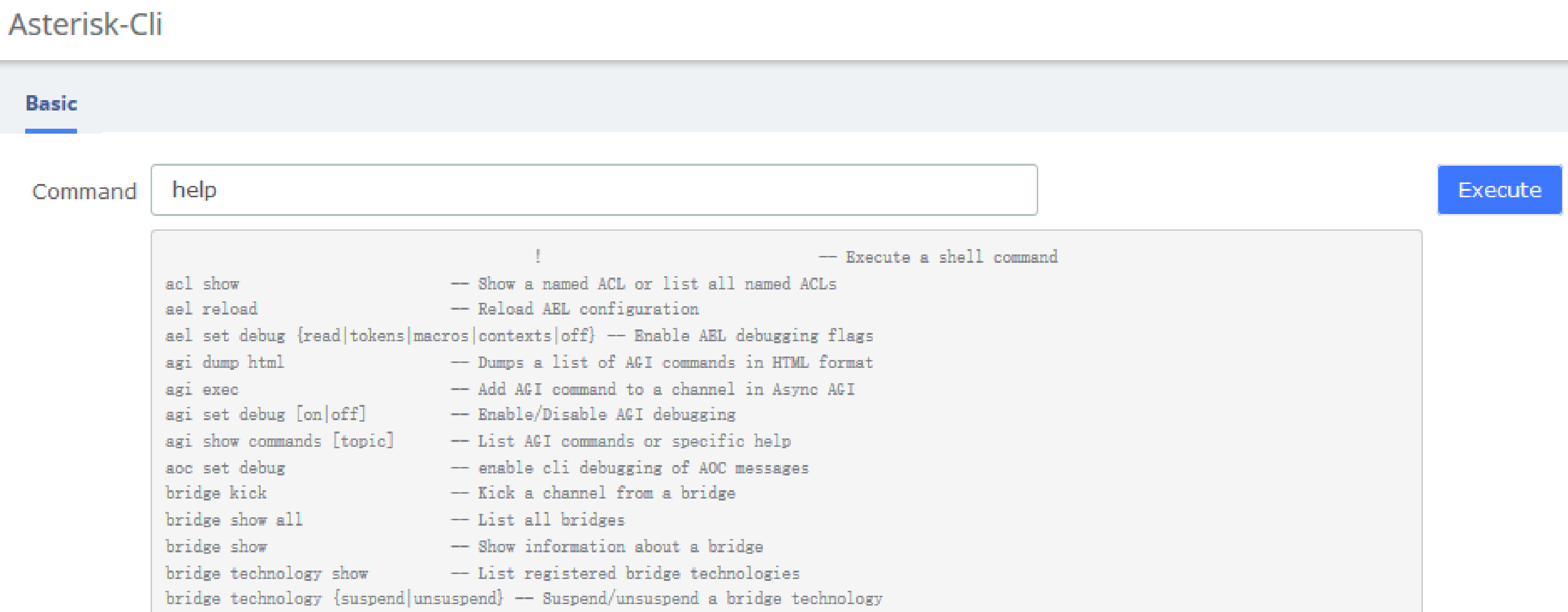


Voicemails Admin interface

3.8 Tools

3.8.1 Asterisk-Cli

The option “Asterisk-Cli” of the Menu “**Tools**” in SMB PBX lets us execute Asterisk commands.



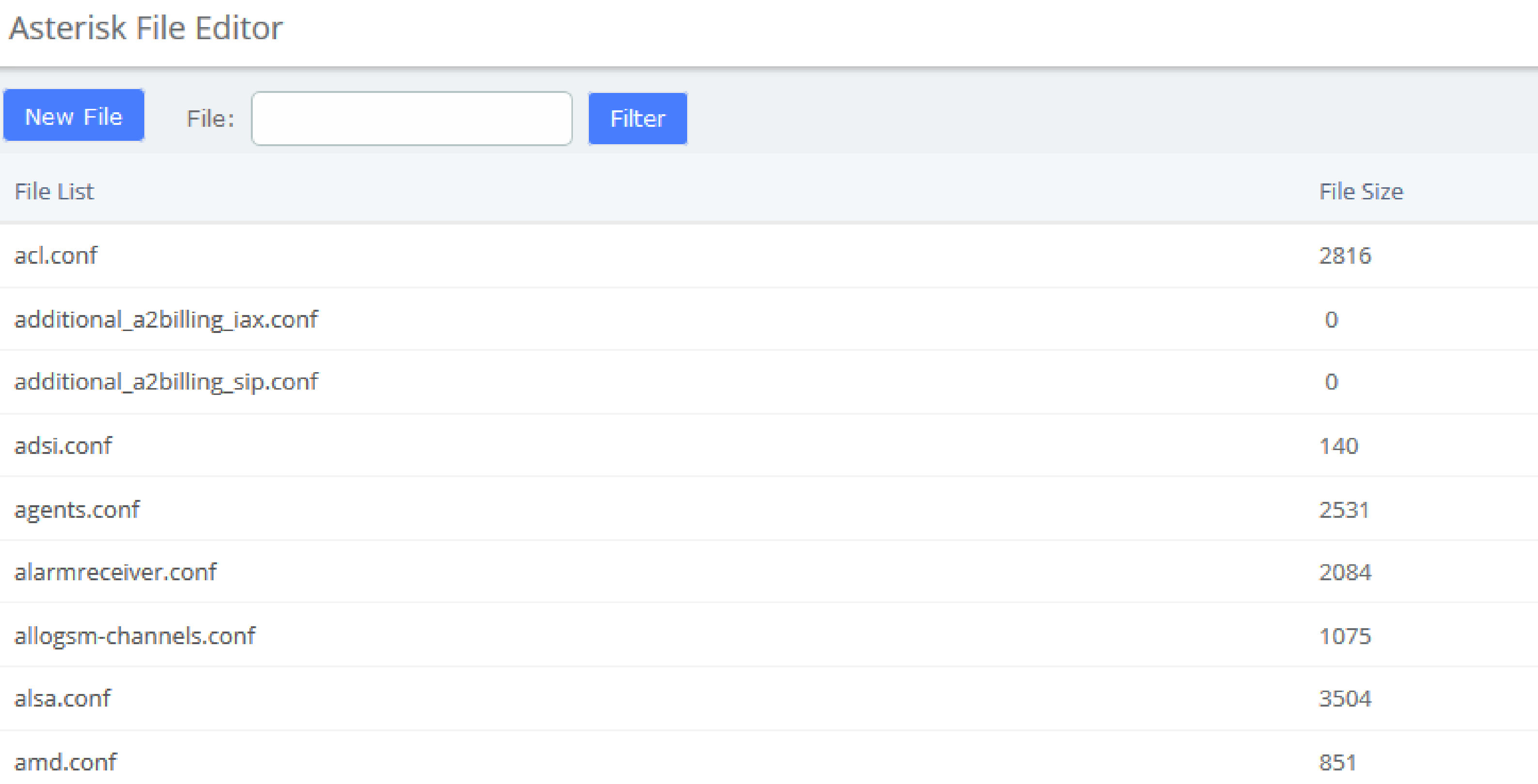
Asterisk-Cli interface

To execute a command, input the same in the Command field and click on the 

button.

3.8.2 Asterisk File Editor

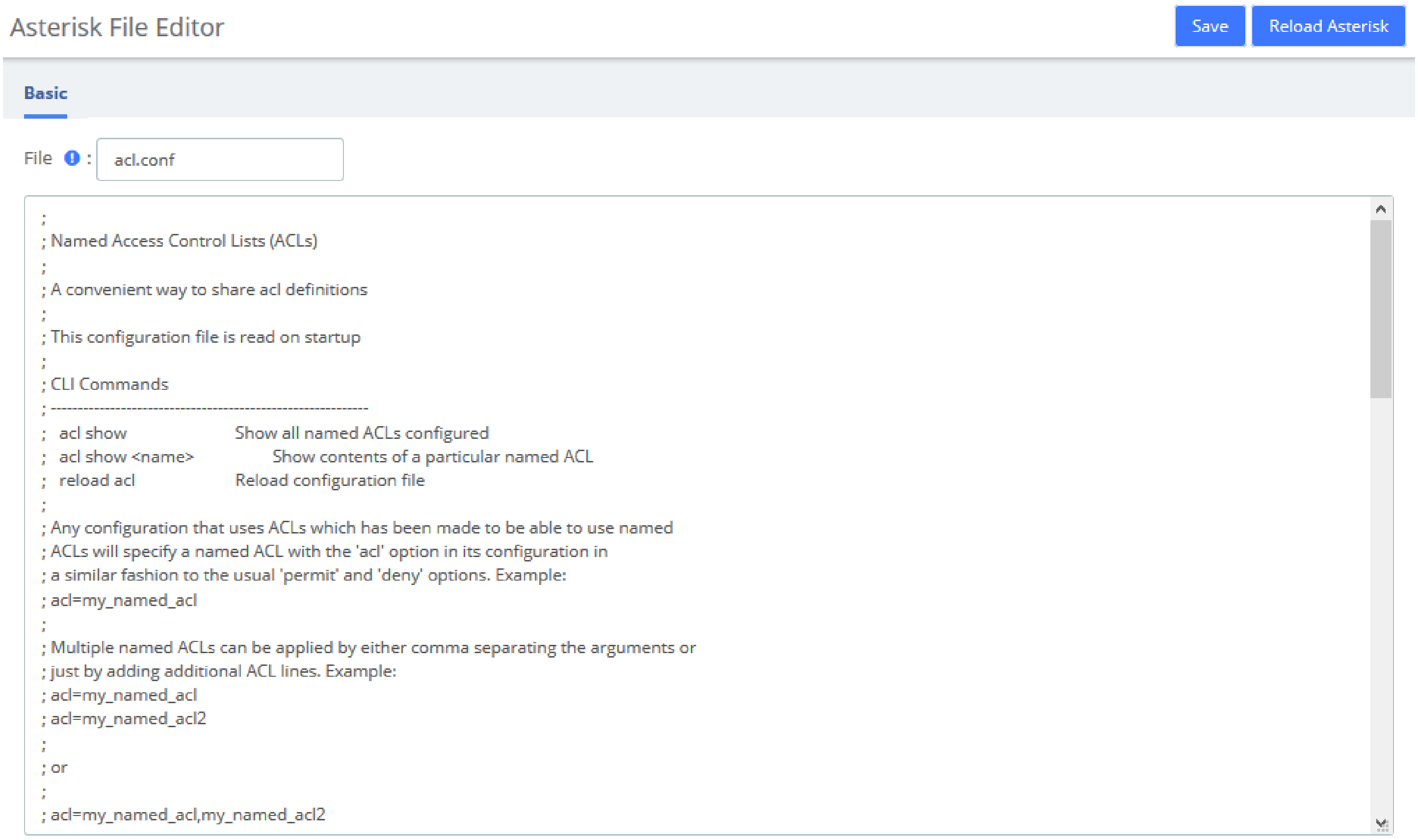
The module "Asterisk File Editor" of the Menu "Tools" in SMB PBX lets us edit easily the configuration files of device, while you have to enter the developer mode before use it. The path of the files you can modify is /etc/asterisk/.



Asterisk File Editor interface

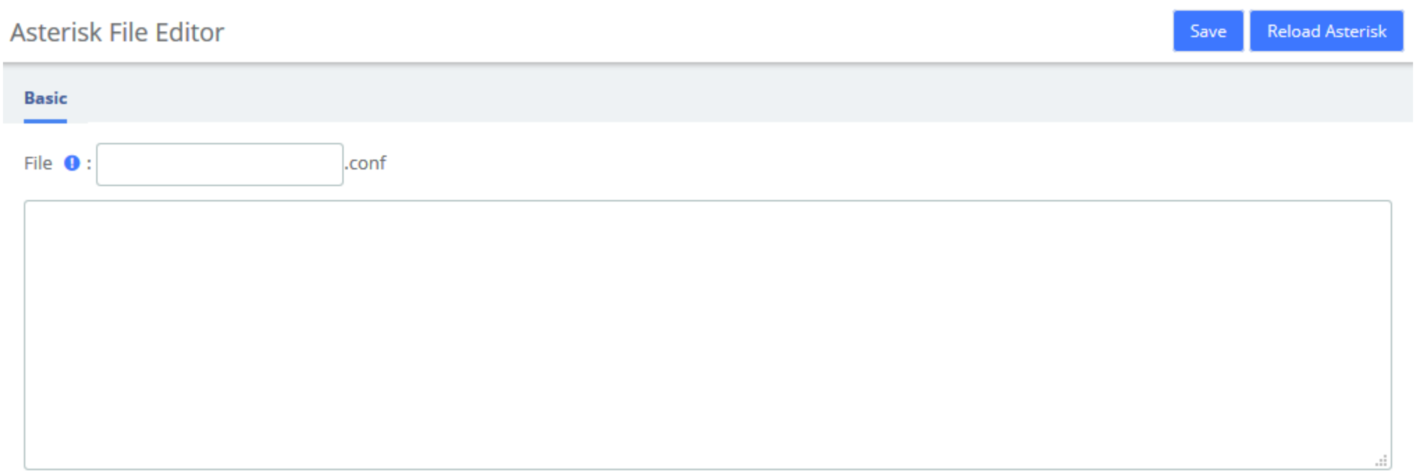
Editing a file

You can find a file by entering the name in the filter field. To edit the file, click on the name to go to the edit mode. Click on "Save" button to save changes and "Reload Asterisk" if necessary.



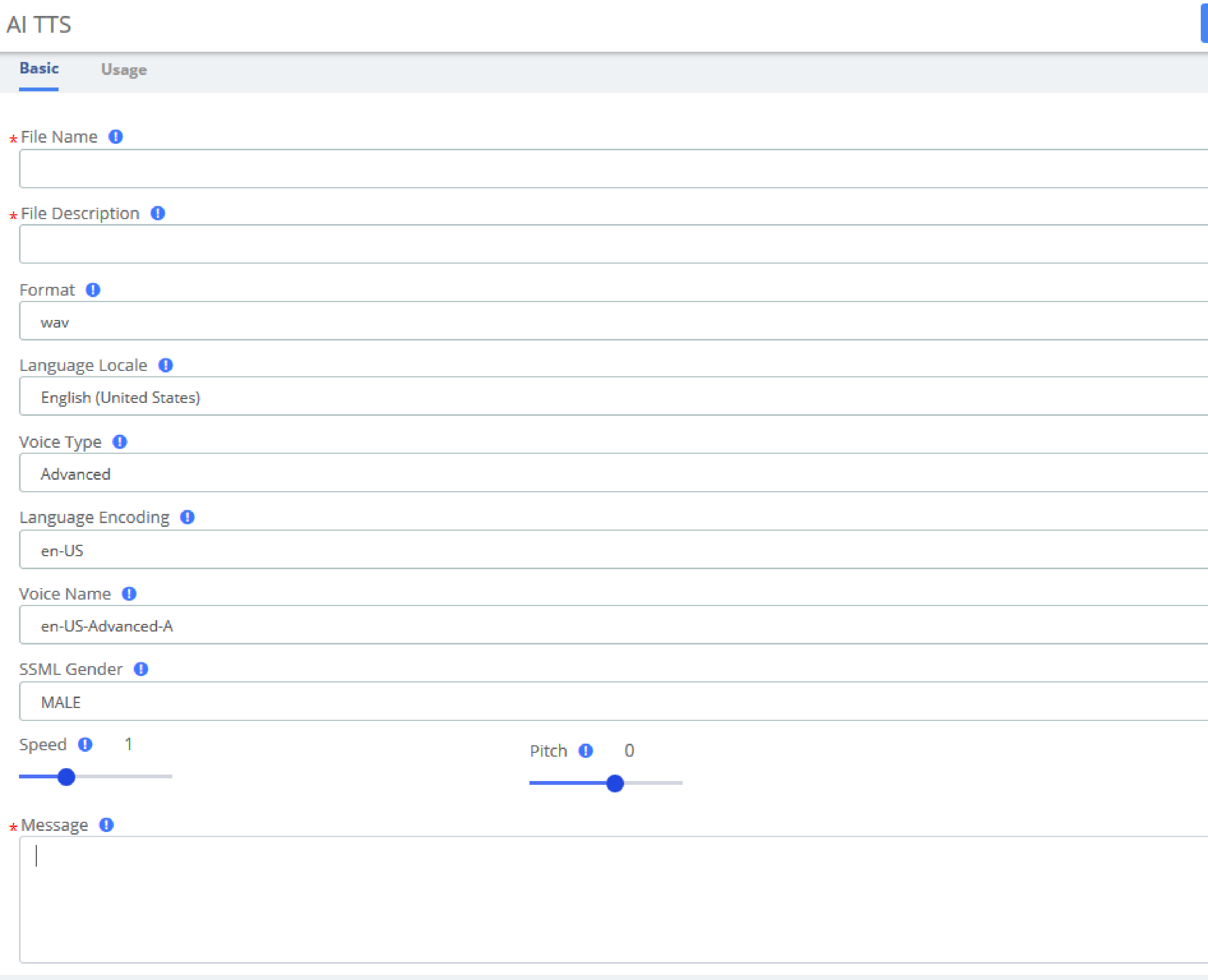
Creating a file

Also you can create a new file by clicking on "New File" button. This file will be created with the extension ".conf" in /etc/asterisk/.



3.8.2 AI TTS

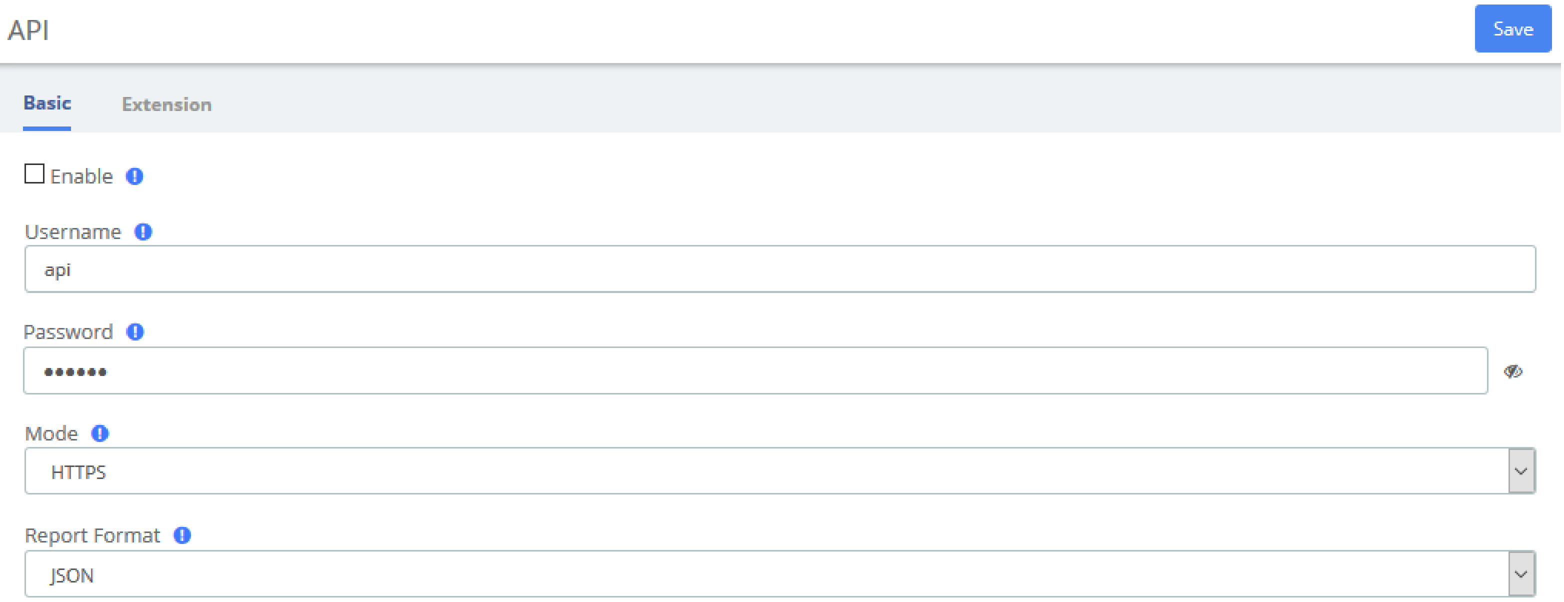
Text can be converted to audio in the "AI TTS" function module. The output format of this file can be ".wav". Write the information you want to convert, select the output format, and click the "Generate Audio File" button. The system will automatically save the file to the location of your hard drive as you requested.



Text to Wav interface

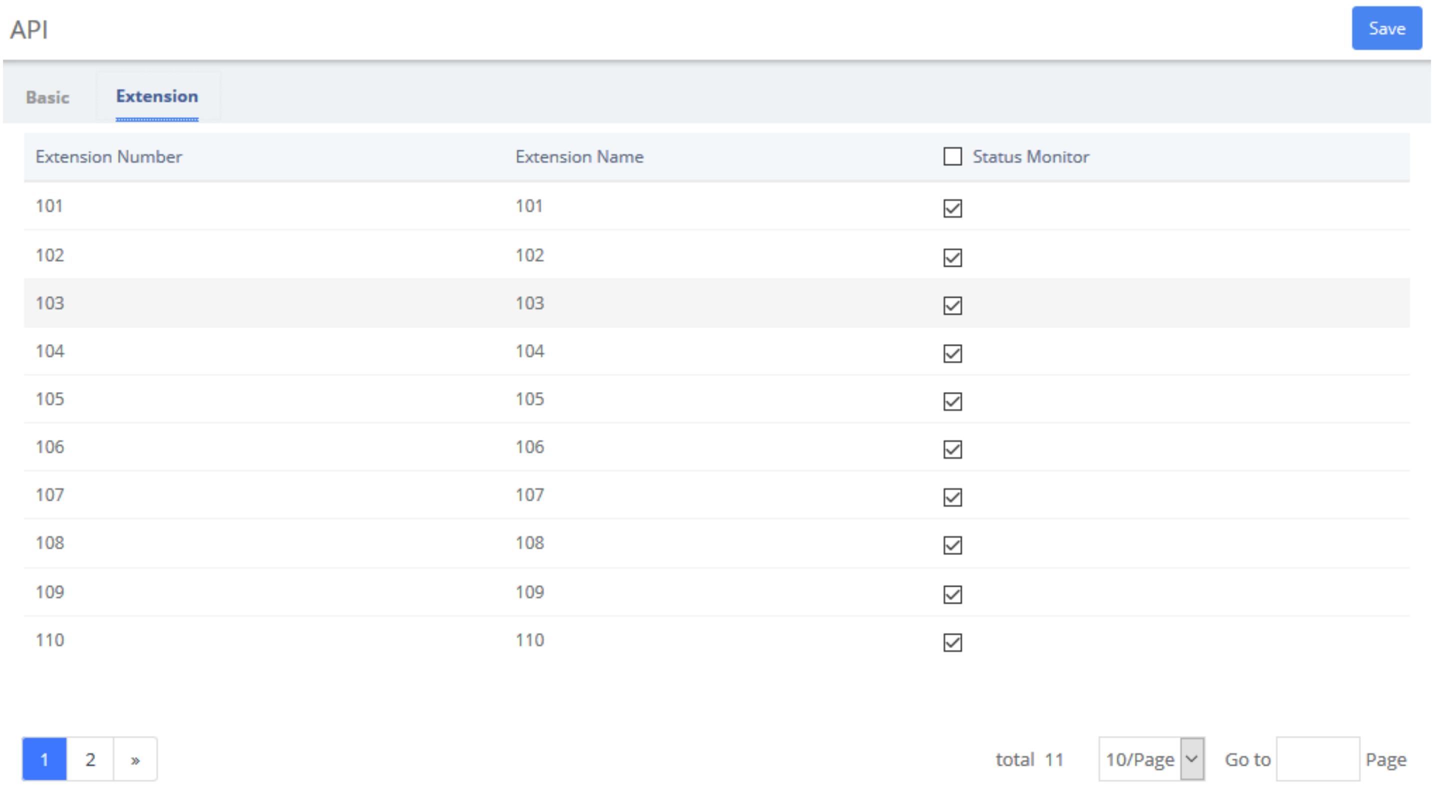
3.8.4 API

This VoIP PBX provides the API interfaces for you to integrate a third-party software or device. You need enable API to access on the PBX and Set the Username and Password, click Save and Apply. The 3rd-party software should use the user name and the password to connect to the PBX API.



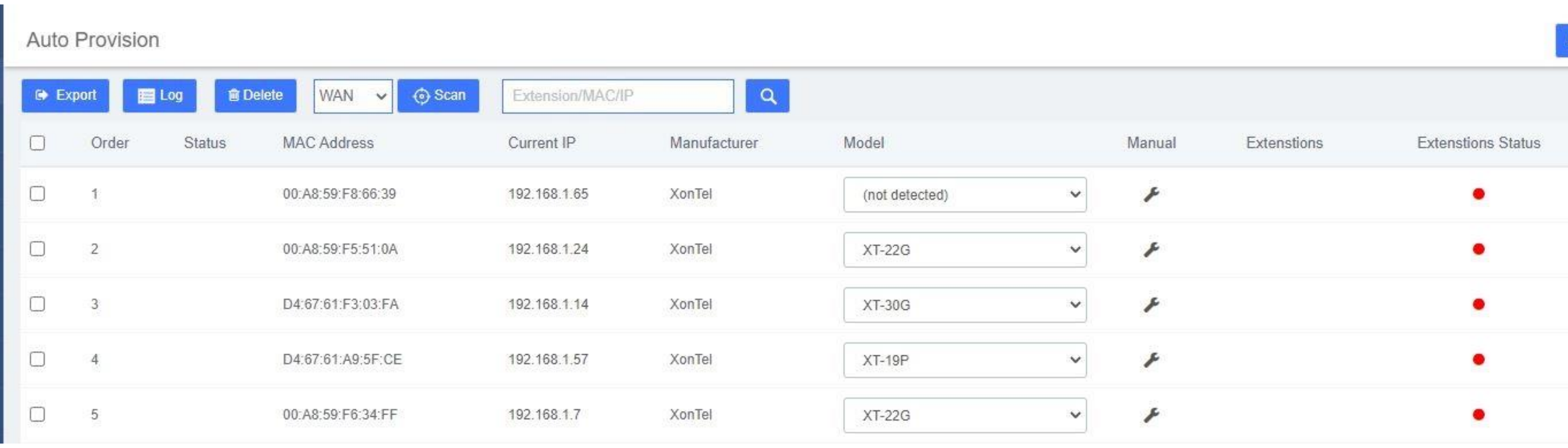
In Extension tab, you can set whether to monitor extension status.

If the extension status is changed, the PBX will send report to the 3rd-party application server.



3.9 Auto Provision

The "Endpoint Configurator" module enables automatic remote configuration of supported endpoints. With this module, the SMB PBX administrator can point supported endpoints to the SMB PBX as their telephony server.



Interface description

Main listing

This is the listing of all endpoints that have been detected or entered. Unlike the old implementation, any endpoints detected or uploaded in past sessions will be kept and displayed until they are explicitly erased. The main listing contains the following columns:

Item	Description
Status	This displays the status of the endpoint as one or more icons. The available flags are as follows: **Scroll icon**: the endpoint has not been scanned, but rather defined in an upload. **Disk icon**: the endpoint configuration has been updated in the database but not yet applied to its configuration files. **Person icon**: the endpoint has at least one endpoint assigned.
MAC Address	This is the main identifier for the endpoint. Configurations in the database and uploaded files are considered to refer to the same endpoint if they reference the same MAC address.

Current IP	If the endpoint was detected through a scan, this field will show the IP at which the endpoint was found. This field is a link to the HTTP configuration interface (if supported) of the phone.
Manufacturer	This displays the detected manufacturer of the endpoint.
Model	This displays the detected model of the endpoint. Since automatic model detection is not (yet) implemented for some manufacturers, this field allows the user to correct the model via a drop-down list. Accurate model detection is required for many other features (such as account assignment) to work.
Options	This link displays a modal dialog on which common options for the endpoint can be manually configured.

Endpoint scan toolbar button

This widget contains a textbox with a network/netmask definition, and a magnifying glass icon. By default, the network definition will be filled with the network definition of the first Ethernet interface of the server. The user may correct this definition to restrict the scan, and then click on the icon to start the scan. When scanning, the toolbar will change to a spinning icon and a Cancel button. As endpoints are detected, they will be added to the main listing, along with their detected manufacturer and model. The toolbar will revert to its default state when the scan is done, or if the scan is aborted with the Cancel button.

Endpoint configuration toolbar button

Select a phone that needs to be configured, click and the following window will pop up, you can clearly see some of the phone's attributes:

Click Accounts to set the extension, and check the line 1/2/3/4 to set the line independently without affecting other lines.

You can also click Network to adjust the network parameters of the phone. When selecting static IP, please make sure to manually enter IP and other parameters.

Apply **Configuration apply toolbar button**

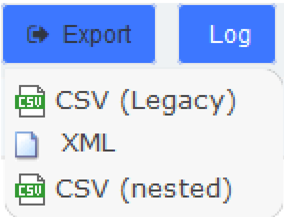
Clicking on this button will start applying the configuration for all selected endpoints (all endpoints for which the checkbox is set). When applying the configuration, the toolbar will change to a progress bar. As endpoints are configured, the progress bar will update, and the toolbar will revert to the default state when the configuration is done. During configuration, a log is generated, and can be viewed by clicking on the Configuration Log toolbar button.

Log **Configuration Log toolbar button**

Clicking on this button will open a modal dialog in which a log of the last configuration run will be shown. This is useful for diagnosing issues with the module failing to configure an endpoint.

Delete **Remove configuration toolbar button**

Clicking on this button will (after a confirmation dialog) remove the database records for the selected endpoints, as well as any generated configuration files for these endpoints. It will NOT, however, contact the endpoints themselves in any way.



Download toolbar button

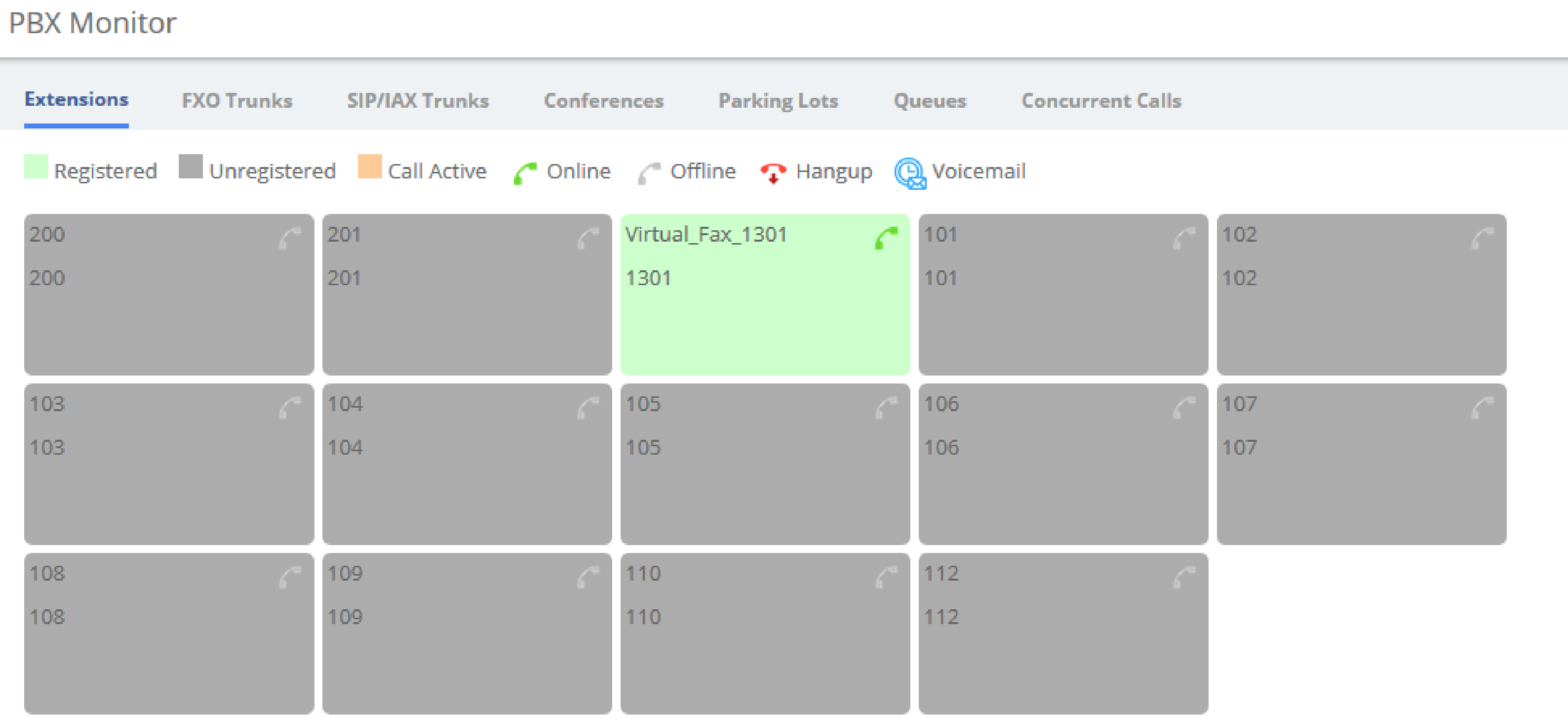
Clicking on this button will display a list of links to download the list of endpoints stored on the database, in three different formats. The supported formats are:

- CSV (Legacy). This is the format used by the old Endpoint Configurator.
- XML. This format allows the definition of endpoints with multiple accounts and properties, as an XML document.
- CSV (Nested). This format can be generated by careful editing in a spreadsheet, and uses indentation to group multiple accounts and properties per endpoint.

3.10 PBX Monitor

The " PBX Monitor " module of the menu "PBX" in SMB PBX allows managing the telephony operations. You can control inbound calls, outbound calls, the order in which the calls are taken, the area that is designated to attend a call, etc.

This module is useful for receptionists who have a general view of the queues, conferences, parking lots, internal extensions, trunks. Here the receptionist can start a call or transfer a call by dragging one extension to another, or include several extensions to a conference room, or a queue. The receptionist can also see the busy extensions, the elapsed time and the caller ID.



Operator Panel interface

Click to forcibly hang up the calling extension.

Click to jump to PBX>Recording>Voicemail

3.11 Conference Panel

The conference panel can help you easily manage and monitor the conference and realize multi-party calls.

3.11.1 Conference List

In the **Conference List**, you can check how many conferences are created on the PBX, and monitor the status of the conferences.

Conference List

Number	Name	Moderator	In-conference	Start Time	Operation
6666	6666	admin	0	----- ----- ----- -----	
8888	8888	admin	0	----- ----- ----- -----	

Conference List Interface

Click the icon to enter the operation panel, as shown below:

Conference List

Invite Selected | Delete | Password Settings | + Add | Open Contacts | Save Contacts

	No.	Caller Number	Name	Duration	Operations
☐		102	102		
☐		106	106		
☐		107	107		

Conference Operation Interface

Click the icon to invite the extension into the conference room. Of course, you can also select multiple extensions and click one key to invite the selected extensions. In addition, extensions can also directly dial the conference room number to actively join the conference room.

Click the icon to kick the extension out of the conference room.

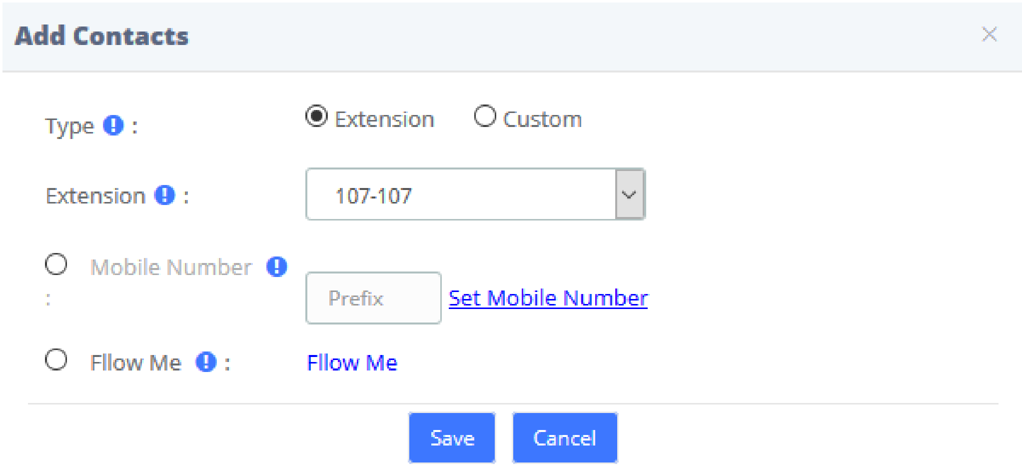
Click the icon, the extension will be muted, and other extensions will not hear the sound of this extension.

Click the icon, the extension will be unmuted, and other extensions can hear the voice of this extension.

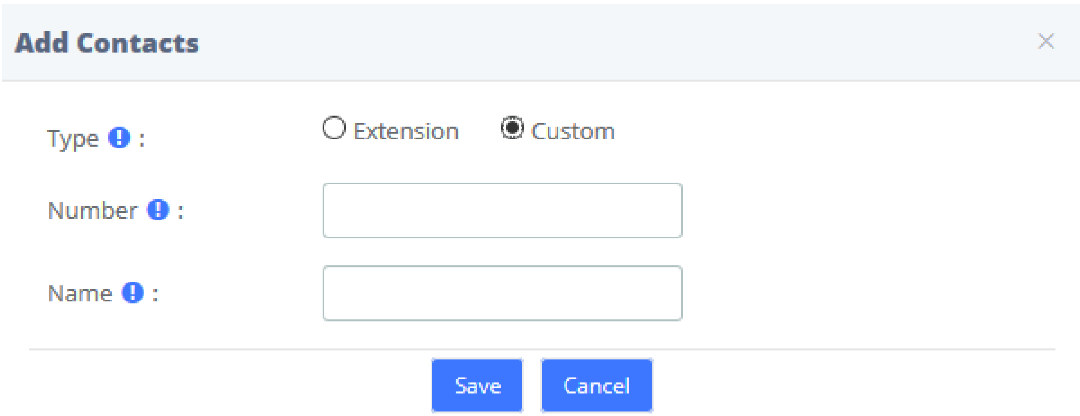
Click the icon to delete the extension from the conference operation panel. Of course, you can also select multiple extensions and Delete the selected extension with one click.

Click Password Settings to change the password of the participant (non-administrator). Leave it blank to indicate that no password is required to enter the meeting. After the change, you need to click **Apply** to take effect.

Click + Add to add a new participant member, as shown below:

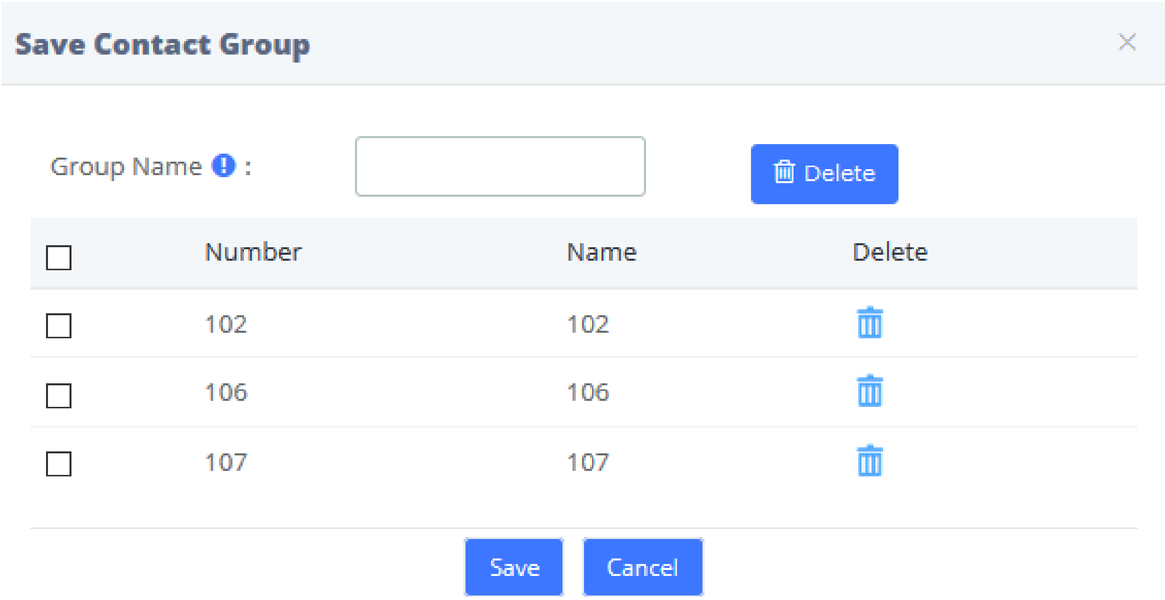


You can also check **Custom** to enter other numbers. This number can be a mobile phone number, and the number can be called from an outside line.



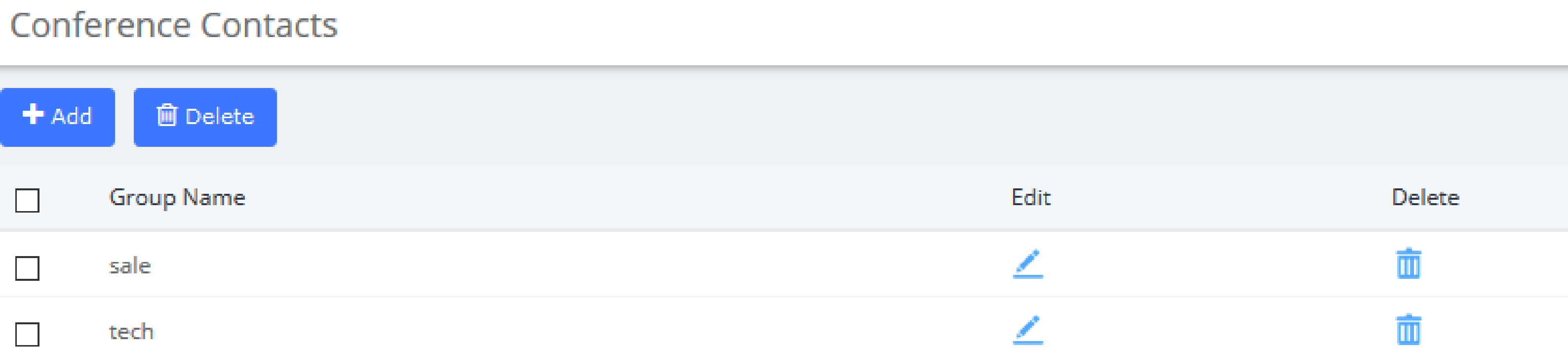
Click Open Contacts, you can select a contact group member to import into the current conference in batches.

Click Save Contacts to save the participants of the current conference to a contact group, as shown in the figure below:

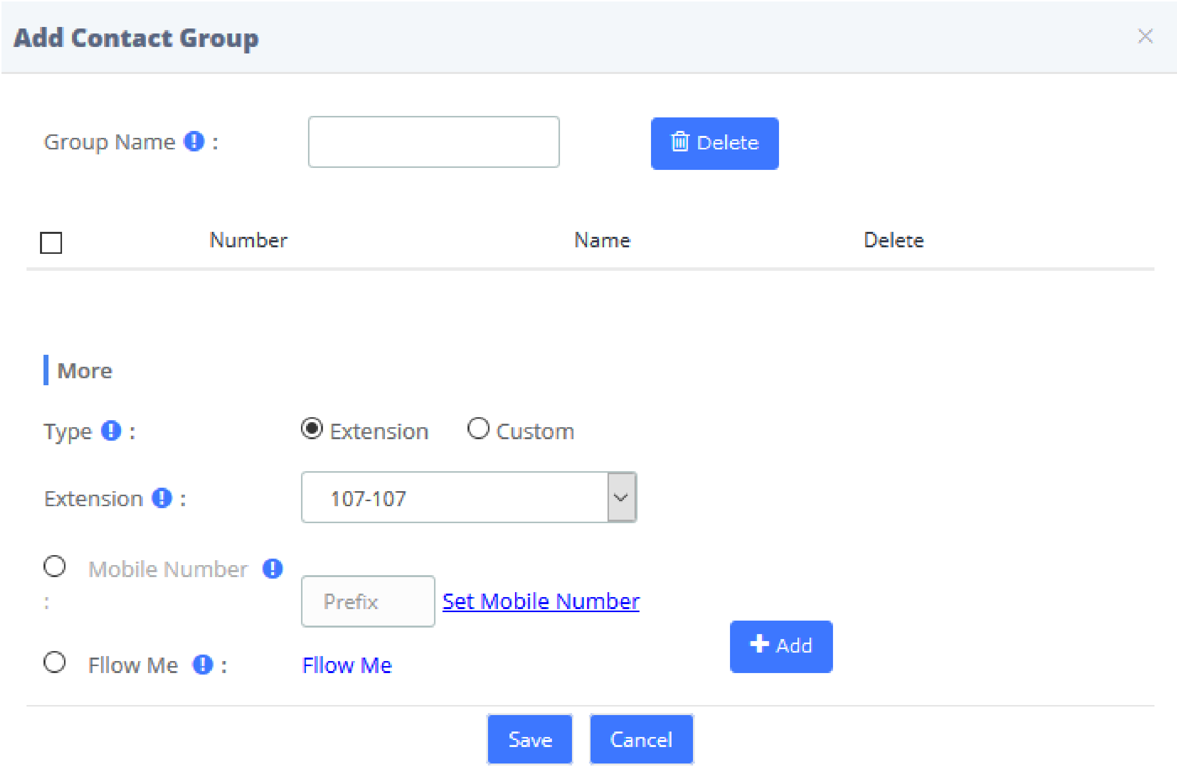


3.11.2 Conference Contacts

In this module, you can manage and add contact groups.



Click + Add to add a contact group, as shown below:

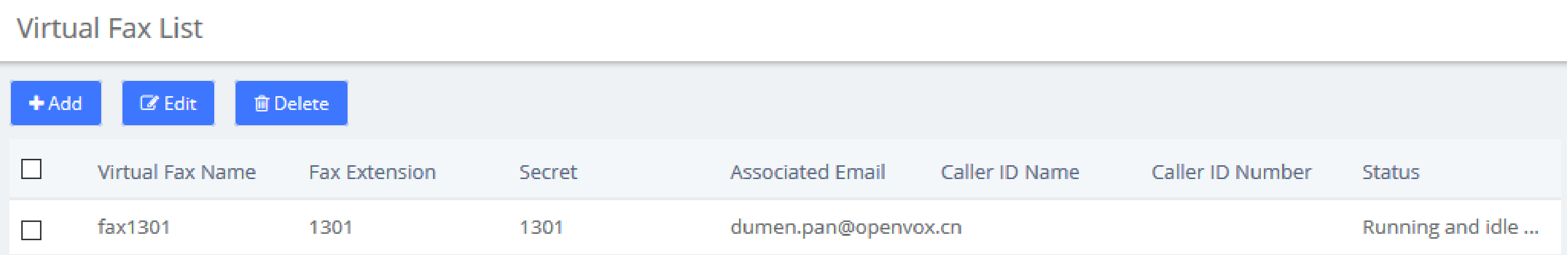


4 Fax

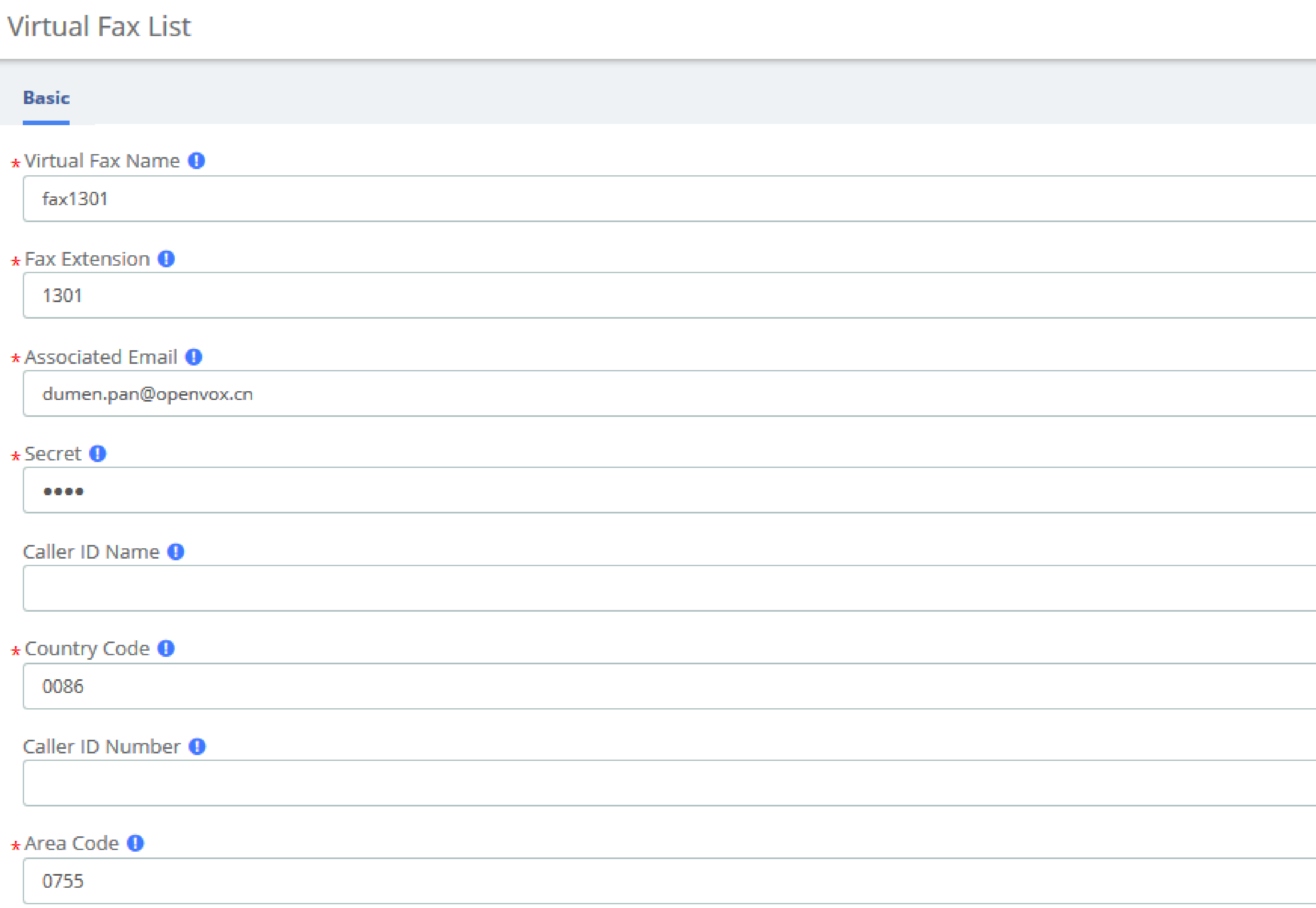
4.1 Virtual Fax

4.1.1 Virtual Fax List

The option **Virtual Fax List** of the Menu **FAX** in SMB PBX lets us verify the list of all the virtual faxes, including the status of each one.

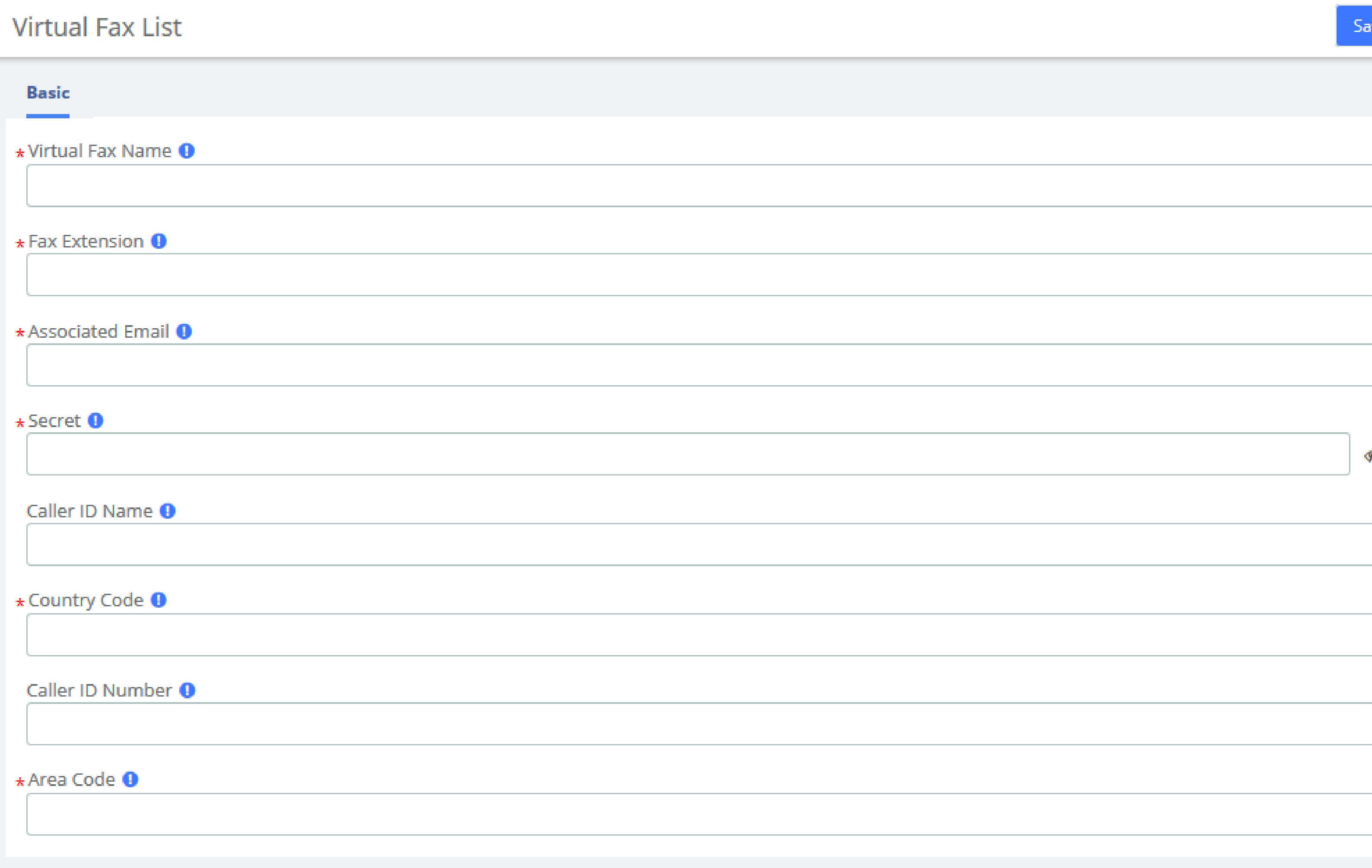


You can **Edit** and **Delete** the Virtual Fax.



Edit Virtual Fax interface

Click +Add you can create a new virtual fax. You should have previously created an IAX extension in **PBX > Extensions > Add IAX2 Extension**.

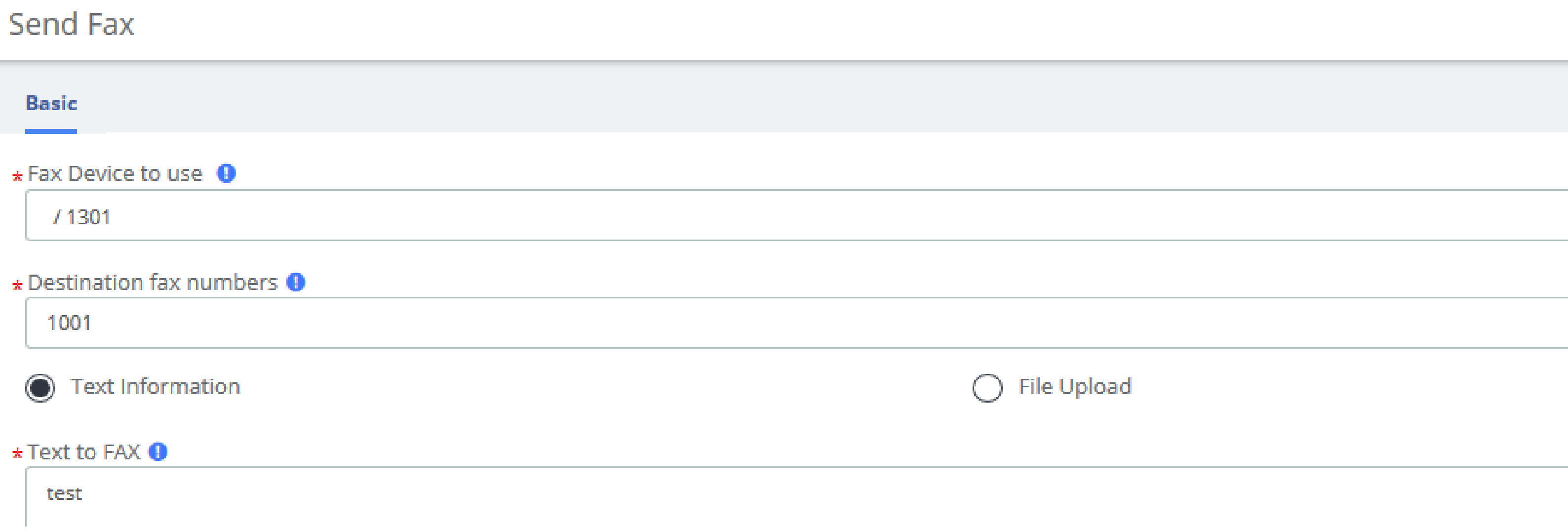


New Virtual Fax interface

To create a new virtual fax, enter the name, e-mail, extension, secret code, country code and area code for the virtual fax (these are the mandatory fields). After this information is added, click on the Save button to save the virtual fax.

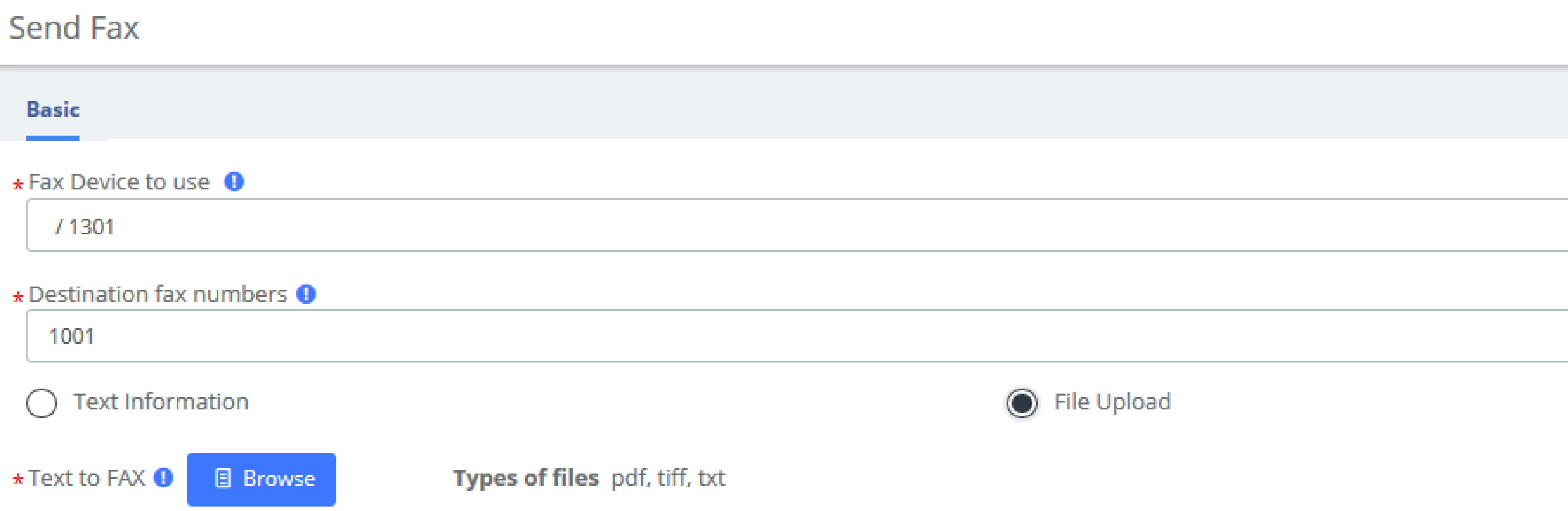
4.1.2 Send Fax

The option **Send Fax** of the menu **Fax** in PBX allows sending faxes to one or more numbers. Here you can enter the text you want to send and click on Send button. (Make sure that you have configured PBX outbound route)



Send fax with text information

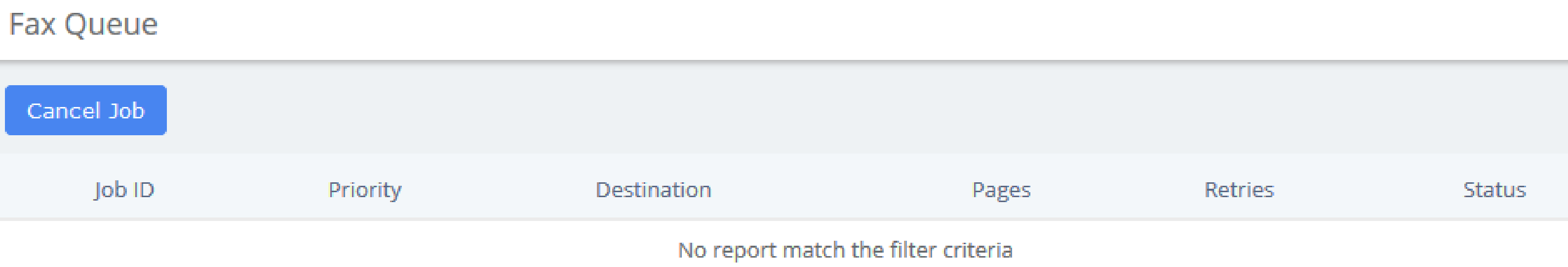
Also, you can send files in the format .pdf, .tiff and .txt



Send fax with File Upload

4.1.3 Fax Queue

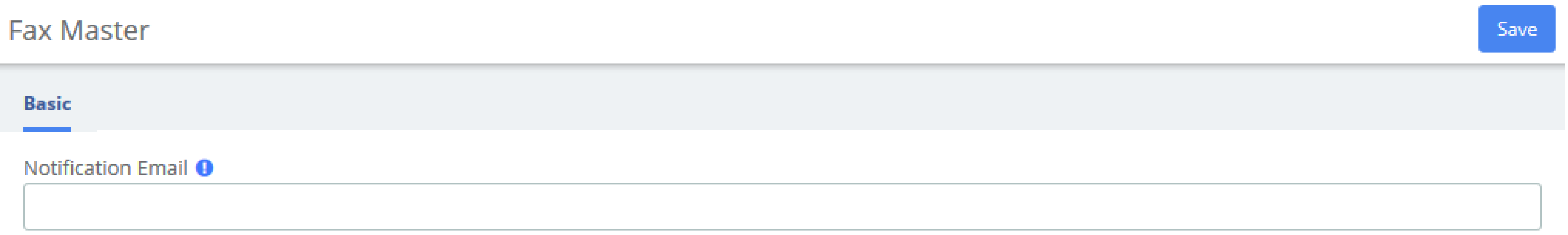
The option **Fax Queue** from the Menu **FAX** in PBX shows the list of faxes that are awaiting its turn to be sent. All the jobs have an ID and a status so you can monitor the sending of the faxes. If you want to cancel a job, just select the job and click on Cancel Job button.



Fax Queue interface

4.2 Fax Master

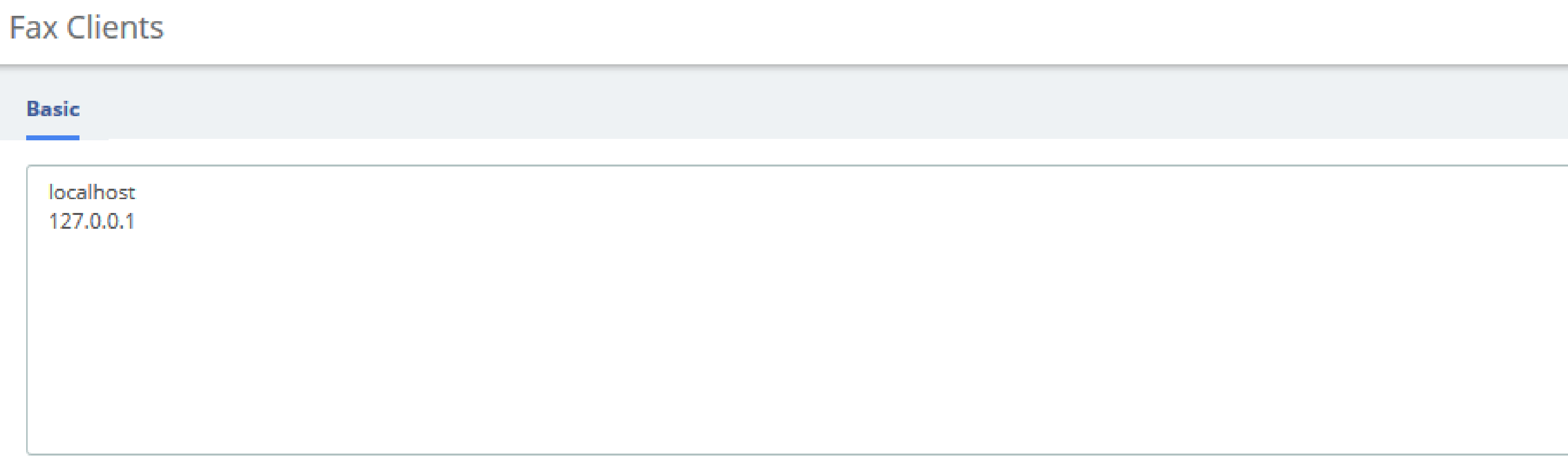
The option **Fax Master** of the Menu **FAX** in SMB PBX lets us input the email address of the administrator of the Fax, and this email will receive notifications of the messages received, errors and other activities of the Fax Server.



Fax Master Interface

4.3 Fax Clients

The option **Fax Clients** of the Menu **FAX** in PBX lets us input the IPs that have permission to send faxes through PBX.



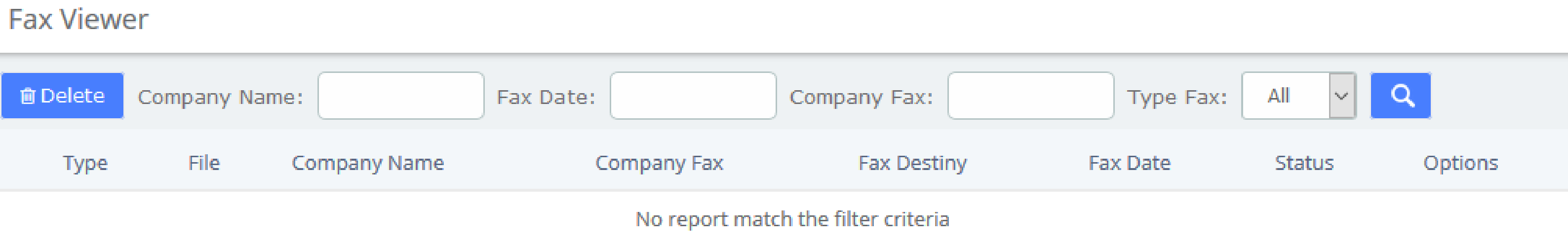
Fax Client interface

Proceed to input the IPs, one IP per line and click on the Save button.

It is recommended that you input the IP 127.0.0.1 and localhost in the configuration because some processes might need to use them.

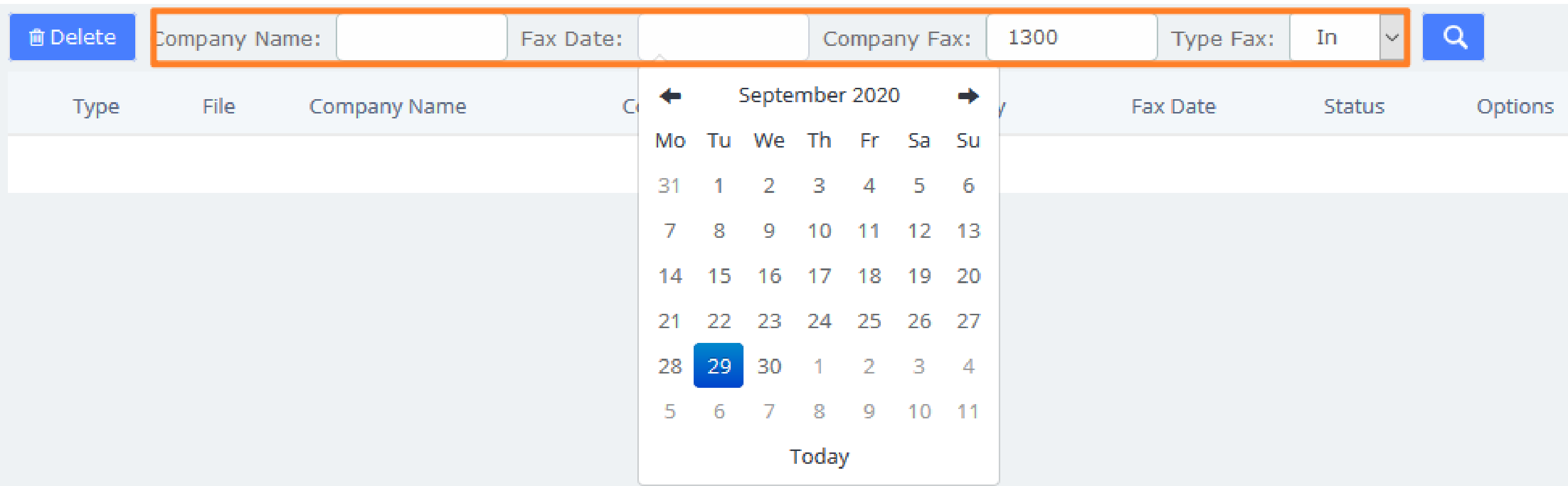
4.4 Fax Viewer

The option **Fax Viewer** of the Menu **Fax** shows a list with all the faxes that have been sent and received in the virtual Faxes. You can download the faxes if you click on the name of the file.



Fax Viewer interface

By the default all the files are shown, but you can filter according to company name, company fax, fax date or type fax.

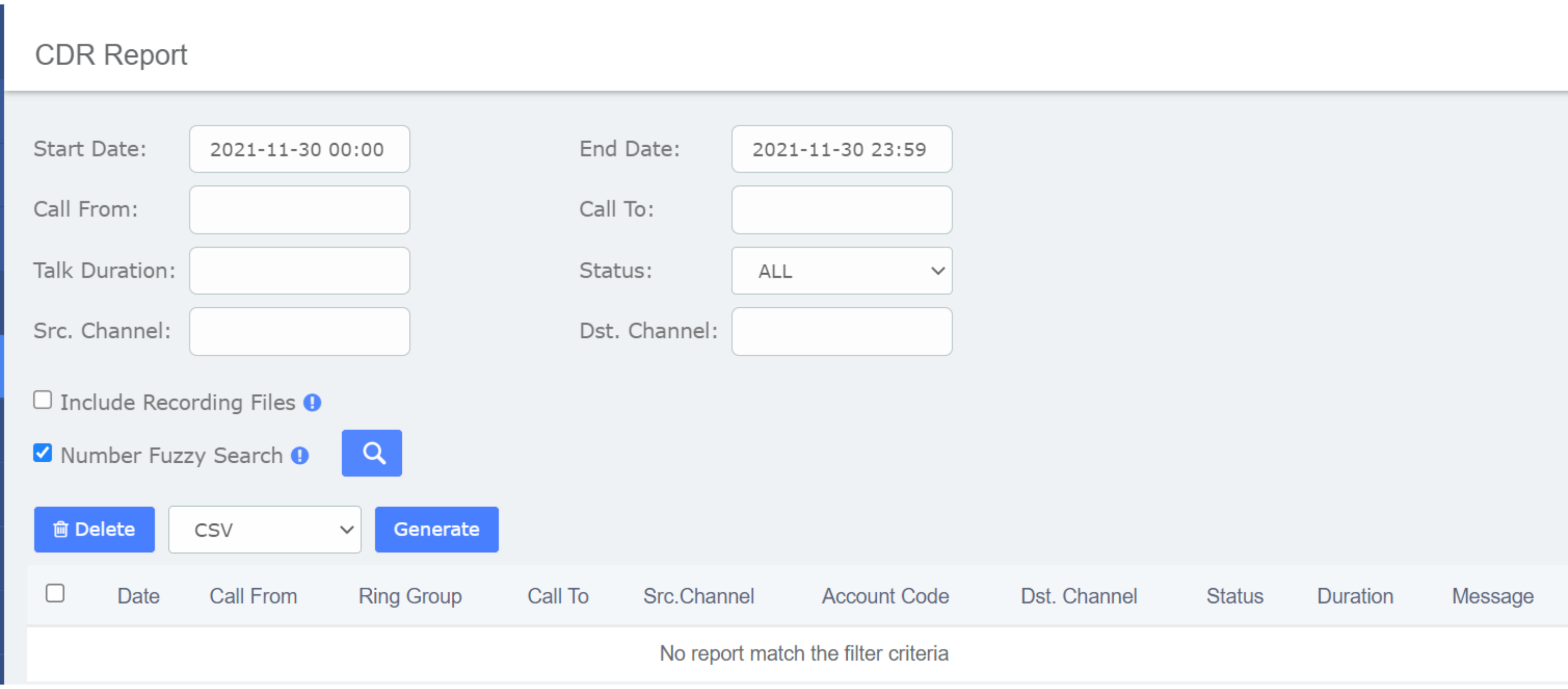


Fax Viewer show filter

5 Reports

5.1 CDR Report

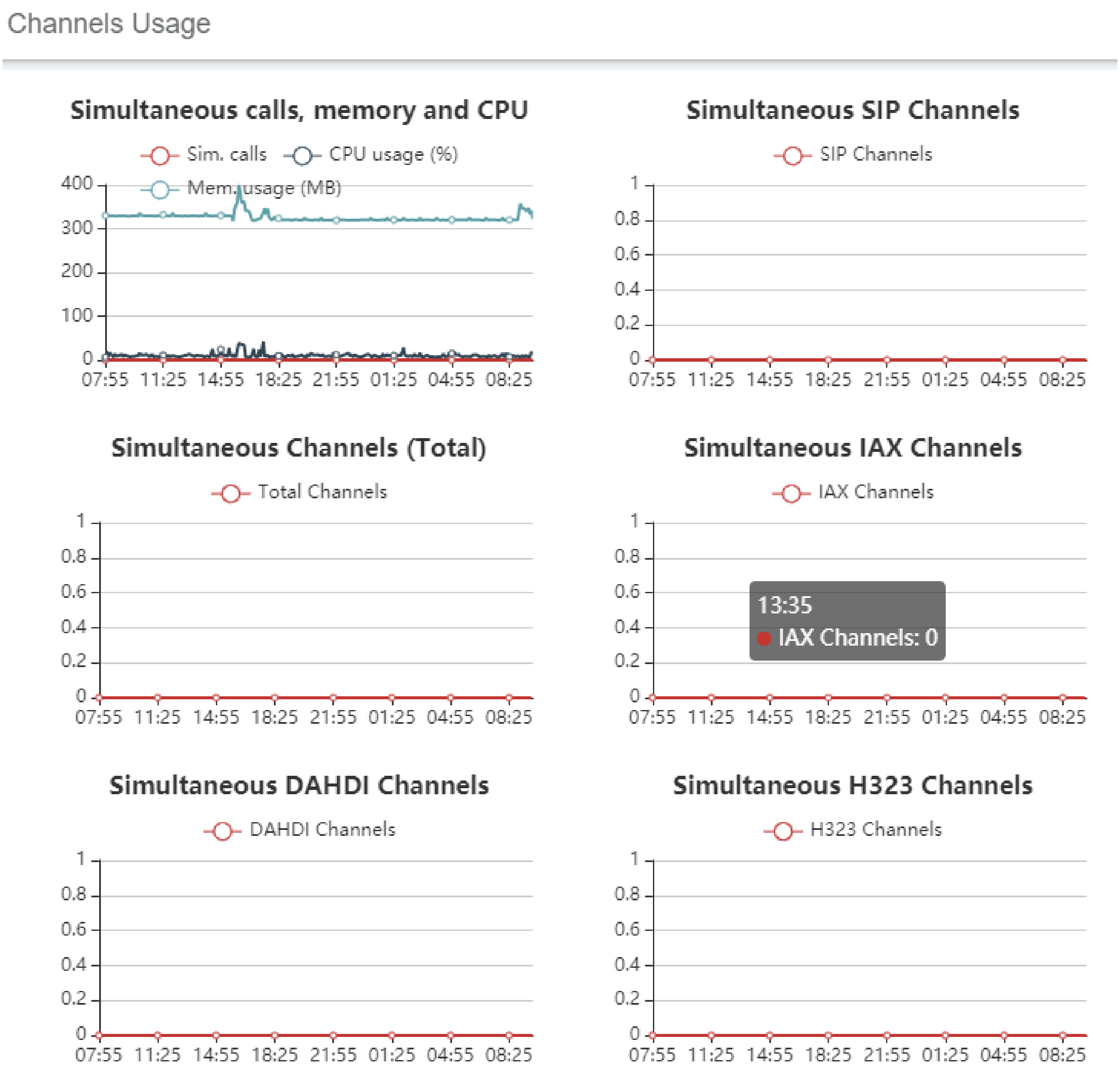
The option **CDR Reports** of the Menu **Reports** in PBX lets us view a list with the details of the calls. You can download this list in different format files such as CSV, XLS and PDF.



CDR Report interface

5.2 Channels Usage

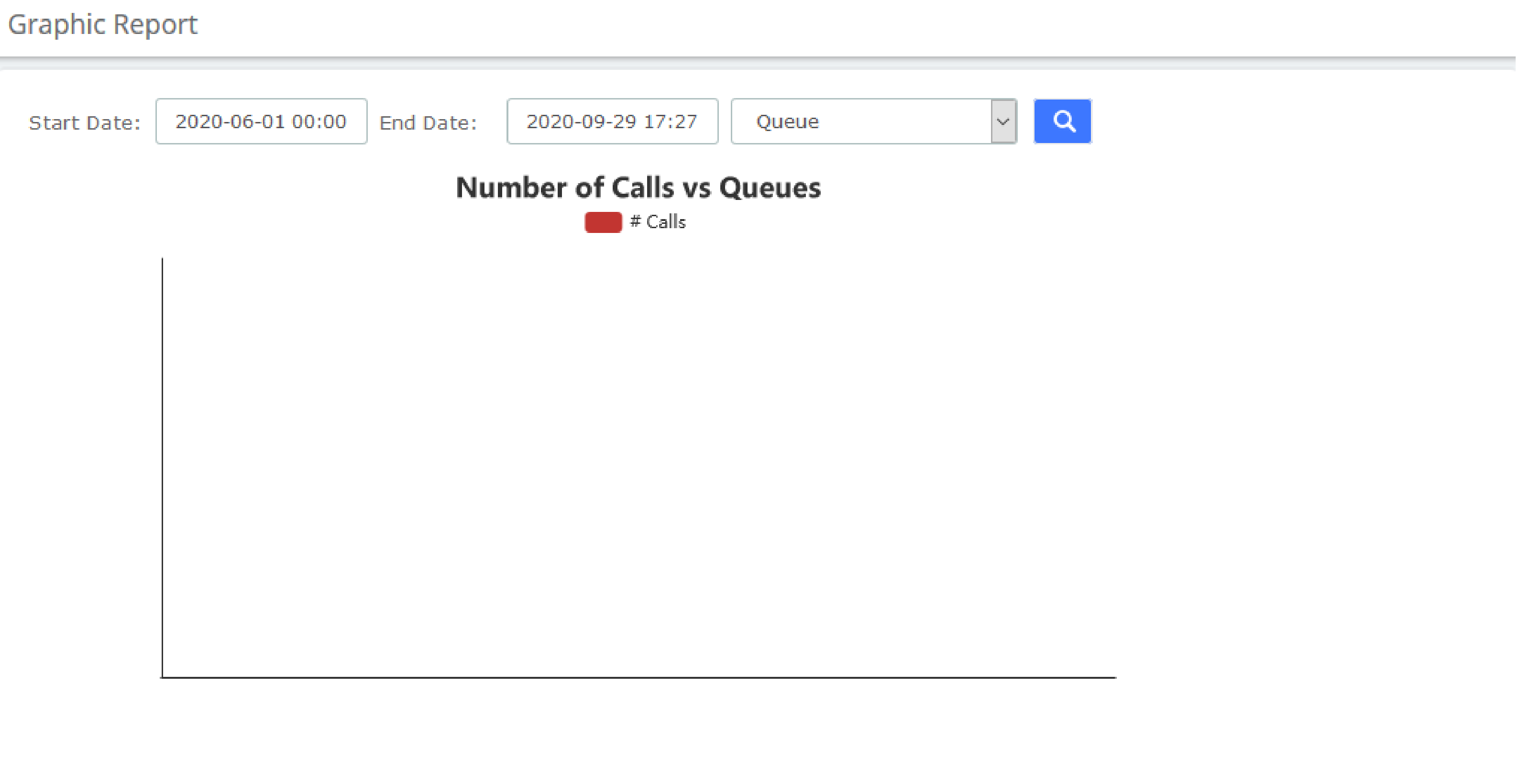
The option **Channels Usage** of the menu **Reports** in SMB PBX allows us view graphically the number of simultaneous calls for each channel.



Channel usage interface

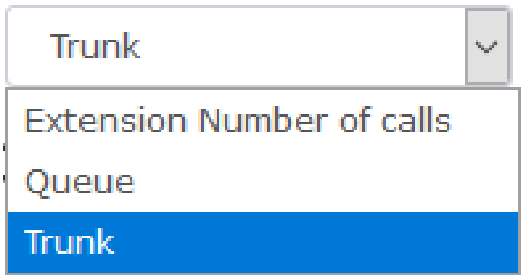
5.3 Graphic Report

The option **Graphic Report** of the **Reports** module allows visualizing graphically information about the number of calls, queues and trunks of the system both in quantity and percentage.



To see the information of a specific extension, select **Extension (Number)** and then click on the link fill in the extension number and click .

It is possible to generate a graphic of Number of Calls in Queues. To do this just select **Queue** from the dropdown menu.



5.4 Summary

The option **Summary** of the menu **Reports** in PBX shows a report of each Extension registered in the server. You can see the number of incoming and outgoing calls, the duration of the calls, the caller id and the dialing number. Use the filter to find an extension or user.

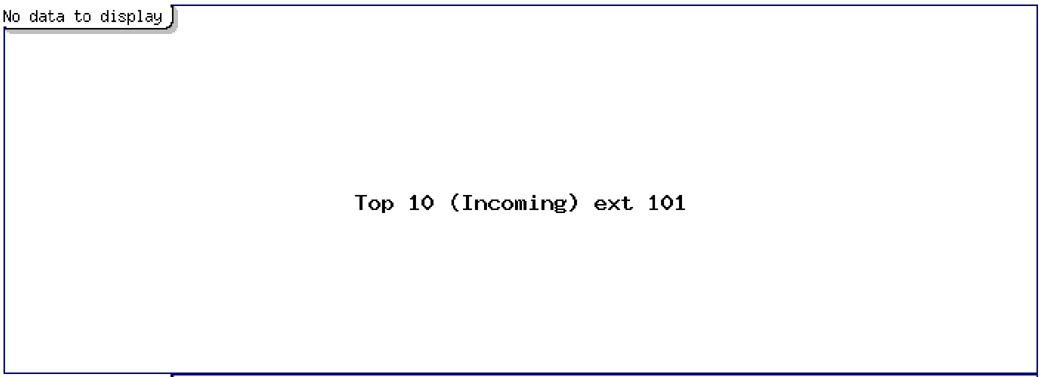
Summary

Start Date: 2020-09-29 00:00 End Date: 2020-09-29 23:59 Filter by: Extension

Extension ▾	User Name	#Incoming Calls	#Outgoing Calls	Total time (Incoming Calls)	Total time (Outgoing Calls)	Details
101	101	0	0	00h. 00m. 00s	00h. 00m. 00s	Call Details
102	102	0	0	00h. 00m. 00s	00h. 00m. 00s	Call Details
103	103	0	0	00h. 00m. 00s	00h. 00m. 00s	Call Details
104	104	0	0	00h. 00m. 00s	00h. 00m. 00s	Call Details
105	105	0	0	00h. 00m. 00s	00h. 00m. 00s	Call Details
106	106	0	0	00h. 00m. 00s	00h. 00m. 00s	Call Details
107	107	0	0	00h. 00m. 00s	00h. 00m. 00s	Call Details
108	108	0	0	00h. 00m. 00s	00h. 00m. 00s	Call Details
109	109	0	0	00h. 00m. 00s	00h. 00m. 00s	Call Details
110	110	0	0	00h. 00m. 00s	00h. 00m. 00s	Call Details

1 2 » total 14 10/Page Go to Pag

Click on **Call Details** to see more information of an extension.

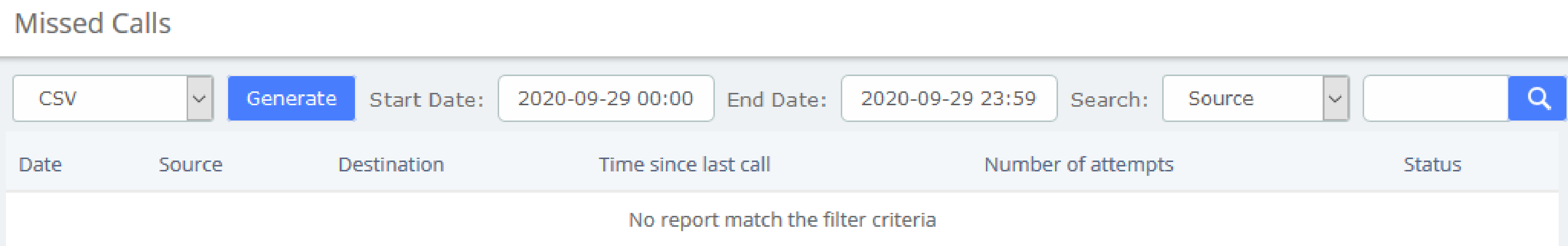


5.5 Missed Calls

The option **Missed Calls** of the menu **Reports** in SMB PBX shows a report of the missed calls of all extensions so you can know when an extension has been receiving calls. You can download this report by clicking on "Download" button. The available formats for this file are csv, xml and pdf

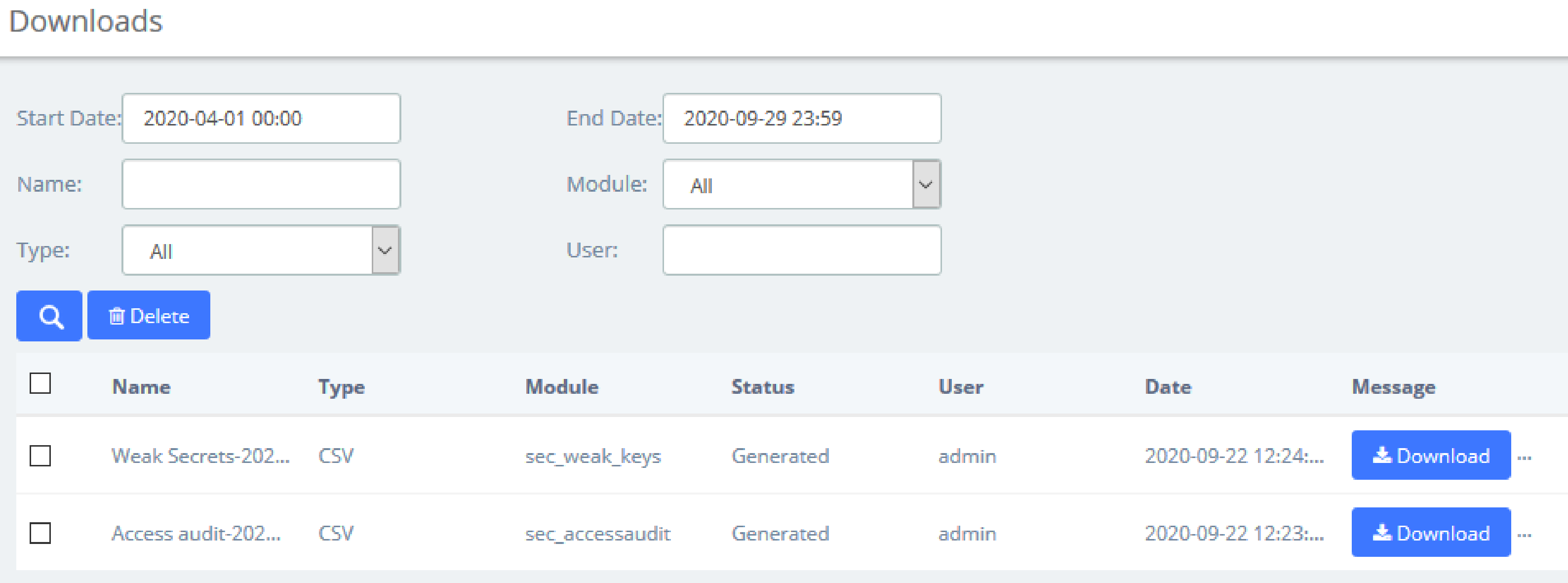
You can filter the results by:

- **Start Date:** Find missed calls from this date.
- **End Date:** Find missed calls until this date.
- **Search:** You can filter the results by these parameters:
  - ***Source:*** Number that made the call.
  - ***Destination:*** Number that received the call.



5.6 Downloads

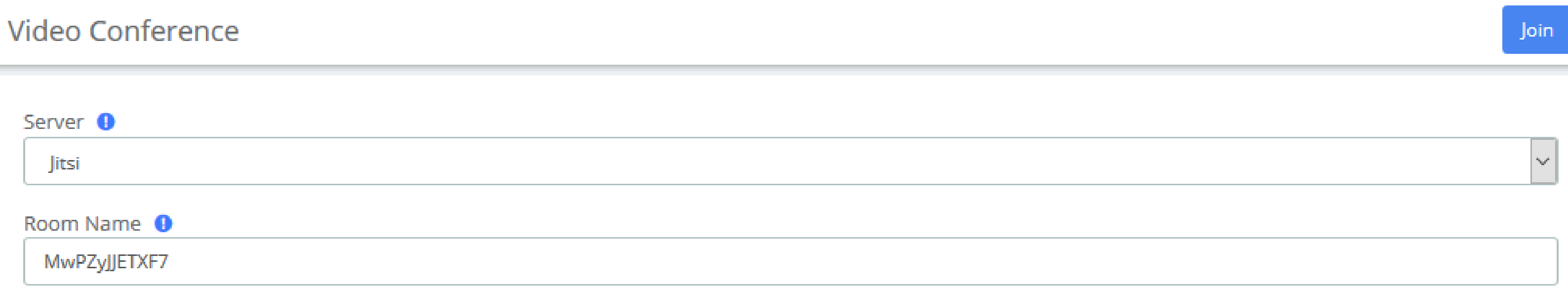
In **Downloads**, users can find and download reports generated in previous modules by themselves, including **CDRs**, **call recordings**, **event logs**, **missed calls**, **booking list**, **audits**, and more.



6 Extras

6.1 Video Conference

Users can create video conferences in the SMB PBX system, allowing multiple people to participate at the same time.



Video Conference interface

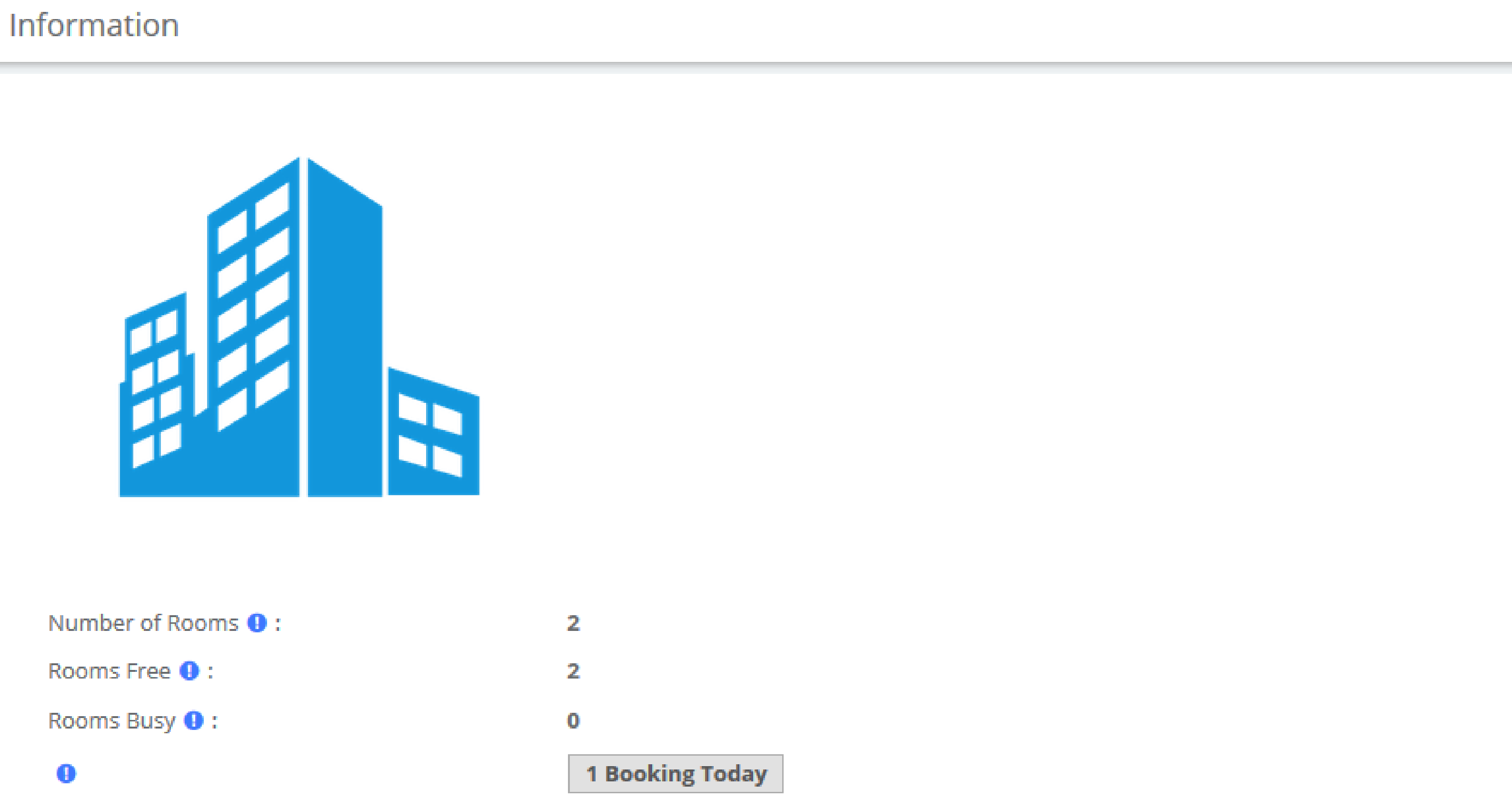
6.2 Hotel

6.2.1 Information

On the **Extras** > **Hotel > Information** page, you can see some information directly on this page.

The total rooms, how many available rooms (rooms free), or not (rooms busy), if you have some booking today, if your hotel is full or potentially full (caused by the booking).

If there's a booking today, just click on the 1 Booking Today button to going to **Booking list** menu directly.



Hotel Information

6.2.2 Service

Room List

You can see the status of the rooms currently in your hotel. The guest name, the room name, if it's free or busy, cleaned or not. If the guest used the mini-bar or not. If the room is on DND (Do Not Disturb) status or not. And you can see if the room is included in a group or not.

If the phone device is a SIP phone, you can know if the phone is connected or not. In this case, you have a small yellow triangle beside the phone number.

Room List

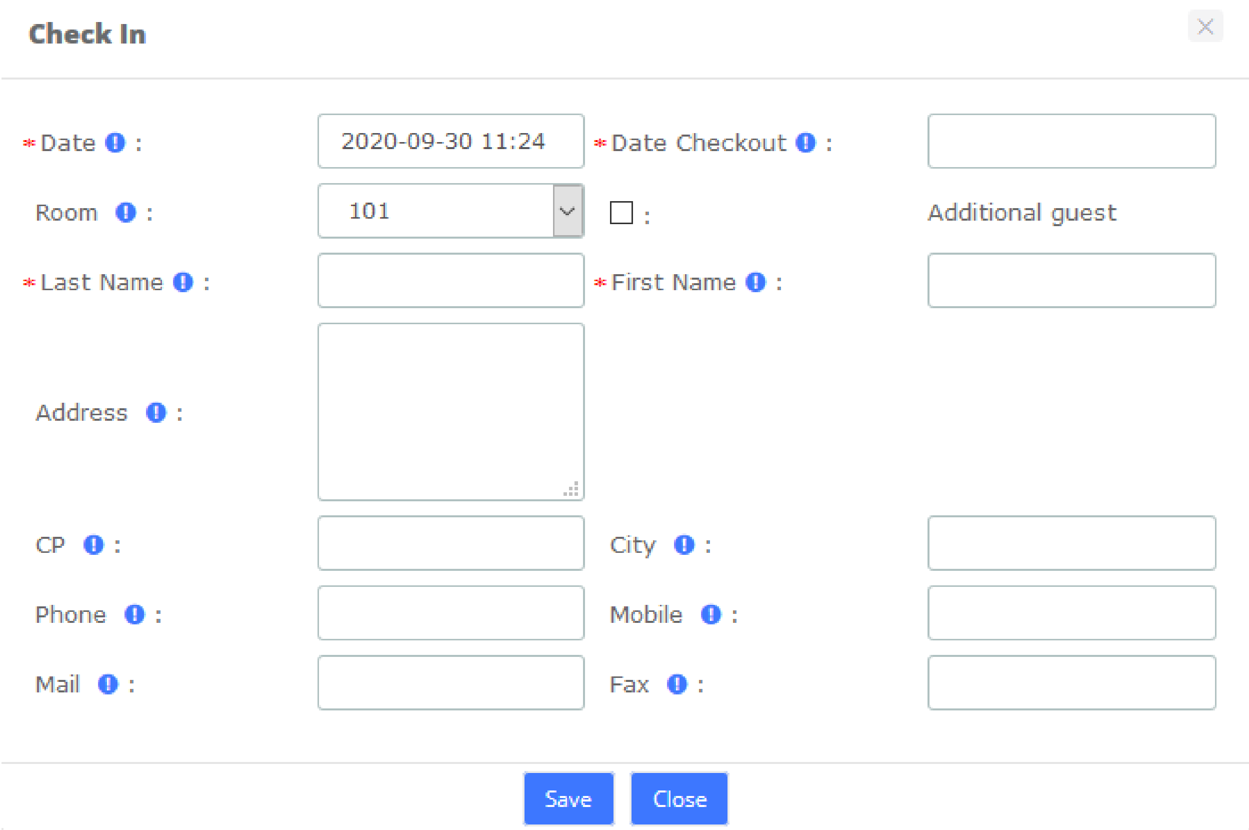
Search: Name

Details	Name	Room Name	Extension	Model	Group	Transfer	Clean	Check In/Check Out	Mini bar	DND	Wakeup
	Free	301	102 ⚠	Double			✓	Check In			
	2 30	302	103 ⚠	Double			✓	Check Out			
	Free	303	104 ⚠	Double			✓	Check In			
	Free	304	105 ⚠	Double			✓	Check In			
	Free	305	106 ⚠	Double			✓	Check In			
	Free	306	107 ⚠	Double			✓	Check In			
	Free	307	108 ⚠	Double			✓	Check In			
	Free	308	109 ⚠	Double			✓	Check In			
	Free	309	110 ⚠	Double			✓	Check In			
	Free	201	201 ⚠	Simple			✓	Check In			

Room List

Check In

You can Check In for a new customer. You can see some fields to enter different values.

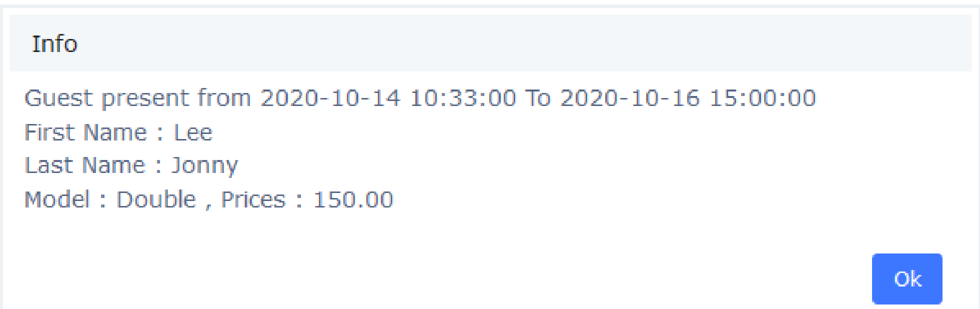


Check-In option

Date, it's the Check-in date (the current date by default). **Date Checkout**, is needed to have a reference in the case where another room will be booked. This information is not used for billing. **Room**, displays all available rooms into this list. Of course, you must enter a **Last Name** and **First Name** to making a check-in.

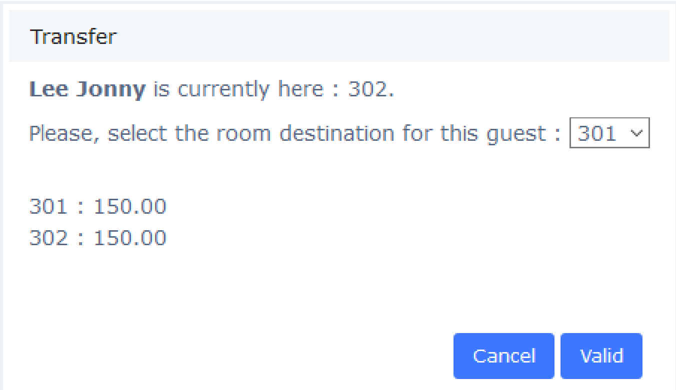
The other fields below are optional. However, one field is needed in the case where you want to sending the billing by mail. In this case, you must enter the **Mail** field. No billing will be sent by fax yet.

Once the guest is checked-in, you can click  and view the Guest's info. Here you can see the customer's name, check-in time, check-out time, room price and other information.



Guest Info

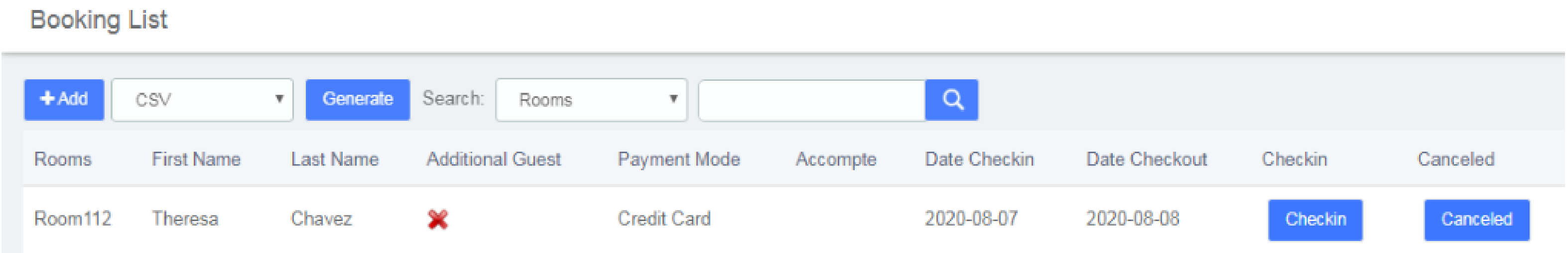
When the customer needs to change the room, select the  icon.



Transfer

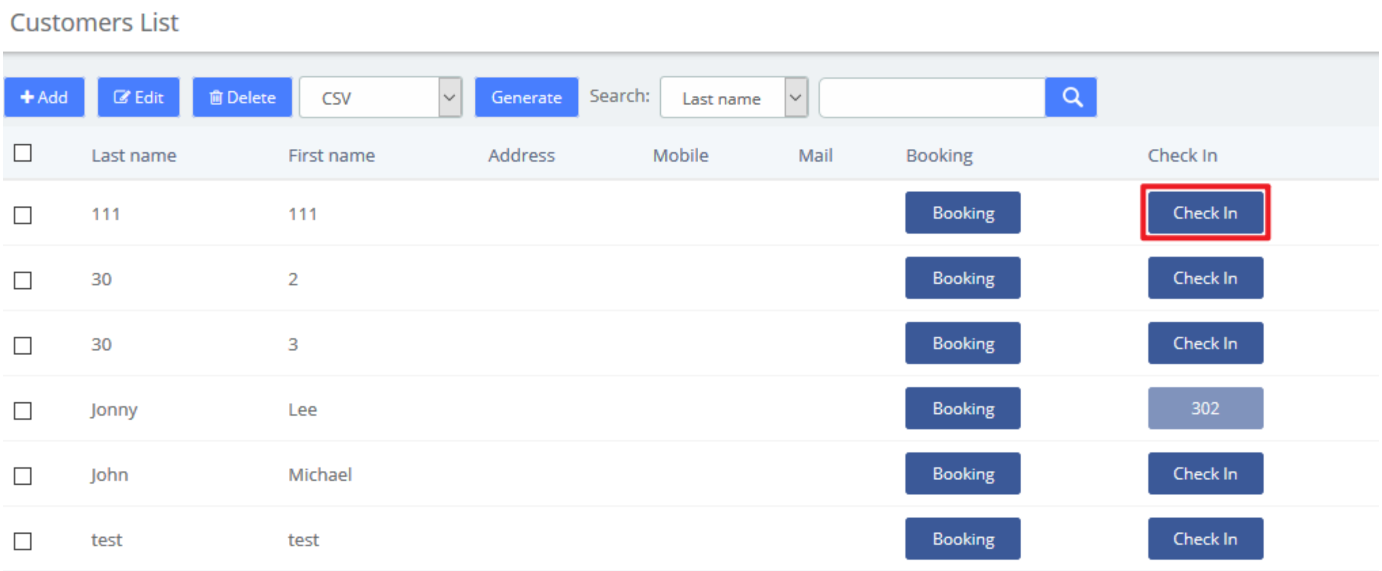
If the customer has purchased a Mini-Bar item or made an outside call, the system will automatically transfer the customer's purchase and call costs to the new room.

And you also can check-in for a customer who have booked in **Booking List**:



Rooms	First Name	Last Name	Additional Guest	Payment Mode	Accompte	Date Checkin	Date Checkout	Checkin	Canceled
Room112	Theresa	Chavez	✖	Credit Card		2020-08-07	2020-08-08	Checkin	Canceled

Or check-in for regular customers in **Customers List**:



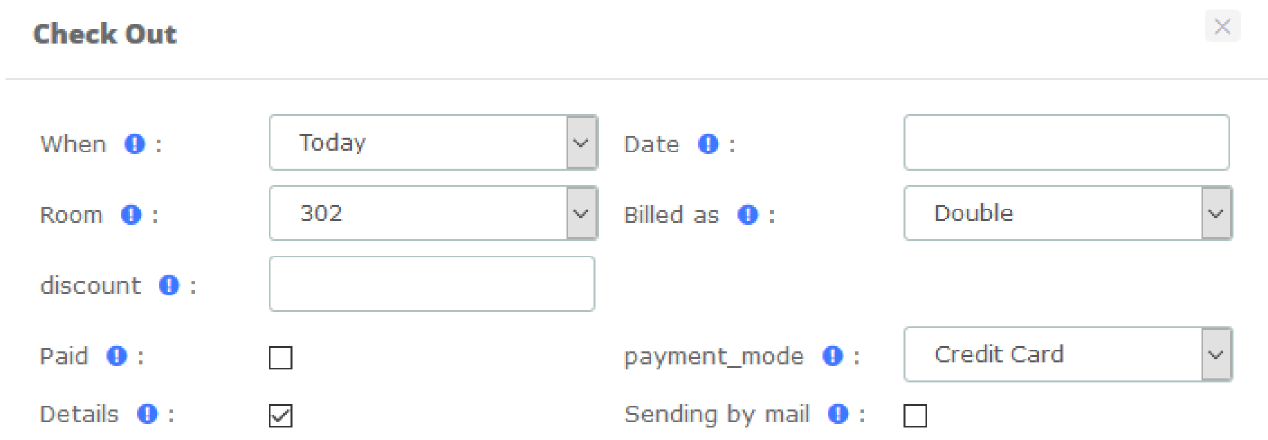
☐	Last name	First name	Address	Mobile	Mail	Booking	Check In
☐	111	111				Booking	Check In
☐	30	2				Booking	Check In
☐	30	3				Booking	Check In
☐	Jonny	Lee				Booking	302
☐	John	Michael				Booking	Check In
☐	test	test				Booking	Check In

Check Out

You can do 2 types of checkout. A classic checkout by room, and a checkout by group.

When the customer needs to check out, select Check Out icon.

Check out by room



Check out by room options

If paid is checked, the billing is paid by the guest, else, this billing is tagged like not paid.

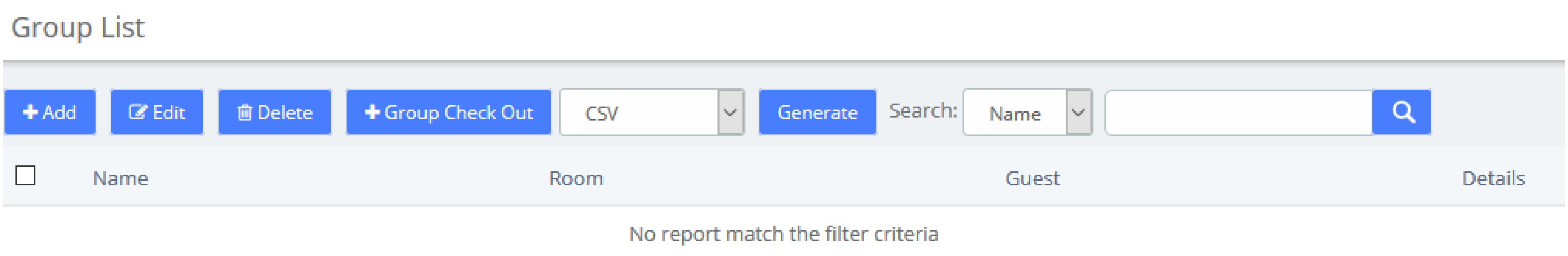
If you want to have all calls details for the room, check  .

If you want that guest receive its billing by mail, check  .

After checking out, you can check the billing report in **Report > Billing Report**.

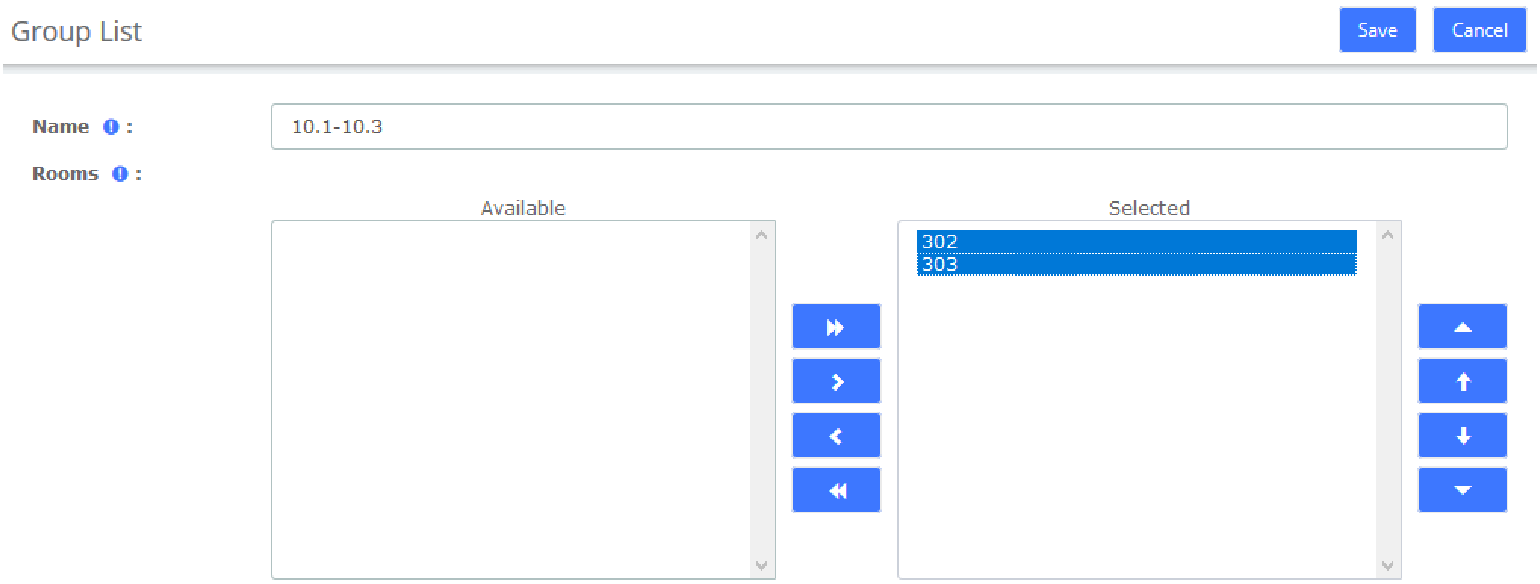
Group List

Group List is used for unified management of customers who check-in in groups.



Group List

Here, you can see all group already existing, and you can add lots of checked rooms into a group in the same time. Just selecting several rooms maintaining, press the *shift* key and click on the rooms that you want.

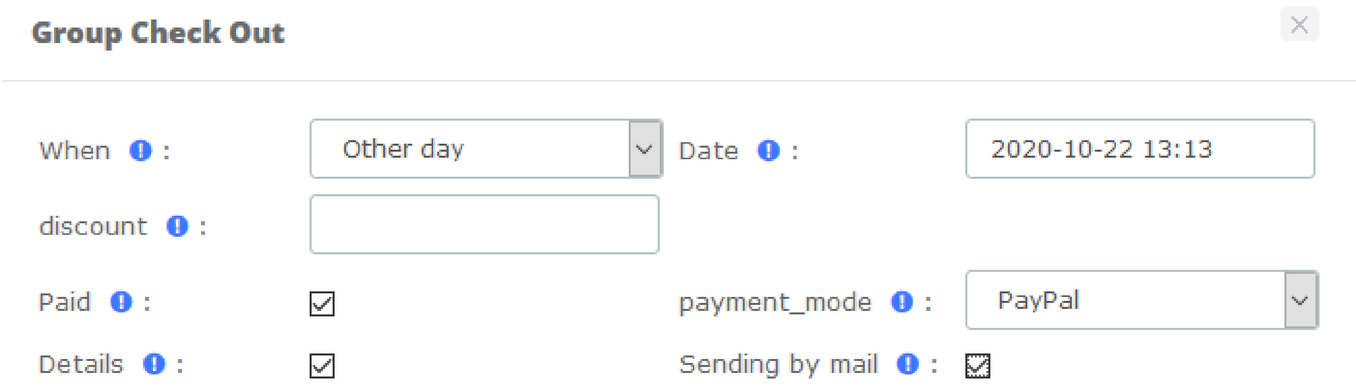


Group List options

Check Out

Checkout by group will take all room in group, and will make the checkout, room by room.

Check the group you want check out and click **+ Group Check Out**. Checking out of *10.1-10.3* group means that both Room 302 and Room 303 will check out.

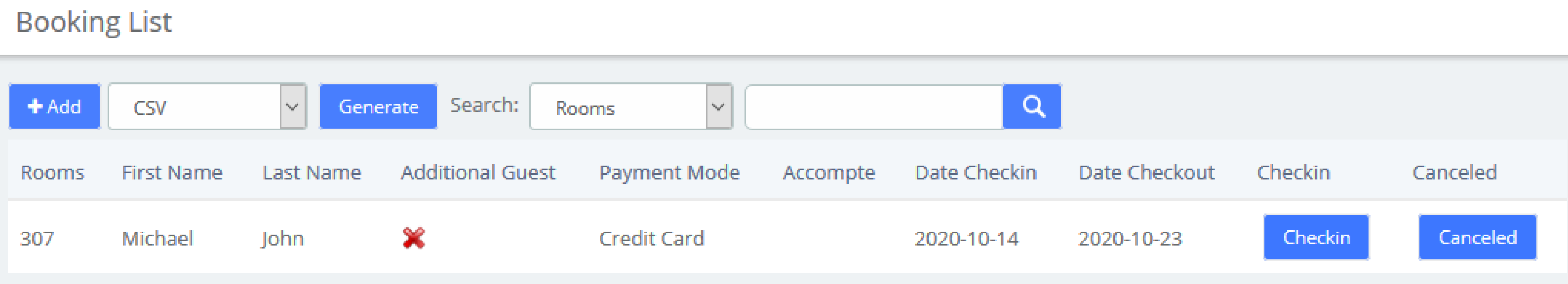


Check out by group option

When checking out, check **Details** to see the call bill.

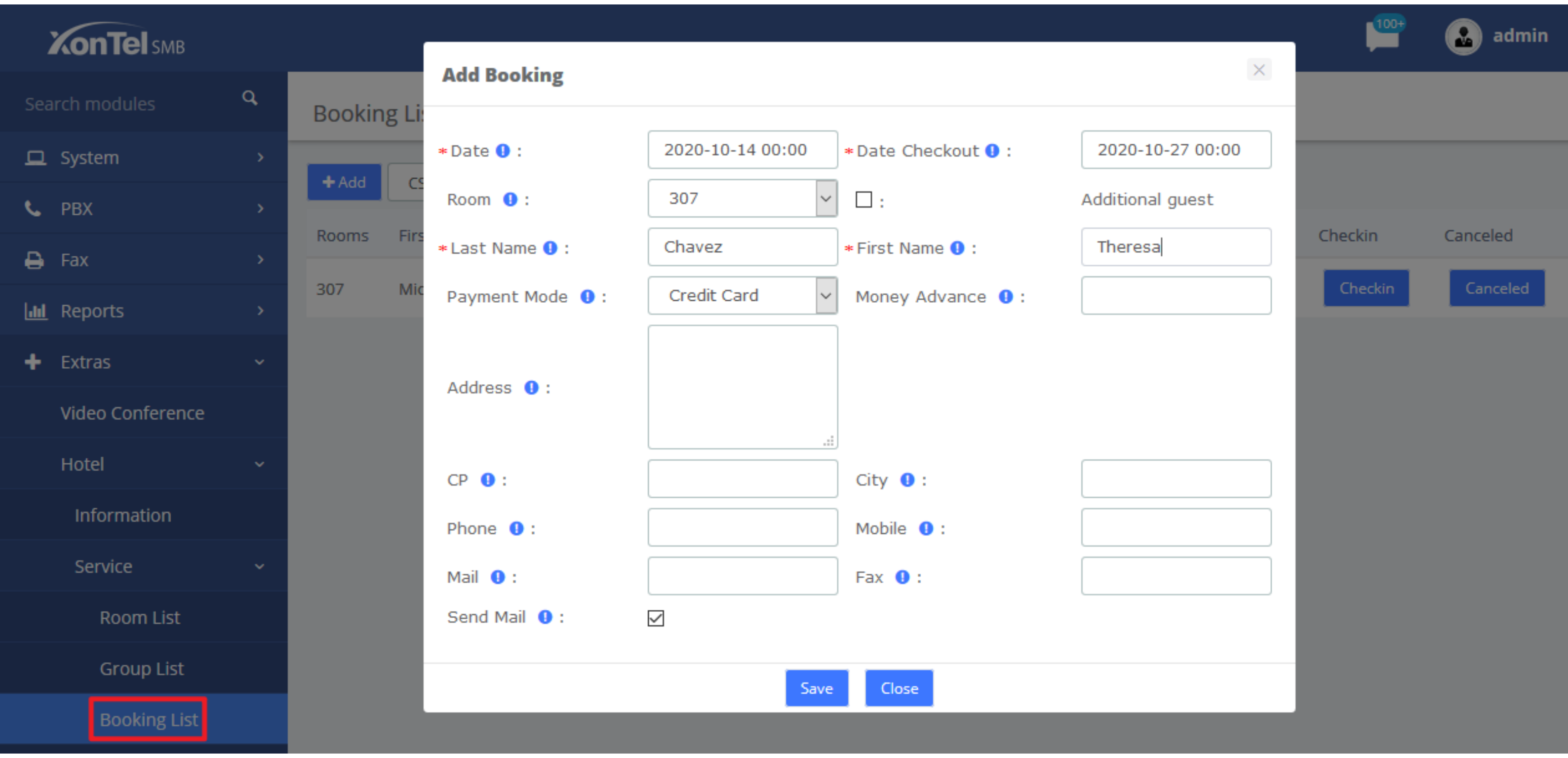
Booking List

Here, you have all booking which currently entered into Hotel. You can do a view between 2 dates.



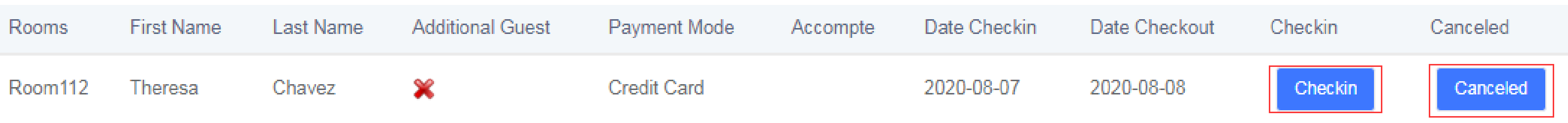
Booking List

Booking room for a new customer:

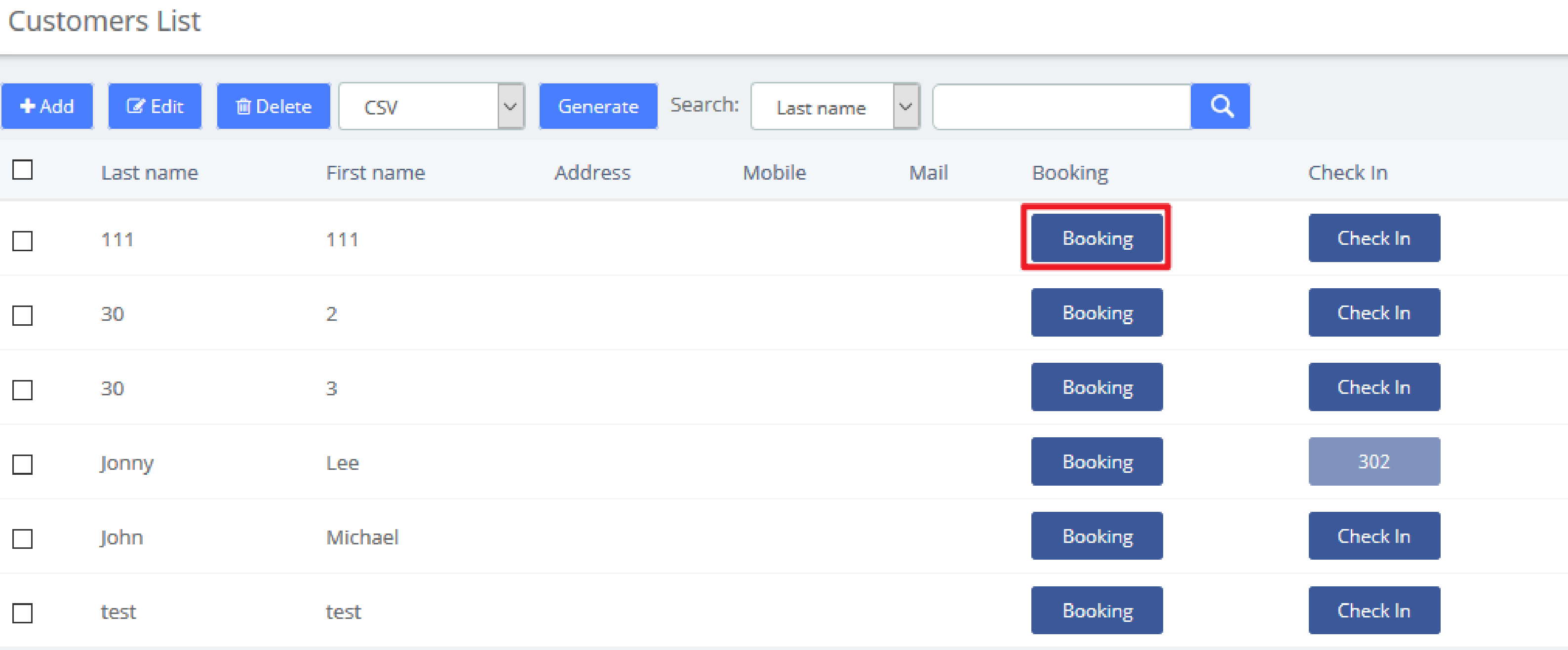


Add Booking

To make a checking on a booked room, click the **Checkin** box, and if you want to cancel a booking, check the **Canceled** box.



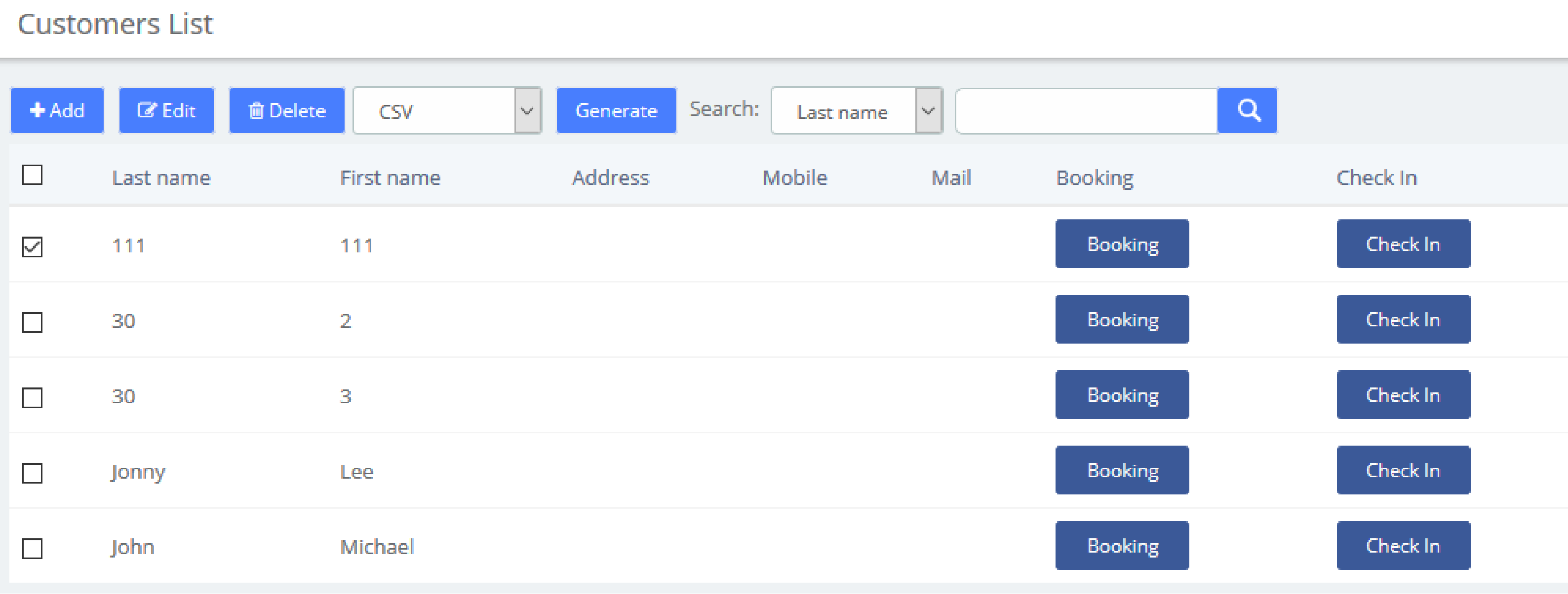
Booking room for a regular customer in the Customers List:



Check-in/Customers List

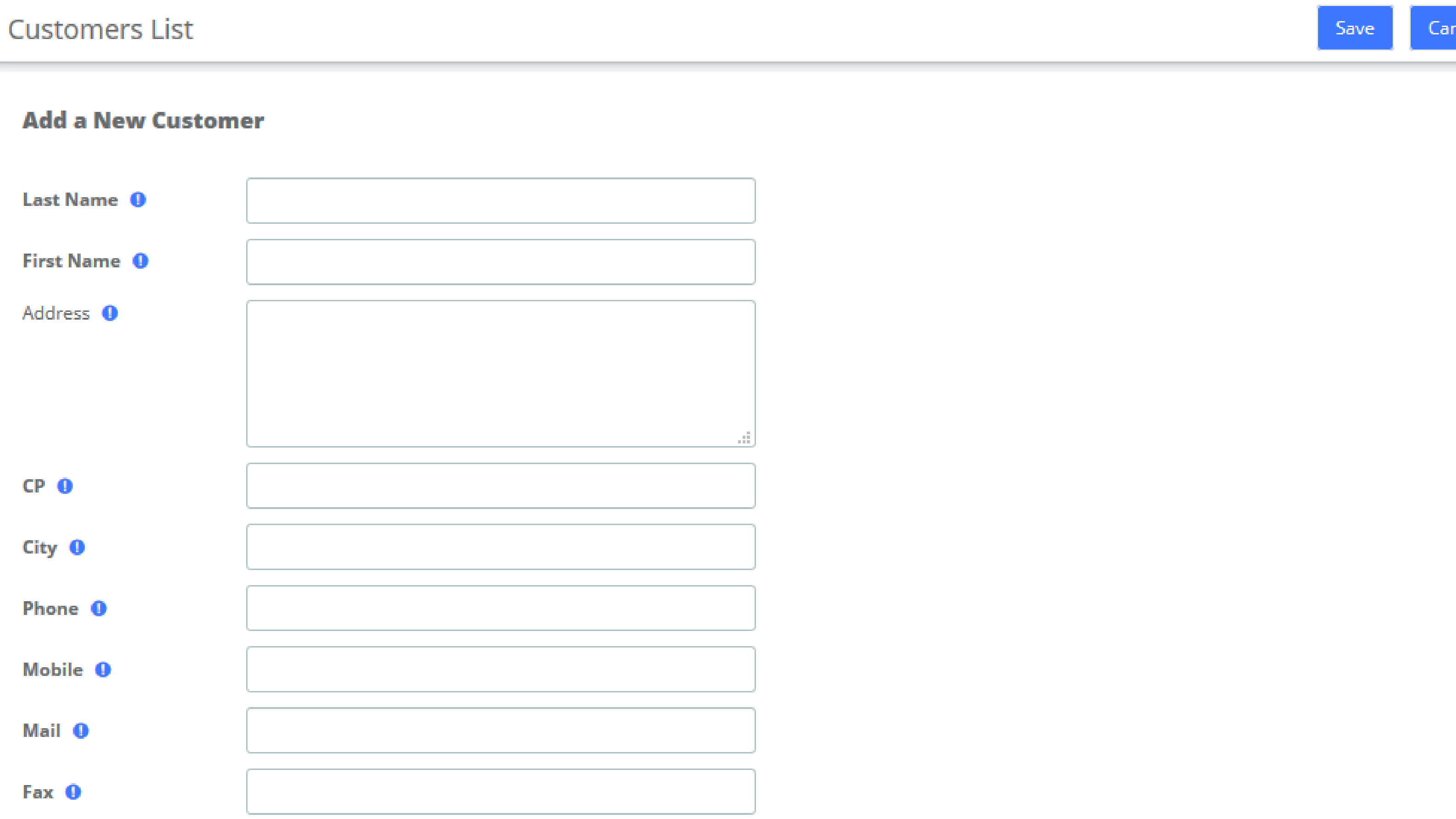
Customers List

All check-in and booking information will be entered into the customer list. This module can also customize customer information. You can book room or check-in for them.



Customers List

Also, you can add a new customer.

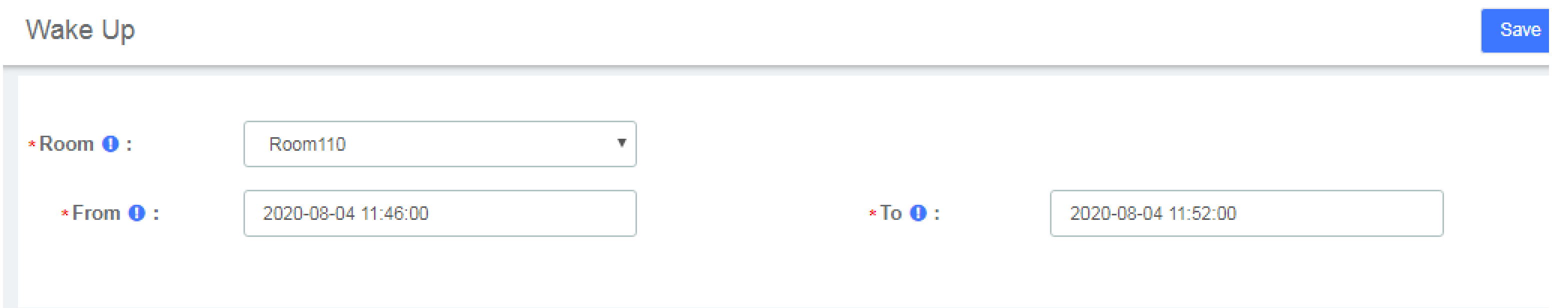


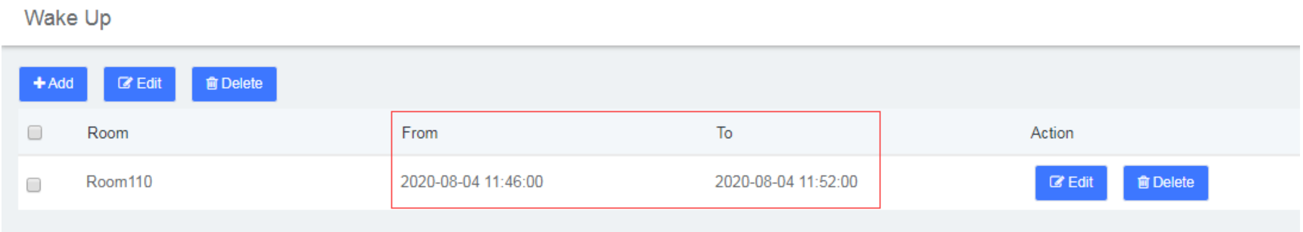
Add a New Customer

Wake Up

This module allows you to set up a wake-up call service for specific customers. The wake-up service will ring for 30 seconds every 30 seconds for the duration of the wake-up service.

In the following example, the wakeup service is set up for Room110's customers from 2020-08-04 11:46 to 2020-08-04 11:52.



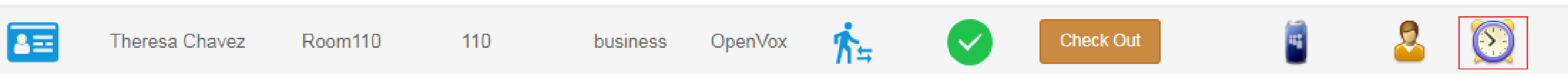


Wake Up Interface

The extension bound to Room110 will ring from 2020-08-04 11:46 to 2020-08-04 11:52

Wakeup Service

It can also be viewed or edited in the Room List module.



Room List/Wake Up

6.2.3 Configuration

Billing Settings

When the customer needs to use the extension to make an outside call, the service charge for the call is calculated based on the set call rate.

Billing Rates

If there is no matching rate list, the default rate will be used.

After initialized the rates, you can create a new billing rate or edit the existing rate.

Billing Rates

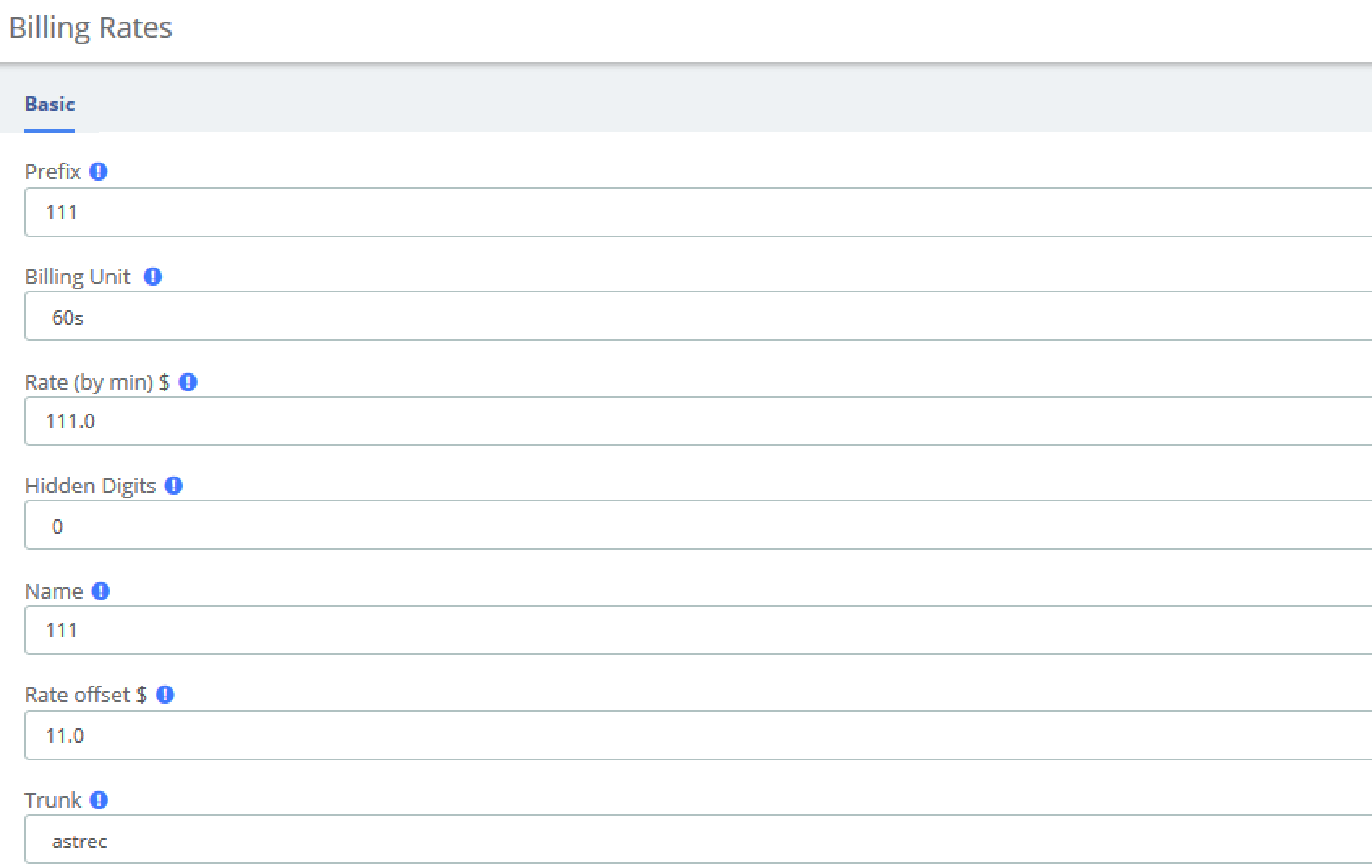
Add | Edit | Delete | Import File | Download

	Prefix	Billing Unit	Name	Rate	Rate Offset	Hidden Digits	Trunk	Creation Date	View
	*		Default	0.6	0.7	0	*	2020-08-03 05:31:28	View
	111	60	111	111.0	11.0	0	astrec	2020-10-10 10:52:37	View

Billing Rates

Note: Only trunks allowed by Billing Trunk can be displayed in the tariff list.

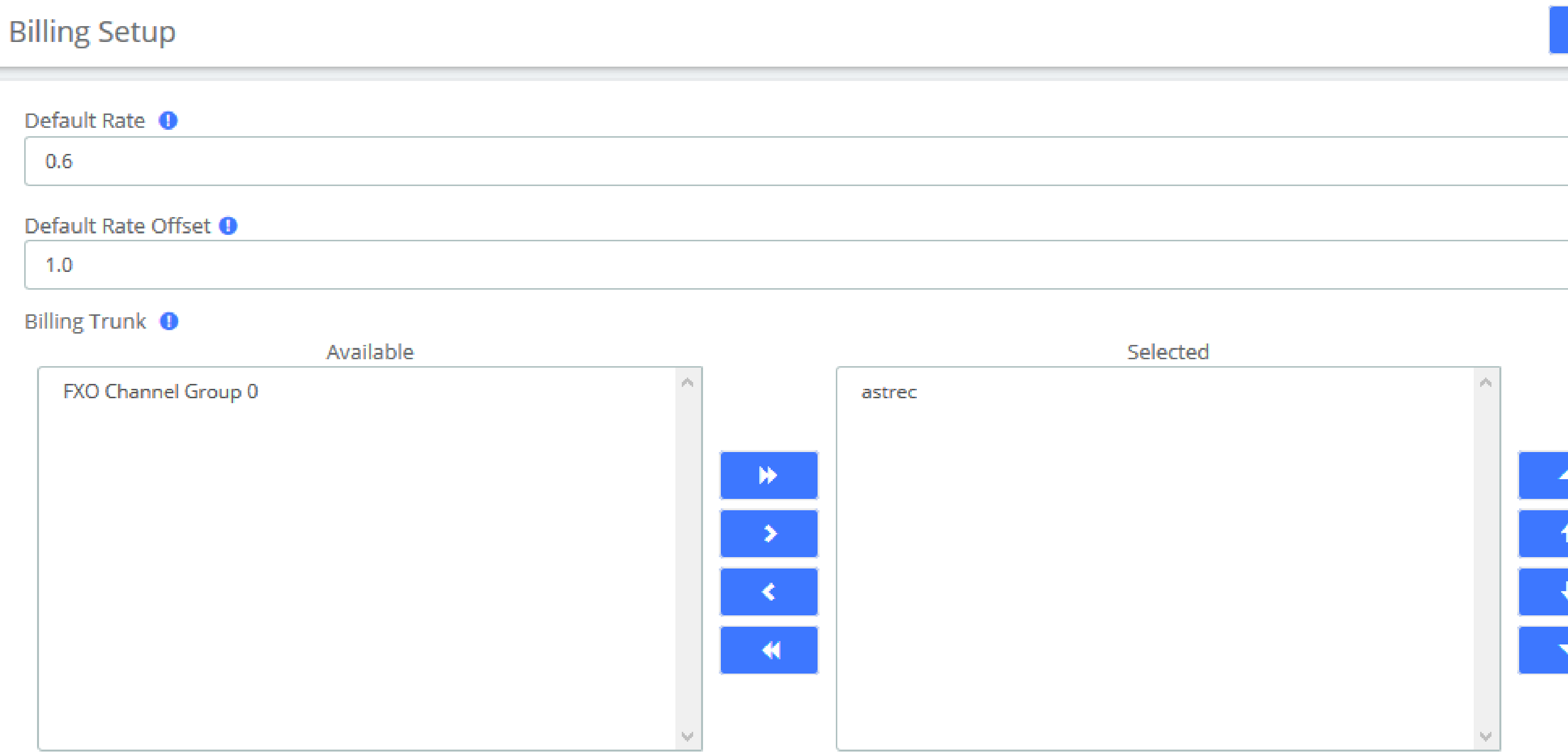
Then create a new billing Rates:



Add/Edit Billing Rates

Billing Setup

You should initialize the billing rates before setting billing rules:



Billing Setup

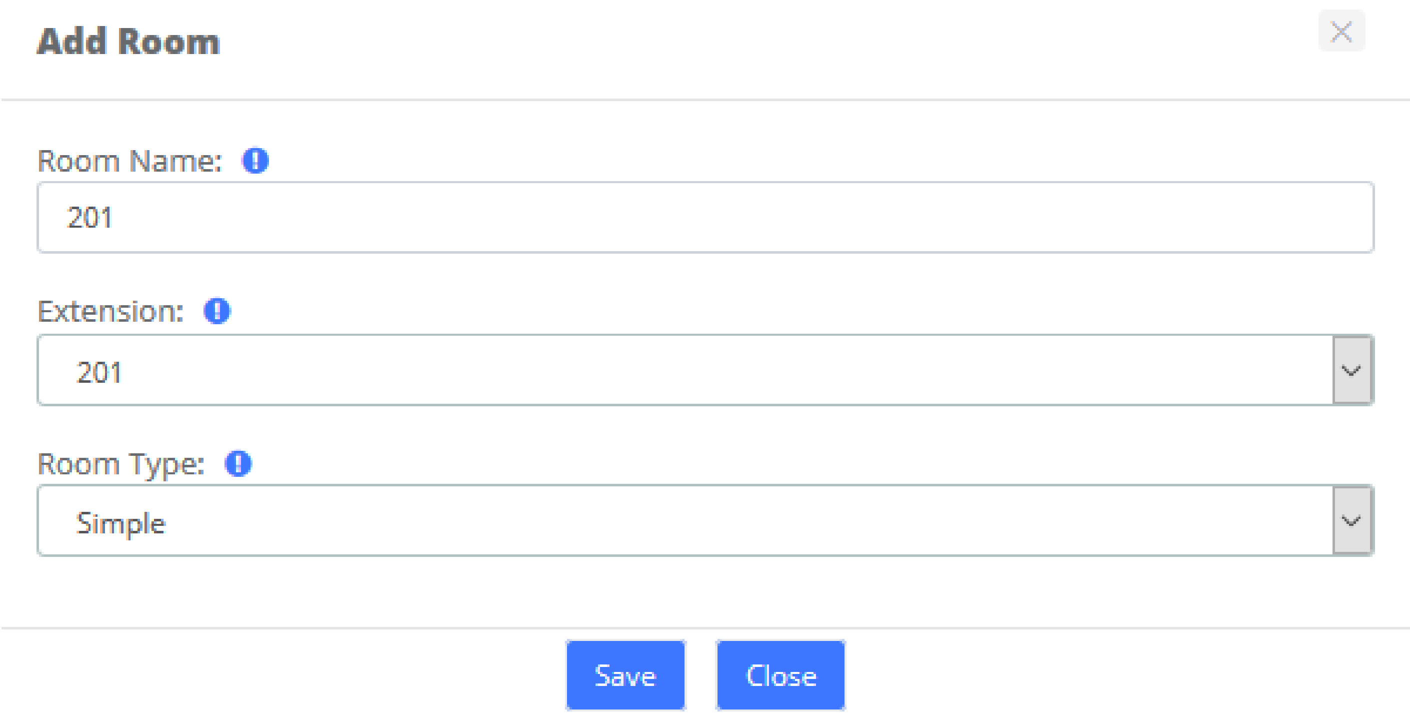
Default Rate and **Default Rate Offset** set the rate of the **Default** billing rate. **Billing Trunk** sets the billing trunks allowed when creating new billing rates.

Room Setting

Room Setup

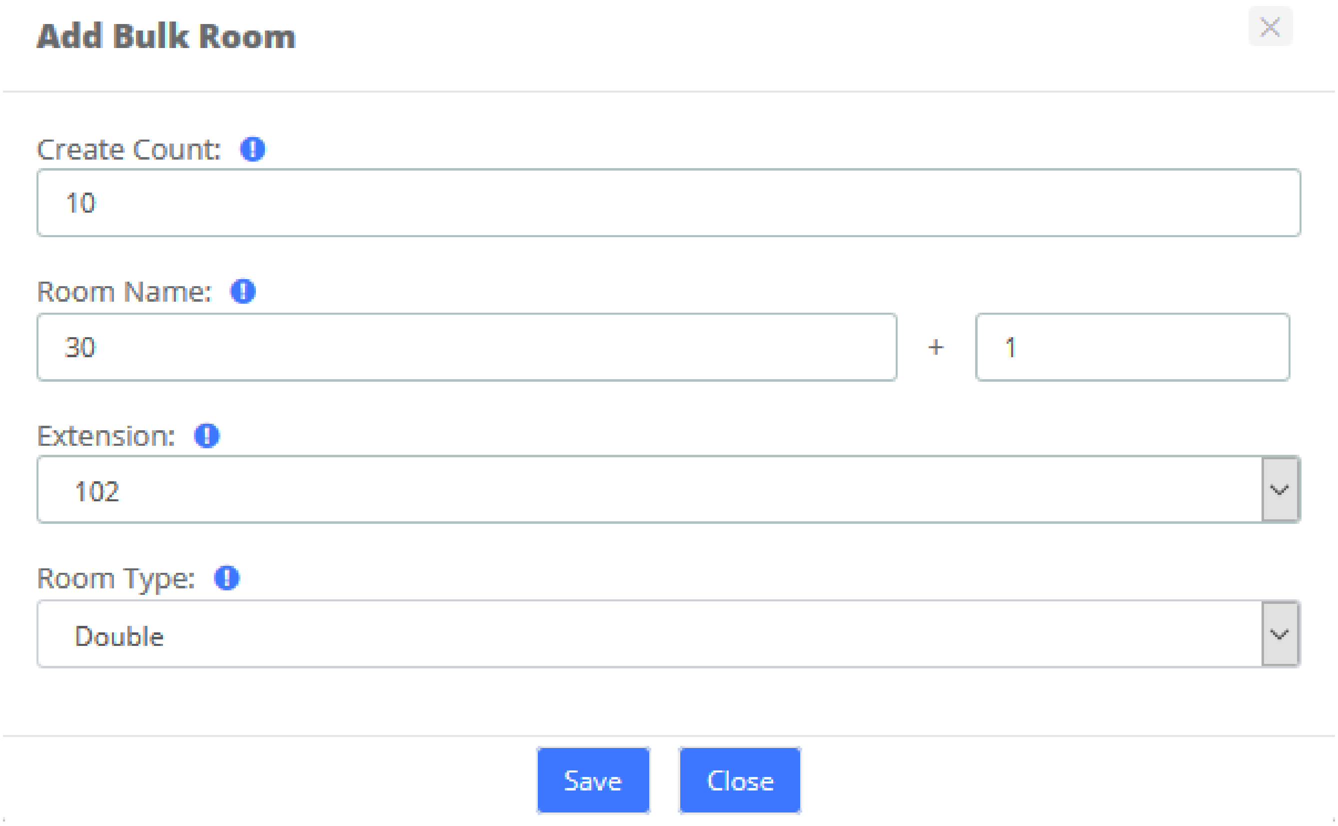
Once we have created the room type, we can generate the hotel room. Don't forget, try to prepare a good list of names for each room. (e.g.: room 100, room 101.etc). This name will be use by Hotel if no name is entered.

Click **+ Add** and create room.



Add Room

Besides, you can click **+ Add Bulk** to batch create rooms.



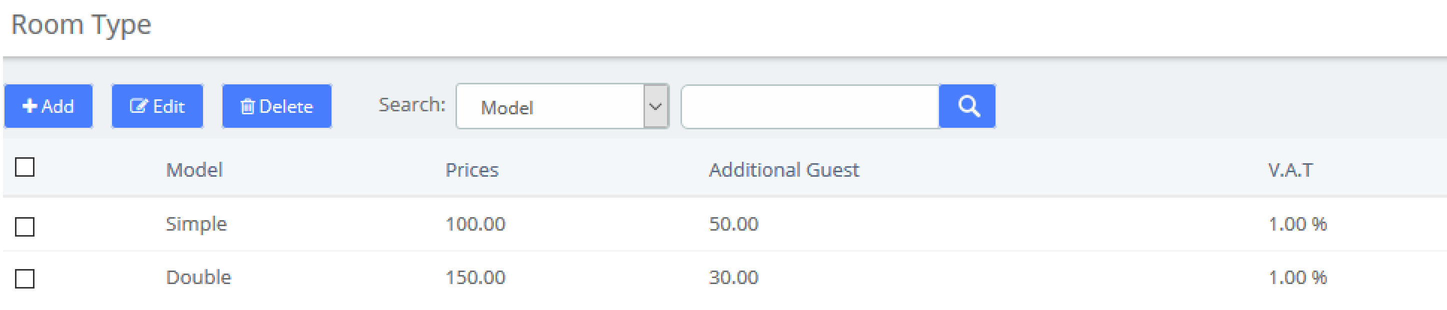
Batch initialization of rooms

Room Type

Room Type display all types already recorded into Hotel configuration.

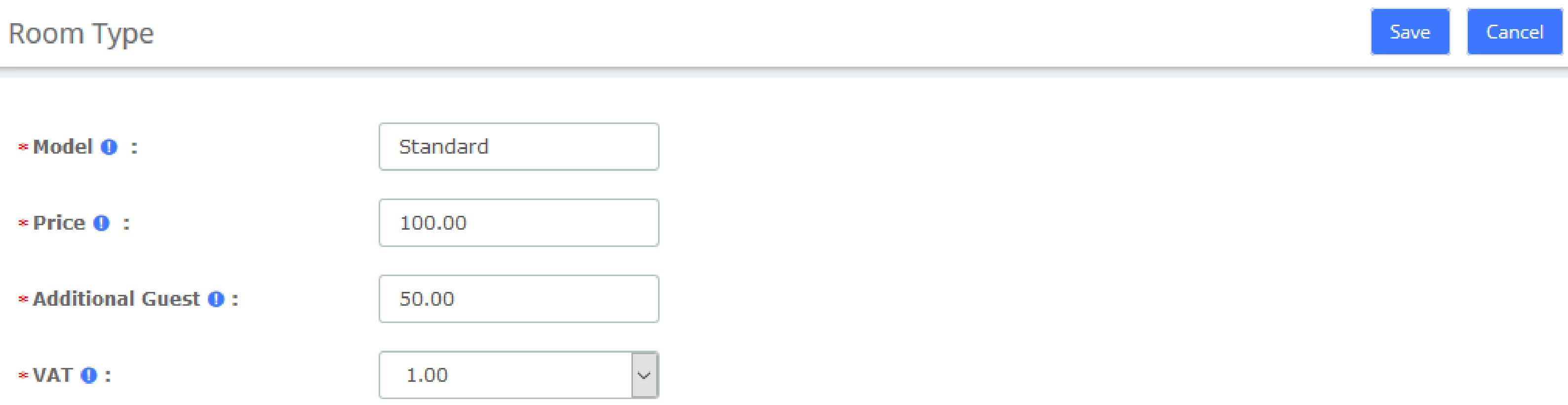
Before to add any room, you must create some room type to putting them on each room. You can create hotel room types, such as common Standard Rooms, Double Rooms, Business Room, King Rooms, etc.

You could delete a room or more just selecting the checkbox at left of row.



Room Types

Just putting a type with its price, enter a price to additional guest if you want, and select the V.A.T used by this room. (2 V.A.T. are enabled).

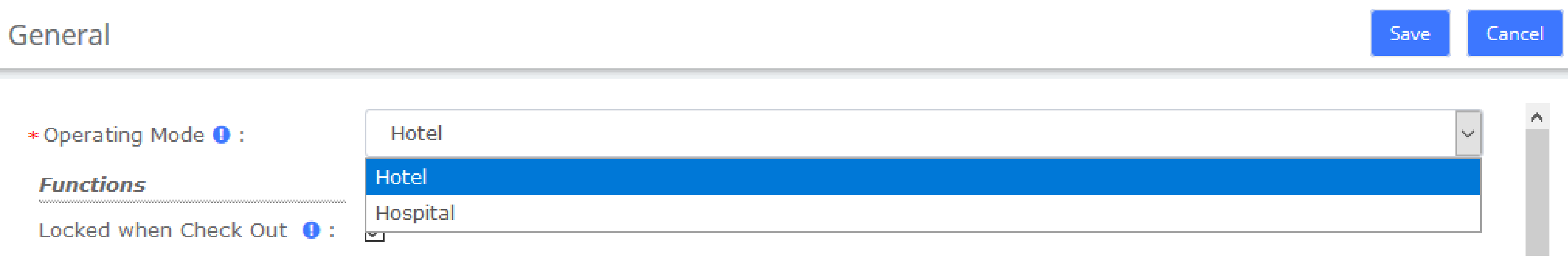


Room Types options

General

Before we begin, we should initialize the configuration of the hotel system, which includes customizing the company logo, company information, configuring emails, etc.

Here, you could select 2 operating mode (Hotel and Hospital). Now, only one operating mode is enabled.



General/Operating Mode

You could select 3 basic functions in Hotel.

- **Locked when checkout**. When the room will be billed, this room will be locked. So impossible to calling a number.
- **Calling between rooms**: When checked, the room is able to call another room, but only if this room is included into the same group as the called room.
- **Room must be clean**: The room appear into the list of available room only if the room is cleaned. Else, the room will not appear into this list. However, this room could appear if you need to make a booking about this room.

Functions

Locked when Check Out : ☑

Calling between rooms : ☑

Room must be clean : ☑

General/Functions

You can customize your company header, like the logo (png, or jpg file extension), the company address, and the professional mail of company.

Company

Logo : Browse

700*500

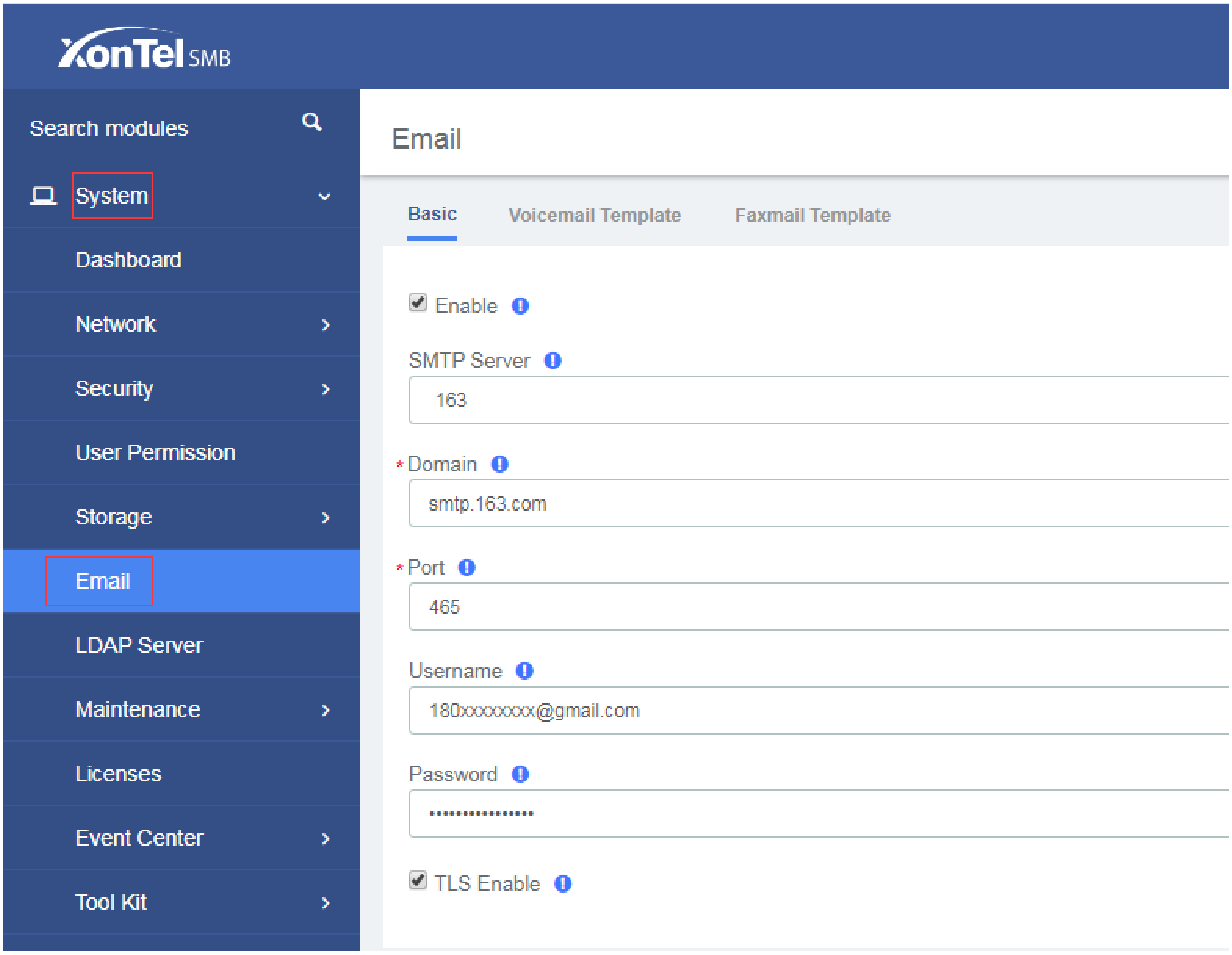
* Company :

Hotel name:
Address:
Tel :
Email:

Mail :

General/Company

Note: Mail is used to send Booking and Check-out reminders, and you need to configure the SMTP service in System->Email before using it.



SMTP settings

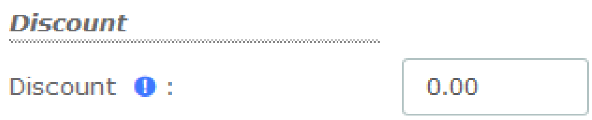
You can customize or change the prefix of each hotel function. 3 Prefix exist right now.

- **Mini-bar**, is able to add some drinks on the room, and will used during the billing. When the chambermaid will clean the room, she could check the mini-bar and enter all drink used by the guest.
- **Room Clean Prefix** will used when the room will cleaned by the chambermaid.
- **Reception** is here to giving a phone number to the reception..

General/Hotel Dial Plan

Two tax values can be entered. The first value is used by the outbound calls during the billing.

General/Tax

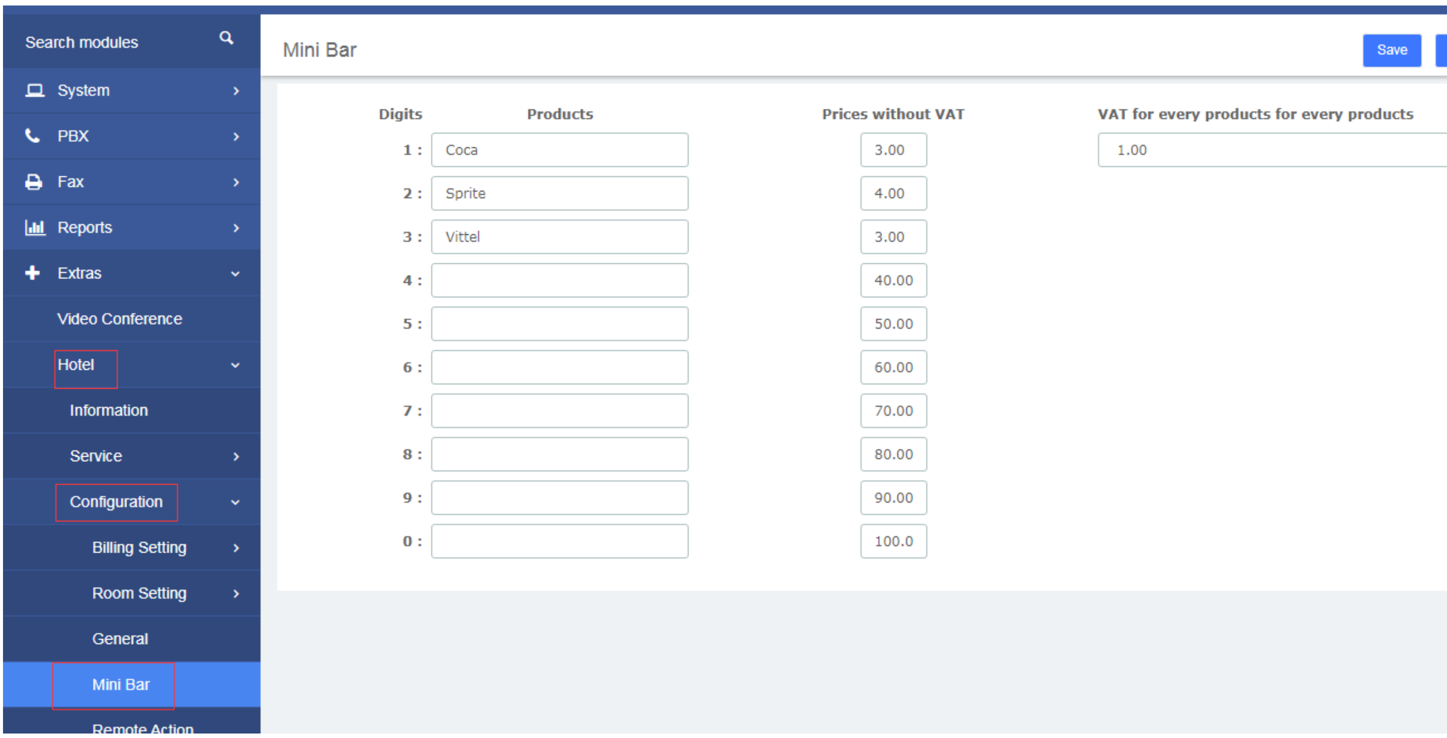


General/Discount

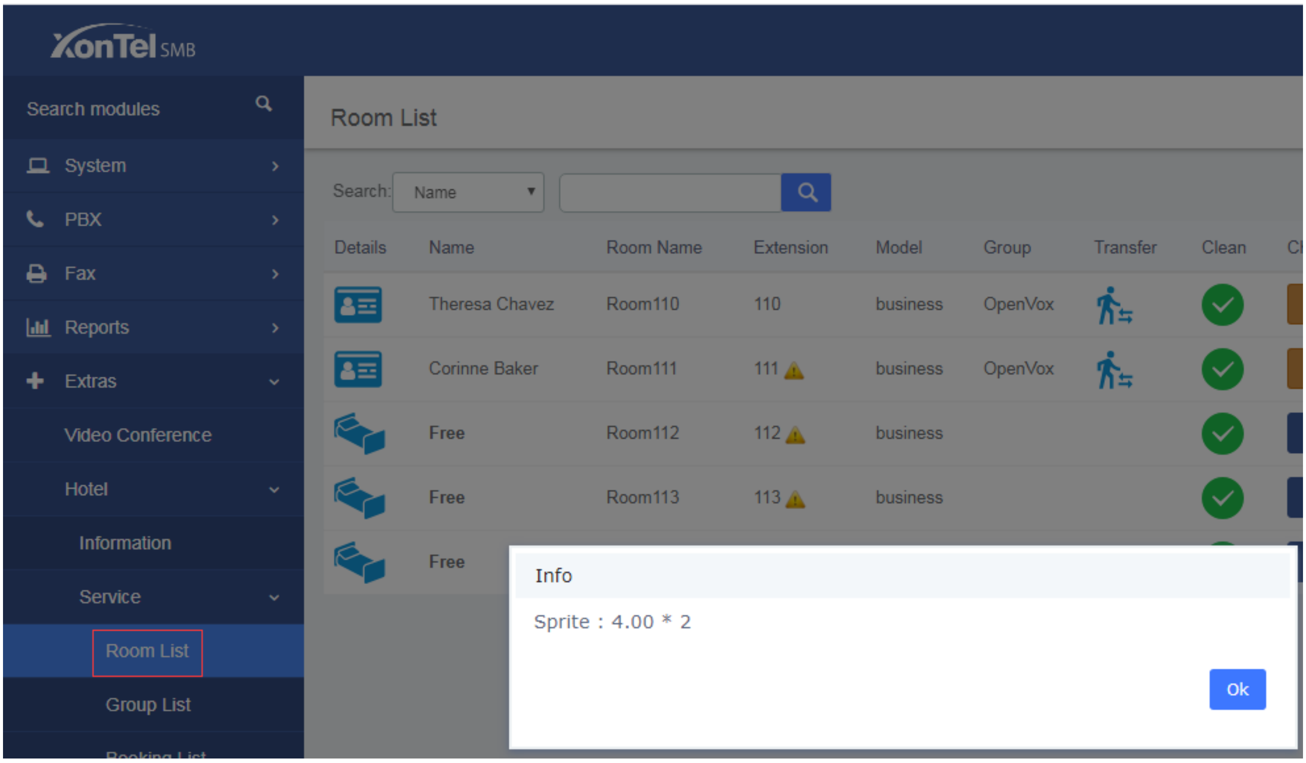
Mini Bar

Set up the items and VAT in the **Mini-Bar**. The waiter can dial to record the items purchased by the customer, dial the prefix (*37 by default) and press the number of the product used, ending with the * key. If you do not press the * key, the purchase will not be recorded. For example: if the customer has purchased three copies of Sprite, use the room' extension to dial *37222*

This menu affecting a product on each key with its price. You can enter 10 different products on this module. 2 V.A.T can be selected.

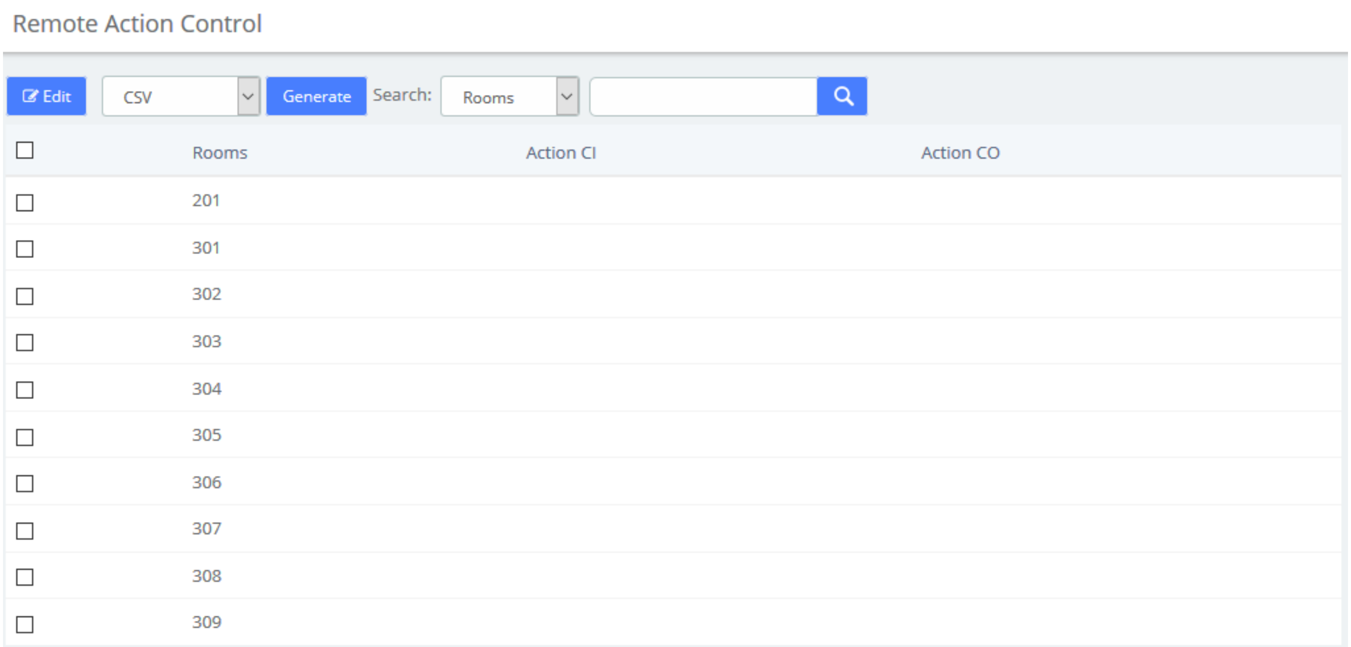


If a customer has purchased an item, we can check it in the room list.

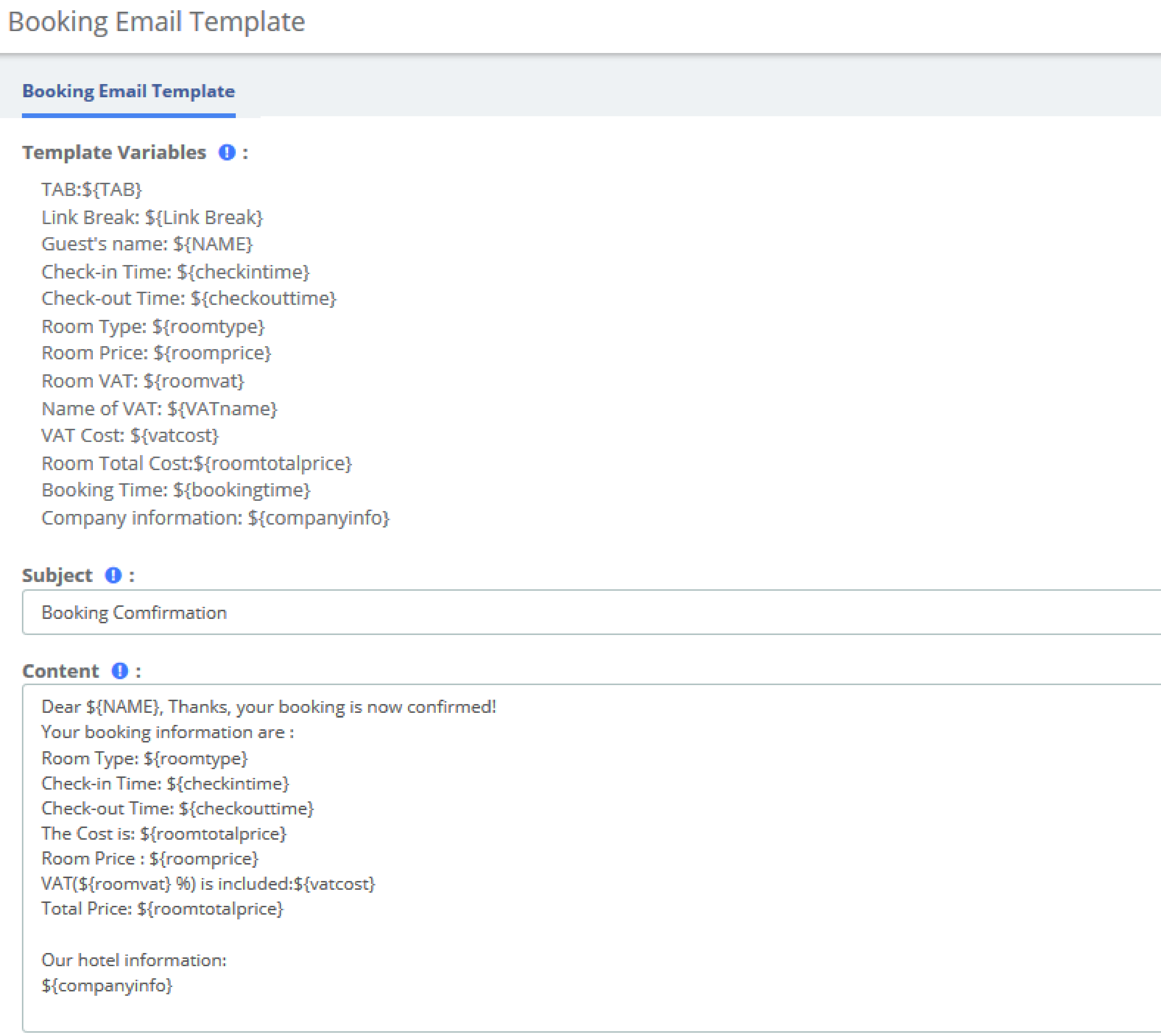


Room List/Mini Bar

Remote Action Control



Booking Email Template



Booking Email Template

6.2.4 Report

Billing Report

You can check customer consumption in the Billing Report after checking out.

Apply | Delete | CSV | Generate | Search: Checkin Date

	Checkin Date	Checkout Date	Room	Guest	Paid	View	Delete
☐	2020-09-30 12:16:00	2020-10-22 13:13:00	302	Lee Jonny	✔		Delete
☐	2020-09-30 12:08:00	2020-10-03 12:30:00	302	2 30	☐		Delete
☐	2020-09-30 12:11:00	2020-10-03 12:30:00	303	3 30	☐		Delete

Billing Report list

Click you can view the billing report, and click can export this report.

Wed 5 Aug 2020 Number : 3720200805

Hotel name:Hotel name
Address: Russia
Tel : 180xxxxxxxx
Email: xxxxx@gmail.com

Theresa Chavez

5139116296680983

2020-08-04 11:12:00 to 2020-08-05 04:39:46.

Sale

Service	Q.T.	PU HT	VAT	Price
Nights with room's model: business	1	150.00 $	0.75 $	150.75 $
Mini Bar :				
Coca	1	3.00 $	0.03 $	3.03 $
Sprite	1	4.00 $	0.04 $	4.04 $

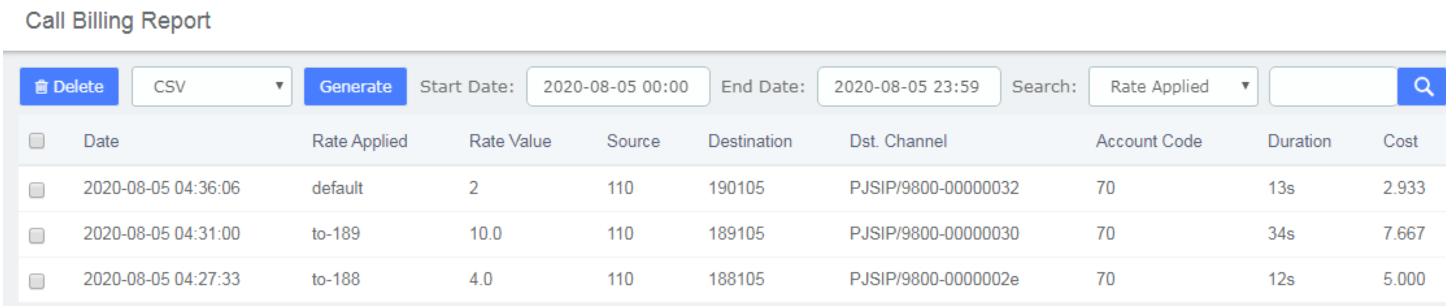
Details calls :

Date - Time - Call to	Call	Duration	Price
2020-08-05 04:27:33 - to-188	188105	0 m 12 s	5.00 $
2020-08-05 04:31:00 - to-189	189105	0 m 34 s	7.66 $
2020-08-05 04:36:06 - default	190105	0 m 13 s	2.93 $

Billing Report

Call Billing Report

After the external call is over, the call record will be displayed in **Hotel->Report->Call Billing Report**.

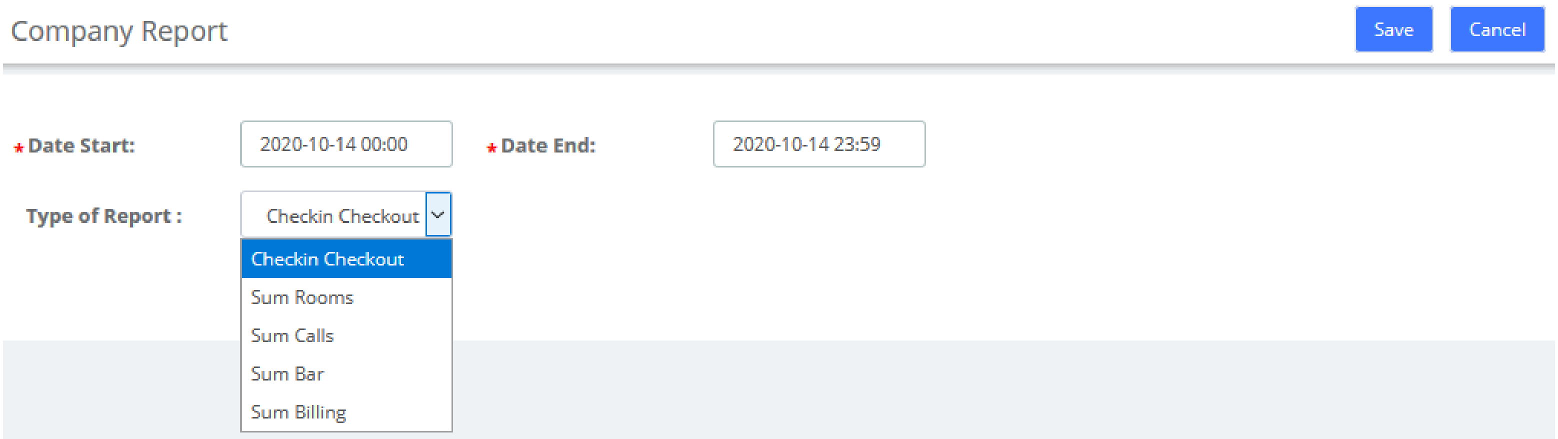

Call Billing Report

	Date	Rate Applied	Rate Value	Source	Destination	Dst. Channel	Account Code	Duration	Cost
	2020-08-05 04:36:06	default	2	110	190105	PJSIP/9800-00000032	70	13s	2.933
	2020-08-05 04:31:00	to-189	10.0	110	189105	PJSIP/9800-00000030	70	34s	7.667
	2020-08-05 04:27:33	to-188	4.0	110	188105	PJSIP/9800-0000002e	70	12s	5.000

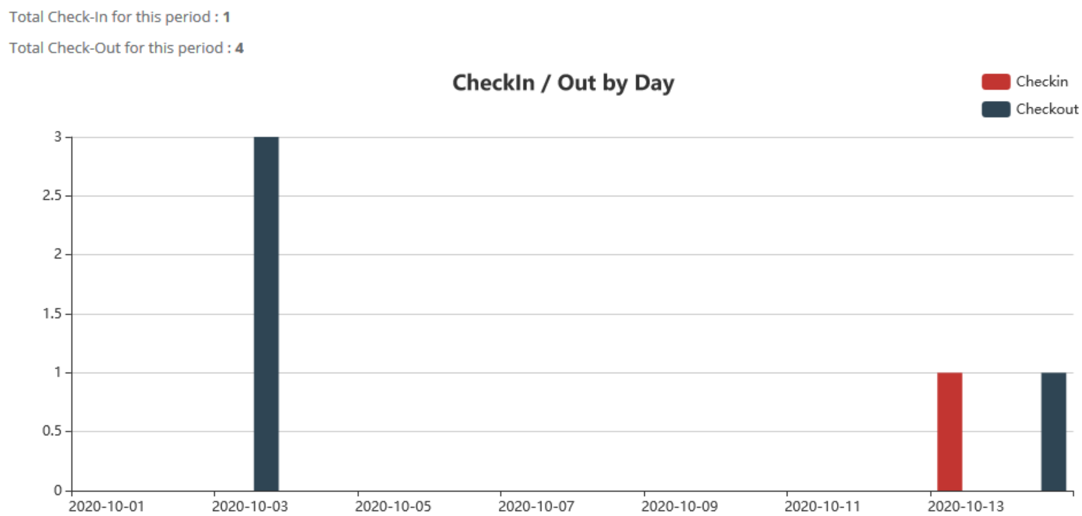
Call Billing Report list

Company Report

You can realize some company report, like how many checks in and checkout by day between two dates. Type of report include Check-in and Check-out info, Sum Rooms, mini-bar, calls, and billings.



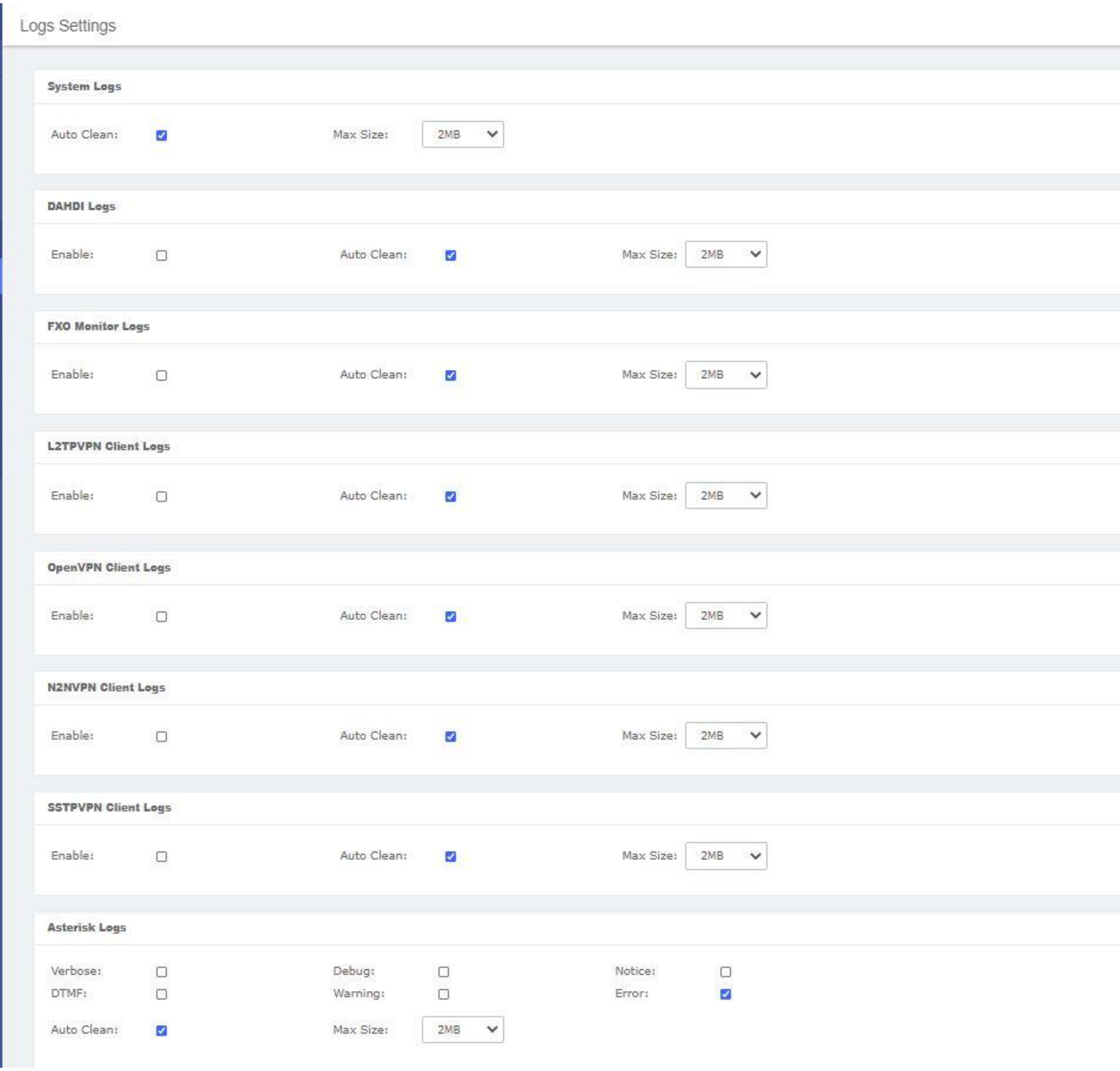
You can realize some company report, like how many check-in and checkout by day between two dates.



7 Logs

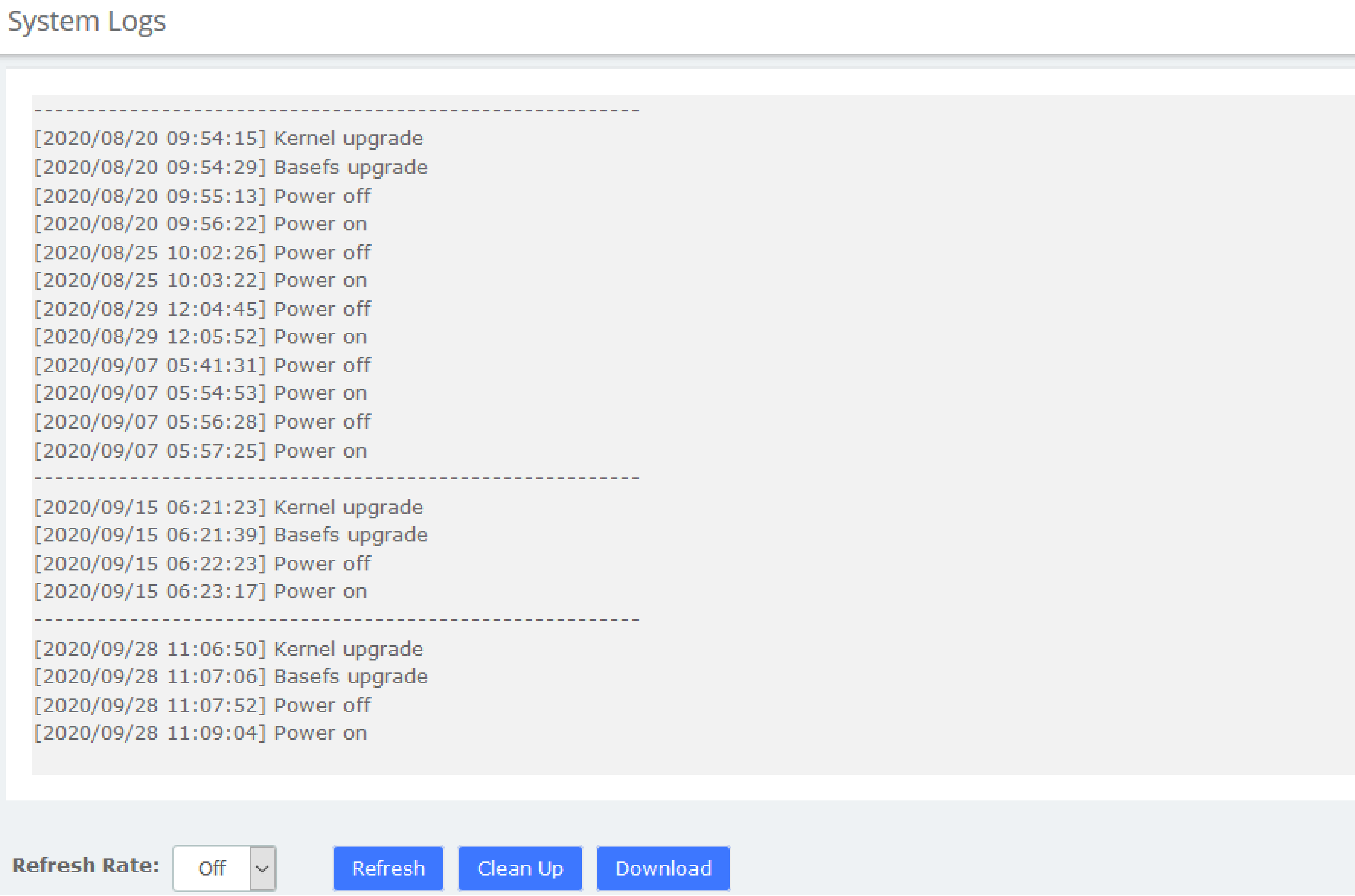
7.1 Logs Settings

Here you can set the maximum size of XonTel SMB PBX logs (System logs, DAHDI logs, FXO Monitor logs, L2TP VPN client logs, Open VPN client logs, N2N VPN client logs, SSTP VPN client logs and Asterisk logs).



7.2 System Logs

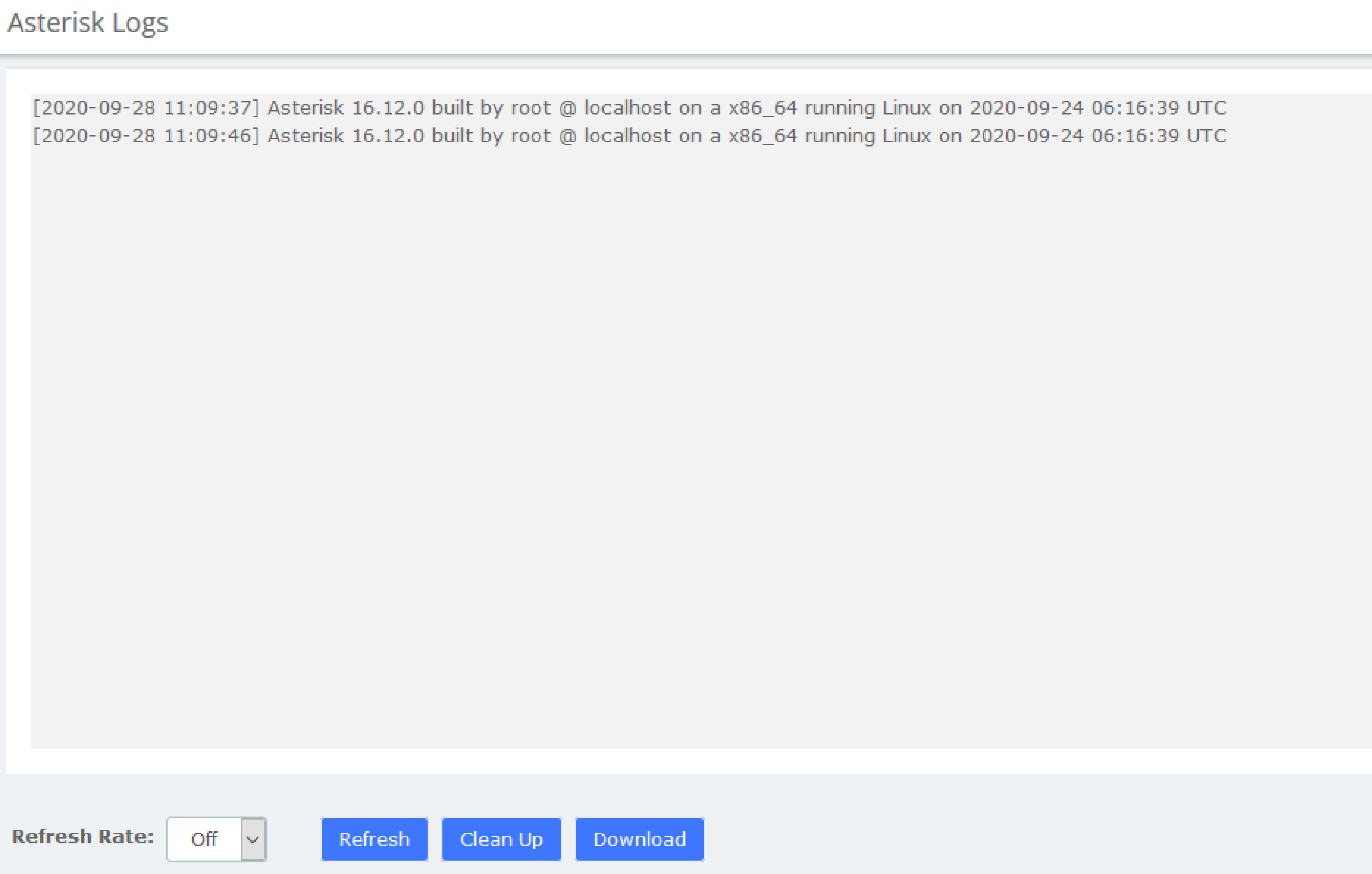
Here you can check the XonTel SMB PBX system logs



System Logs interface

7.3 Asterisk Logs

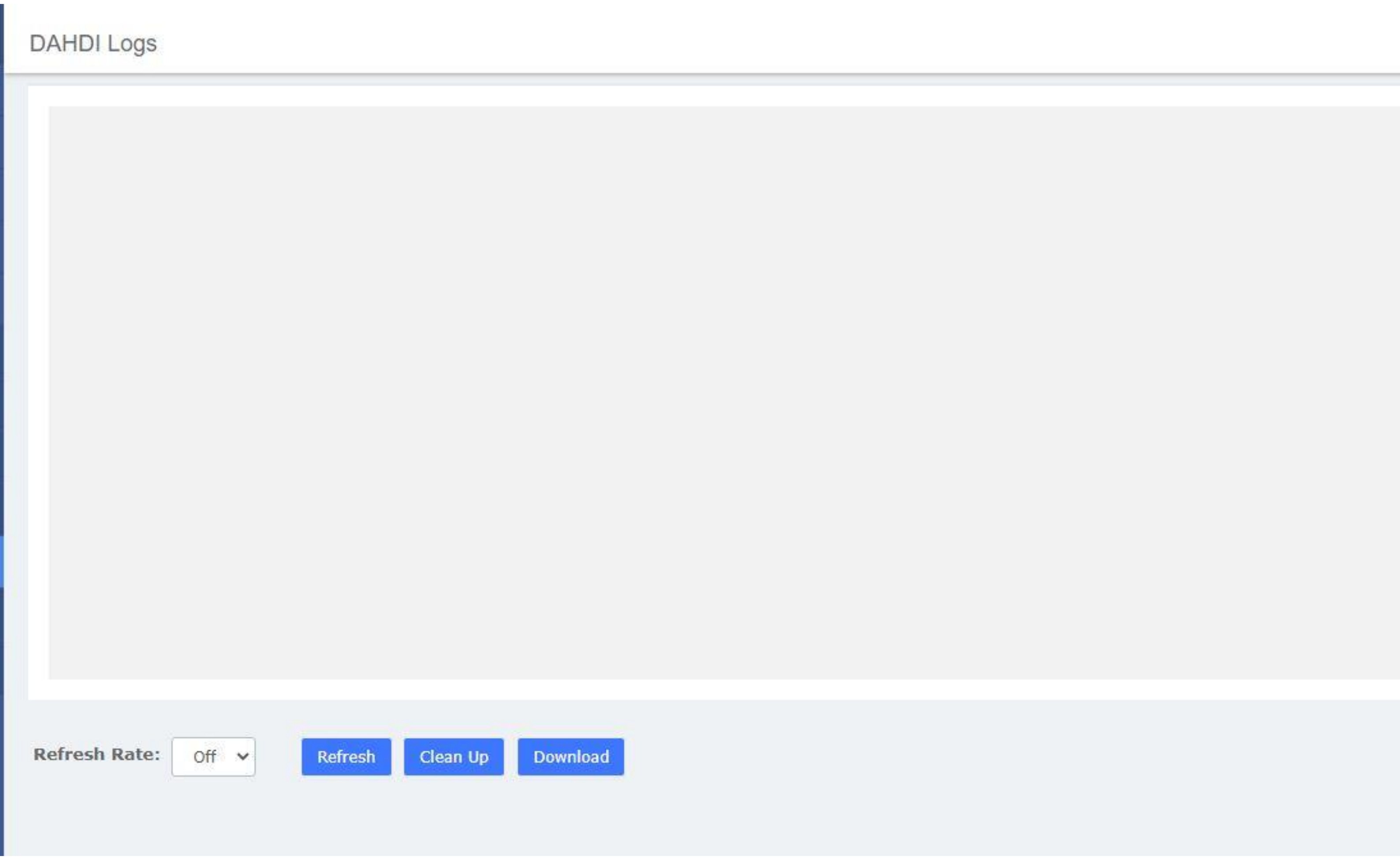
Here you can check the XonTel SMB PBX Asterisk logs



Asterisk logs interface

7.4 DAHDI Logs

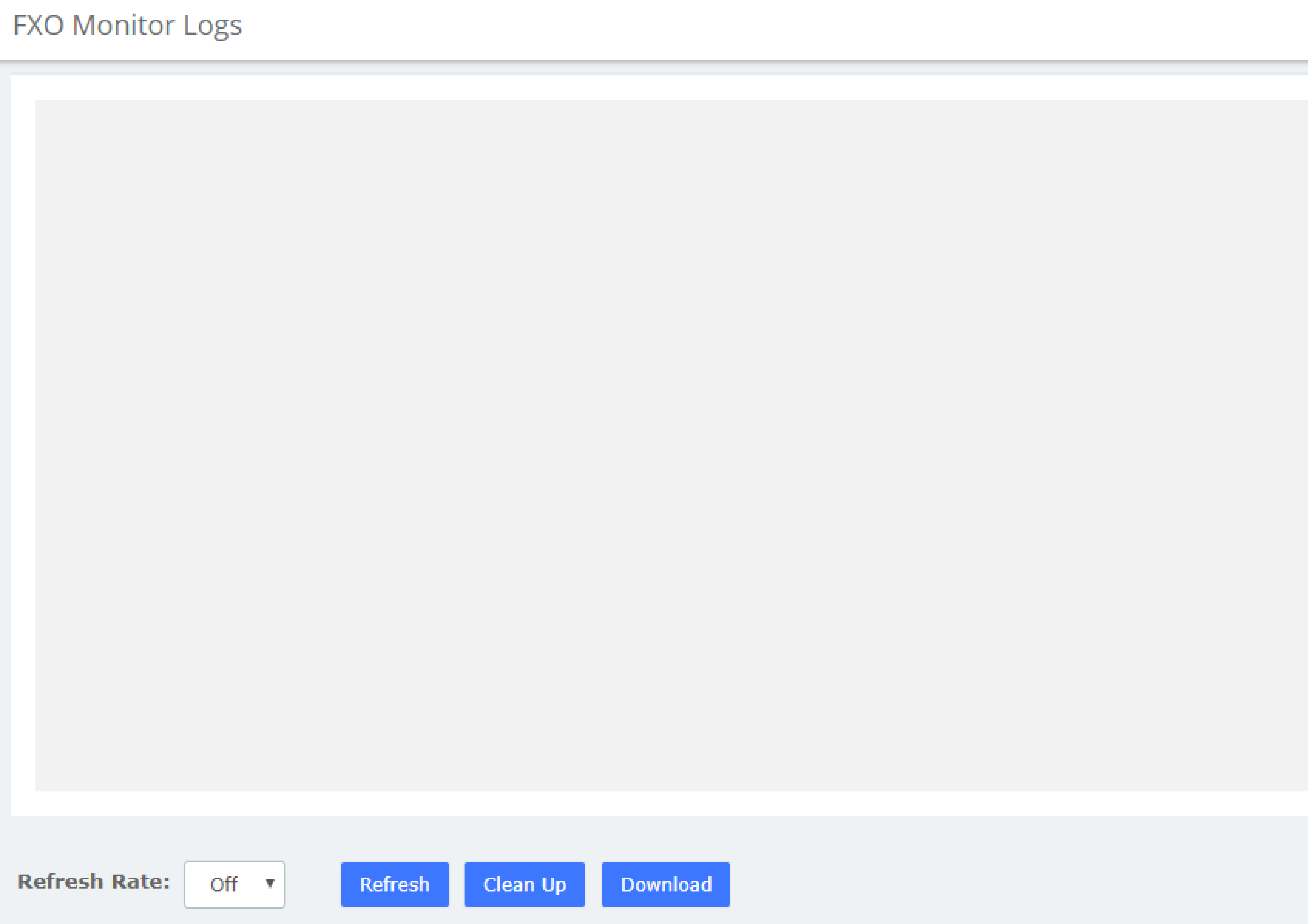
Here you can check the XonTel SMB PBX DAHDI logs.



DAHDI logs interface

7.5 FXO Monitor Logs

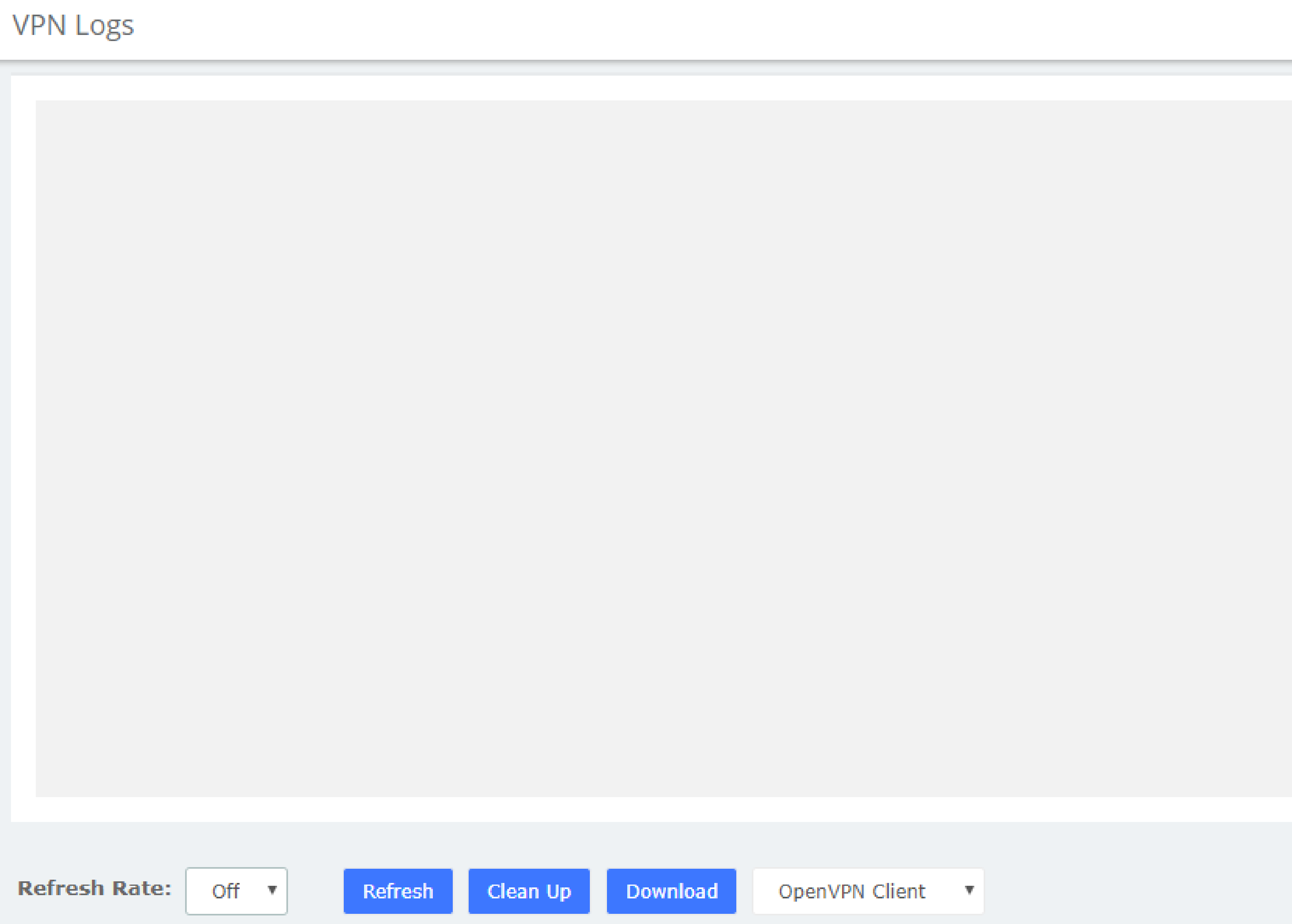
Here you can check the XonTel SMB PBX FXO Monitor logs



FXO Monitor logs interface

7.6 VPN Logs

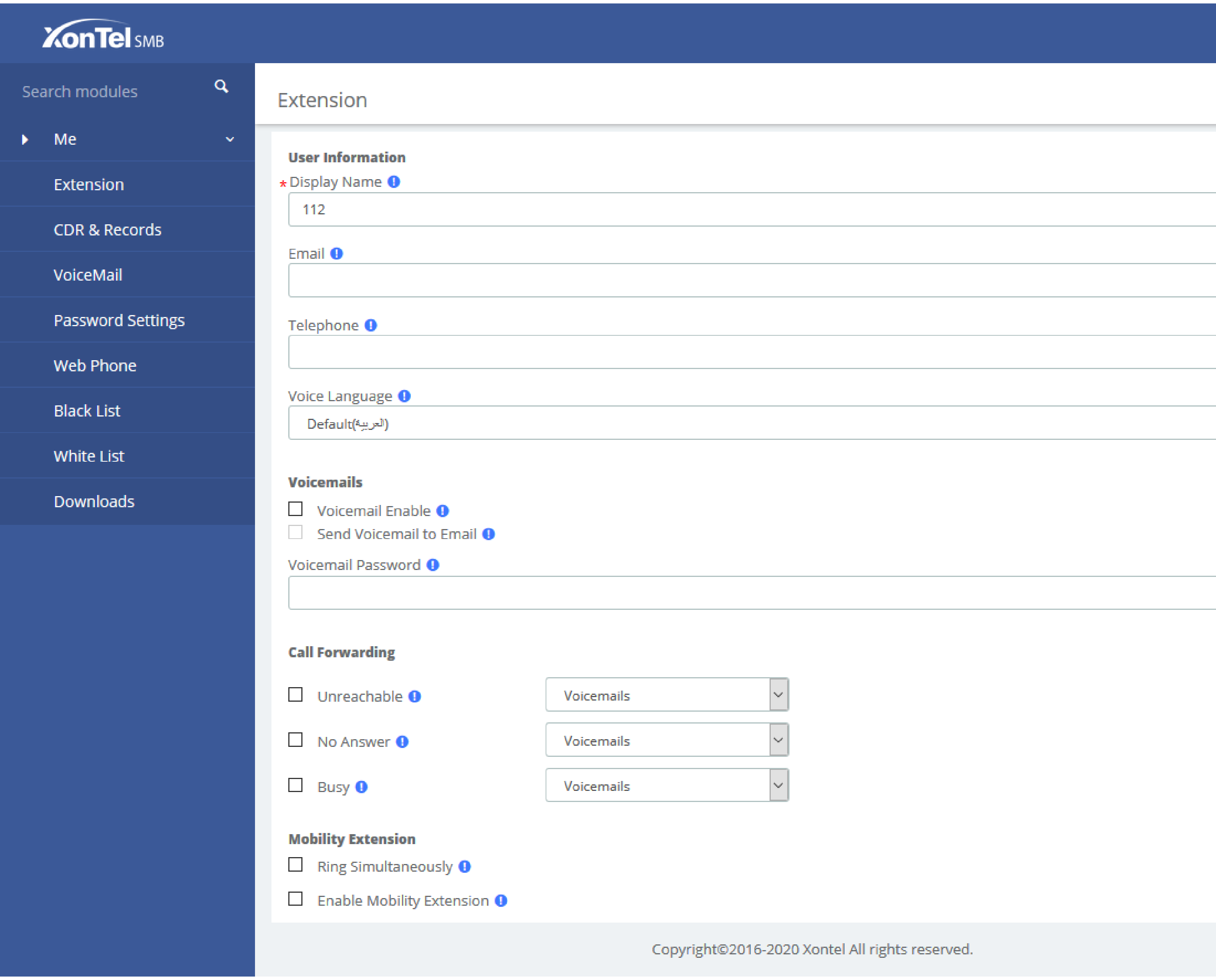
Here you can check the XonTel SMB PBX VPN logs



VPN logs interface

8 Me Bar

As mentioned in **2.4 User Permission**, all SIP extensions will be given **Me** module permissions by default. Enter **Me Bar** interface requires login with extension account. On the login page of PBX, enter the **Extension/User Password** and click login. Note that the extension login uses the **User Password**, not the **Registration Password**. After the extension is logged in, as shown below

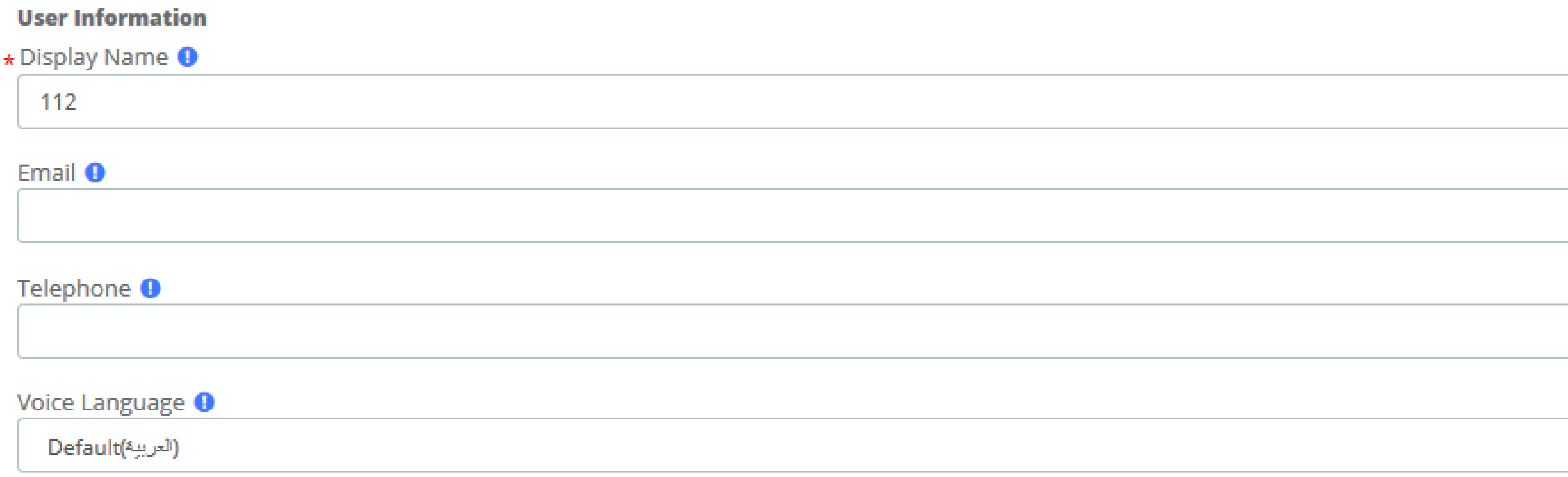


If the extension has set permissions in **System>User Permission**, the corresponding module will appear in the menu after the extension logs in. You can set the extension permissions flexibly and reduce the burden of the XonTel SMB PBX administrator. For details, see **2.4 User Permission**.

8.1 Extension

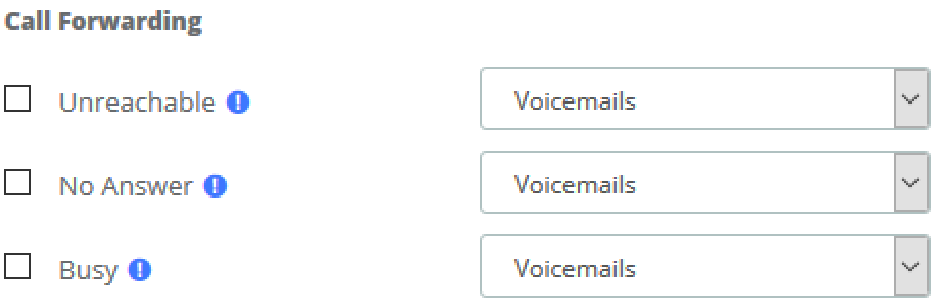
Most of the parameters on this page can also be configured under the Administrator's management page (**PBX>Extension>Edit**).

You can set the basic information of the extension.

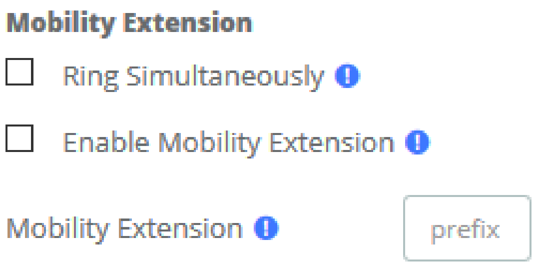


You can also set the Voicemail feature of the current extension:

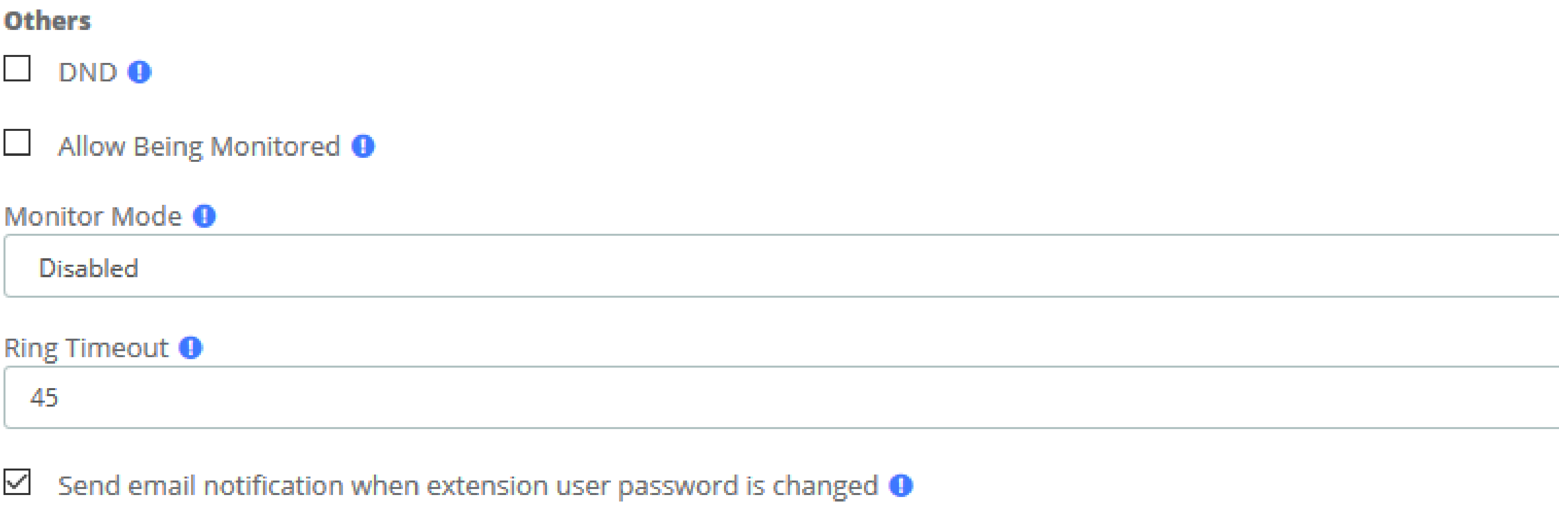
You can also set the Call Forwarding feature for the current phone, click the drop-down list to select Voicemails/Extension:



You can also set whether to enable the Mobility Extension feature (i.e. mobile phone number) for the current extension.

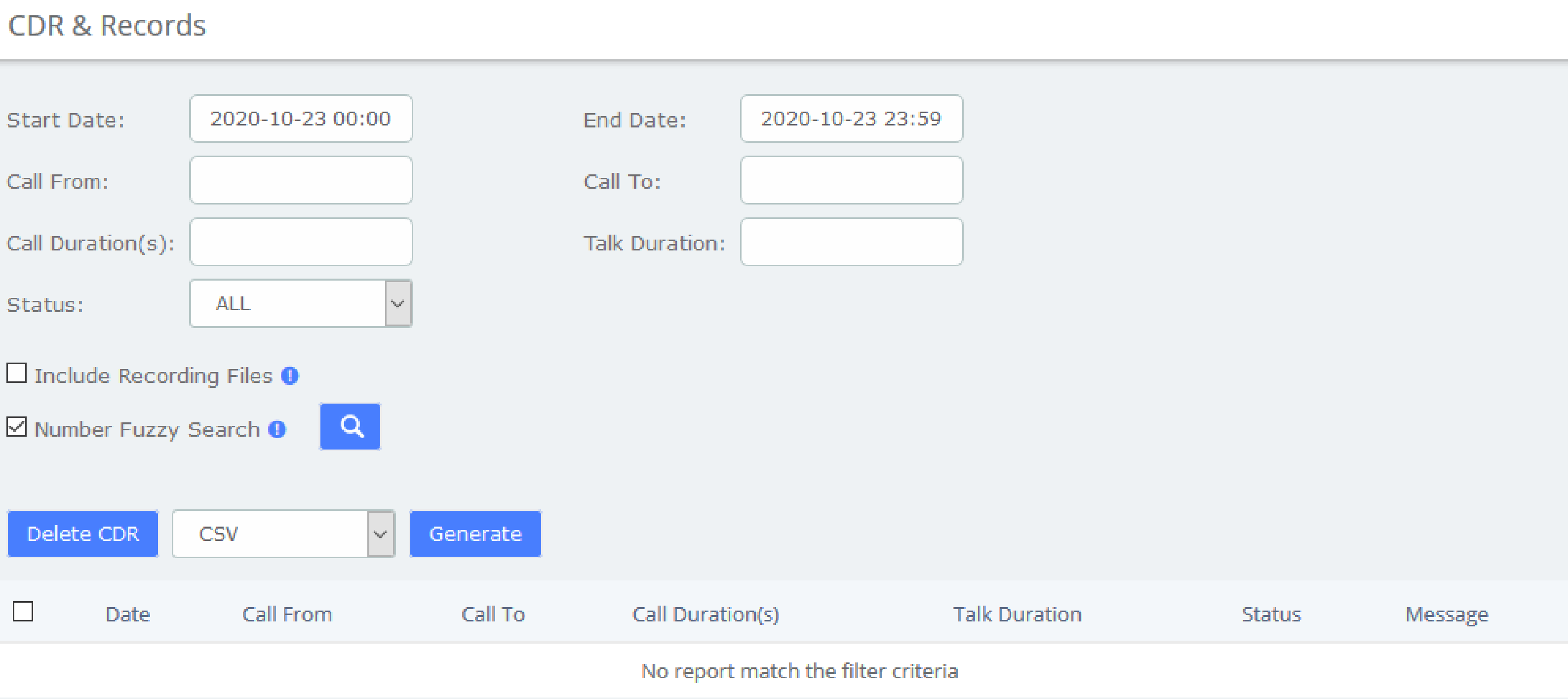


You can also set other functions of this extension.

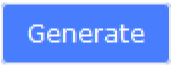


8.2 CDR & Records

You can view the call records and recordings related to the current extension



CDR & Records

When the extension is given permission to download CDRs, 

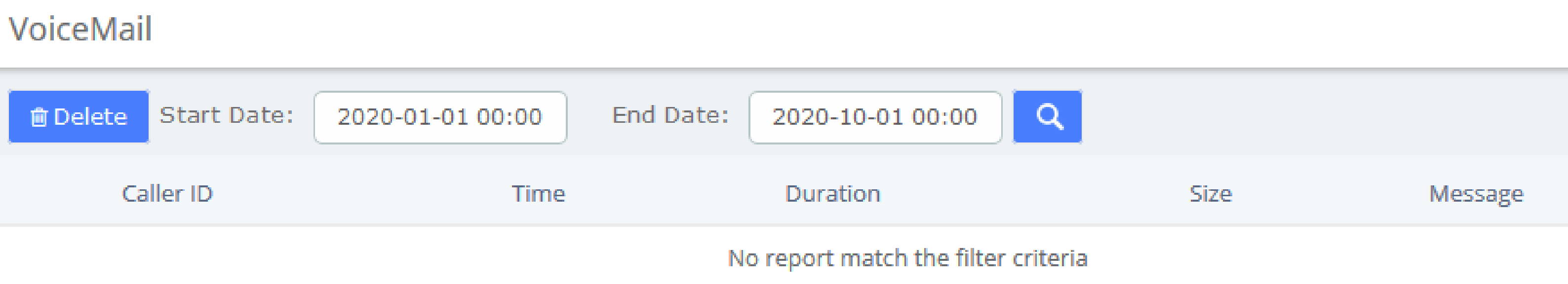
button will appear, which can generate call detail records and you can download them on the Downloads page.

When the extension is given permission to delete the CDR, the 

button will appear. You can click the call log in the check box and click the button to delete it.

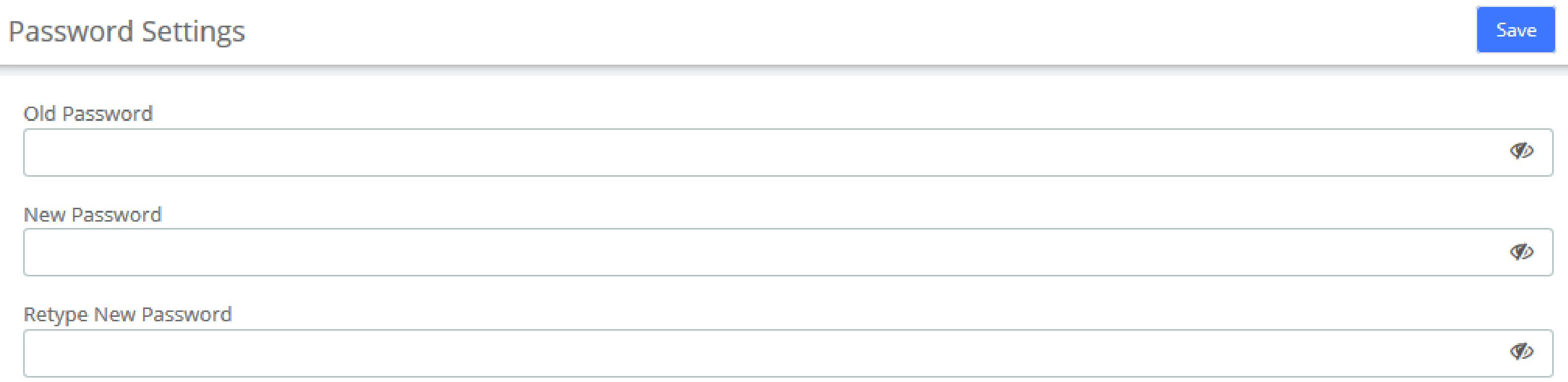
8.3 Voicemail

You can also check the voice messages of the changed extension



8.4 Password Settings

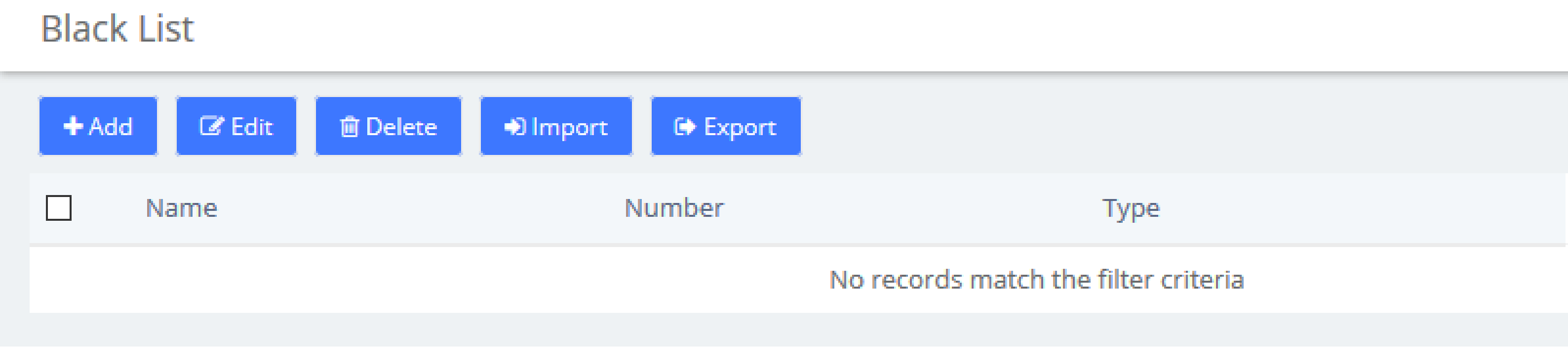
You can reset the login password of the extension



8.5 Black List

The Black List module is the same as the blacklist function under the admin account. You can add a phone number to a blacklist or remove a phone number from a blacklist. You can also choose to blacklist any blocked or unknown calls.

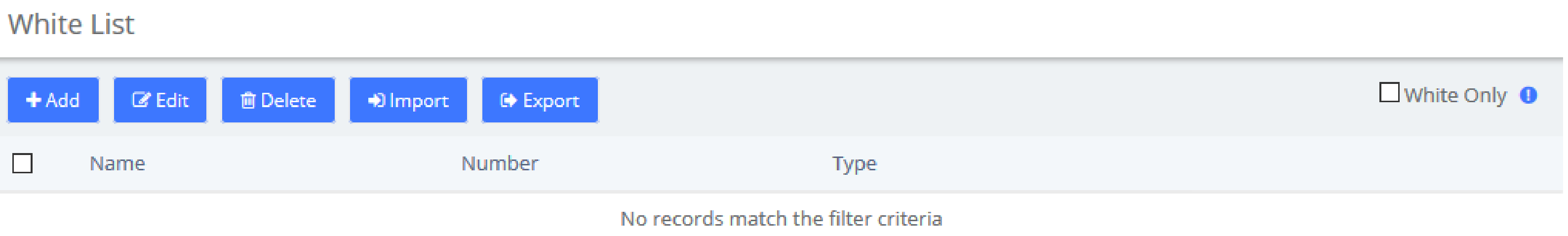
When a number is blacklisted, any calls with that number in the Caller ID field received by the system will be routed to the disconnected record.



Black List

8.6 White List

The White List module is the same as the whitelist function under the admin account.



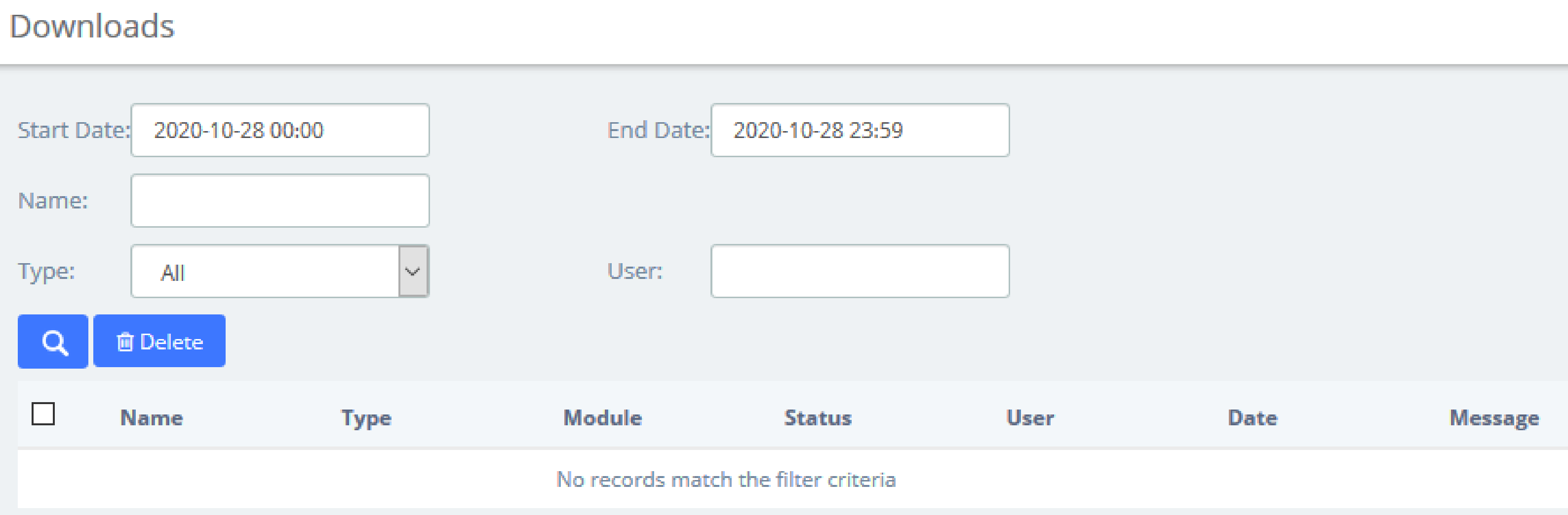
White List

If enabled **White Only**, the incoming call will be limited. For example, if you add a 1000 in whitelist, and the type is Inbound, then only 1000 can dial in and reach this extension.

8.7 Downloads

The call records generated on the CDR page or the CDR & Records page of the Me Bar can be downloaded on the Downloads page.

If the extension is given permission to view and download, the download content of the specified extension can also be viewed on the page. For details, see **2.4 User Permission**.

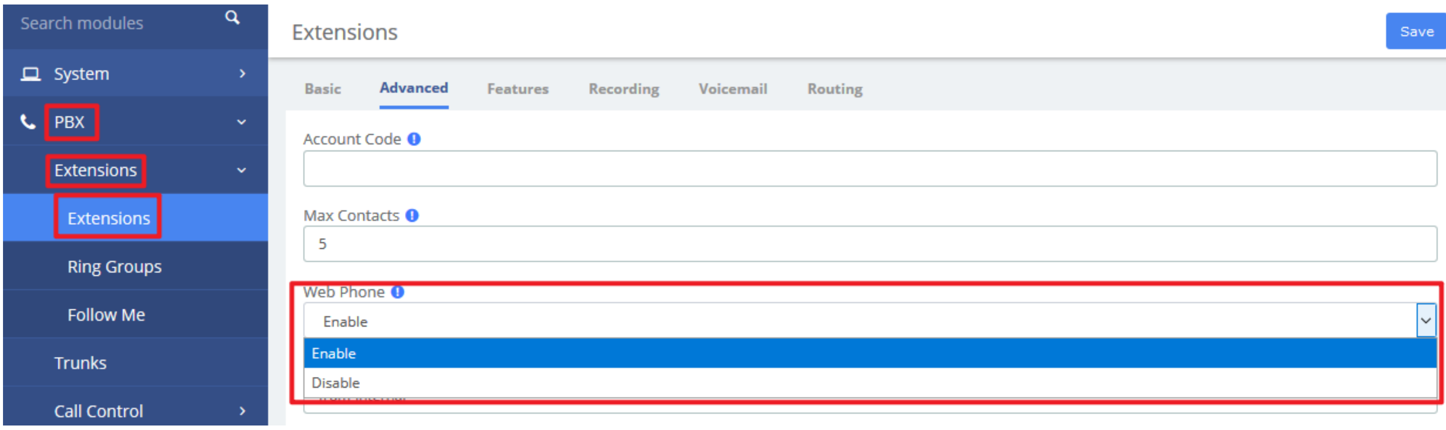


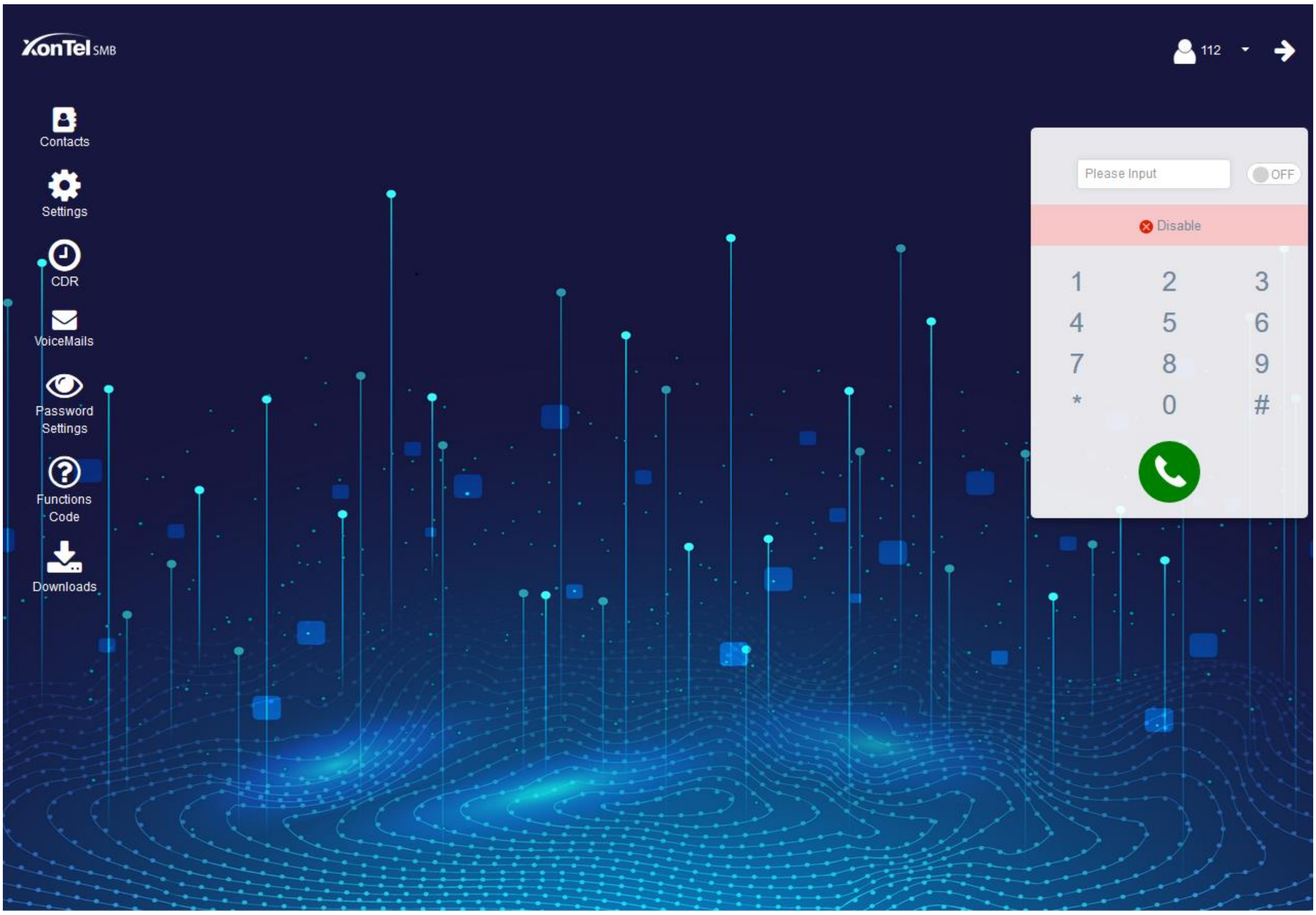
Downloads

9 Web Phone

If you have enabled Web Phone in **PBX** > **Extensions** > **Extensions** module under the admin account, you can enter the Web Phone module which supports all the functions of Me Bar, and use it to make calls directly. It should be noted that, since the underlying transmission of the VoIP uses the WSS protocol, this means that you cannot use Web Phone and other phones at the same time.

Recommend using Chrome browser.

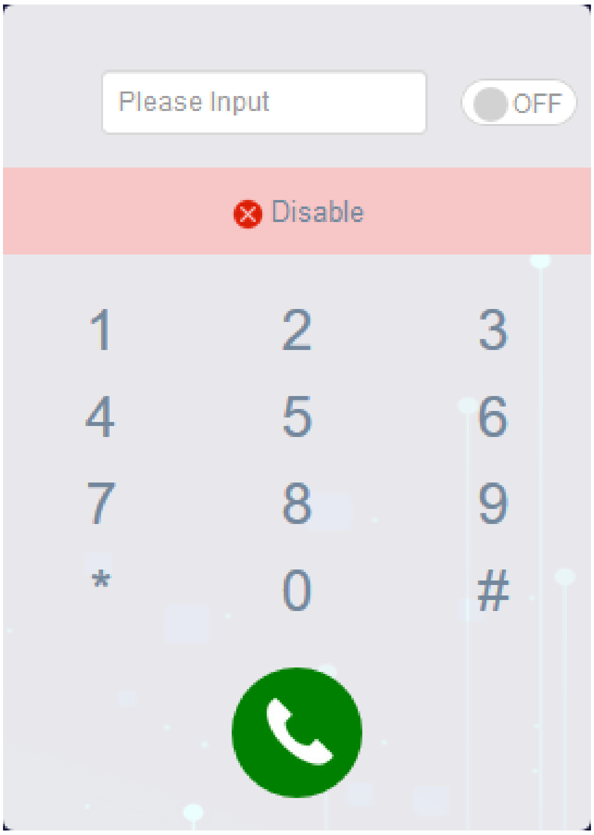




9.1 Web Dialing

When entering the module for the first time, the extension is not registered and the dial cannot be used. You need to slide the 

switch to turn it on. After it is turned on, the extension's transmission protocol will automatically become WSS and be registered.

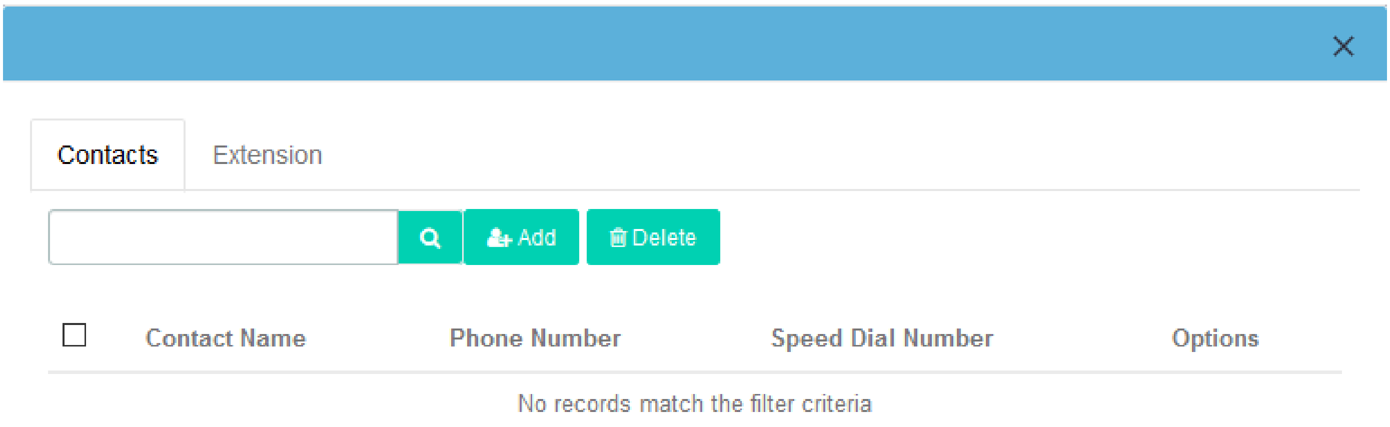


Dial

You can tap the dial pad on the page to dial, or input the number you want to dial, and then tap  to initiate the call. If the browser asks whether to enable the microphone, please allow it.

9.2 Contacts

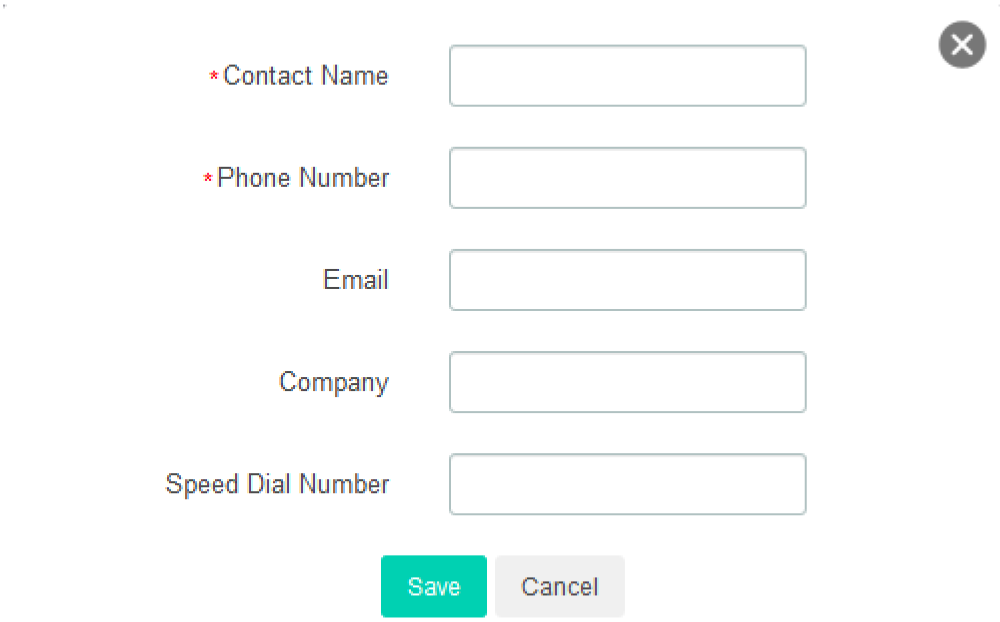
Contacts can be understood as a phone book, and you can add frequently used contacts to this phone book to achieve speed dialing. The added contact is only visible to the current extension.



Contacts

Click 

to add a new contact, "Contact Name" and "Phone Number" options are required.



Add Contact

After checking the box, click 

to delete contacts in batches. Of course, you can also click  to delete this contact.

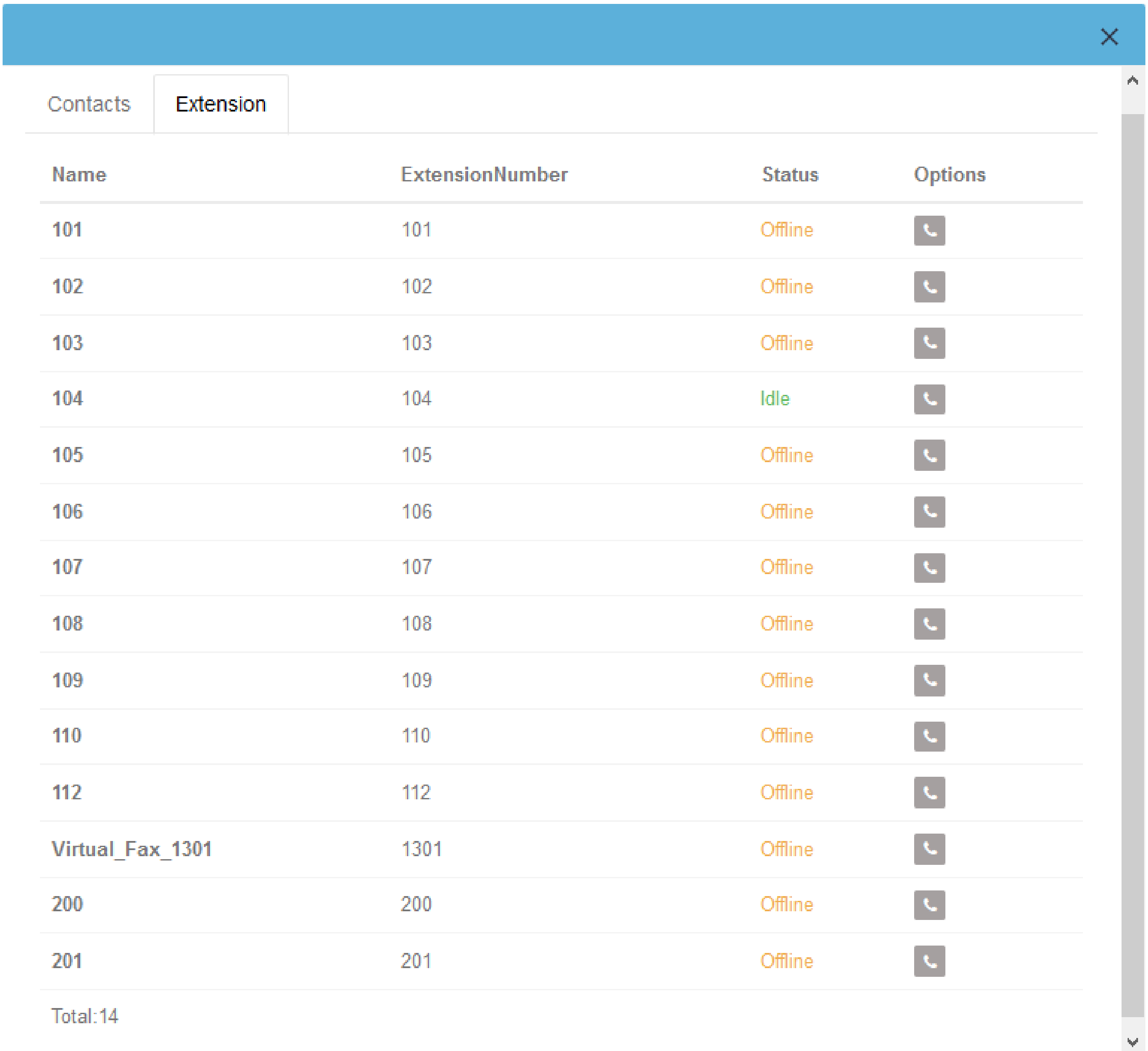
Click to directly dial the contact without entering the number.

Click to edit this contact.

In this interface, you can also search for a contact, enter his/her name in the input box, or enter the phone number, and click the button. If the contact is in the "Phonebook", the contact will be displayed, otherwise it will prompt "No records match the filter criteria".

The Extension tab will display all extensions in the SMB PBX system, and will display the status of the extensions (Idle, Offline, Busy).

Click to directly dial the currently online extension.

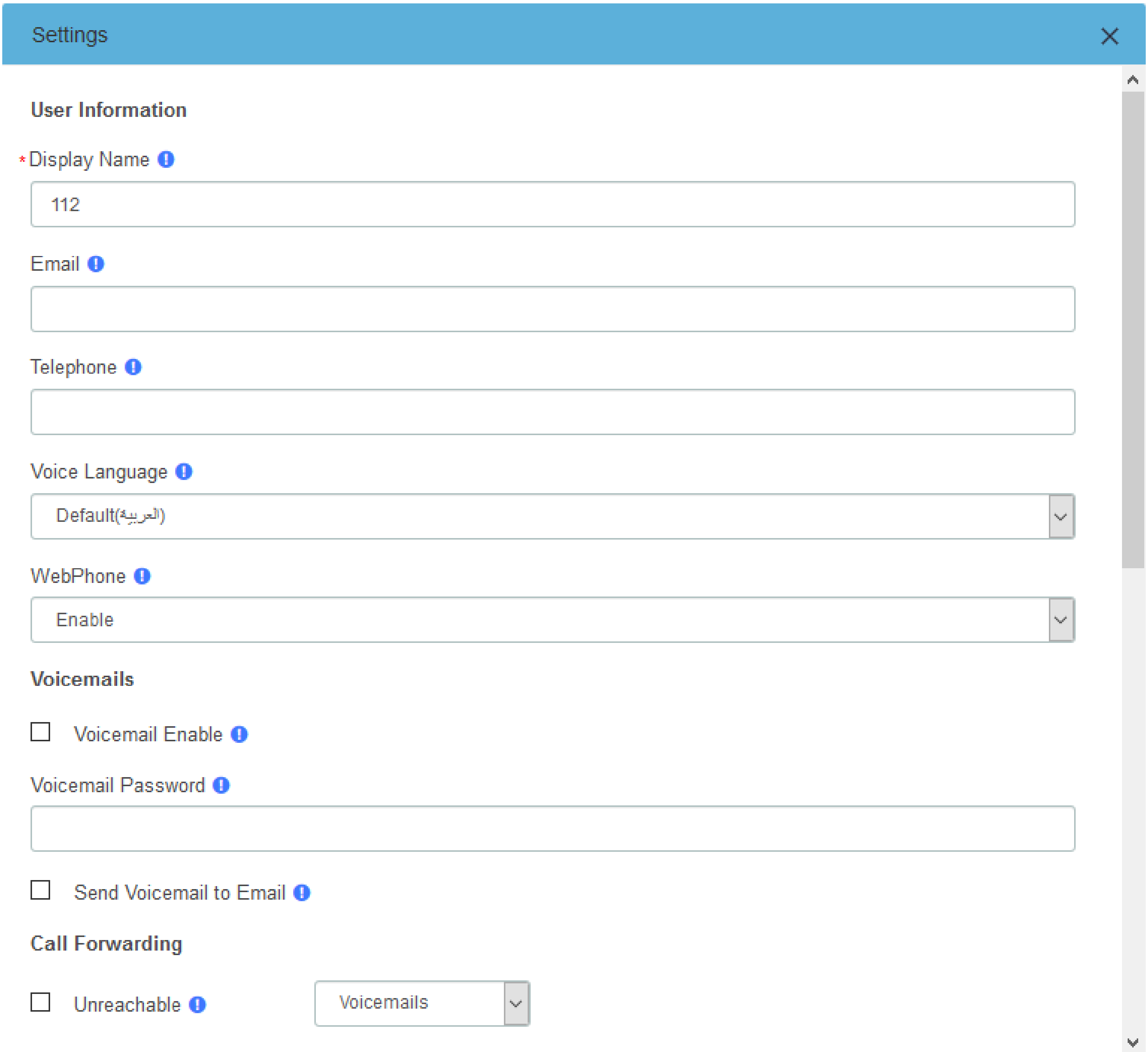


Name	ExtensionNumber	Status	Options
101	101	Offline	
102	102	Offline	
103	103	Offline	
104	104	Idle	
105	105	Offline	
106	106	Offline	
107	107	Offline	
108	108	Offline	
109	109	Offline	
110	110	Offline	
112	112	Offline	
Virtual_Fax_1301	1301	Offline	
200	200	Offline	
201	201	Offline	

Total:14

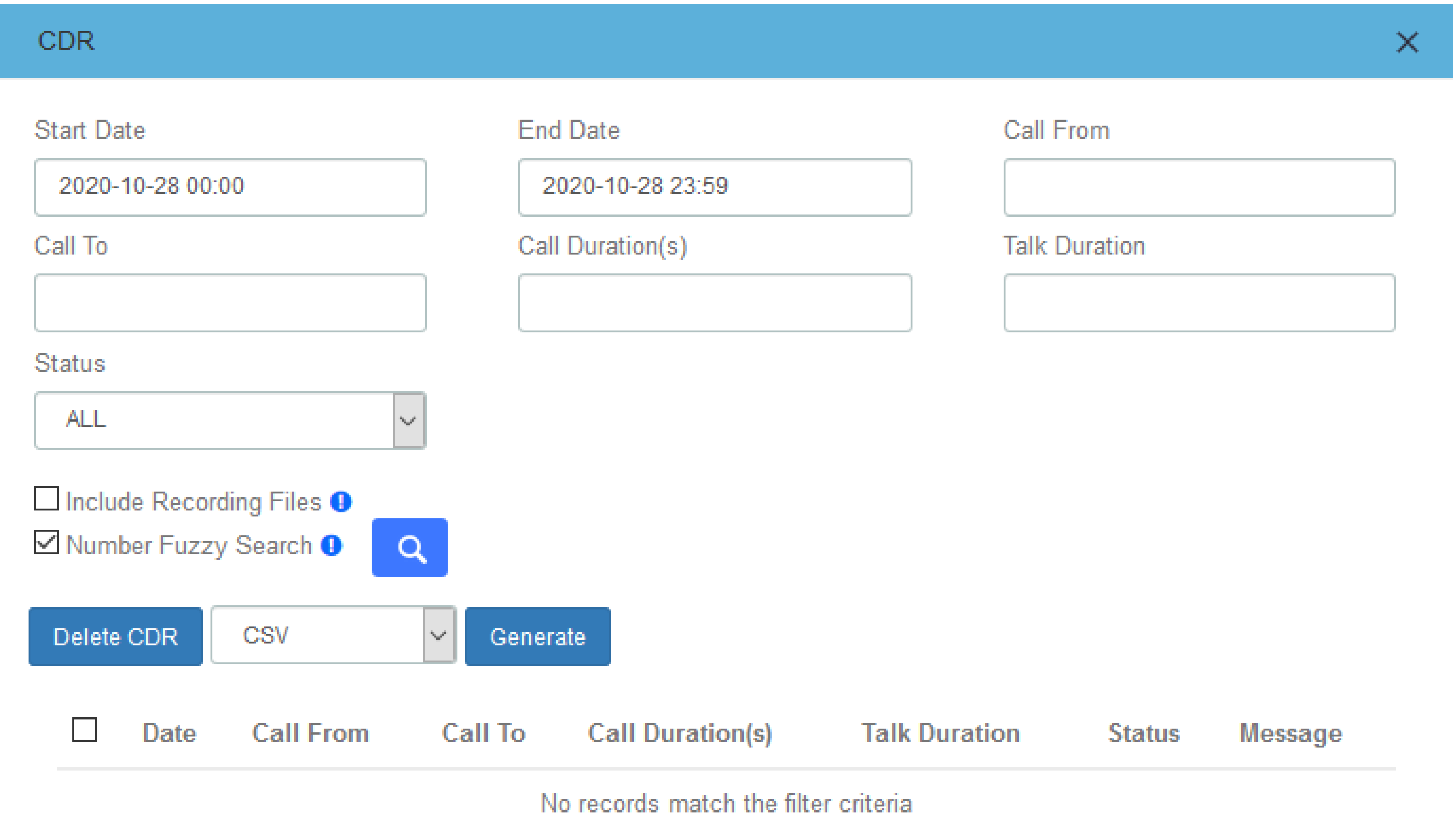
9.3 Settings

This setting page is basically the same as the **Extension** page in **Me Bar**.



9.4 CDR

You can view the call details records and related recordings of the current extension

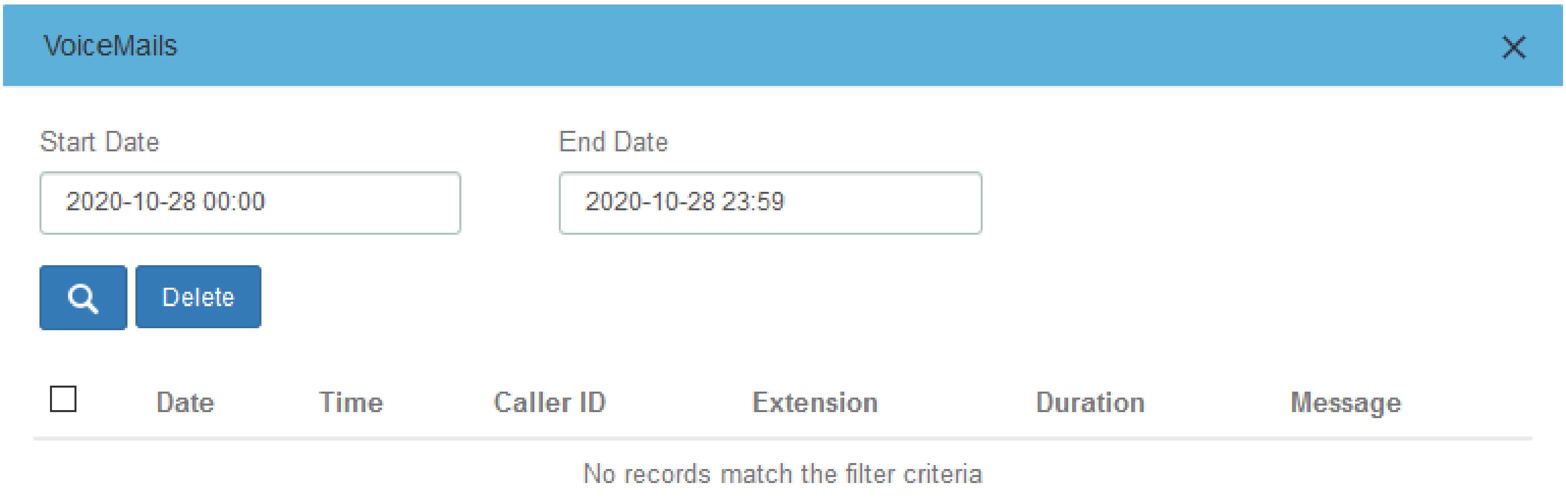


When the extension is given the permission to download call records, Generate button will appear, you can generate call records and download them in the download content.

When the extension is given the permission to delete call records, Delete CDR button will appear, you can select call records and delete.

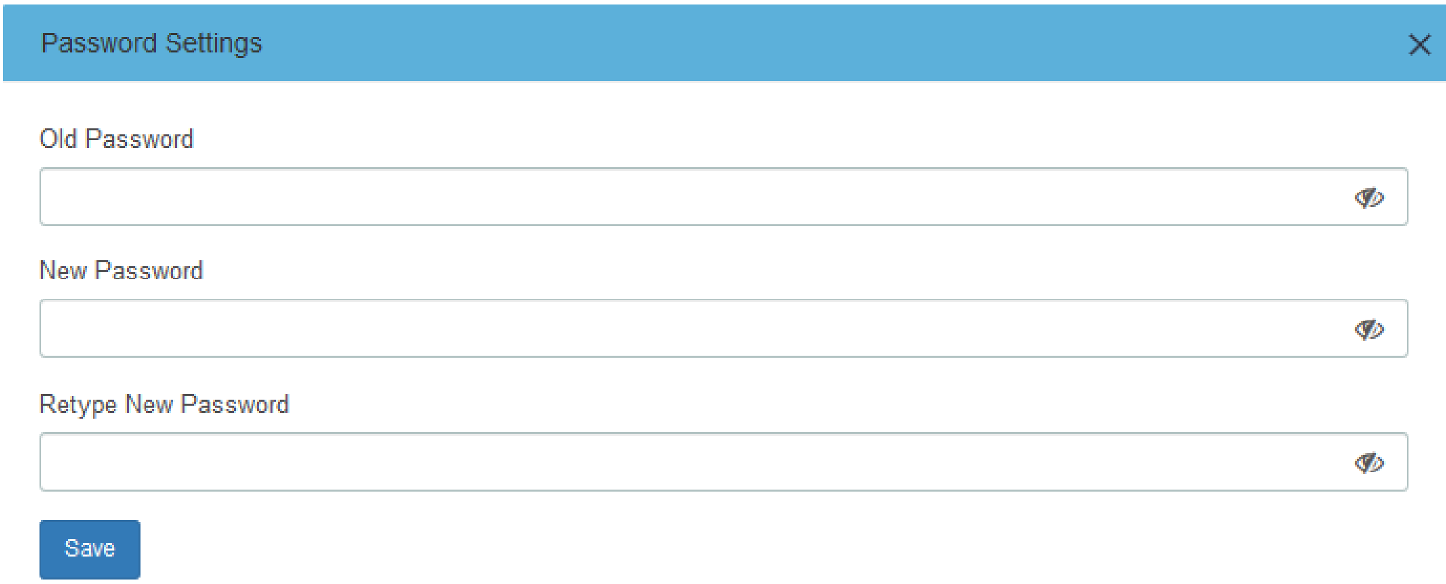
9.5 VoiceMails

You can view the VoiceMail of the current extension.



9.6 Password Settings

You can modify the login password of the extension.



9.7 Functions Code

The function code page lists all the function codes that can be used on SMB PBX. You can quickly configure/use the basic functions of system according to different function codes. For details, please refer to **PBX>Settings>Function Codes**.

9.8 Downloads

The call records generated on the CDR page or the **CDR & Records** page of the **Me Bar** can be downloaded on the **Downloads** page.

If the extension is given permission to view and download, the download content of the specified extension can also be viewed on the page. For details, see **2.4 User Permission**.