OpenVox Communication Co., Ltd

UC501 User Manual

Version 1.0

OpenVox Communication Co.,Ltd

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Revision History

Issue version	Issue date	Detail of change
1.0	Apr. 9th, 2019	Initial

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FCC Part 68

1. This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. On the bottom of this equipment is a label that contains, among other information, a product identifier in the format US:AAAEQ##TXXXX. If requested, this number must be provided to the telephone company.

2. A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA. A compliant telephone cord and modular plug is provided with this product. It is designed to be connected to a compatible modular jack that is also compliant. See installation instructions for details.

3. If this equipment [US: G4DIS01AUC501] causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

4. The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

5. If trouble is experienced with this equipment [US: G4DIS01AUC501], for repair or warranty information, Service can be facilitated through our office at:

   U.S. Agent Company name: WinWealth Tech Inc.

   Address: 170 W Pomona Ave Monrovia, CA 91016 USA

   Tel: +1 6262400785, +1 6265741300

   Fax: +1 626 574 1300

   If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

6. Please follow instructions for repairing if any (e.g. battery replacement section); otherwise do not alternate or repair any parts of device except specified. For repair procedures, follow the instructions outlined under the limited warranty.

7. Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.

8. If your home has specially wired alarm equipment connected to the telephone line, ensure the installation of this UC501 does not disable your alarm equipment. If you have questions about what will disable alarm equipment, consult your telephone company or a qualified installer.

9. If the telephone company requests information on what equipment is connected to their lines, inform them of:
   a) The ringer equivalence number [0.1]
   b) The USOC jack required [RJ11C]
   c) Facility Interface Codes ("FIC") [02LS2, 04DU9-BN, 04DU9-DN, 04DU9-1KN, 04DU9-1SN]
   d) Service Order Codes ("SOC") **[N/A]**
   e) The FCC Registration Number [US: G4DIS01AUC501]

10. The REN is used to determine the number of devices that may be connected to a telephone line. Excessive RENs on a telephone line may result in the devices not ringing in response to an incoming call. In most but not all areas, the sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company.

The REN for this product is part of the product identifier that has the format US:AAAEQ##TXXXX. The digits represented by ## are the REN without a decimal point. For this product the FCC Registration number is [US: G4DIS01AUC501] indicates the REN would be 0.1.

11. We suggest the customers use a surge arrestor to protect the device.

Caution - To ensure proper operation, this equipment must be installed according to the enclosed installation instructions. To verify that the equipment is operating properly and can successfully report an alarm, this equipment must be tested immediately after installation, and periodically thereafter, according to the enclosed test instructions.

Caution - Verification of Line Seize capability should be made immediately after installation, and periodically thereafter, in order to ensure that this equipment can initiate a call even when other equipment (telephone, answering system, computer modem, etc.) connected to the same line is in use.

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1 Overview

1.1 Introduction

UC501 IPPBX is an upgraded version of UC500. It can be pre-installed with OpenVox IPPBX system or other open-source communication system chosen by customers. It has built-in Uninterruptible Power Supply (UPS) and full PBX functions to meet different usage scenarios.

The UC501 is equipped with up to 8 analog ports and 2 Ethernet interfaces for seamlessly integrating VoIP trunks and your existing PSTN lines. In addition, UC501 supports a wide selection of codecs 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 UC Series appliances support industry standard SIP trunks, IAX2 trunks, analog PSTN trunks, and analog station trunks. In addition, the UC501 is modular in design, equipped with 1FXO/1FXS/4FXO/4FXS modules, and with a detachable chassis, users can easily change the port type or expand the system.

The UC series IPPBX 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 CallManager, Broadsoft, Huawei IMS 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 UC series delivers a full-featured IP Telephony solution. By supporting intelligent communication functions such as mobile phone extensions, instant multi-party conferences, call history, click-to-dial, and customer information management, it not only facilitates seamless communication between enterprise employees and customers, but also provides a solid basis for enterprises to analyze core business data.

Specification

Table 1-1-1 Product Specification

Item	Description
System Capacity	Up to 800 extension registers
	100 concurrent calls with G.729 codec
	300 concurrent calls with G.711 codec
Max Network Interface	2×10/100M LAN port
Max FXS/FXO Interface	8
USB Port	1×USB 2.0 for external storage or disaster recovery system
External Storage	1×SD slot, support up to 128G
Telephony Interface	FXS/FXO interface, Optional
RAM	DDR3 1GB
Storage	16GB Onboard Flash

Power Consumption	12V/1.33A 16W Maximum

Features

General

- Up to 8 FXS/FXO (PSTN/POTS) Analog Port
- Support SIP & IAX2
- Abundant HD voice codecs: OPUS, AMR-NB/WB, G.722, SILK and VP8
- HD Video Calls
- Echo Canceller

System

- Simple and Convenient Configuration via Web GUI
- User Portal
- Extension User Privileges
- System Administrators Monitor
- Event Notification
- Support Backup/Restore
- Remote Management
- Hot Standby

Network

- Network configuration
- Support Static Route
- Support Fail2ban
- Secure SIP calling (TLS encryption)
- Support Multiple VPN protocols including OpenVPN, L2TP, N2N, SSTP

PBX

- Import/Export Extensions
- Call Transfer
- Follow-Me/Ring Group/Queue
- Quickly Auto Provision IP Phones
- Support IMS
- Flexible Inbound/Outbound Route
- Blacklist
- AutoCLIP
- Time Condition
- PIN List
- Automated Attendant (IVR)

- Phonebook
- LDAP Service
- Wakeup Service
- DISA (Direct Inward System Access)
- Conference
- Call Back
- Call Parking
- Paging and Intercom
- Speed Dial
- Call Recording
- Music On Hold
- Support Open API Protocol (based on Asterisk)
- Click2call
- Support WebRTC
- Access Control Interface based on ACL
- AI TTS
- SIP Instant Messaging

Email

- Voicemail
- Missed Calls Notification
- Email Server
- Antispam support
- Support Mail Relay
- Fax to Email

Report

- Call Detail Records (CDR)
- Billing Report

1.2 Compatible Endpoints

- Any SIP compatible IP Phone (Desktop Phones and Soft Phones for Windows, Linux, iOS and also Android platforms). Desktop phone examples include: CooFone Series IP Phones provided by OpenVox, and also Cisco, Grandstream, Yealink, Polycom, Snom, Akuvox, Escene, Favil, HTek etc. Soft Phone examples include 3CX, CooCall, Linphone, X-Lite, Zoiper etc.
- IAX compatible endpoints, for example, CooFone IP Phones provided by OpenVox and also Zoiper softphone.
- Analog Phones and Fax Machines
- Web Extensions (WebRTC)

1.3 Module Combination

Users can choose two module accessories to customize.
- FXO-100+FXO-100
- FXO-100+FXS-100
- FXO-100+FXO-400
- FXO-100+FXS-400
- FXS-100+FXS-100
- FXS-100+FXO-400
- FXS-100+FXS-400
- FXO-400+FXO-400
- FXO-400+FXS-400
- FXS-400+FXS-400

1.4 Log in to the Web GUI

- **Step 1**
  Use a CAT5 cable to connect the device to the local network where the PC is connected, or connect the device 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
- **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 (e.g. 192.168.2.218);
- **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 password is admin.

Getting Started!

Website Login

Default IP: 172.16.101.1

Username: admin

Password: admin

Figure 1-3-1 Login interface

1.5 Web GUI overview

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

Figure 1-4-1 Web GUI layout
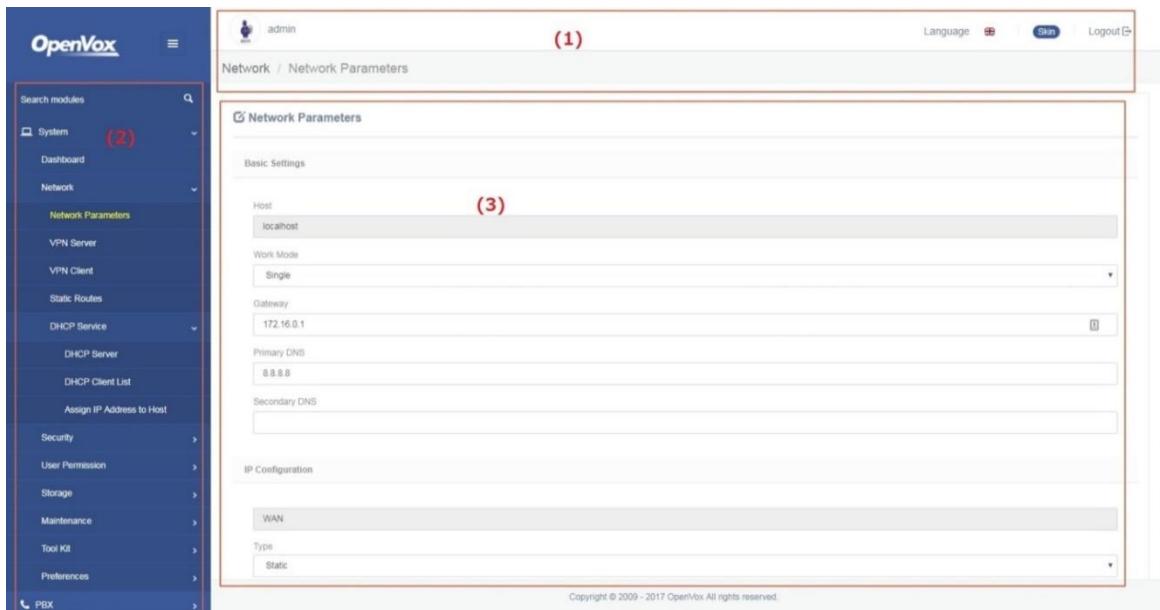
Table 1-4-1 Web Management Interface Layout

| Item	Description
(1) System button area	Contains buttons such as Reboot, Logout. Product information; and displays the identity of 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 configuration.

2 System

2.1 Dashboard

The option "Dashboard" of menu "System" in UC series 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 UC series is running. It allows to check out the history of CPU and Memory usage over the time.

Figure 2-1-1 System Resource
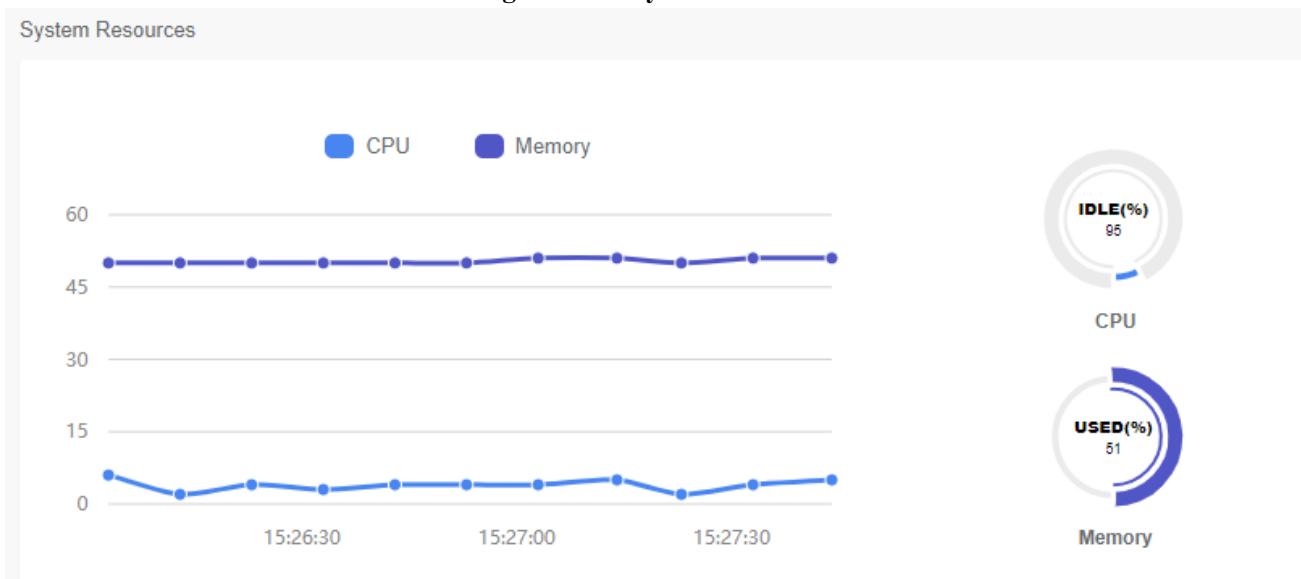
Processes Status: It shows the enabled and disabled processes. Here you can start, stop and restart these processes.

Figure 2-1-2 Processes Status
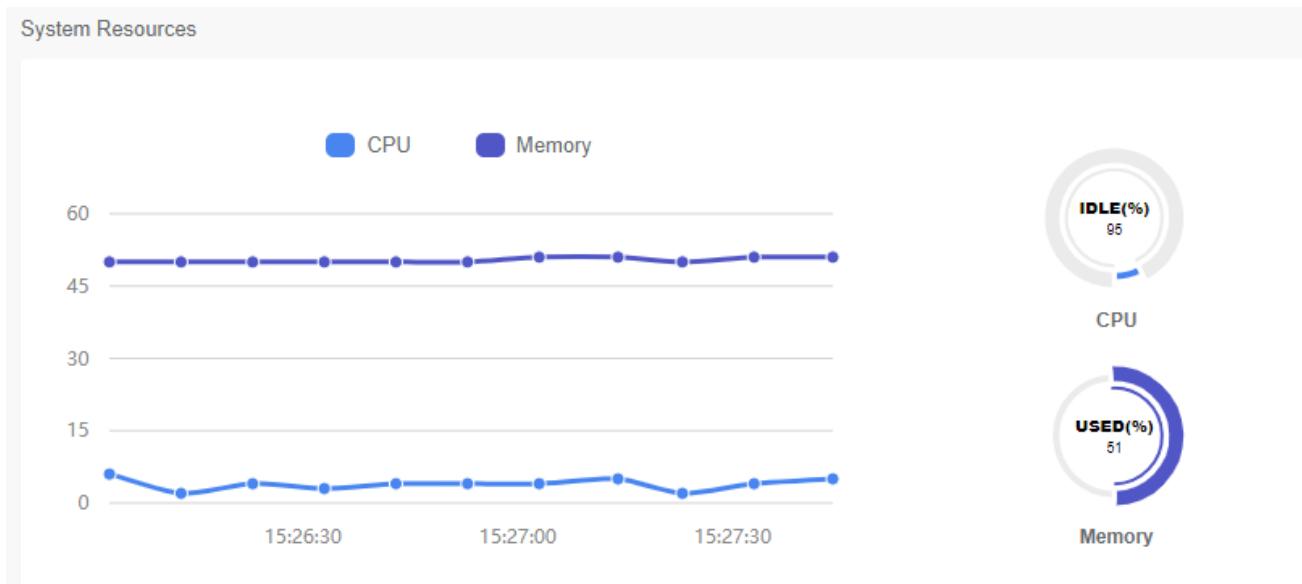
Hard Drives: Hard Drives shows the free and used space of the hard drives installed on your server.

Figure 2-1-3 Hard Drives
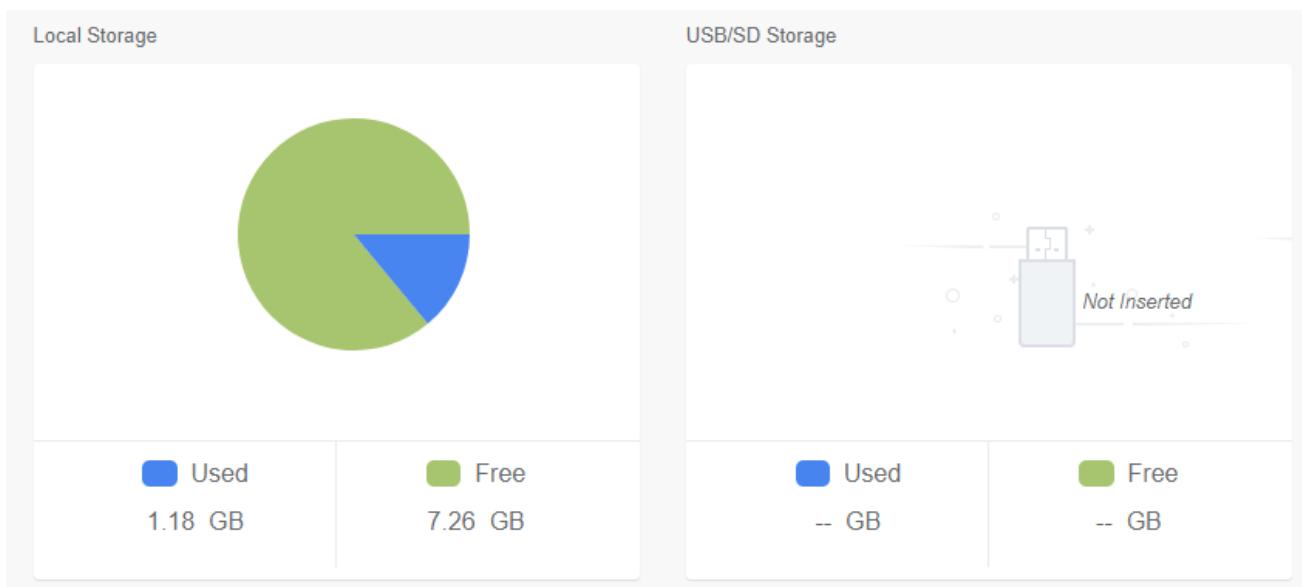
Communication Activity: This applet shows the number of extensions, trunks and calls currently on sip server.

Figure 2-1-4 Communication Activity
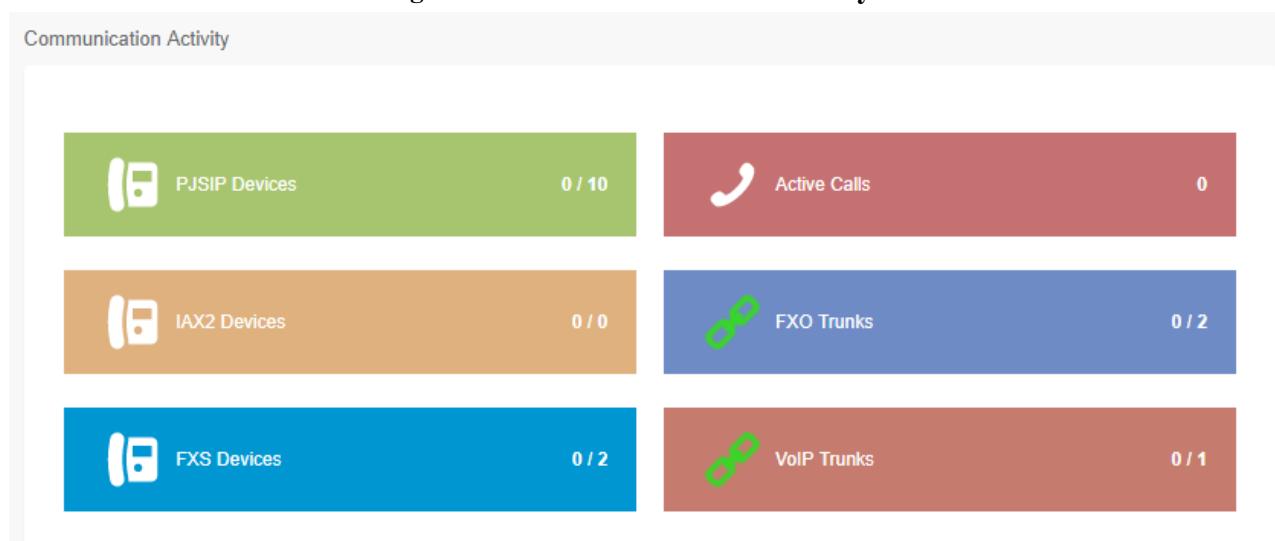
2.2 Network

2.2.1 Network Parameters

The option "Network Parameters" of the Menu "Network" in UC series series 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 2-1-5 Network Parameters Interface

This corresponds to the general network parameters of the server.

Table 2-1-1 Description of Edit Network Parameters

Item	Description
Basic Settings	
Host	Server Name, for example: pbx.subdomain.com
Work Mode	Optional work modes: Single/Double
Gateway	IP Address of the Port of Connection (Default Gateway)
Primary DNS	IP Address of the Primary Domain Name Server (DNS)
Secondary DNS	IP Address of the Secondary or Alternative Domain Name Server (DNS)
IP Configuration	
Type	The type of IP address that the Interface has, which could be STATIC when the IP address is fixed or DHCP when the IP address is obtained automatically from a DHCP server.
IP Address	IP Address assigned to the Interface
Mask	The Network Mask assigned to the Interface
MAC	Physical Address of the network Interface
Status	Shows the physical status of the Interface, if it's connected or not

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.
You are supposed to upload an OpenVPN Client config file downloaded from the VPN server if you chose OpenVPN mode.

Figure 2-1-7 VPN Client Interface

N2N

Basic

N2N VPN Settings

Enable

| Disable | ▼ |

Server IP Address

Server Port

Local IP

Subnet Mask

User Name

User Password

Connect Status: Disabled

Save

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L2TP

2.2.3 Static Routes

The Static Routes module of the menu "Network" lets users view and add the routing rules.

Figure 2-1-8 Static Routest Interface

Destination	Subnet Mask	Gateway	Metric	Interface
0.0.0.0	0.0.0.0	172.16.0.1	1024	WAN
172.16.0.0	255.255.0.0	0.0.0.0	0	WAN

Table 2-1-2 Description of Static Routes

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 intreface, denfined the interface used to send packet for the specific destination.

2.2.4 DHCP Service

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 UC series 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 UC series's DHCP service so it can assign IP addresses in the network.

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

Figure 2-1-9 DHCP Server interface

Here the description of each field.

Table 2-1-3 Description of DHCP Server

Item	Description
Status	It indicates if the DHCP service is enabled or disabled.
Starting IP Address	This will be the beginning of the IP range that the server will provide.
Ending IP Address	This will be the ending of the IP range that the server will provide.
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/Update]. The service can be started by clicking on [Enable DHCP].

DHCP Client List

This module shows a list of DHCP clients and leased IP addresses.
Navigate to **System > Network > DHCP Client List** and you will see a list of all devices receiving their IP address from the UC series system.

Figure 2-1-10 DHCP Client List interface

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

Assign IP Address 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 > Assign IP Address to Host.**

Figure 2-1-12 Assign IP Address to Host

To create a new association, click **Assign IP Address** button. Fill out the required information and click on **Save** button.

Figure 2-1-13 Add Assign IP Address
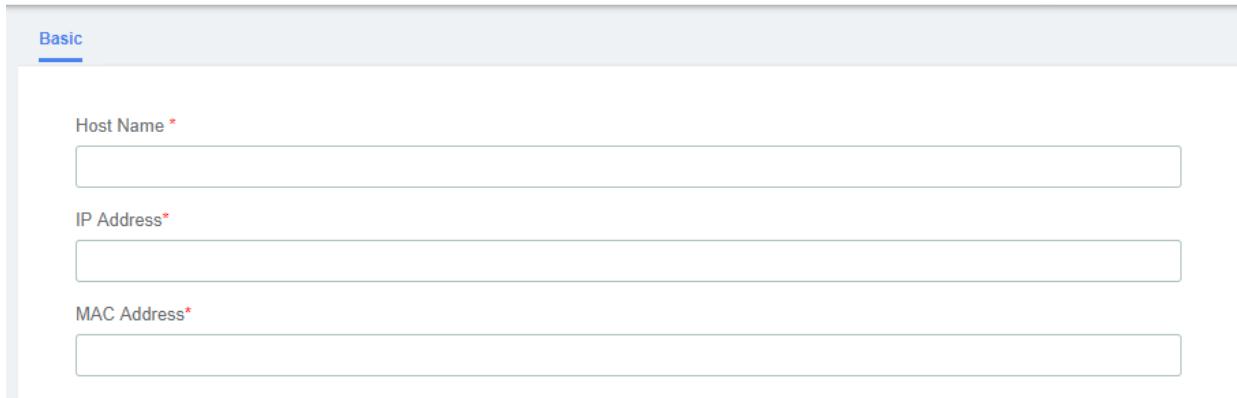
The following table shows the description of each field:

Table 2-1-4 Description of 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" in UC series 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.

Figure 2-1-14 Audit interface

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

2.3.2 Weak Keys

The module "Weak Keys" of the menu "Security" lets us identify the keys that are not enough strong for the extensions created in UC series (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.

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Figure 2-1-15 Weak keys interface

Weak Keys
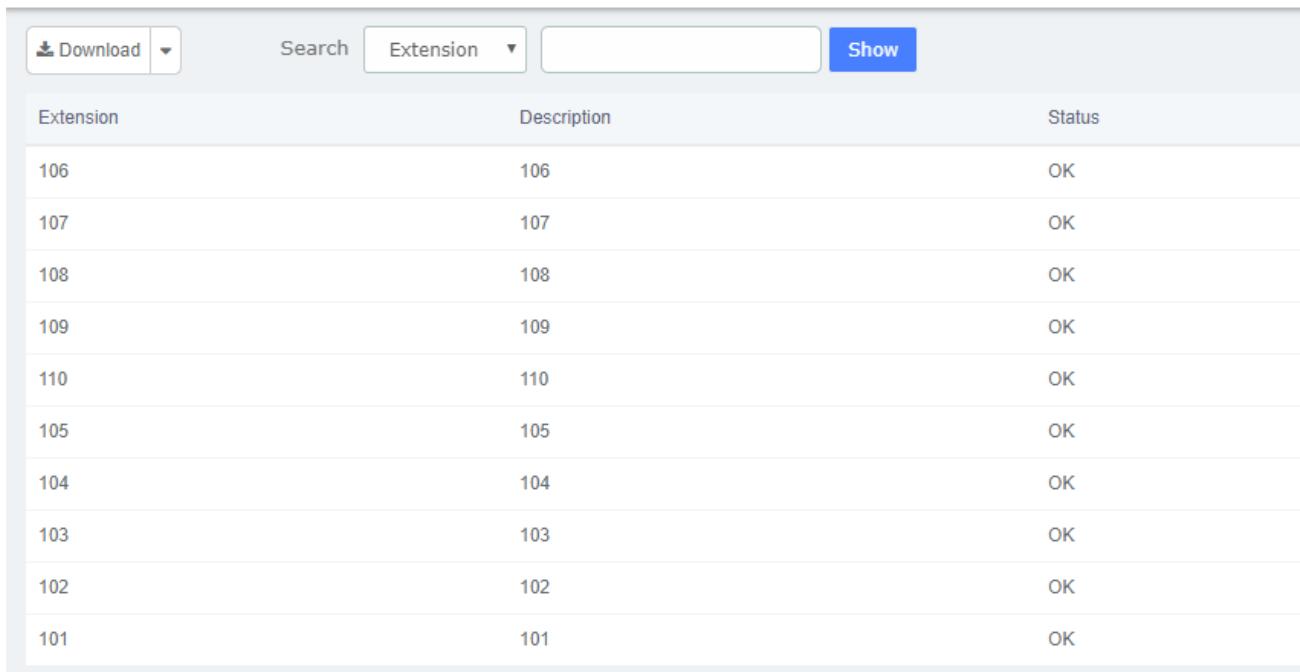
Extension	Description	Status
106	106	OK
107	107	OK
108	108	OK
109	109	OK
110	110	OK
105	105	OK
104	104	OK
103	103	OK
102	102	OK
101	101	OK

You can download the results in different formats such as PDF, XML and CSV by clicking on the "Download" button.

Change Key

If the key of an extension is not enough strong, you will be notified through the Status column and a link called "Change Secret" will be available to change the key. Once you click on this link, you will see a form where you can set the new secret. The secret must be at minimum six characters in length of which must contain at least two numbers and two letters. When the new secret is set, click on the "Save" button to apply the changes.

2.3.3 Certifications

The module "Certifications" of the menu "Security" implement greatly enhances the security. It's also rather confusing to get it working, and create or add a certificate on the asterisk server. There are three Client Methods for us to choose, it configures the clients to use TLS.

Figure 2-1-16 Certificate setting interface

To use TLS, you need to understand the principle of it. Check the TLS configuration parameters below, you would be rewarded about it.

We can create the certificate when inputted the Key Name, Organization, IP address and the Password. After client and server mutual authentication, with a license, it can be allowed to access.

There are several basic steps we need to do:

1. Your asterisk server needs a certificate.

We must create or add a certificate on the asterisk server. Creating a server key - We need to create a digital key for our server, and the key.pem is your server key. The key.pem file is your server key and the request.pem is your certificate request.

2. Add some configuration settings into the sip.conf file.

3. Configure the clients to use TLS

2.3.4 Firewall

Firewall Rules

UC series system has been preconfigured with a built-in firewall which prevents your IP phone system from unauthorized access, phone calls and other attacks. To manage the firewall, navigate to web menu **Security->Firewall**.

The option "Firewall" of the menu "Security" in UC series allows building iptables rules to control the packets that send and receive the UC series.

Figure 2-1-18 Deactivate firewall rules

Firewall Rules
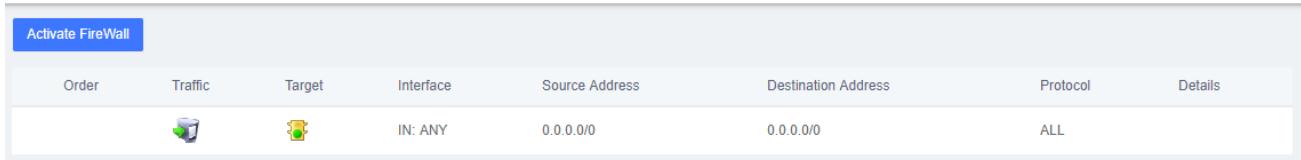
Order	Traffic	Target	Interface	Source Address	Destination Address	Protocol	Details
			IN: ANY	0.0.0.0/0	0.0.0.0/0	ALL	

To use this module the firewall must be enabled with the rules that appear by default. It can be done by clicking on "Activate Firewall" button. Once the firewall is enabled, you can create, delete, edit, disable and reorder the iptables rules.

Figure 2-1-19 Firewall rules interface

Every time you create or edit the rules, you have to save the changes. You can do this by clicking on "Save" button that will appear automatically when you modify something. If you don't save the changes the rules won't take effect in the system.

Adding a New Rule

To add a new rule click on the **New Rule** button and a form will appear with some data to fill out. The form can vary depending on the parameters you select for Traffic and Protocol.

Figure 2-1-20 Add a new rules interface

The ports used when you select the protocol TCP, UDP, ICMP and IP, are obtained from the module "Define Ports" in the same menu. Therefore, make sure the port you want to use is previously defined if you want to create a new rule.

In the source and destination address fields you have to enter the IP with the format x.x.x.x/y, where y is the network mask and should be a number between 0 and 32. If you let the default IP address (0.0.0.0) the netmask will be 0. If you let the netmask in blank it will not be taken into account. To enter a specific IP address, just let in blank the netmask value.

Once you created the rule, click on "Save" button and the new rule will appear in the list. Make sure

you save the changes so they take effect in the system after creating a new rule.

Editing a Rule

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

Figure 2-1-21 Edit a rules interface

Deleting a Rule

To delete a rule just select the checkbox corresponding to the rule at the left side and click on "Delete" button. Make sure you save the changes so they take effect in the system after deleting a rule.

Reordering the Rules

You can modify the order of the rules by clicking on the blue arrows in the column Order. If you click on the up arrow of a rule, this rule will go up one position and the one which was in that position will go down. If you click on the down arrow of a rule, this rule will go down one position and the one which was in that position will go up. Make sure you save the changes so they take effect in the system after modifying the position of the rules.

Activate /Deactivate a rule

You can activate or deactivate a rule by clicking on the light bulb corresponding to the rule. When it is ON the rule is activated, when it is OFF the rule is deactivated. Make sure you save the changes so they take effect in the system after doing this action.

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Define Ports

The module "Define Ports" of the menu "Security" in UC series allows creating, editing, and deleting ports that are used for the module "Firewall Rules". These ports can be from the protocols TCP, UDP, ICMP and IP. This module shows a list of all the existing ports and the results can be filtered by name and protocol.

Figure 2-1-22 Define Ports interface

Define Ports
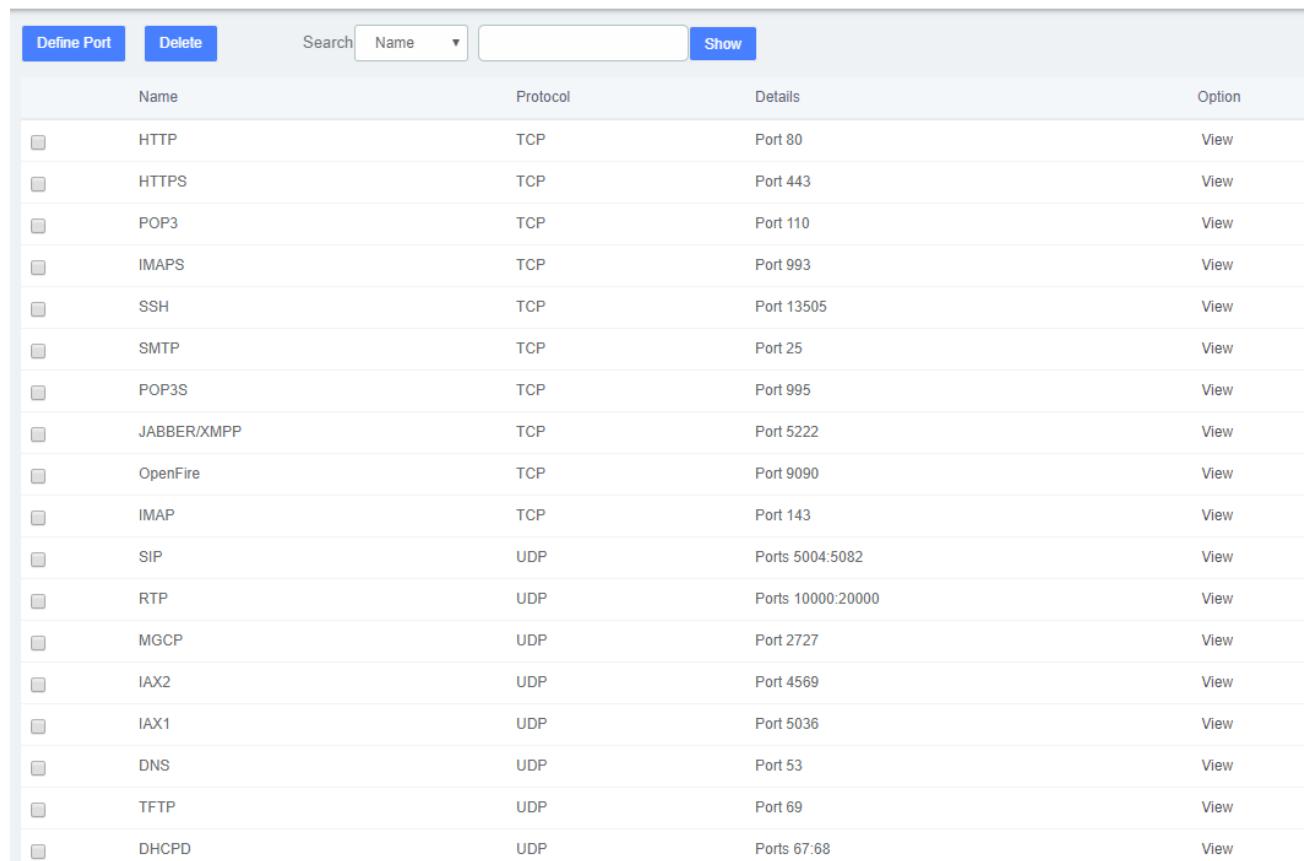
Name	Protocol	Details	Option
HTTP	TCP	Port 80	View
HTTPS	TCP	Port 443	View
POP3	TCP	Port 110	View
IMAPS	TCP	Port 993	View
SSH	TCP	Port 13505	View
SMTP	TCP	Port 25	View
POP3S	TCP	Port 995	View
JABBER/XMPP	TCP	Port 5222	View
OpenFire	TCP	Port 9090	View
IMAP	TCP	Port 143	View
SIP	UDP	Ports 5004:5082	View
RTP	UDP	Ports 10000:20000	View
MGCP	UDP	Port 2727	View
IAX2	UDP	Port 4569	View
IAX1	UDP	Port 5036	View
DNS	UDP	Port 53	View
TFTP	UDP	Port 69	View
DHCPD	UDP	Ports 67:68	View

Define Port

To define a new port, click on the **Define Port** button and a form will appear with some parameters to fill out. The form can vary depending on the parameters you select for the field Protocol. Once the information is filled, click on "Save" button.

View Port

To view an existing port, click on the "View" link located in the row corresponding to the port. Here you can see the information of the port and edit it if needed.

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Edit Port

To edit a port, click on the "View" link corresponding to the port you want to modify and then click on "Edit" button. A form will appear with the parameters of the port ready to be modified.

Delete a port

To delete a port just select the checkbox located at the left side corresponding to the port and click on "Delete" button.

2.3.5 Fail2Ban

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

Fail2Ban is able to reduce the rate of incorrect authentications attempts however it cannot eliminate the risk that weak authentication presents. Configure services to use only two factor or public/private authentication mechanisms if you really want to protect services.

The option "Fail2Ban" allows configuring Fail2ban service so it can prevent the uc series from

malicious attacks.

Navigate to **System > Security > Fail2Ban** to configure rules**.**

Figure 2-1-23 Fail2Ban interface

"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 IPPBX system will block the IP which exceeds max retry. These settings don't take effect on any allowed addresses.

Figure 2-1-24 Fail2Ban add whitelist

Add whitelist allows you to add a trusted IP addresses or network addresses to the system IP whitelist. The IPs in the whitelist will always be treated as trusted IP's and will not be filtered by the firewall rules.

2.4 User Permission

System > Users Permission allows you to create users and modify the permissions of users accessing the web interface. If the selected user group is **Administrator**, all function permissions are enabled by default; or the user group is set to **Custom**, and the user web permissions are customized. Click the "Create" button to create a new user, fill in the necessary information, and click Save:

Figure 2-1-26 Create New User

Click [Apply] to save the configuration.

2.5 Storage

2.5.1 Storage Devices

The option "Storage" of the menu "Storage" allows you to format your TF/SD card and set the default storage device.

Figure 2-1-30 Storage Devices Interface
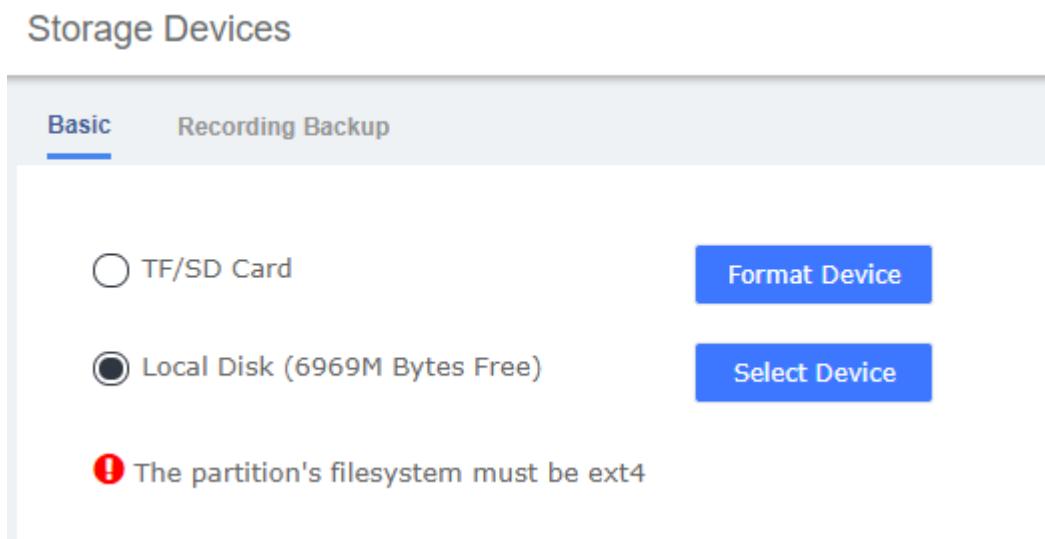
2.5.2 Auto Clean Up

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

Figure 2-1-31 Auto Clean Up Interface
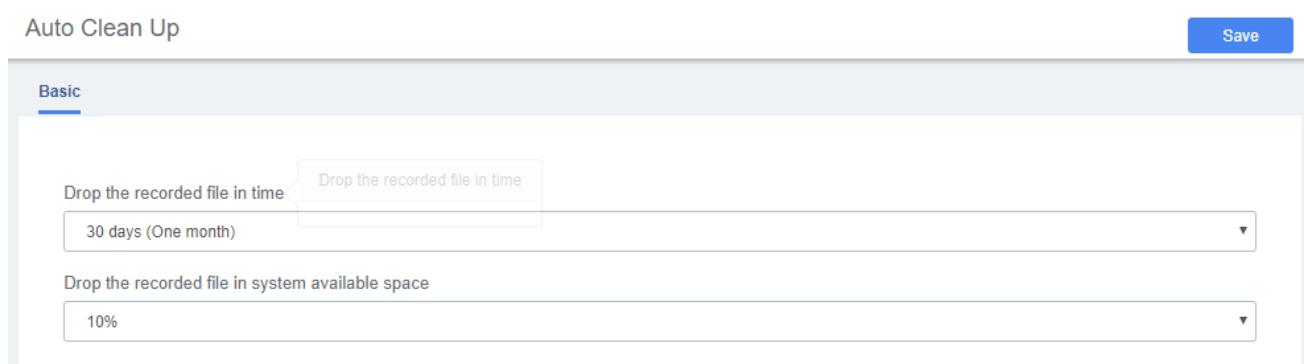
2.6 Email

The module **Email** allows adding a SMTP Server generally used to send messages from a mail client of a different mail server.
The fields to configure an Email are:

Figure 2-3-7 Email

Table2-3-2 Definition of Email

Item	Definition
Status	Status of connection SMTP Server.
SMTP Server	Remote email server.
Domain	Domain of SMTP Server.
Port	Port to establish the connection with SMTP Server.

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User	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). Some SMTP servers like Gmail requires these certificates.

2.7 LDAP Service

LDAP (Lightweight Directory Access Protocol) is a protocol for accessing directory services. It is generally used as a phone book on IPPBX. Based on the available LDAP services, it meets the requirements for fast search of phone directories. You can set up UC IPPBX as a server.
Once LDAP is set up, you can search the LDAP directory and find contacts on your IP phone.

Figure 2-3-7 LDAP Service

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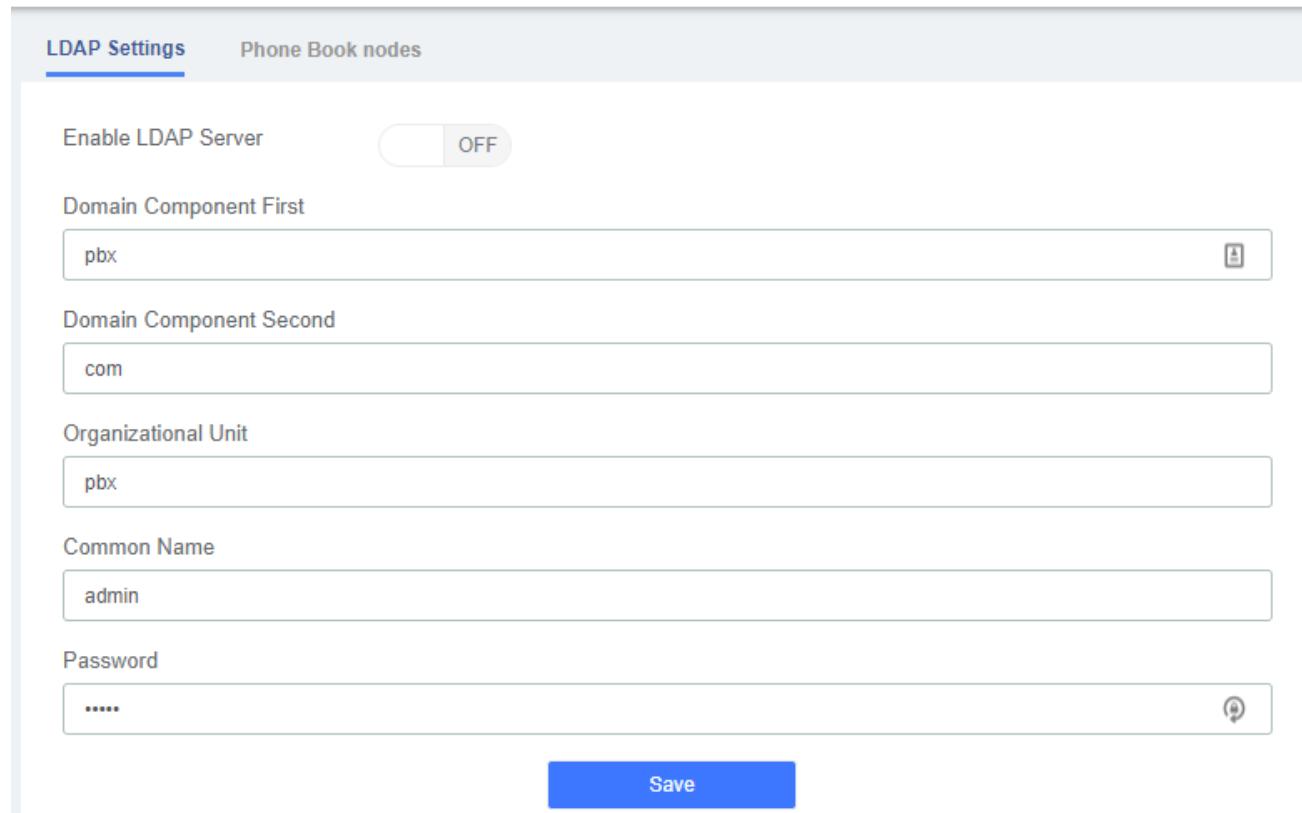
2.8 Maintenance

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 as well as update firmware online.

Figure 2-1-32 Firmware Update
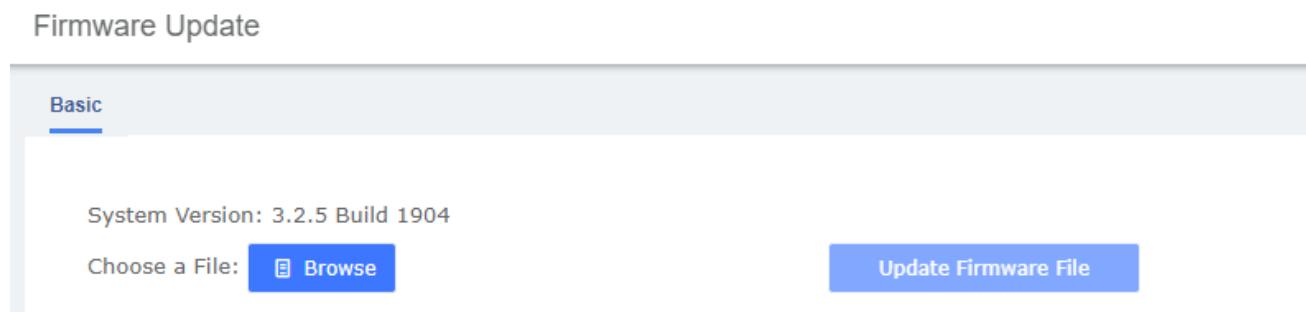
2.8.2 Backup & Restore

The option "Backup & Restore " of the menu "Maintenance" allows you to back up and restore the configuration of UC series, besides, you can also reset device to the default settings. If you have made a backup any time before this will appear in the list. To download a backup from the list, just click on the name of the tar file.

Figure 2-1-33 Backup & Restore
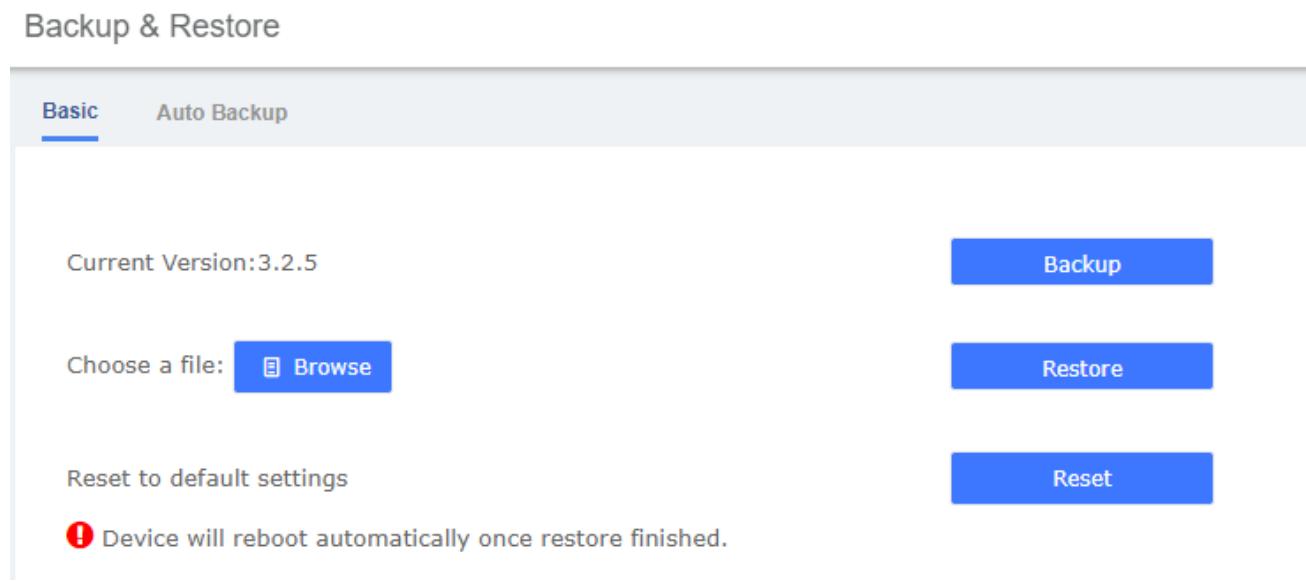
To enable auto backup, navigate to **System > Maintenance > Backup & Restore > Auto Backup**, change the disable option to the frequency you want . There are three media you could select to back up your config file : SD Card, FTP and CIFS.
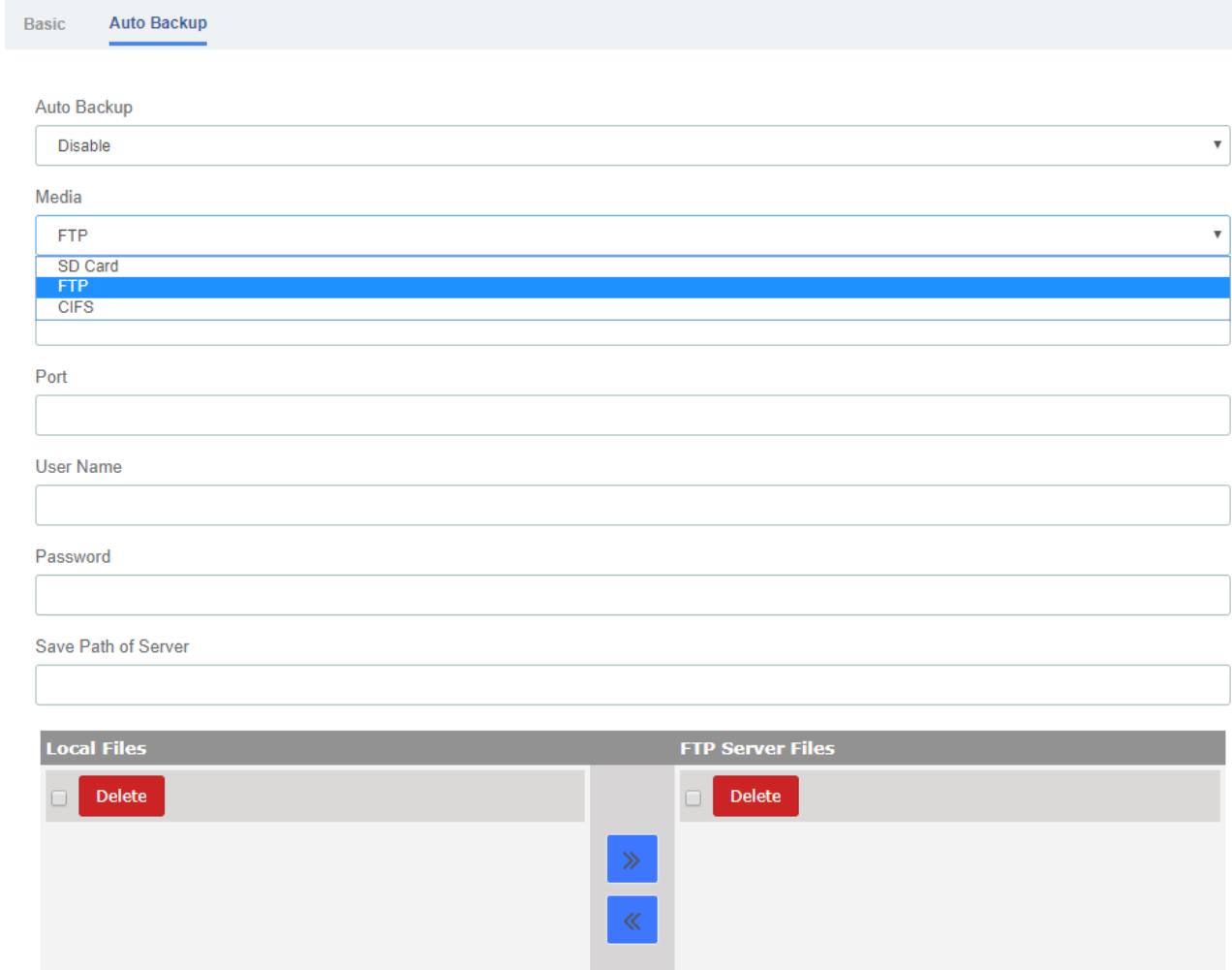
2.8.3 Login Settings

The option "Login Settings" of the Menu "Maintenance" in UC series lets us configure the login settings.

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

Figure 2-1-35 Login Settings Interface

By default, the SSH port is 13505. Generally, it is recommended that the SSH be disabled.

To enable SSH, enter the developer mode, navigate to **System > Maintenance > Login Settings > SSH Settings**, switching the enable to the on.

Figure 2-1-36 SSH Settings interface

2.8.4 Reboot Settings

This option allows for the rebooting of the IP-PBX series. Upon choosing whichever of the two options, you will be prompted to confirm the action.

Navigate to **System > Maintenance > Reboot Settings**.

Figure 2-1-34 Reboot Settings Interface

2.9 Event Center

The UC system provides time monitoring and prompting functions. Users can set events and notifications that need to be monitored, and add notification contacts. You can send monitoring alerts by sending emails or dialing extensions.

2.9.1 Event Settings

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2.9.2 Event Logs

In the upper right corner of the notification bar and **Event Center > Event Log** page, you can view the relevant logs of monitoring events.

2.10 Tool Kit

2.10.1 Network Capture

The UC series have been supplied a network packets capture in the web for ease of user to analysis,

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capture and monitor the network status, RTP flows, protocol analysis and so on.

Figure 2-1-37 Capture interface

2.10.2 Port Monitor

The UC series also supplied port monitor module for user to monitor and record the port steam.

Figure 2-1-38 Port Monitor interface

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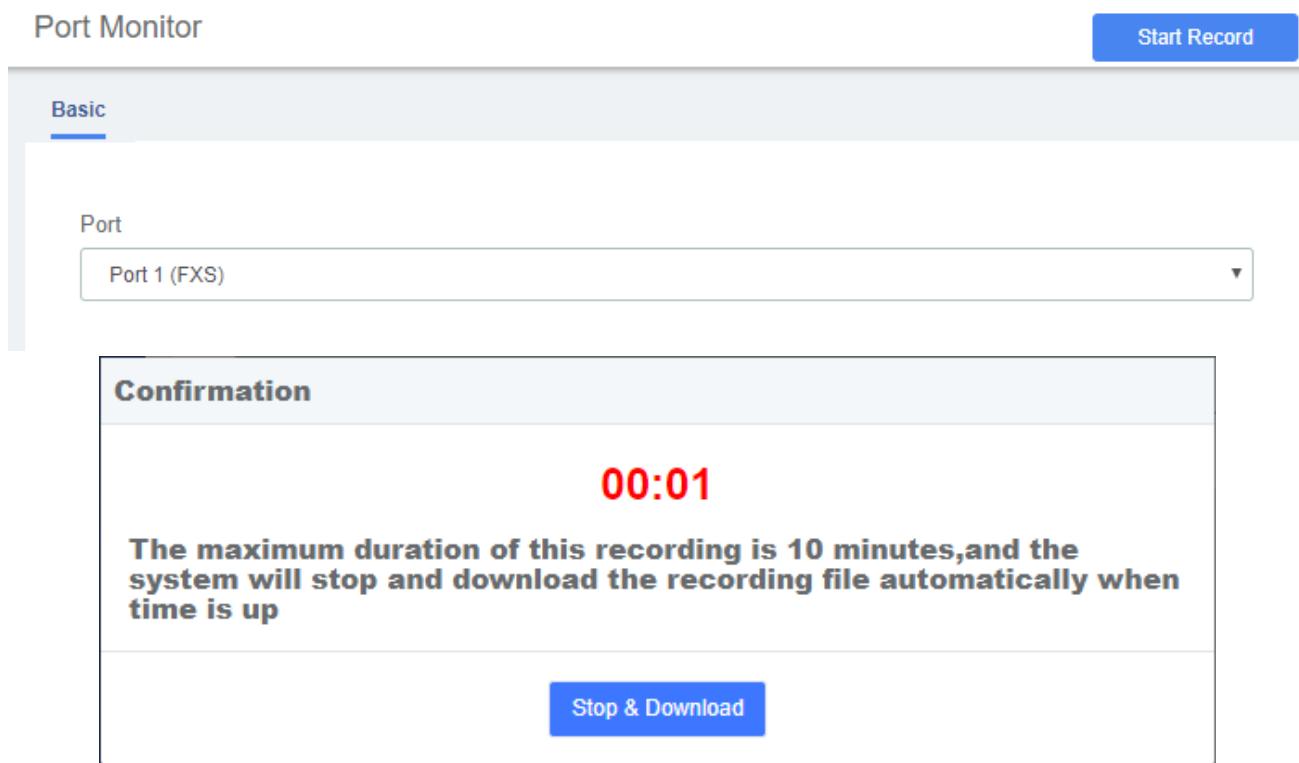
2.10.3 IP Ping and Traceroute

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

Figure 2-1-39 IP Ping and Traceroute interface
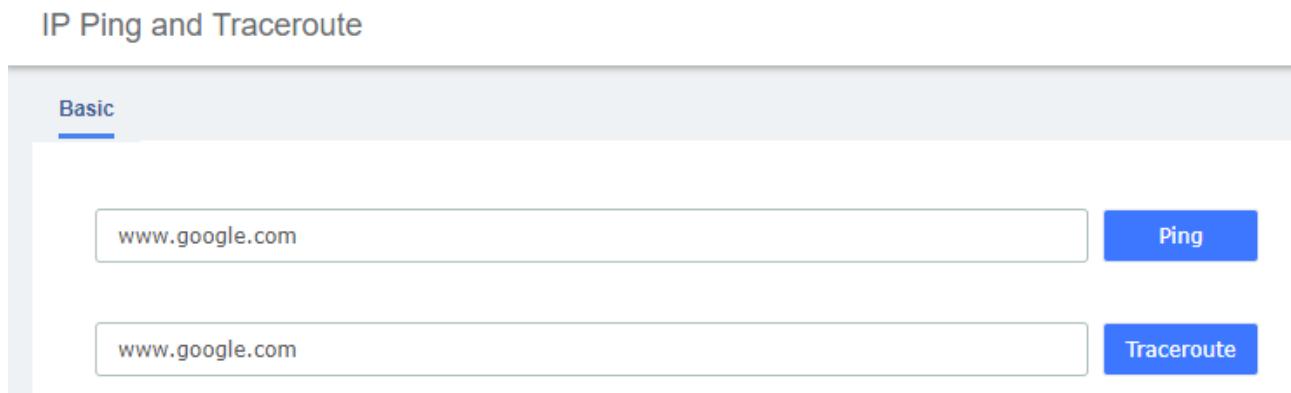
2.11 Preference

2.11.1 Language

The option "Language" of the Menu "Preferences" in UC series lets us configure the language for the UC series Web Interface.
Select the language from the list and click on the "save" button.
You can also download or upload languages you need.

Figure 2-1-40 Language setting
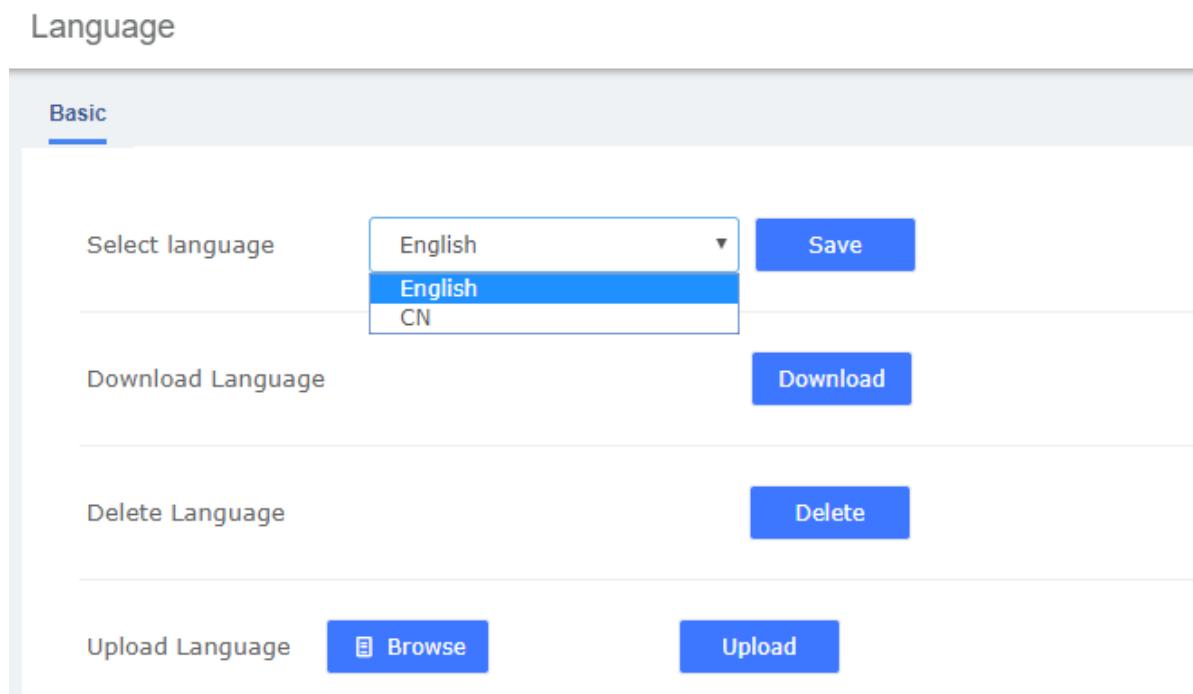
2.11.2 Date/Time

The option "Date/Time" of the Menu "Preferences" in UC series lets us configure the Date, Hour and Timezone for the UC series Web Interface.

Select the new date, hour and timezone and click on the "Apply changes" button.

Navigate to **System > Preferences > Date/Time** to deploy time server.

Figure 2-1-41 Date/Time Interface

2.11.3 Currency

The option "Currency" of the menu "Preferences" lets us change the currency for Reports in UC series Web Interface.

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

Figure 2-1-42 Currency Setting interface
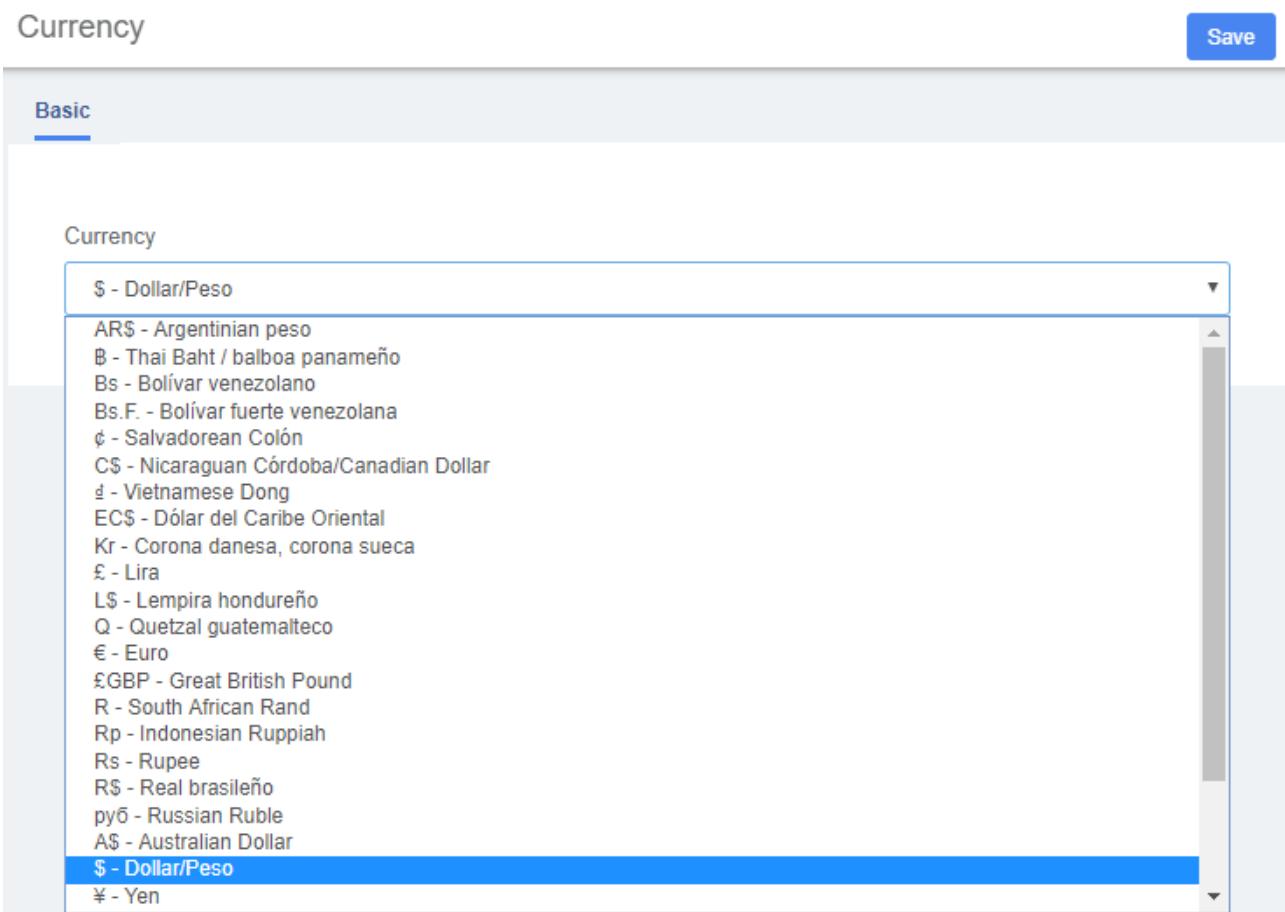
2.11.4 About

Navigate to **System >About**, lets us view some information of UC series about firmware version and other useful information.

Figure 2-1-43 About information

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2.11.5 Develop Mode

You can enter the developer mode by clicking the Hardware Version 5 times. This process is irreversible so please be cautious.

Developer Mode Options   ×

Are you sure you are entering developer mode? Please contact our customer service staff if you have any questions

Developer Mode: ON

Cancel   Save
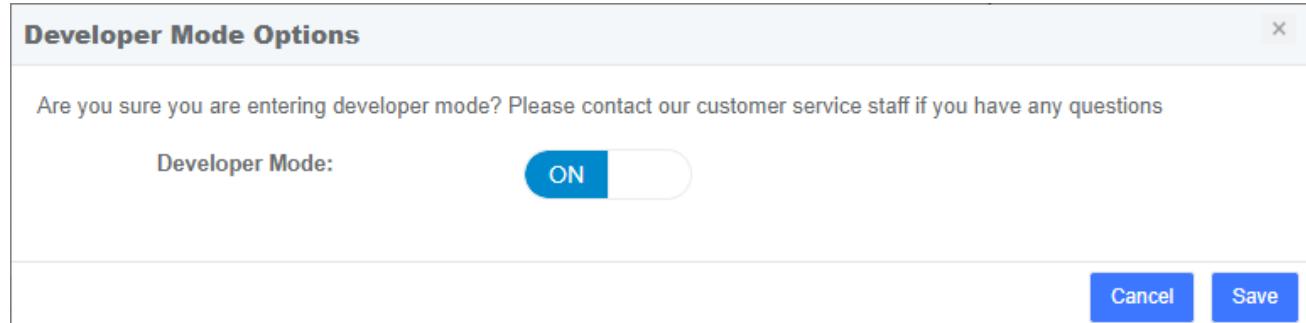
3 PBX

The Menu "PBX" lets us configure extensions, trunks, routes, dialplan, queues, IVR and so on for UC series.

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.

Figure 2-2-1 Add an Extension interface
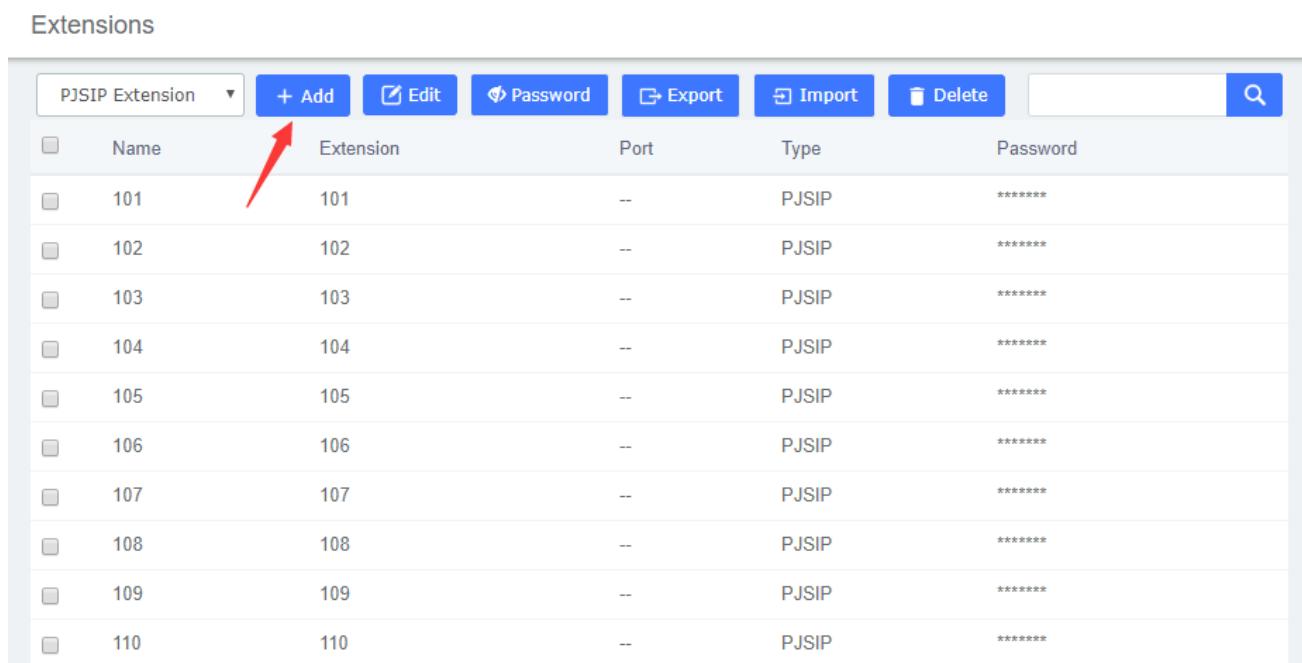
Extensions

	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	*******

Click one of extensions number and edit it:

Figure 2-2-2 Extension parameter interface

Table 2-2-1 Definition of Extension parameter

Item	Description
Basic	
Extension	The extension number to dial to reach this user.
Display Name	The CallerID name for calls from this user will be set to this name. only enter the name , NOT the number.
Secret	Password (secret) configured for the device. Should be alphanumeric with at least 2 letters and numbers to keep secure.
Advanced	
Dtmfmode	The DTMF signaling mode used by this device, usually rfc2833 for most phone.
Canreinvite	Re-Invite policy for this device, see Asterisk documentation for details.
Context	Asterisk context this device will send calls to. Only change this is you know what you are doing.
Host	Host settings for this device, almost always dynamic for endpoint.
Trustrpid	Whether Asterisk should trust the RPID settings from this device. Usually should be yes for CONNECTEDLINE( ) functionality to work if supported by the endpoint.
Sendrpid	Whether Asterisk should send RPID (or PAI) info to the device. Usually should be enabled to the settings used by your device for CONNECTEDLINE( ) functionality to work if supported by the endpoint.
Prack	The PRACK request plays the same role as ACK, but for provisional responses.

Type	Asterisk connection type, usually friend for endpoint.
NAT	NAT seting, see Asterisk documentation for details. Yes usually works for both internal and external devices. Set to No if the device will always be internal.
Port	Endpoint port number to use, usually 5060. Some 2 ports devices such as ATA may used 5061 for the second port.
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.
Qualifyfreq	Frequency in seconds to send qualify messages to the endpoint.
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 websockets (such as WebRTC) then you must select an option that includes WS.
Avpf	Whether to Enable AVPF. Defaults to no. the WebRTC standard has selected AVPF as the audio video profile to use for media streams. This is not the default profile in use by Asterisk. As a result the following must enabled to use WebRTC.
Icesupport	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 WebRTC setups.
Dtlsenable	Whether to enable DTLS for this peer. Defaults to no.
Dtlsverify	Whether to verify that the provided peer certificate is valid. Defaults to no.
Dtlssetup	Behavior on DTLS incoming and outgoing connections, defaults to actpass.
Dtlscertfile	Path to certificate file to present.
Dtlscafile	Path to cafile file to present
Dtlsprivatekey	Path to private key for certificate file.
Encryption	Whether to offer SPTR 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. Defaults to no.
Callgroup	Callgroup(s) that this device is part of, can be or more callgroups, e.g. '1,3-5' would be in groups 1,3,4,5.
Pickupgroup	Pickupgroup(s) that this 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.

Allow	Allow specified codecs, the available codecs are on the left options bar and the selected on the right.
Dial	How to dial device, this should not be changed unless you know what you are doing.
Accountcode	Accountcode for this device.
Mailbox	Mailbox for this device. This should not be changed unless you know what you are doing.
Vmexten	Asterisk dialplan extension to reach voicemail for this device. Some devices use this to auto-program the voicemail button on the endpoint. If left blank, the default vmexten setting is automatically configured by the voicemail module. Only changed this on devices that may have special needs.
Deny	IP Address range to deny access to, in the form of network/netmask.
Permit	IP Address range to allow access to, in the form of network/netmask. This can be a very useful security option when dealing with remote extensions that are at a known location (such as a branch office) or with a known ISP range for some home office situations.
Email Address	The email address that completed dictations are sent to.
Language Code	This will cause all messages and voice prompts to use the selected language if installed.
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. A common example is a team of support people who would like their internal CallerID to display the general support number(a ringgroup or queue). There will be no effect on external calls.
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 a addition to the users extension to call them.
Features	
Outbound CID	Override the callerid when dialing out a trunk. Any setting here will override the common outbound callerid set in the trunk admin. Format: **"caller name"** <#######> Leave this filed blank to disable the outbound callerid 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.
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	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

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Limit	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 callerid 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 callerID. 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 CallerID.
Pinless Dialing	Enabling Pinless Dialing will allow this extension to bypass any pin codes normally required on outbound calls.
Emergency CID	This callerid will always be set when dialing out an Outbound Route flagged ad Emergency. The Emergency CID overrides all other CallerID 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 affects 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'.
DID Description	A description for this DID, such as "Fax".
Add Inbound DID	A direct DID that is associated with this extension. The DID should be in the same format as provider (e.g. full number, 4digits for 10x4, etc).
Add Inbound CID	Add a CID for more specific DID+CID routing. A DID must be specified in the above Add DID box. 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.
Recording	
Inbound External Calls	Recording of inbound calls from external sources.
Outbound	Recording of outbound calls from external sources.

External Calls	
Inbound Internal Calls	Recording of calls received from other extensions on the system.
Outbound Internal Calls	Recording of calls made to other extensions on the system.
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 can not 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.
Email Address	The email address that Voicemails are sent to.
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 voicemailbox (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.
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, ringgroups, 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, ringgroups, 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 UC series. This makes it easy the migration of data.

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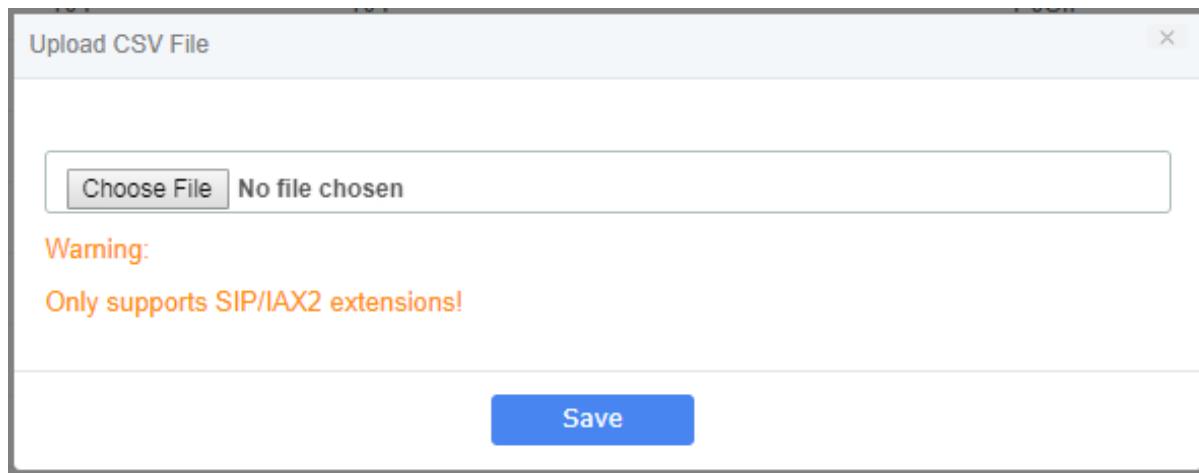
To download a CSV file with all the extensions created in UC series, click on the [Export] button and save the file into your local hard drive.

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.
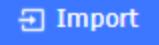
Make sure the following indications are taken into account:

- 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:

Figure 2-2-3 Ring groups interface
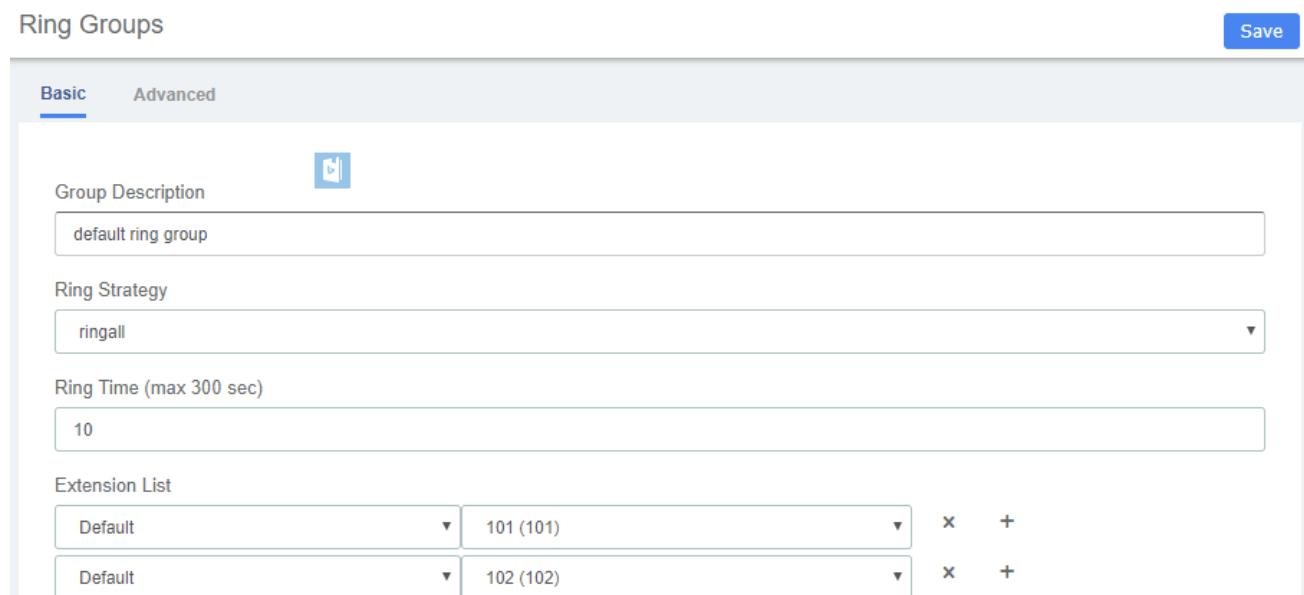
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Table 2-2-2 Definition of add Ring groups interface

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 1 st and 2 nd and 3 rd 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 FreePBX CF unconditional, then all will be rung **First available:** ring only the first available channel **Firstnotonphone:** ring only the first channel which is not offhook-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.
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 callerid 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 CallerID
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 "+".
Record Calls	You can always record calls that come into ring group, never record them, or allow the extension that answers to do on-demand recording. If recording is denied then one-touch on demand recording will be blocked.

3.1.3 Follow Me

Follow Me (also known as **Find Me / Follow Me** or **FMFM**) 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.**
You will be presented with the following screen:

OpenVox Communication Co. , LTD.   URL: www.openvox.cn

Figure 2-2-4 Follow Me interface

Select the extensions that you want to define.

Figure 2-2-5 Follow Me User interface

OpenVox Communication Co. , LTD.   URL: www.openvox.cn

Table 2-2-3 Definition of Follow Me

Item	Definition
Basic	
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 FreePBX DND, it won't be rung. If the primary is FreePBX 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
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 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	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 CallerID
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 "+".

3.1.4 Endpoint Configurator

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

Figure 2-2-6 Endpoint Configurator interface

	Order	Status	MAC Address	Current IP	Manufacturer	Model	Manual	Extenstions	Extenstions Status
☐	1		00:1F:C1:1D:BC:53	172.16.1.85	Hanlong	(not detected) ▾	🔧		🔴
☐	2	🗑 👤	0C:38:3E:1E:D5:44	172.16.1.67	Fanvil	X6 ▾	🔧		🔴
☐	3	🗑	A0:98:05:17:02:8E	172.16.1.161	OpenVox	C401 ▾	🔧		🔴
☐	4		00:A8:59:F5:E0:79	172.16.1.209	VOPTech	(not detected) ▾	🔧		🔴
☐	5		00:A8:59:F7:8C:83	172.16.1.210	VOPTech	(not detected) ▾	🔧		🔴
☐	6		0C:38:3E:1B:5B:EA	172.16.1.212	Fanvil	X4 ▾	🔧		🔴
☐	7		00:0E:30:00:91:EA	172.16.1.69	VOPTech	VI2007 ▾	🔧		🔴

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:

Table2-2-4 Description of **Interface description**

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 Elastix 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

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.

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.

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.
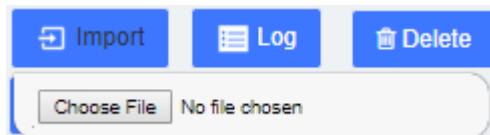
Upload toolbar button

Clicking on this icon will display a small dialog in which the user may specify an endpoint list file to upload to the server. The file format is automatically detected.

3.2 Trunks

The "Trunks Module" is used to connect your FreePBX/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 FreePBX/ 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.

Figure 2-2-7 Add trunk interface

Figure 2-2-8 Add SIP Trunk

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Table2-2-5 Definition of add a SIP trunk

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.
Outbound Proxy Server	Example: proxy.provider.domain:port
Host	Host settings for this device, almost always dynamic for endpoint.
Username	Username configured for this trunk.
Secret	Password (secret) configured for the device. Should be alphanumeric with at least 2 letters and numbers to keep secure.
Port	Endpoint port number to use, usually 5060. Some 2 ports devices such as ATA may used 5061 for the second port.
Register String	Most VoIP providers require your system to REGISTER with theirs. Enter the registration line here. example: username:password@switch.voipprovider.com. Many providers will require you to provide a DID number, ex: username:password@switch.voipprovider.com/didnumber in order for any DID matching to work.
Codec	Allow specified codecs, the available codecs are on the left options bar and the selected on the right.
Advanced	
Outbound CallerID	CallerID for calls placed out on this trunk Format: <#######>. You can also use the format: "hidden" <#######> to hide the CallerID 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 the auto-generated context: as the inbound trunk's context. (see extensions_additional.conf) Leave blank to specify no maximum.
Type	Asterisk connection type, usually friend for endpoint.
Outbound Dial	The outbound dialing prefix is used to prefix a dialing string to all outbound calls placed on this

Prefix	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.
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.
Qualifyfreq	Frequency in seconds to send qualify messages to the endpoint.
Nat	NAT seting, see Asterisk documentation for details. Yes usually works for both internal and external devices. Set to No if the device will always be internal.
Insecure	Specifies how to handle connections with peers. Default no (authenticate all connections).
Dtmfmode	The DTMF signaling mode used by this device, usually rfc2833 for most phone.
Trustrpid	Whether Asterisk should trust the RPID settings from this device. Usually should be yes for CONNECTEDLINE( ) functionality to work if supported by the endpoint.
Sendrpid	Whether Asterisk should send RPID (or PAI) info to the device. Usually should be enabled to the settings used by your device for CONNECTEDLINE( ) functionality to work if supported by the endpoint.
Prack	The PRACK request plays the same role as ACK, but for provisional responses.
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 messages 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 websockets (such as WebRTC) then you must select an option that includes WS
Avpf	Whether to Enable AVPF. Defaults to no. the WebRTC standard has selected AVPF as the audio video profile to use for media streams. This is not the default profile in use by Asterisk. As a result the following must enabled to use WebRTC.
Icesupport	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 WebRTC setups
Dtlsenable	Whether to Enable DTLS for this peer. Defaults to no.
Dtlsverify	Whether to verify that the provided peer cerificate is valid. Defaults to no.
Dtlssetup	Behavior on DTLS incoming and outgoing connections. Defaults to actpass.
Dtlscertfile	Path to certificate file to present
Dtlscafile	Path to cafile file to present
Dtlsprivatekey	Path to private key for certificate file.
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. Defaults to no.

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
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.

Figure 2-2-9 Add FXO Trunk

Table2-2-6 Definition 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). The default setting is g0 (group zero).
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	CallerID for calls placed out on this trunk Format: <########>. You can also use the format: "hidden" <#######> to hide the CallerID 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
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.

Figure 2-2-10 Add IAX2 Trunk

Trunks   Save

Basic   Codec   Advanced

Enable Trunk

On

Trunk Mode

IP Authentication

Trunk Name*

Host*   Port

:

Username

Secret

Table 2-2-7 Definition of Add IAX2 Trunk

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.
Username	Username configured for this trunk.
Secret	Password (secret) configured for the device. Should be alphanumeric with at least 2 letters and numbers to keep secure.
Port	Endpoint port number to use, usually 5060. Some 2 ports devices such as ATA may used 5061 for the second port.
Register String	Most VoIP providers require your system to REGISTER with theirs. Enter the registration line here. example: username:password@switch.voipprovider.com. Many providers will require you to provide a DID number, ex: username:password@switch.voipprovider.com/didnumber in order for any DID matching to work.
Codec	Allow specified codecs, the available codecs are on the left options bar and the selected on the right.
Advanced	
Outbound CallerID	CallerID for calls placed out on this trunk Format: <#######>. You can also use the format: "hidden" <#######> to hide the CallerID 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.
Type	Asterisk connection type, usually friend for endpoint.
Outbound Dial Prefix	The outbound dialing prefix is used to prefix a dialing string to all outbound calls 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.
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.
Qualifyfreq	Frequency in seconds to send qualify messages to the endpoint.
Nat	NAT seting, see Asterisk documentation for details. Yes usually works for both internal and external devices. Set to No if the device will always be internal.
Insecure	Specifies how to handle connections with peers. Default no (authenticate all connections).
Dtmfmode	The DTMF signaling mode used by this device, usually rfc2833 for most phone.
Trustrpid	Whether Asterisk should trust the RPID settings from this device. Usually should be yes for CONNECTEDLINE( ) functionality to work if supported by the endpoint.
Sendrpid	Whether Asterisk should send RPID (or PAI) info to the device. Usually should be enabled to the settings used by your device for CONNECTEDLINE( ) functionality to work if supported by the endpoint.
Prack	The PRACK request plays the same role as ACK, but for provisional responses.
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 messages 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 websockets (such as WebRTC) then you must select an option that includes WS
Avpf	Whether to Enable AVPF. Defaults to no. the WebRTC standard has selected AVPF as the audio video profile to use for media streams. This is not the default profile in use by Asterisk. As a result the following must enabled to use WebRTC.
Icesupport	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 WebRTC setups
Dtlsenable	Whether to Enable DTLS for this peer. Defaults to no.
Dtlsverify	Whether to verify that the provided peer cerificate is valid. Defaults to no.
Dtlssetup	Behavior on DTLS incoming and outgoing connections. Defaults to actpass.
Dtlscertfile	Path to certificate file to present
Dtlscafile	Path to cafile file to present
Dtlsprivatekey	Path to private key for certificate file.
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. Defaults to no.
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.
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.

Figure 2-2-11 Add CUSTOM Trunk interface

Table 2-2-8 Definition of 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	CallerID for calls placed out on this trunk Format: <#######>. You can also use the format: "hidden" <#######> to hide the CallerID 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
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.

Figure 2-2-12 Add incoming Route interface

Table 2-2-9 Definition of Add incoming Route

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 CallerID 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 suppoesd 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 CallerID 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 CallerID 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.
Call Recording	Controls or overrides the call recording behavior for calls coming into this DID. Allow will honor the normal downstream call recording settings. Record on Answer starts recording when the call would otherwise be recorded ignoring any settings that say otherwise. Record Immediately will start recording right away capturing ringing, announcements, MoH, etc. never will disallow recording regardless of downstream settings.
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 FreePBX/Asterisk system which numbers your phones are permitted to call and which Trunk to send the calls to.

Generally, a FreePBX/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 routing110 calls, and a route for ordinary calls.   A phone system might also have special routes for interoffice calls, international calls, and other special circumstances

Figure 2-2-13 Outbound Routes interface

Table2-2-10 Definition of Outbound Routes

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 'longdistance').
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 Password	Optional: A route can prompt users for a password before allowing calls to progress. This is 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 CallerID is supplied, the dialed number will only match the prefix + match pattern if the CallerID being transmitted matches this. When extensions make outbound calls, the CallerID will be their extension number and NOT their Outbound CID. The above special matching sequences can be used for CallerID 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 **callerid** 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.
Call Recording	Controls or overrides the call recording behavior for calls coming into this DID. Allow will honor the normal downstream call recording settings. Record on Answer starts recording when the call would otherwise be recorded ignoring any settings that say otherwise. Record Immediately will start recording right away capturing ringing announcements, MoH, etc. Never will disallow recording regardless of downstream settings.
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 Blacklist

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.

Figure 2-2-14 Blacklist interface

OpenVox Communication Co. , LTD.   URL: www.openvox.cn

Inbound blacklist   **Outbound blacklist**

Blacklist
Enable   ☑

The blacklist (based on CalleeID) is used for all outbound routes.

Country Codes

⊞·▢ North America
⊞·▢ South America
⊞·▢ Europe
⊞·▢ Asia and the Middle East
⊞·▢ Africa
⊞·▢ Oceania

Blacklist Manage

Add Blacklist Rule

[   ]   Add

Delete

☐	Continent	Country	Blacklist Rule

3.3.4 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.

Figure 2-2-15 Call flow control interface

Table 2-2-12 Definition of Call flow control

Item	Definition
Call Flow Toggle 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

3.3.5 Time Conditions

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

Figure 2-2-16 Time Conditions interface

Table 2-2-13 Definition of add Time Conditions

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.

3.3.6 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 p.m and end at 1:00 p.m. 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.

Figure 2-2-17 Time Groups interface

3.3.7 PIN Sets

FreePBX 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.

Figure 2-2-18 PIN Sets Interface

Table 2-2-14 Definition of add PIN Set

Item	Definition
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.

3.3.8 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.

Figure 2-2-19 Add FXO Channel interface、

Table 2-2-15 Definition of Add FXO Channel

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

3.3.9 AutoCLIP Route

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 prompting 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.

Figure 2-2-20 IVR interface
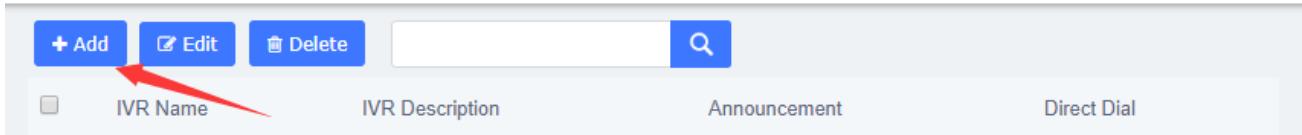
Table 2-2-16 Definition of add IVR

Item	Definition
Basic	
IVR Name	Name of this IVR
IVR Description	Description of this ivr
Announcement	Greeting to be played on entry to the IVR.
Direct Dial	Provides options for callers to direct dial an extension. Direct dialing can be: ● Completely disabled ● Enabled for all extensions on a system
Timeout	Amount of time to be considered a timeout
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	After playing Invalid Retry Recording the system will replay mail IVR Announcement

Announcement on Invalid	
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)
Invalid Destination	Destination to send the call to after Invalid recording is played.
Timeout Retries	Number of times to retry when no DTMF is heard and the IVR choice time out.
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)
Timeout Destination	Destination to send the call to after Timeout Recording is played.
Return to IVR after VM	If checked, upon exiting voicemail a caller will be returned to this IVR if they got a user voicemail
Return	Return to IVR
Delete	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 immediately before this.

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.

Figure 2-2-21 Queues interface

Table 2-2-17 Definition of add Queues

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 dialplan 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 CallerID 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 FreePBX as you are by-passing the normal dialplan. 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 FreePBX 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.
<td colspan="2" align="center">**General Queue Options**</td>
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.
Call Recording	Incoming calls to agents can be recorded. (saved to /var/spool/asterisk/monitor)
Recording Mode	Choose to 'Include Hold Time' in the recording so it starts as soon as they enter the queue, or to defer recording until 'After Answered' and the call is bridged with a queue member.
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 timeoutpriority. 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 ringtime 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 Wrapup 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 can not 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 more strict. Simply speaking, if no agent could answer the phone then don't admit them. If agents are infuse 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 FreePBX dialplan). ● **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 FreePBX dialplan) ● **Strict** Same as Yes but more strict. Simply speaking, if no agent could answer the phone then have them leave the queue. If agents are inuse 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 holdtime (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!
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 randomized (avoiding peak business hours, when possible). Please note: randomized schedules will be rescheduled (randomly) every time ANY backup is saved.
Penalty Members Limit	the phone(s) that are rung

Queues by default will sort callers with a first in, first out order. The Queue Priority module allows you weight some callers differently from others. By giving certain callers a higher priority, they are allowed to bypass all of the other callers with a lower priority to receive faster service. The default setting is for all callers to have a priority of zero. Callers with a higher number will be placed in front of priority zero callers. Queue priorities are often used when providing service level agreements (SLAs).

Figure 2-2-22 Queue Priorities interface

Table 2-2-18 Definition of add Queue Priorities

Item	Definition
Description	The descriptive name of this Queue Priority instance
Priority	The Queue Priority set

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.

Figure 2-2-23 Phonebook interface
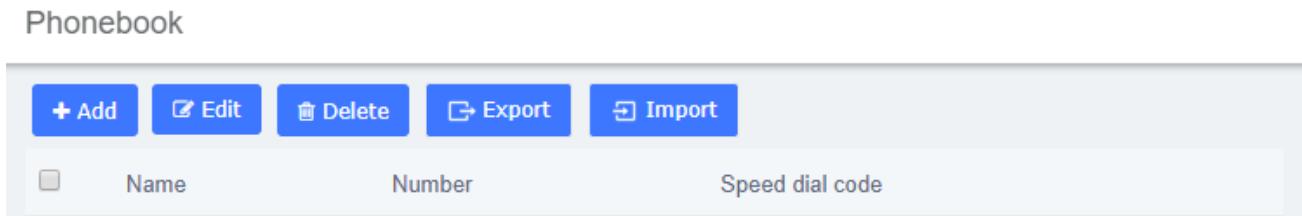
Navigate to **PBX > Call Control > Phonebook**, Add a speed dial number by using the following information.

Table 2-2-19 Definition of Phonebook

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

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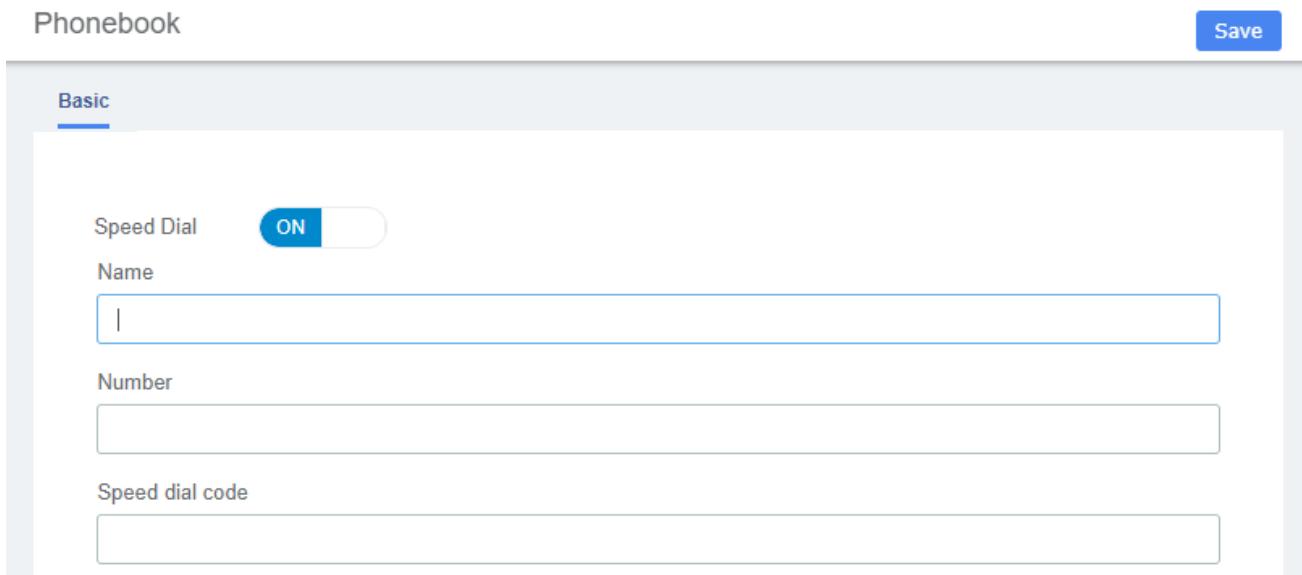
To dial this speed dial number, we dial *088, where *0 is to access the speed dial system's feature and 88 is the speed dial code we entered.

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; Speeddial

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

3.4.4 Wakeup Service

User can enable the Wakeup service and set the time and date, members, and receive the call reminder after the time.

Figure 2-2-24 Wakeup Service Interface

3.4.5 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.

Figure 2-2-24 DISA Interface

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.

Table 2-2-20 Definition of add DISA

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.

3.4.6 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.

Figure 2-2-25 Conference interface

Below a description of each parameter:

Table 2-2-22 Definition of add Conference

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 sends events on the Manager Interface identifying the channel that is talking. The talker will also be identified on the output of the meetme 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.

3.4.7 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**
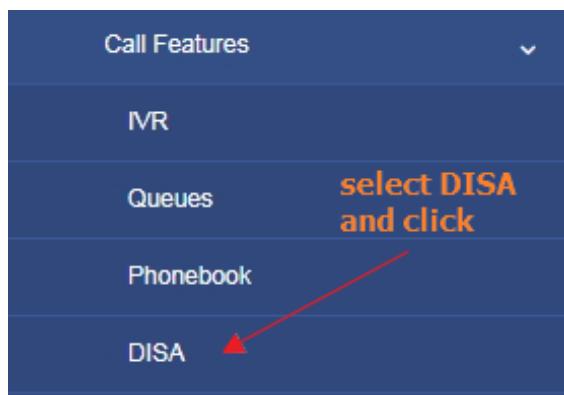
a. DISA name: MyMobile
b. Response Timeout:10
c. Digit Timeout:5

d. Caller ID:0400123456 (My Mobile Number)

e. Context: from-internal

Figure 2-2-28 Set on DISA
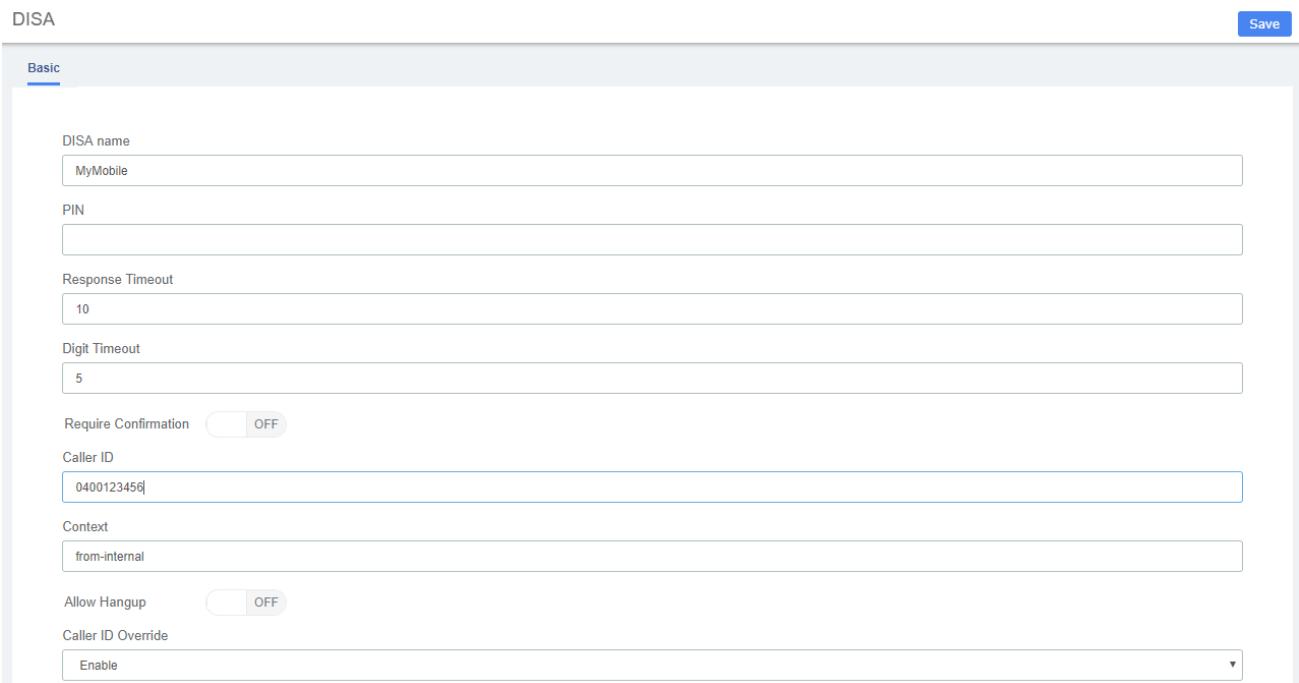
Click Save button.

2. Setup Callback
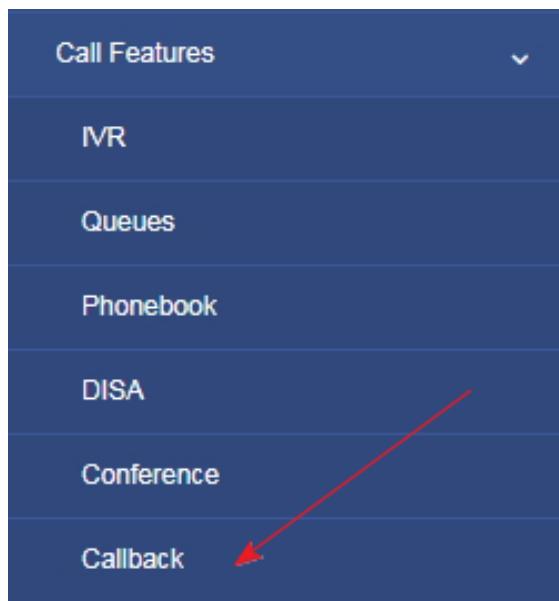
a. Callback Description: My Mobile

b. Callback Number: 0400123456 (My mobile Number)

c. Delay Before Callback:10

d. Destination after Callback: IVR – Residence (or Office IVR)

Figure 2-2-29 Callback interface

Click Save button

3. Inbound Routes

a. Description: Callback-MyMobile

b. DID Number:61247324100 (My DID number)

c. Caller ID Number: 0400123456 (My mobile Number)

d. Set Destination to: Callback – MyMobile

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 be then get DISA where I can make an external call at no cost to my Mobile.

Table 2-2-23 Definition of add 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 CallerID Number.
Delay Before Callback	Optional: Enter the number of seconds the system should wait before calling back.

3.4.8 Parking Lot

Figure 2-2-30 Parking Lot interface

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.

Table 2-2-24 Example usage of 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)

3.4.9 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.

Figure 2-2-31 Voicemail Blasting interface

Table 2-2-25 Definition of add VMBlast Group

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.

3.4.10 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.

Figure 2-2-32 Paging and Intercom interface
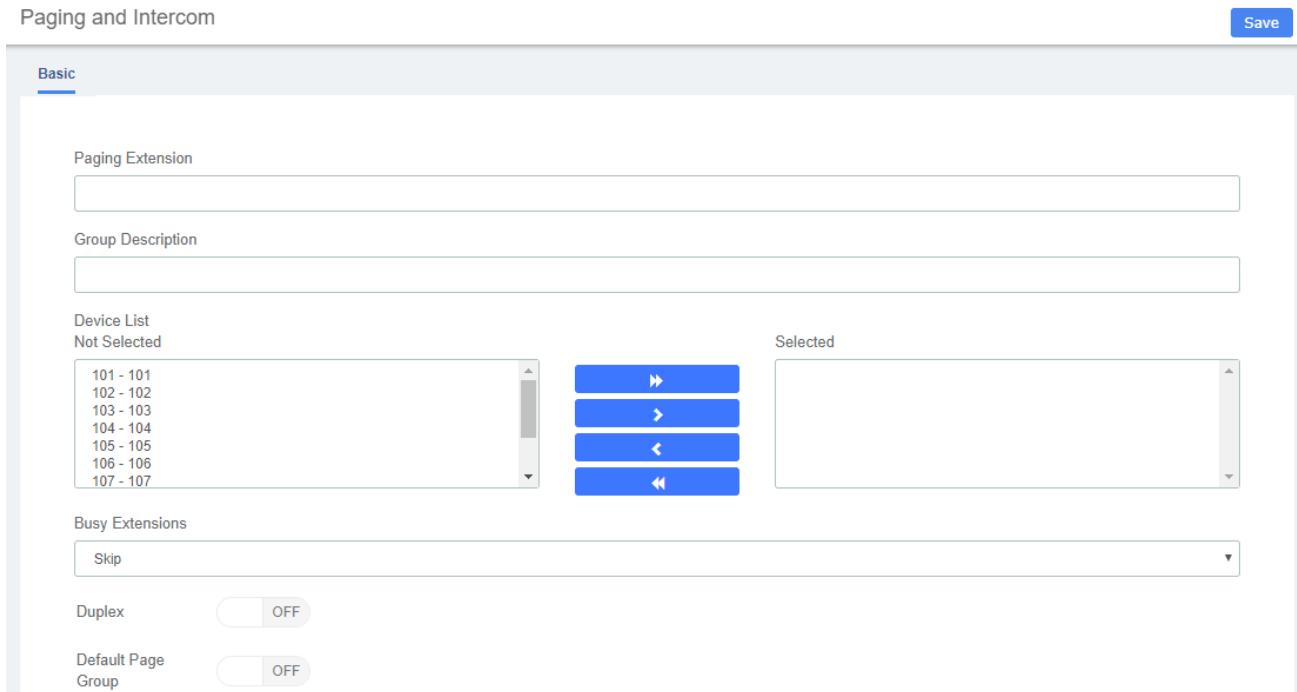
This module is for specific phones that are capable of Paging or Intercom. This section is for configuring group paging, intercom is configured through Feature Codes. Intercom must be enabled on a handset before it will allow incoming calls. It is possible to restrict incoming intercom calls to specific 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.

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.

Figure 2-2-33 Languages interface

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 it's 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.

Table 2-2-26 Definition of add Language

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.

3.5.2 System Recordings

The System Recordings module is used to record or upload messages that can then 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 make a selection. 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.

Figure 2-2-34 System Recordings interface

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System Recordings

Record or upload

— If you wish to make and verify recordings from your phone, please enter your extension number here:

| Extensions ▼ | ==choose== ▼ | Go |

— Alternatively, upload a recording in any supported asterisk format. Note that if you're using .wav, (eg, recorded with Microsoft Recorder) the file **must** be PCM Encoded, 16 Bits, at 8000Hz

| Choose File | No file chosen | Upload |

Name

Name this Recording

— Click "SAVE" when you are satisfied with your recording

Save

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.

Figure 2-2-35 Announcements interface

Announcement   Save

Basic

Name

test

Recording

test   ▼

Repeat button

Disable   ▼

Allow Skip   [ON]

Return to IVR   [OFF]

Don't Answer Channel   [OFF]

Destination

Custom Destinations   ▼

Query Current IP   ▼

Table 2-2-27 Definition of Announcements

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 IVR	If the announcement came from an IVR and this box is checked, the destination below will 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.

3.5.4 Route Congestion Messages

Figure 2-2-36 Route Congestion Messages interface

Table 2-2-28 Definition of Route Congestion Messages

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." Hangupcause 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."Hangupcause is 28

3.5.5 Music On Hold

The volume adjustment is a linear value. Since loudness is logarithmic, the linear lever will be less of an adjustment. You should test out the installed music to assure it is at the correct volume. This feature will convert MP3 files to WAV files. If you do not have mpg123 installed, you can set the parameter: Convert Music Files to WAV to false in Advanced Settings.

Figure 2-2-37 Music on Hold Interface

3.6 Settings

3.6.1 Global Settings

Global Settings

Basic   Device Settings   Dialplan and Operational   Features Settings

Asterisk Manager

Asterisk Manager Password   111111

Asterisk Manager User   admin

System Setup

Aggresively Check for Duplicate Extensions   ○ True   ● False

User & Devices Mode   extensions

Call Recording Format   wav

Convert Music Files to WAV   ● True   ○ False

Voice Prompts

Default language   en

Refresh Page

3.6.2 Analog Settings

Figure 2- 2-38 general Configuration

Table 2-2-29 Instruction of General

Options	Definition
Tone duration	How long generated tones (DTMF and MF) will be played on the channel. (in milliseconds)
Codec	Set the global encoding: mulaw, alaw.
Impedance	Configuration for impedance.
Flash/Wink	Turn on/off Flash/wink.
Min flash time	Min flash time.(in milliseconds).
Max flash time	Max flash time.(in milliseconds).
"#"as Ending Dial Key	Turn on/off Ending Dial Key.
Hang up on polarity switch	Turn on/off Hangup on polarity switch

Figure 2-2-39 Hardware gain

Hardware gain

FXS Rx gain

| 0 | ▼ |

FXS Tx gain

| 0 | ▼ |

FXO Rx gain

| 0 |

FXO Tx gain

| 0 |

Table 2-2-30 Instruction of 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.

Figure 2-2-40 Fax

Fax

Maximum Transmission Rate

| 14400 | ▼ |

Minimum Transmission Rate

| 14400 | ▼ |

Ecm   ON

Table 2-2-31 Definition of 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.

Figure 2-2-41 Send Caller ID

Send CallerID

The pattern of sending CID

send CID after first ring ▼

Waiting time before sending CID

100

Send polarity reversal(DTMF Only)   OFF

Start code(DTMF Only)

Stop code(DTMF Only)

Table 2-2-32 Instruction of 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 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.

Figure 2-2-42 CallerID Detection

Table 2-2-33 Instruction of CallerID detect

Options	Definition
Use Callerid	Turn on/off callerid detect function
Hide Callerid	Turn on/off callerid detect function

Figure 2-2-43 Country

Country

Country

| China | ▾ |

Dial tone

| 450 |

Busy tone

| 450/350,0/350 |

Congestion tone

| 450/700,0/700 |

Record tone

| 950/400,0/10000 |

Ring cadence

| 1000,4000 |

Ring tone

| 450/1000,0/4000 |

Call waiting tone

| 450/400,0/4000 |

Dial recall tone

| 450 |

Info tone

| 450/100,0/100,450/100,0/100,450/100,0/100,450/400,0/400 |

Stutter tone

| 450+425 |

Table 2-2-34 Definition of 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.)
Stutter tone	

Figure 2-2-44 Silence detect

Silence detect

Silence detect   OFF

Silence threshold

250

Silence length

300

Silence framesize

1024

Table 2-2-35 Definition of 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.

Figure 2-2-45 Special tone

Special tone

Custom Busy
Tone detect   OFF

Busy Tone count: 3

Busy Tone pattern: 450/350,0/350

Add New Busy Tone Pattern

Table 2-2-36 Instruction of 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

3.6.3 RTP Settings

3.6.4 IAX2 Settings

Figure 2-2-52 Audio Codecs

Table 2-2-43 Audio Codecs

Options	Definition
Codec Priority	Asterisk: codecpriority. 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)reqonly 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 effect 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.

Figure 2-2-53 Jitter Buffer Settings

Table 2-2-44 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: jitterbuffer. 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.

Figure 2-2-54 Advanced General Settings

Table 2-2-45 Advanced General Settings

Options	Definition

Language	Default Language for a channel, Asterisk: language
Bind Address	Asterisk: bindaddr. 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: bindport. 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: delayreject. 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 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.

Figure 2-2-55 Feature code admin interface

3.6.6 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 FreePBX 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 not longer provided.

Figure 2-2-56 Misc Destinations interface
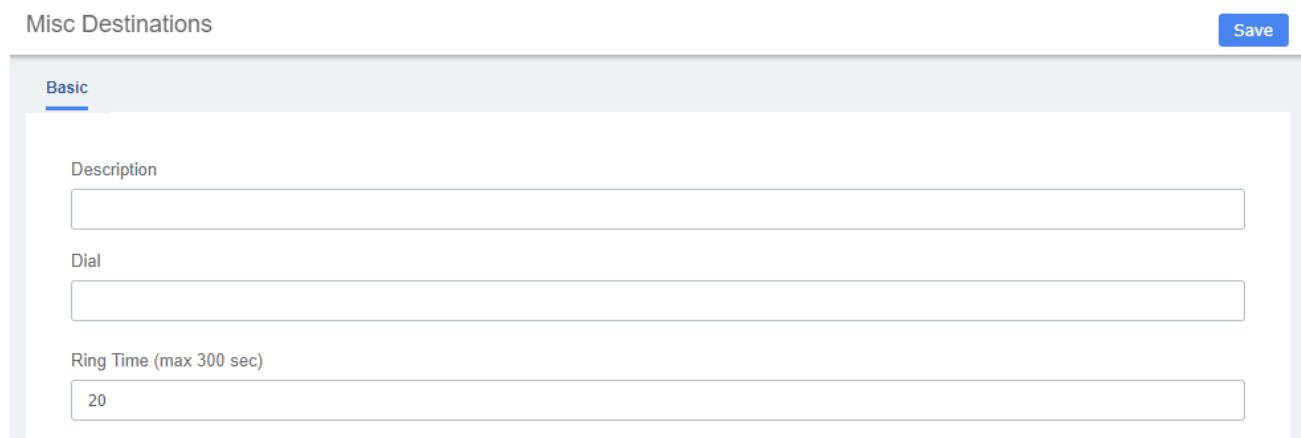
Table 2-2-46 Definition of add Misc Destination

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.

3.6.7 PJSIP Settings

3.6.8 AMI

Figure 2-2-58 Add AMI User interface

OpenVox Communication Co. , LTD.   URL: www.openvox.cn

Figure 2-2-58 AMI Settings interface

3.7 Recording

3.7.1 Call Recordings

The option "Calls Recordings" of the Menu "Recordings" in UC series 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.

Figure 2-2-59 Calls Recordings interface
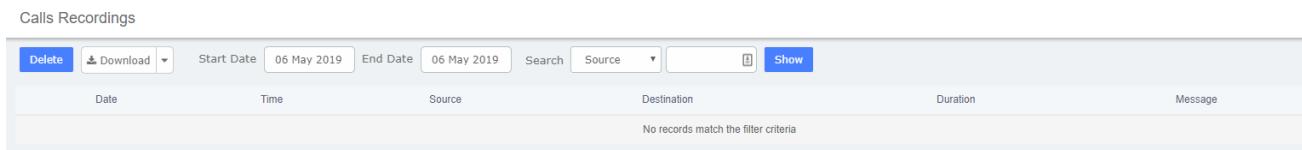
3.7.2 VoiceMails

The option "Voicemail" of the Menu "Recordings" in UC series lets us view a list with details of the voicemails for the extension of the logged user.

Figure 2-2-60 Voicemails interface

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

Table 2-2-49 Definition of Show 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.

Figure 2-2-61 Voicemails Admin interface

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3.8 Tools

3.8.1 Operator Panel

The option "Operator Panel" of the menu "Tools" in UC series 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 extension to a conference room, or a queue. The receptionist can also see the busy extensions, the elapsed time and the caller ID.

Figure 2-2-62 Operator Panel interface
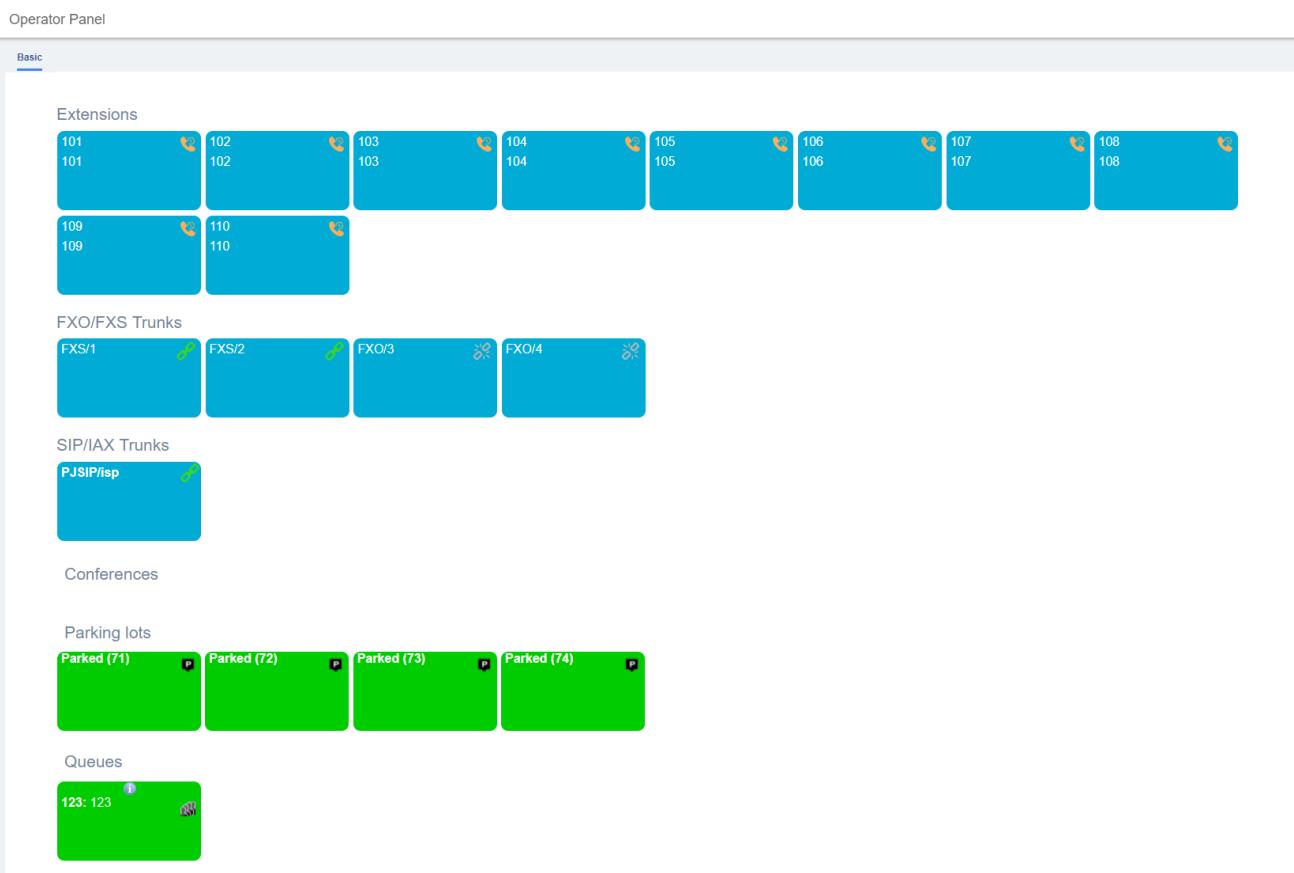
3.8.2 WebRTC

WebRTC is a new feature. You can use any web browser that supports WebRTC to register an

extension number to your UC series system without any plugins.

To register the extensions used for WebRTC please follow the steps below:

Step 1:

Create a SIP Extension

To create a extension, navigate to **PBX > Extensions > Extensions**. Click on "Add" button to add a new sip extension.

Step 2:

Configure the SIP Extension and Settings

As you can see, these extensions use different protocols for signaling and media (WS/WSS) and they are not ordinary SIP extensions that can use IP phones or softphone to register so must be treated differently. There are some options that are supposed to be changed of extensions as follows:

Transport

| WSS Only | ▼ |

Avpf

| Yes | ▼ |

Icesupport

| Yes | ▼ |

Dtlsenable

| Yes | ▼ |

Encryption

| Yes (SRTP only) | ▼ |

Besides, some configurations on the sip settings page are as follows:

IP Configuration

| Public IP | ▼ |

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Step 3:
Register a Web Extension
After completing the process above you can access the WebRTC extension register interface.
Navigate to **PBX > Tools > WebRTC**, you will see the web extension register interface.
Please complete the register credentials as below:

(Note:172.16.208.33 should be your IPPBX IP address)
Next, press Login and the web extension will be registered and is ready for phone calls just like any other standard extension.
WebRTC can even be adapted to an enterprise website which can help an enterprise serve their

customers with direct voice communication via their website. For more advanced WebRTC settings please refer to the WebRTC manual.

3.8.3 Asterisk-Cli

The option "Asterisk-Cli" of the Menu "Tools" in UC series lets us execute Asterisk commands.

Figure 2-2-64 Asterisk-Cli interface
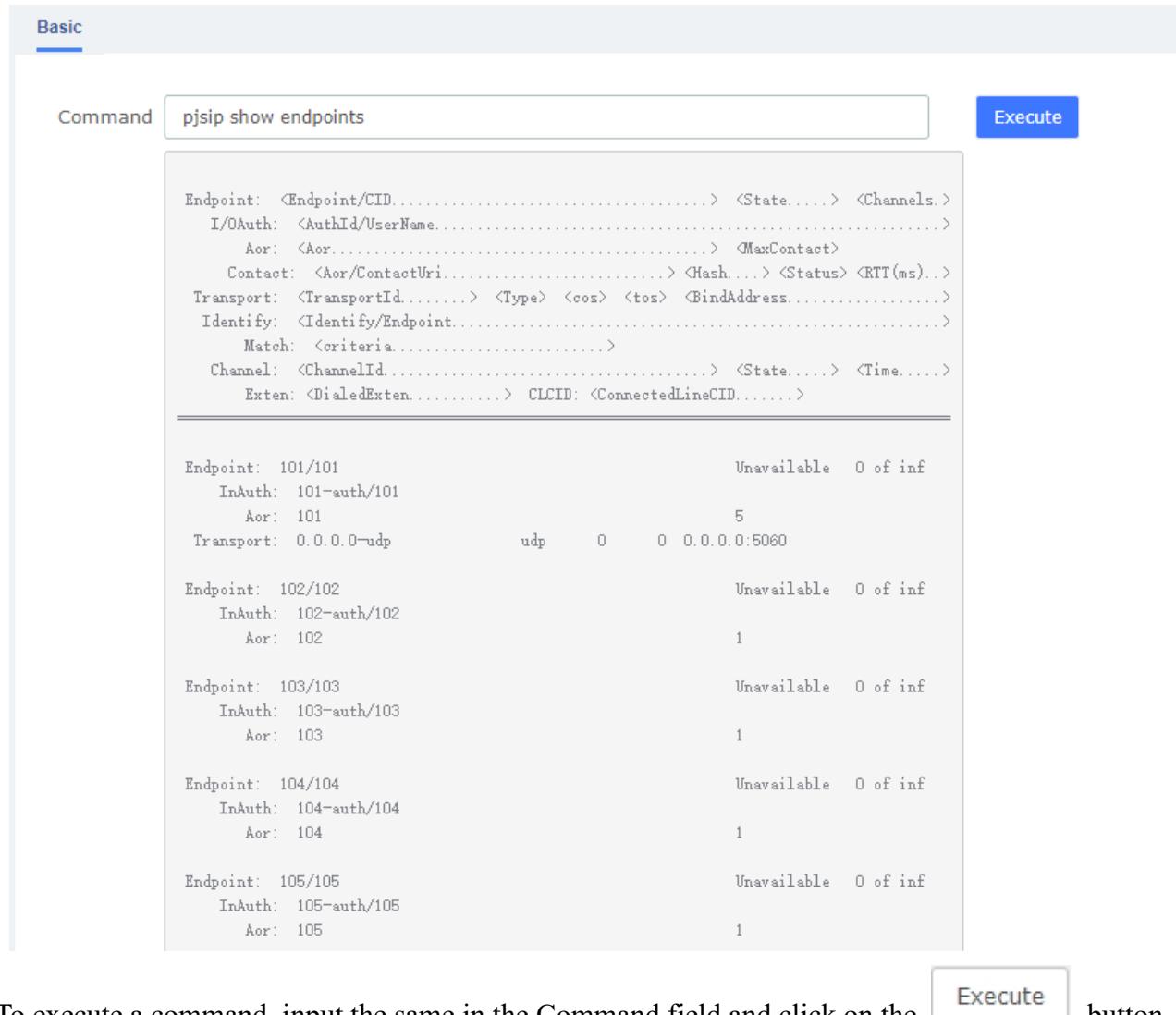
To execute a command, input the same in the Command field and click on the [Execute] button.

3.8.4 Asterisk File Editor(developer mode)

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

Figure 2-2-65 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.

Figure 2-2-66 Editing a file interface

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/.

Figure 2-2-67 Create a file interface
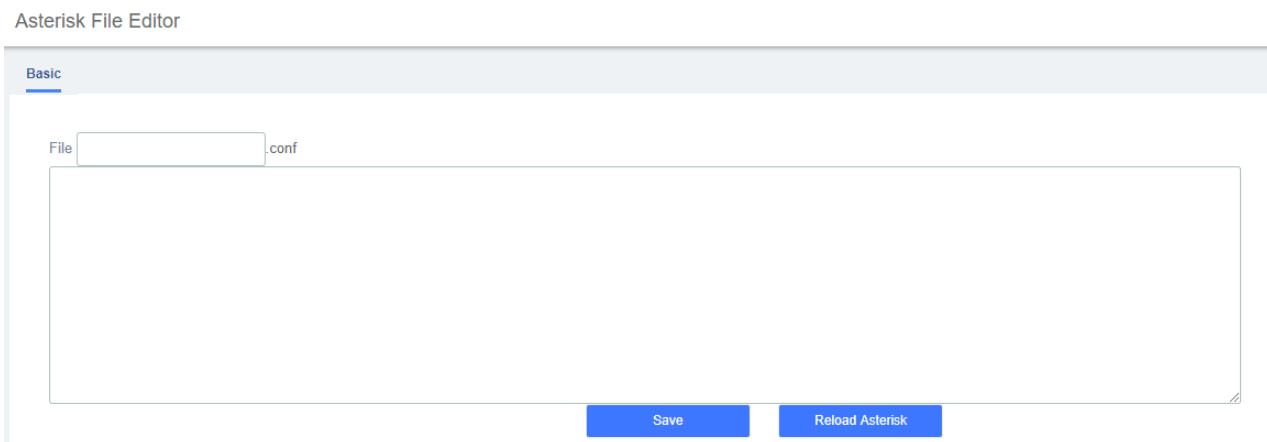
3.8.5 AI TTS

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.

Figure 2-2-68 Text to Wav interface

4 Fax

4.1 Virtual Fax

4.1.1 Virtual Fax List

The option "Virtual Fax List" of the Menu "FAX" in UC series lets us verify the list of all the virtual faxes, including the status of each one.

Figure 2-4-1 Virtual Fax List interface

Virtual Fax List

Virtual Fax Name	Fax Extension	Secret	Associated Email	Caller ID Name	Caller ID Number	Status
1001	1001	pbx1001	1001@openvox.cn	1001	1001	Running and idle on ttyIAX3

Also, clicking on the virtual fax's name displays its information:

New Virtual Fax   Edit

Basic

Virtual Fax Name

1001

Fax Extension (IAX)

1001

Associated Email

1001@openvox.cn

Secret (IAX)

pbx1001

Caller ID Name

1001

Country Code

086

Caller ID Number

1001

Area Code

0755

Delete

Here you can Edit and Delete the Virtual Fax.

4.1.2 New Virtual Fax

The option "New Virtual Fax" from the Menu "FAX" in UC series lets us create a new virtual fax. You should have previously created an IAX extension in "PBX => Extensions =>Add IAX2 Extension".

Figure 2-4-2 New Virtual Fax

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.

4.1.3 Send Fax

The option "Send Fax" of the menu "Fax" in UC series allows sending faxes to one or more numbers.

Here you can enter the text you want to send and click on [Send] button.

Figure 2-4-3 Send fax with text information

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

Figure 2-4-4 Send fax with File Upload

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4.1.4 Fax Queue

The option "Fax Queue" from the Menu "FAX" in UC series 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.

Figure 2-4-5 Fax Queue interface

Fax Queue

Job ID	Priority	Destination	Pages	Retries	Status
No records match the filter criteria					

[Cancel job]

4.2 Fax Master

The option "Fax Master" of the Menu "FAX" in UC series 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.

Figure 2-4-6 Fax Master Interface

Fax Master [Save]

Basic

Write the email address which will receive the notifications of received messages, errors and activity summary of the Fax Server

user@openvox.cn

4.3 Fax Clients

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

Figure 2-4-7 Fax Client interface

Fax Clients [Save]

Basic

localhost
127.0.0.1

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.

Figure 2-4-8 Fax Viewer interface
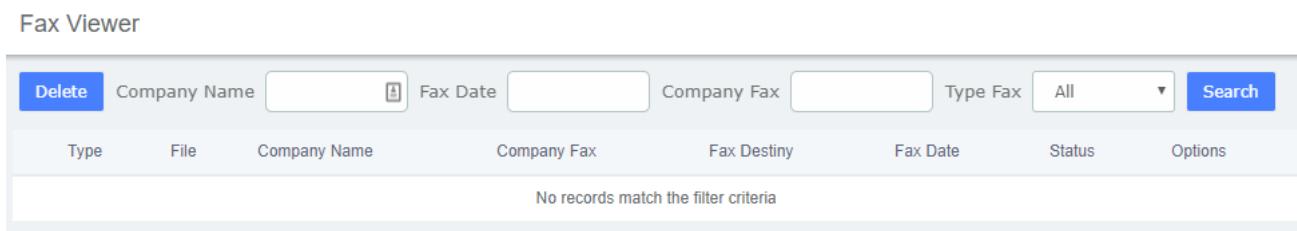
Type	File	Company Name	Company Fax	Fax Destiny	Fax Date	Status	Options
			No records match the filter criteria				

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

5 Reports

5.1 CDR Report

The option "CDR Reports" of the Menu "Reports" in UC series 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.

Figure 2-5-1 CDR Report interface
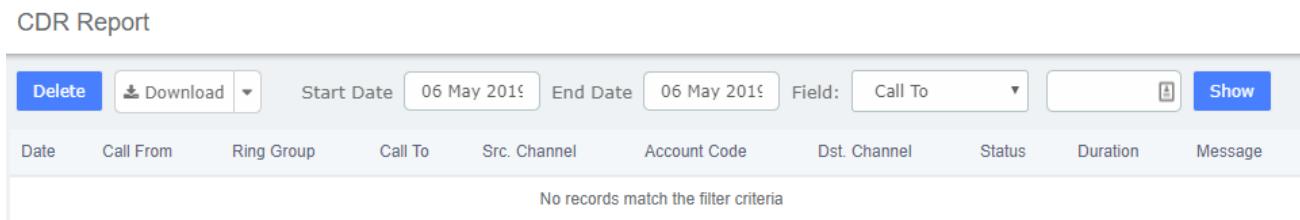
5.2 Channels Usage

The option "Channels Usage" of the menu "Reports" in UC series lets us view graphically the number of simultaneous calls for each channel.

Figure 2-5-2 Channel usage interface
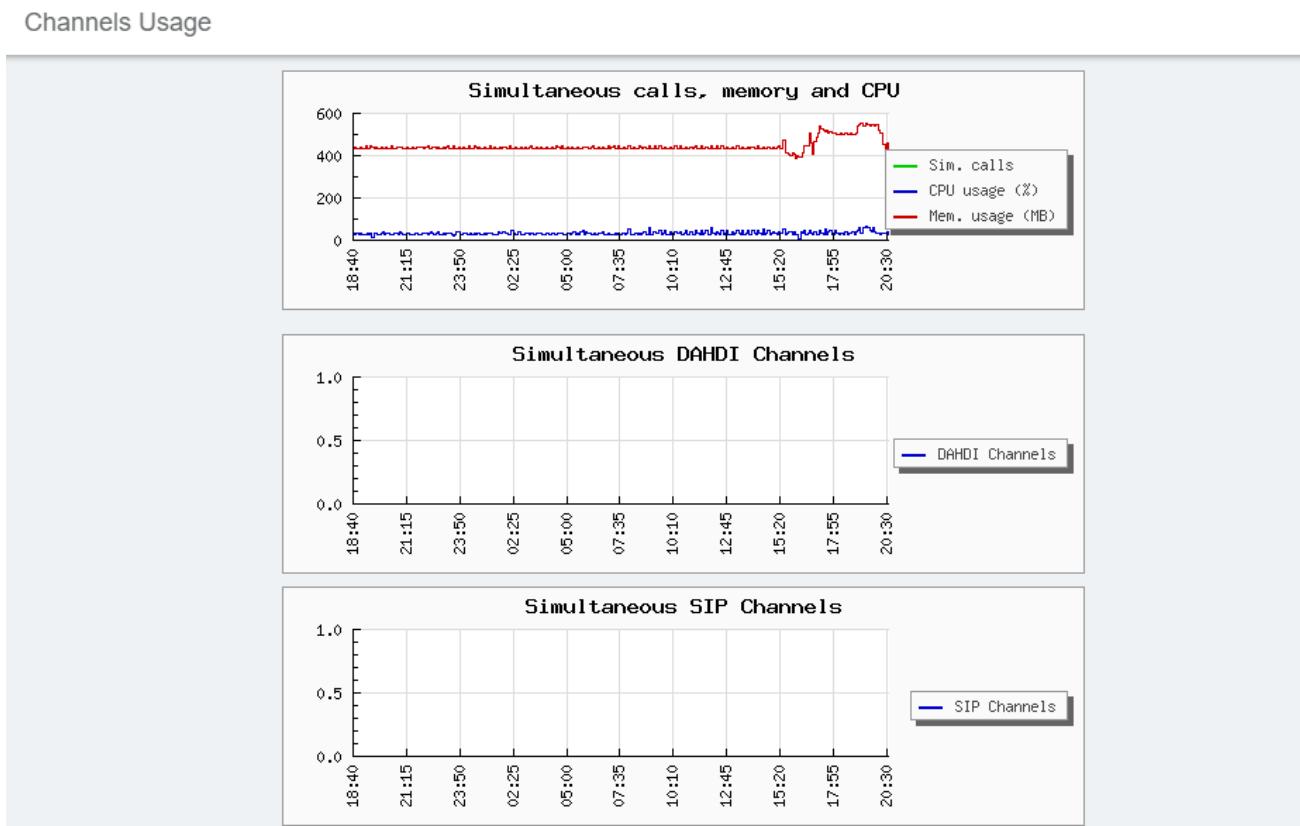
5.3 Billing

5.3.1 Destination Distribution

The "Destination Distribution" option of the "Billing" Menu in UC series lets us view graphically the distribution of the outgoing calls grouped by rate. The graph will change depending on the values of the filter:

Figure 2-5-3 Destination Distribution interface

Billing / Destination Distribution

☑ Destination Distribution

| Start Date | 27 Oct 2017 | End Date | 27 Oct 2017 | Criteria | Distribution by Time ▼ | Save |

no message
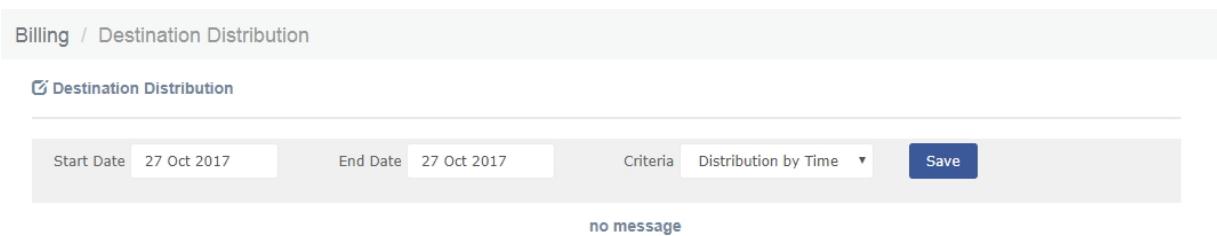
Table 2-5-1 Definition of Destination Distribution

Name	Description
Start Date	The start date for calls to be selected.
End Date	The end date for calls to be selected.
Criteria	Criteria for distribution: Distribution by Time, Distribution by Number of Calls, Distribution by Cost.

5.3.2 Rates

The option "Rates" of the menu "Reports" allows creating new rates and editing existing ones for billing.

Figure 2-5-4 Rate interface

☰ Billing Rates

| Create New Rate | Import File |

Prefix	Name	Rate	Rate Offset	Hidden Digits	Trunk	Creation Date	View
*	Default	0.5	1.0	0	*	2017-10-27 11:33:27	View
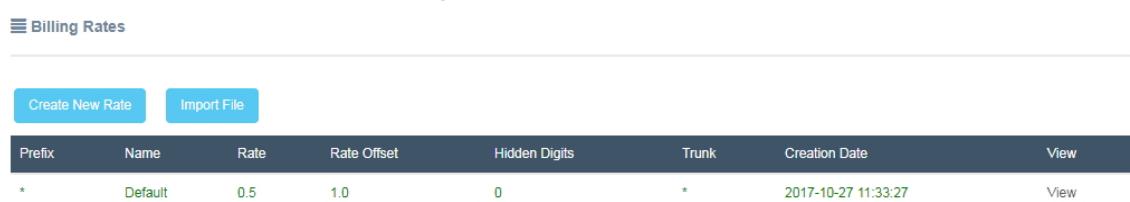
To edit or delete a rate, click on the "View" link from the list

Figure 2-5-5 View Rate interface

Create a new Rate

You can create a new rate by clicking on [Create New Rate] button.

Figure 2-5-6 Create a new Rate interface

Table2-5-2 Definition of Create a new Rate

Item	Description
Prefix	All the numbers that begin with this prefix will apply to this rate.
Name	This is the name to identify the rate.

Rate (by min)	This is the rate that will be apply to every single minute of consumption.
Rate offset	This is the rate assigned for the connection.
Hidden Digits	This indicates the amount of digits you want to hide in the destination number.
Trunk	Select the trunk that will apply for the rate. Make sure the trunk you want to use is enabled. To check this, go to "Billing Setup" module.

Click button **Import File** to import a file into the system.

Figure 2-5-7 Import file interface
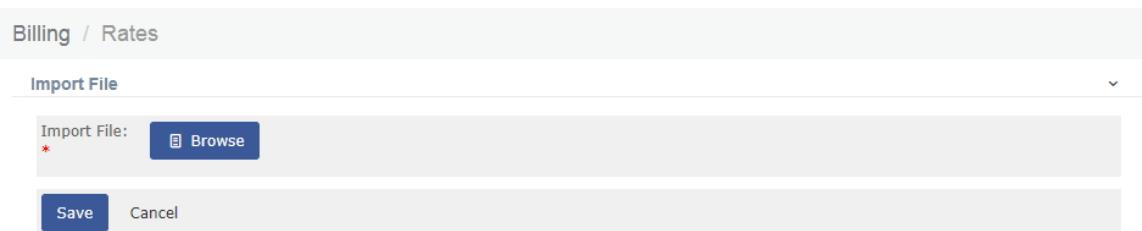
Below a description of each parameter:

5.3.3 Billing Report

The option "Billing Report" in UC series shows a complete report of calls according to a rate established in "Billing Rates". You can filter the results by date, rate applied, duration and so on. Also you can download this report in different formats such as CSV, XLS and PDF.

Figure 2-5-8 Billing report interface
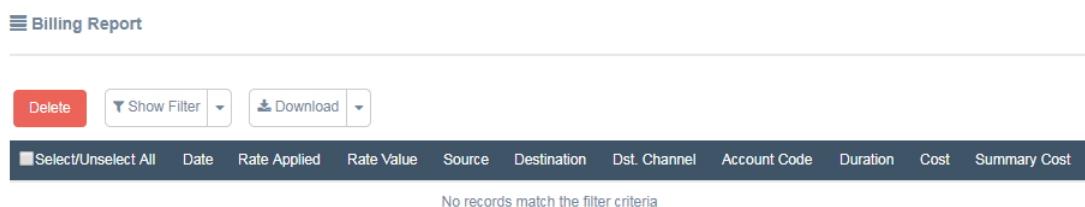
The fields in this report are:

Name	Description
Date	Initial date of call
Rate Applied	Name of Rate applied
Rate Value	Rate value by minutes

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Source	Number or source
Destination	Destination Number
Dst. Channel	Channel Destination (Example: DAHDI/1)
Account Code	Code of account extension
Duration	Duration in seconds of calls
Cost	Cost of call
Summary Cost	Sum of all calls by cost field

5.3.4 Billing Setup

The option "Billing Setup" of the Menu "Billing" in UC series lets us determine the cost per minute of the connection for the route by omission, and also determine which of the trunks will be used for the billing process.

Figure 2-5-9 Billing setup interface

The list shows all of the registered trunks; you should select the ones that will be used for billing and click on the "Billing Capable" button.

5.4 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.

Figure 2-5-10 Graphic report interface

To see the information of a specific extension, select "Extension (Number)" and then click on the link "Here". In the pop-up window, choose the phone number and then click on "Show button".

Figure 2-5-11 Specific extension info
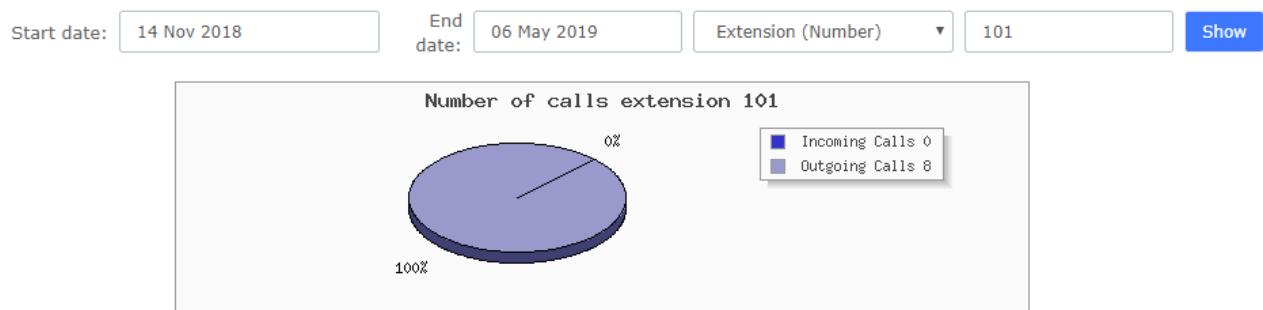
It is possible to generate a graphic of Number of Calls vs. Queues. To do this just select "Queue" from the dropdown menu.

Figure 2-5-12 Queue record
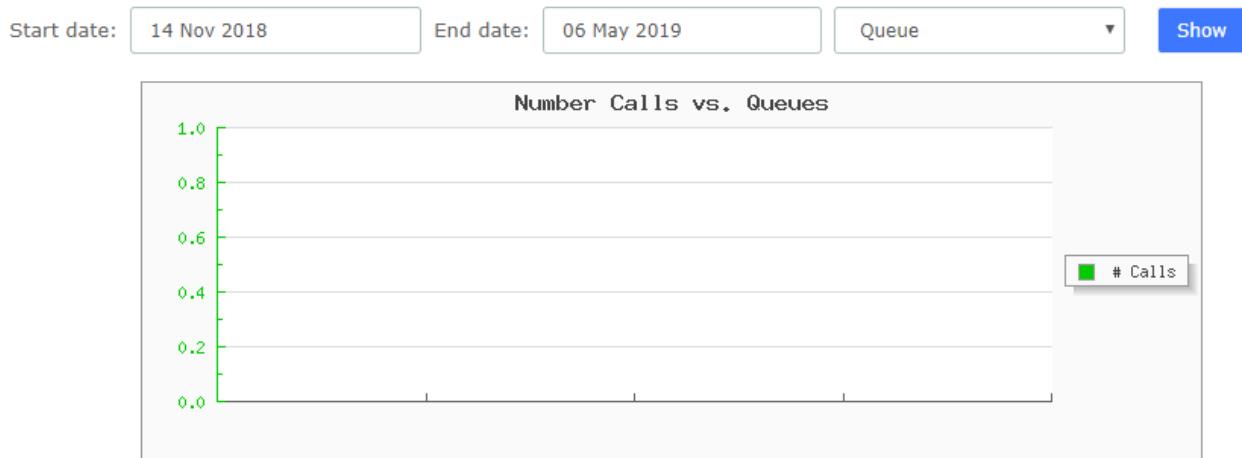
5.5 Summary

The option "Summary" of the menu "Reports" in UC series 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.

Figure 2-5-13 Summary interface

Summary

| Start date | | End date | | Filter by | Extension ▼ | | 🔢 | Show | |

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

Click on "View" to see more information of an extension.

Figure 2-5-14 View extension info

No data to display

Top 10 (Incoming) ext 1001

Top 10 (Outgoing) ext 1001

0%

100%

■ 1002 (2 calls)
■ Other calls (0 calls)
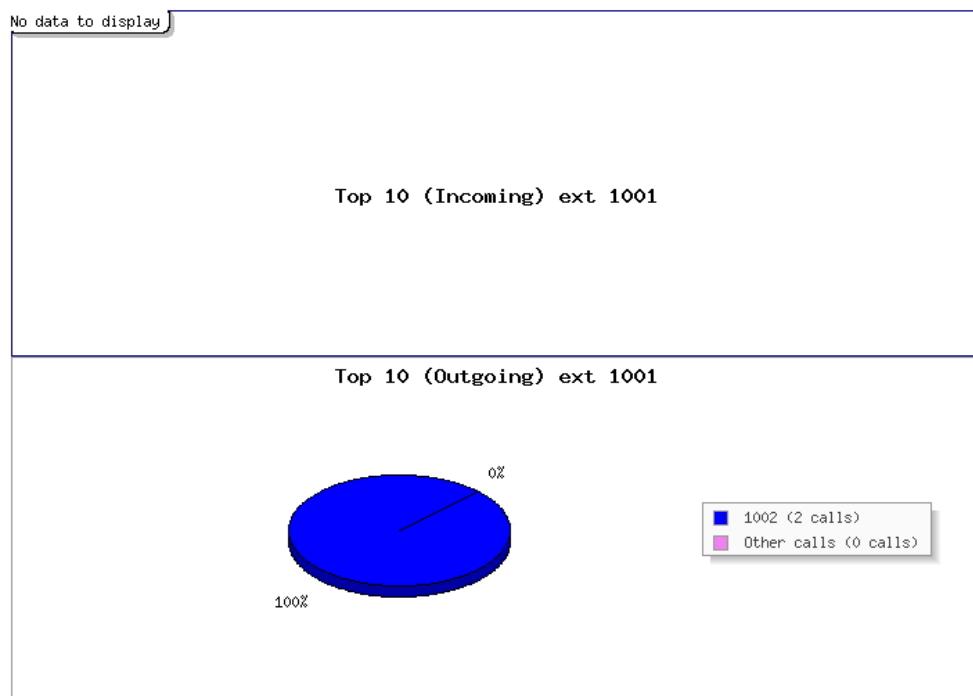
5.6 Missed Calls

The option "Missed Calls" of the menu "Reports" in UC series 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:
    o *Source:* Number that made the call.
    o *Destination:* Number that received the call.

Figure 2-5-15 Missed calls interface

6. AddsOn

6.1 A2billing

A2billing is a full-featured, flexible VoIP billing system and terminal platform that provides the fastest and lowest cost service for VoIP billing customers on the market today. This feature has been added to the new UC, and users who need it can enter the account password to log in.
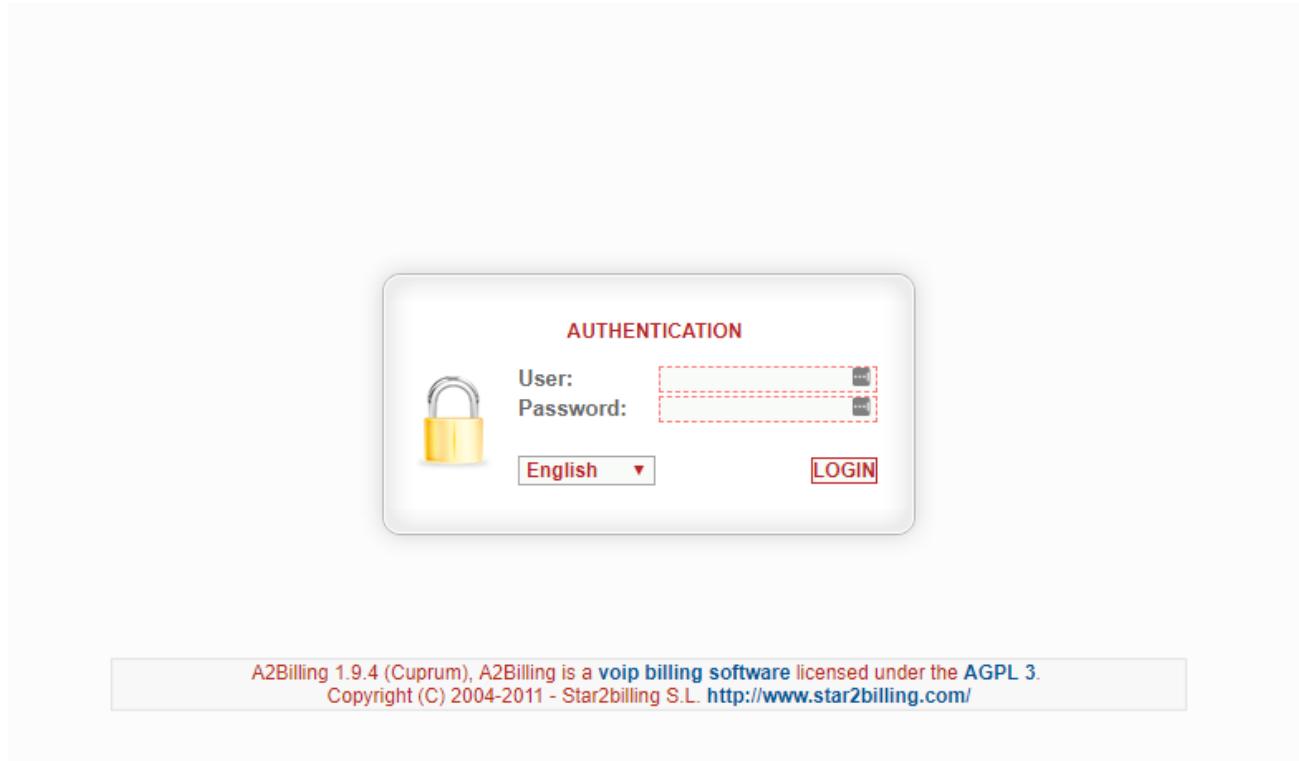
6.2 Video Conference

Users can create video conferences in UC501, allowing multiple people to participate at the same time.

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7 Logs

The option "Logs" of the "Reports" module allows visualizing the content of logs for monitoring the events. You can filter the results by date or strings that are in the content of the logs.

7.1 Logs Settings

Figure 2-6-1 Logs Settings interface

7.2 System Logs

Figure 2-6-2 System Logs interface

7.3 Asterisk Logs

Figure 2-6-3 Asterisk logs interface

7.4 DAHDI Logs

Figure 2-6-6 DAHDI logs interface
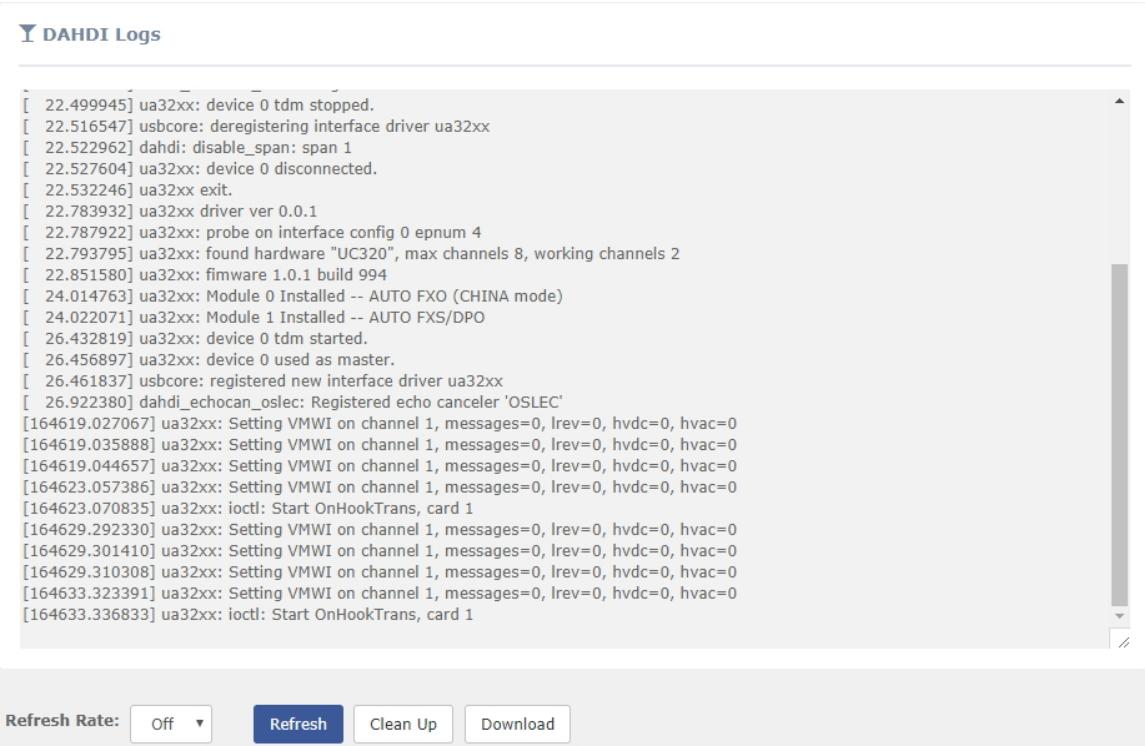
```
Y DAHDI Logs

[  22.499945] ua32xx: device 0 tdm stopped.
[  22.516547] usbcore: deregistering interface driver ua32xx
[  22.522962] dahdi: disable_span: span 1
[  22.527604] ua32xx: device 0 disconnected.
[  22.532246] ua32xx exit.
[  22.783932] ua32xx driver ver 0.0.1
[  22.787922] ua32xx: probe on interface config 0 epnum 4
[  22.793795] ua32xx: found hardware "UC320", max channels 8, working channels 2
[  22.851580] ua32xx: fimware 1.0.1 build 994
[  24.014763] ua32xx: Module 0 Installed -- AUTO FXO (CHINA mode)
[  24.022071] ua32xx: Module 1 Installed -- AUTO FXS/DPO
[  26.432819] ua32xx: device 0 tdm started.
[  26.456897] ua32xx: device 0 used as master.
[  26.461837] usbcore: registered new interface driver ua32xx
[  26.922380] dahdi_echocan_oslec: Registered echo canceler 'OSLEC'
[164619.027067] ua32xx: Setting VMWI on channel 1, messages=0, lrev=0, hvdc=0, hvac=0
[164619.035888] ua32xx: Setting VMWI on channel 1, messages=0, lrev=0, hvdc=0, hvac=0
[164619.044657] ua32xx: Setting VMWI on channel 1, messages=0, lrev=0, hvdc=0, hvac=0
[164623.057386] ua32xx: Setting VMWI on channel 1, messages=0, lrev=0, hvdc=0, hvac=0
[164623.070835] ua32xx: ioctl: Start OnHookTrans, card 1
[164629.292330] ua32xx: Setting VMWI on channel 1, messages=0, lrev=0, hvdc=0, hvac=0
[164629.301410] ua32xx: Setting VMWI on channel 1, messages=0, lrev=0, hvdc=0, hvac=0
[164629.310308] ua32xx: Setting VMWI on channel 1, messages=0, lrev=0, hvdc=0, hvac=0
[164633.323391] ua32xx: Setting VMWI on channel 1, messages=0, lrev=0, hvdc=0, hvac=0
[164633.336833] ua32xx: ioctl: Start OnHookTrans, card 1

Refresh Rate:  Off ▼    Refresh    Clean Up    Download
```

7.5 FXO Monitor Logs

Figure 2-6-7 FXO Monitor logs interface
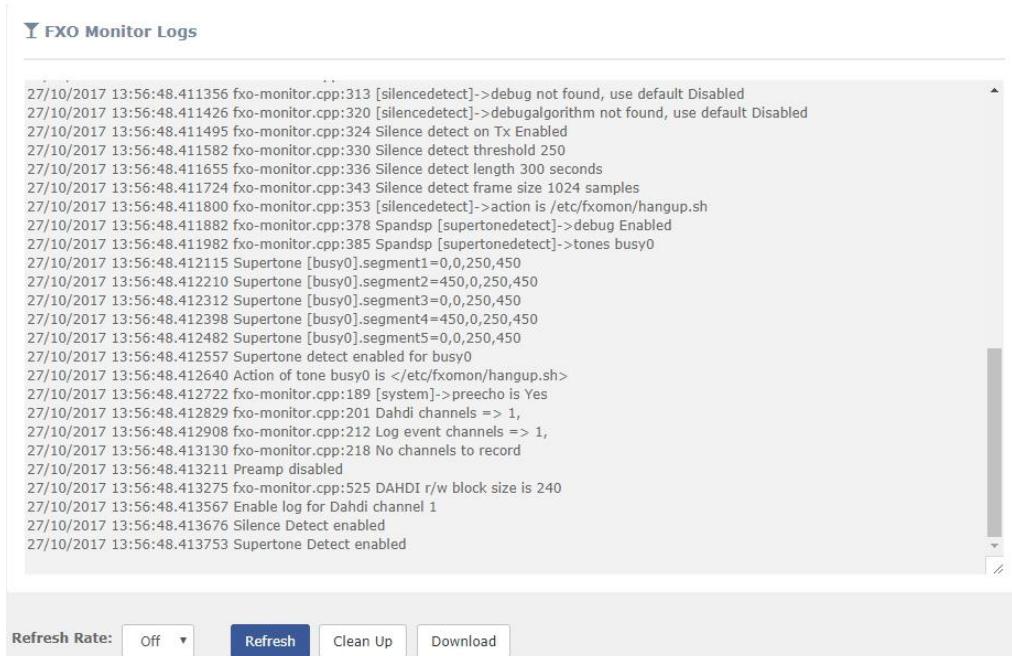
```
Y FXO Monitor Logs

27/10/2017 13:56:48.411356 fxo-monitor.cpp:313 [silencedetect]->debug not found, use default Disabled
27/10/2017 13:56:48.411426 fxo-monitor.cpp:320 [silencedetect]->debugalgorithm not found, use default Disabled
27/10/2017 13:56:48.411495 fxo-monitor.cpp:324 Silence detect on Tx Enabled
27/10/2017 13:56:48.411582 fxo-monitor.cpp:330 Silence detect threshold 250
27/10/2017 13:56:48.411655 fxo-monitor.cpp:336 Silence detect length 300 seconds
27/10/2017 13:56:48.411724 fxo-monitor.cpp:343 Silence detect frame size 1024 samples
27/10/2017 13:56:48.411800 fxo-monitor.cpp:353 [silencedetect]->action is /etc/fxomon/hangup.sh
27/10/2017 13:56:48.411882 fxo-monitor.cpp:378 Spandsp [supertonedetect]->debug Enabled
27/10/2017 13:56:48.411982 fxo-monitor.cpp:385 Spandsp [supertonedetect]->tones busy0
27/10/2017 13:56:48.412115 Supertone [busy0].segment1=0,0,250,450
27/10/2017 13:56:48.412210 Supertone [busy0].segment2=450,0,250,450
27/10/2017 13:56:48.412312 Supertone [busy0].segment3=0,0,250,450
27/10/2017 13:56:48.412398 Supertone [busy0].segment4=450,0,250,450
27/10/2017 13:56:48.412482 Supertone [busy0].segment5=0,0,250,450
27/10/2017 13:56:48.412557 Supertone detect enabled for busy0
27/10/2017 13:56:48.412640 Action of tone busy0 is </etc/fxomon/hangup.sh>
27/10/2017 13:56:48.412722 fxo-monitor.cpp:189 [system]->preecho is Yes
27/10/2017 13:56:48.412829 fxo-monitor.cpp:201 Dahdi channels => 1,
27/10/2017 13:56:48.412908 fxo-monitor.cpp:212 Log event channels => 1,
27/10/2017 13:56:48.413130 fxo-monitor.cpp:218 No channels to record
27/10/2017 13:56:48.413211 Preamp disabled
27/10/2017 13:56:48.413275 fxo-monitor.cpp:525 DAHDI r/w block size is 240
27/10/2017 13:56:48.413567 Enable log for Dahdi channel 1
27/10/2017 13:56:48.413676 Silence Detect enabled
27/10/2017 13:56:48.413753 Supertone Detect enabled

Refresh Rate:  Off ▼    Refresh    Clean Up    Download
```

7.6 VPN Logs

Figure 2-6-8 VPN logs interface

VPN Logs

Refresh Rate: Off ▼   Refresh   Clean Up   Download   openvpn client ▼