

# OpenVox

OpenVox Communication Co., Ltd



## UC300 Series Quick Start Guide

Version 1.0



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## Contents

1. Appearance .....	5
2. Description of account and ports .....	5
3. Device connection .....	6
4. Access to device .....	6
5. Function setting .....	7
1) Shutdown .....	7
2) Language .....	8
3) Import License Authorization File .....	8
4) Add extensions .....	8
5) Add sip trunk .....	10
6) Queue .....	13
7) Add Conference .....	14
8) Add IVR .....	14
9) Outbound Routes .....	16
10) Inbound Routes .....	17
6 Call .....	18
1) Internal Call .....	18
2) Outgoing call .....	18
3) Incoming call .....	18

# 1.Appearance



Figure 1-1 UC300-A11EM1 front panel



Figure 1-2 UC300-A11EM1 back panel



Figure 1-1 UC300-A41EM1 front panel



Figure 1-2 UC300-A41EM1 back panel

# 2.Description of account and ports

Website login

Default IP: 172.16.101.1

Username: admin

Password: admin

Sip account (10 accounts)

Username: 101~110

Password: pbx101~pbx110

UC300-A41EM port

fxs: port5 connects to analog telephony

fxo: port1~port4 connect to outside telephone connection

## 3. Device connection

FXS port connects to analog telephony, FXO connects to outside telephone connection.

## 4. Access to device

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



Figure 4-1 Login interface

Type in the default username: admin, and default password: admin to login.

## 5. Function setting

### 1) Shutdown

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

Navigate to **System > Shutdown**

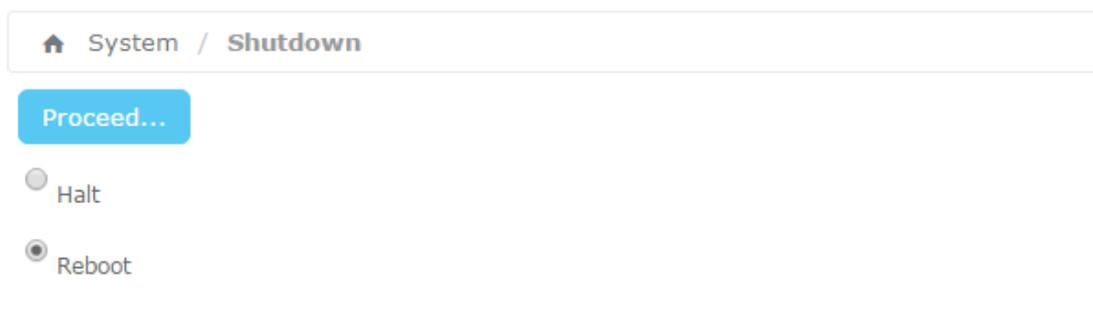


Figure 5-1 Shutdown Interface

## 2) Language

The option “Language” of the Menu “Preferences” in UC300 lets us configure the language for the UC300 Web Interface.

System / Preferences / Language

- Select language: English (en) Save
- Download Language Download
- Delete Language Delete
- Upload Language 选择文件 未选择任何文件 Upload

Figure 5-2 Language setting

## 3) Import License Authorization File

UC300 default license to support the sip extensions are 30, for more sip extensions, please contact OpenVox sales personnel ([sales@openvox.cn](mailto:sales@openvox.cn)) to buy license.

System / Licenses

- Current Licenses Informations:
  - Licenses UID: a0980502a004
  - Max SIP Number: 30
  - Max IAX Number:
  - Max Connected Number:
- Upload Licenses 选择文件 未选择任何文件 Upload

Figure 5-3 Language setting

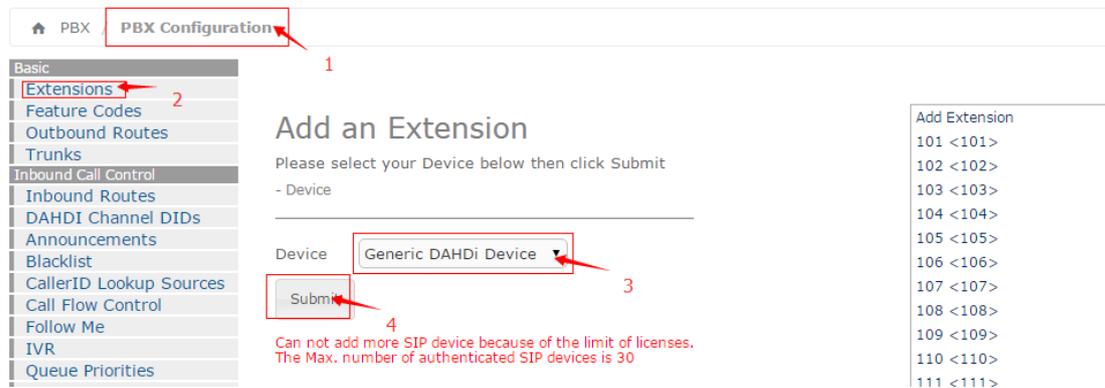
## 4) Add extensions

By default, UC300 has registered 10 sip extensions, you can use sip phone to register directly.

Username: 101~110

Password: pbx101~pbx110

For instance, add DAHDI extensions(222) to the FXS port, navigate to **PBX>PBX configuration > extensions**, click “Add extension”, choose Generic DAHDI Device in the Device, then click “submit”.



**Figure 5-4 add an extension setting**

The configurations interface as bellow:

Outbound CID: 200

User extension: 200

Display Name: fxs

Channel : 5

## Add DAHDI Extension

- Add Extension

User Extension <sup>?</sup>	200 <span style="color: red;">!</span>
Display Name <sup>?</sup>	fxs
CID Num Alias <sup>?</sup>	
SIP Alias <sup>?</sup>	

- Extension Options

Outbound CID <sup>?</sup>	200
Asterisk Dial Options <sup>?</sup>	tr <input type="checkbox"/> Override
Ring Time <sup>?</sup>	Default <input type="button" value="v"/>
Call Forward Ring Time <sup>?</sup>	Default <input type="button" value="v"/>
Outbound Concurrency Limit <sup>?</sup>	No Limit <input type="button" value="v"/>
Call Waiting <sup>?</sup>	Disable <input type="button" value="v"/>
Internal Auto Answer <sup>?</sup>	Disable <input type="button" value="v"/>
Call Screening <sup>?</sup>	Disable <input type="button" value="v"/>
Pinless Dialing <sup>?</sup>	Disable <input type="button" value="v"/>
Emergency CID <sup>?</sup>	
Queue State Detection <sup>?</sup>	Use State <input type="button" value="v"/>

- Assigned DID/CID

DID Description <sup>?</sup>	
Add Inbound DID <sup>?</sup>	
Add Inbound CID <sup>?</sup>	

- Device Options

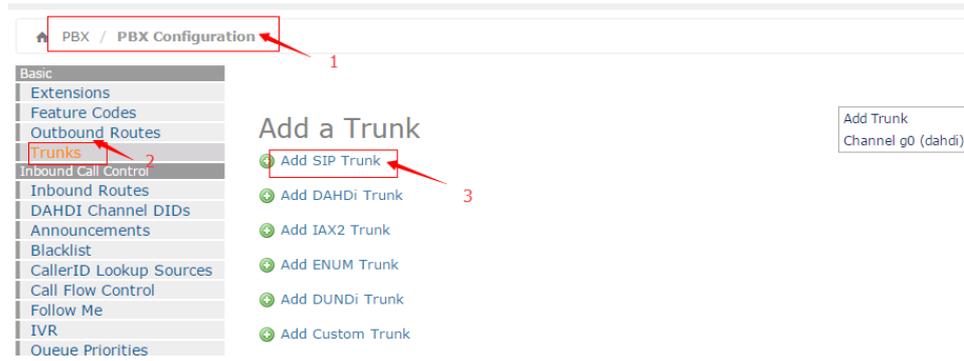
This device uses dahdi technology.

channel <sup>?</sup>	5
----------------------	---

Figure 5-5 DAHDI extensions basic setting

### 5) Add sip trunk

Navigate to **PBX>PBX configuration >Trunk**, click “Add SIP Trunk”.



**Figure 5-6 Add sip trunk interface**

Enter the following configuration as below, in this case, the IP of service 2 is 172.16.101.2.

**Table 5-1 2 service sip connections**

UC300	host=172.16.101.2 type=friend context=from-trunk username=8000 secret=8000 qualify=0 insecure=very canreinvite=no Fromuser=8000
Service 2	host=172.16.101.1 type=friend context=from-trunk username=8000 secret=8000 qualify=0 insecure=very canreinvite=no Fromuser=8000

# Add SIP Trunk

Add  
Cha

## General Settings

Trunk Name <sup>?</sup>:

Outbound CallerID <sup>?</sup>:

CID Options <sup>?</sup>:

Maximum Channels <sup>?</sup>:

Asterisk Trunk Dial Options <sup>?</sup>:   Override

Continue if Busy <sup>?</sup>:  Check to always try next trunk

Disable Trunk <sup>?</sup>:  Disable

## Dialed Number Manipulation Rules

(prepend) + prefix | match pattern

Dial Rules Wizards <sup>?</sup>:

Outbound Dial Prefix <sup>?</sup>:

## Outgoing Settings

Trunk Name <sup>?</sup>:

PEER Details <sup>?</sup>:

```
host=172.16.101.2
type=friend
context=from-trunk
username=8000
secret=8000
qualify=0
insecure=very
canreinvite=no
Fromuser=8000
```

## Incoming Settings

USER Context <sup>?</sup>:

USER Details <sup>?</sup>:

```
secret=***password***
type=user
context=from-trunk
```

Figure 5-7 UC300 add sip trunk interface

## 6) Queue

Navigate to **PBX>PBX configuration >Queue**, adding a queue, static agents are assumed to always be in the extension of queue but not be supposed login the queue, and can't log out of the queue. The dynamic member is an extension or callback number that can log in and out of the queue, you can use the Quick Extension Pick feature set an extension to a static / dynamic agent quickly.

After finish setting, user extensions can dial queue number (600) to join the queue directly. The static members that have been set, such as the 101 extension, you can directly dial the shortcut key (\* 45) to join the queue.

### Add Queue

Add Queue

Queue Number: 600

Queue Name: queue

Queue Password:

Generate Device Hints:

Call Confirm:

Call Confirm Announce: Default ▾

CID Name Prefix:

Wait Time Prefix: No ▾

Alert Info:

Static Agents: 102,0  
103,0

Extension Quick Pick: (pick extension) ▾

Dynamic Members: 101,0

Extension Quick Pick: (pick extension) ▾

Restrict Dynamic Agents: Yes No

Agent Restrictions: Call as Dialed ▾

Figure 5-8 Add queue interface

## 7) Add Conference

Navigate to **PBX>PBX configuration >Conference**, add a conference, user extensions can dial conference number (700) to join the conference directly.

### Add Conference

Add Conference

---

Conference Number:

Conference Name:

User PIN:

Admin PIN:

---

Conference Options

Join Message:

Leader Wait:

Talker Optimization:

Talker Detection:

Quiet Mode:

User Count:

User join/leave:

Music on Hold:

Music on Hold Class:

Allow Menu:

Record Conference:

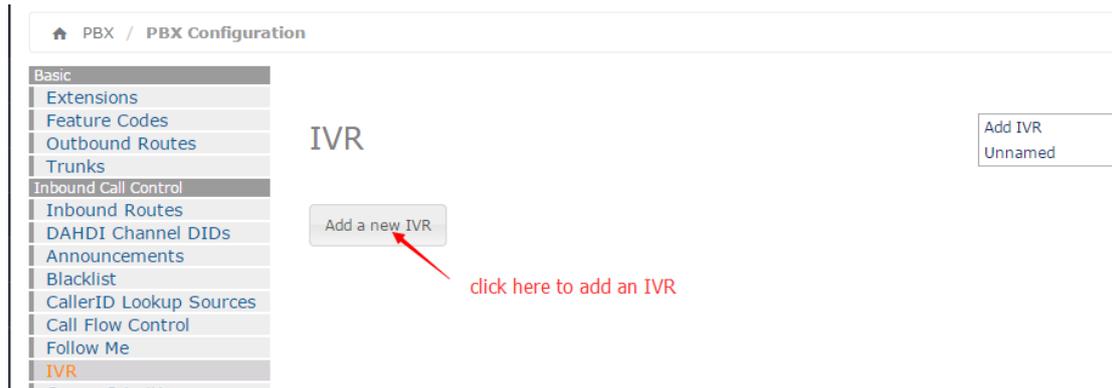
Maximum Participants:

Mute on Join:

Figure 5-9 Add conference interface

## 8) Add IVR

Navigate to **PBX>PBX configuration >IVR**, add an IVR.



Function Description: Press 1 to go to the fxs extension; press 2 to go to the queue, press 3 to join the meeting. In this setup, hang-up directly on illegal input and timeout input.

### Add IVR

- IVR General Options

IVR Name

IVR Description

- IVR Options (DTMF)

Announcement

Direct Dial

Timeout

Invalid Retries

Invalid Retry Recording

Append Announcement on Invalid

Return on Invalid

Invalid Recording

Invalid Destination

Timeout Retries

Timeout Retry Recording

Append Announcement on Timeout

Return on Timeout

Timeout Recording

Timeout Destination

Return to IVR after VM

- IVR Entries

Ext	Destination	Ext	Return	Delete
1	Extensions	<1002> 1002	<input type="checkbox"/>	<input type="checkbox"/>
2	Queues	queue <600>	<input type="checkbox"/>	<input type="checkbox"/>
3	Conferences	conference <700>	<input type="checkbox"/>	<input type="checkbox"/>

Figure 5-10 Add IVR interface

## 9) Outbound Routes

UC300 has set a outbound route begin with 9, which by FXO outbound.

Dial Patterns that will use this Route <sup>?</sup>

---

(  ) + 9  | [ .  /  ] 

+ Add More Dial Pattern Fields

Dial patterns wizards <sup>?</sup>: (pick one) ▼

Trunk Sequence for Matched Routes <sup>?</sup>

---

0  DAHDI/g0 ▼

Optional Destination on Congestion <sup>?</sup>

---

Normal Congestion ▼

Submit Changes Duplicate Route

**Figure 5-11 Outbound Route interface**

Similarly, you can also create a new outbound begin with 8, to call to the sip trunk. Put"." to the match pattern is recommended.

Route Name <sup>?</sup>:

Route CID:   Override Extension <sup>?</sup>

Route Password:

Route Type:  Emergency  Intra-Company

Music On Hold? <sup>?</sup>:

Time Group:

Route Position <sup>?</sup>:

Additional Settings

---

Call Recording <sup>?</sup>:

PIN Set <sup>?</sup>:

Dial Patterns that will use this Route <sup>?</sup>

---

(  ) +  | [  ]  /

+ Add More Dial Pattern Fields

Dial patterns wizards <sup>?</sup>:

Trunk Sequence for Matched Routes <sup>?</sup>

---

Optional Destination on Congestion <sup>?</sup>

Figure 5-12 Outbound Route interface

## 10) Inbound Routes

The incoming calls to the UC300 system, connects to distance, which can be chosen by yourself. A typical example is that you can set all of the incoming calls on an FXO or FXS port on your UC300 to a specific extension, ring group, voicemail, etc. Other routes that are not specifically configured can use the global incoming route for incoming control.

The following figure shows how an incoming call can be routed to the IVR and transferred to the appropriate destination via the IVR (press 1 go to the FXS analog trunk; press 2 go to the queue; press 3 go to the conference).

Call Recording

---

Call Recording ⓘ :

Set Destination

---

Figure 5-13 Inbound Route interface

## 6 Call

### 1) Internal Call

The internal extension user can dial the other party's extension number directly to establish a conversation. For example, the sip extension 101 dial 222 can make a conversation with the dahdi extension (analog telephone).

### 2) Outgoing call

An extension user needs to add put 9 as prefix to the number before exiting via FXO port. For example: dial 9 + your phone number, you can make a call to your phone. Similarly, you can dial a number begin with 8, communicating to server B. The rules for outgoing routing can be found in the "Outgoing Routing" section.

### 3) Incoming call

Use the mobile phone or landline dial outside the number directly, according to the incoming routing rules to IVR, press 1 go to the FXS analog trunk; press 2 go to the queue; press 3 go to the conference, this feature will be achieved in the end. The rules for outgoing routing can be found in the "Inbound Routes" section.