### DINSTAR

UC Advance Configuration Practice Guide



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## **Foreword**



### This course is mainly:

Describe Advanced Features of Dinstar UC IPPBX

Introduce how to configure Advanced Features and Common call scenarios

# Course Objective





**Understand and Know Advanced Features** 

Through this course you will be able to



Learn How to Configure Common Call Scenarios



Understand How to Use New Features

# Contents



1 Advanced Features

2 Call Configuration

3 New Features

# Advanced Features

- 1. Event Report
- 2. Voicemail
- 3. Follow Me
- 4. SCA
- 5. Conference
- 6. LDAP
- 7. Auto Provision
- 8. API

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### • Event reporting type

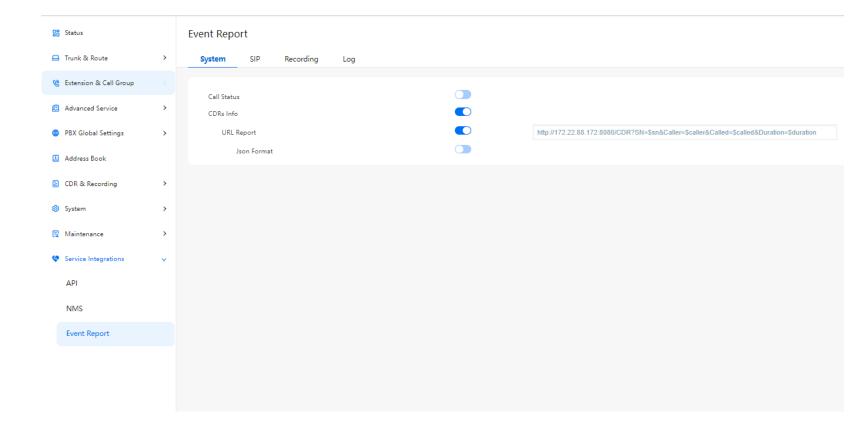
Call Status

CDR

SIP Extension Register/Unregister

SIP Trunk Available/Unavailable

**Voice Recording** 





### System & SIP Event report

# 1. Click on the **Service Integrations-> Event** report->System / SIP

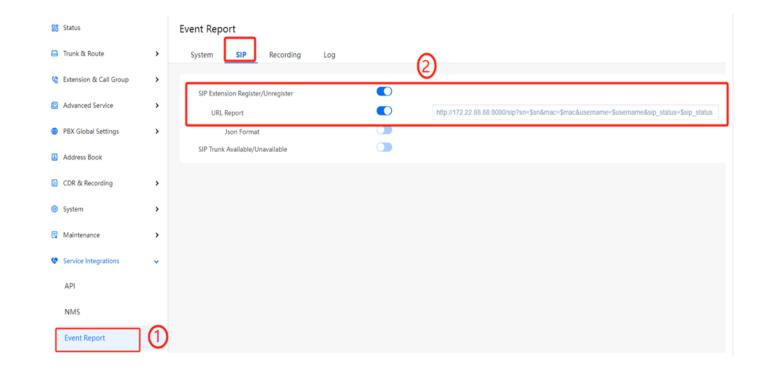
2.Select the projects that need to be reported and fill in the URL in the format of http://ip:port/ <event>?

key1=\$value1&key2=\$value2 <event>:

Corresponding event type, events: callstatus, SIP, SIPTRUNK, CDR. Fill in different event types, and the event messages will be uploaded to different files for saving.

#### **Example:**

http://172.28.7.49:8080/sip?sn=\$sn&mac=\$mac&username=\$username&sip\_status=\$sip\_status





### System & SIP Event report

3. Trigger event reporting conditions and view report results:

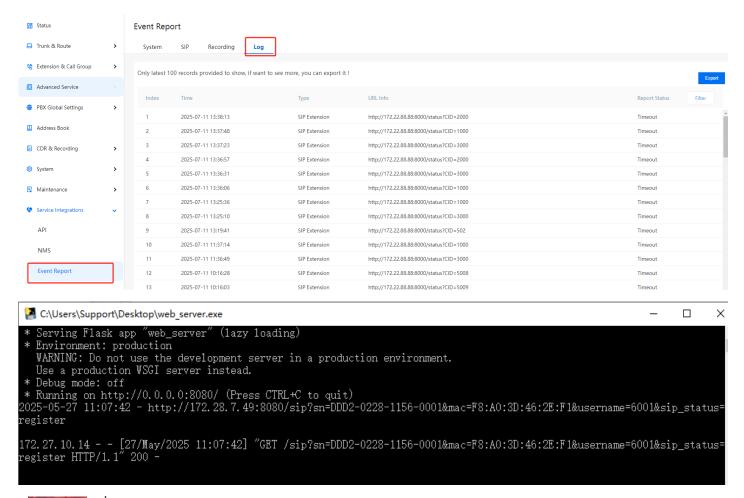
#### UC

The **event report-log** can view the reporting status and information

#### The WEB server

There are corresponding event logs generated in the directory where the script is located Example

SIP extension register/unregister will generate a SIP log, and the testing method for other events is similar



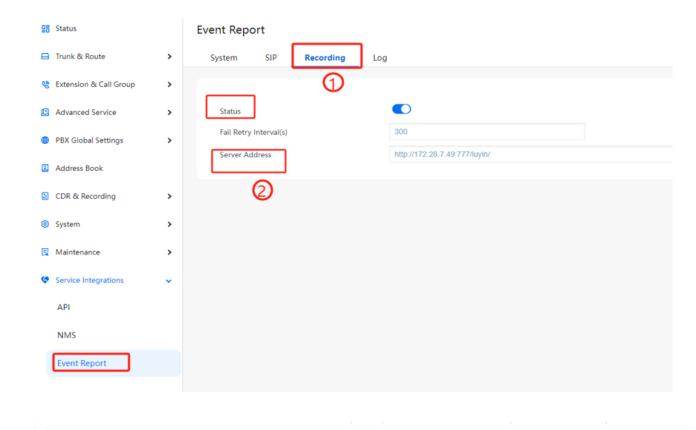


2025-05-27 11:07:42 - http://172.28.7.49:8080/sip?sn=DDD2-0228-1156-0001&mac=F8:A0:3D:46:2E:F1&username=6001&sip\_status=register 2025-05-27 15:15:34 - http://172.28.7.49:8080/sip?sn=DDD2-0228-1156-0001&mac=F8:A0:3D:46:2E:F1&username=6005&sip\_status=register



### • Recording report

- Check Service Integrations-> Event report recording
- 2. Fill in the URL and provide a URL that can save audio files
- Make a phone call to check the recording report status



20250527042020\_SIPP1\_6001\_SIPP2\_6005.wav



#### Main function

**Message Recording/Storage**: Callers can leave verbal messages in the voicemail system and these messages will be recorded and stored by the system.

**Auto-answer/Polite refusal**: Voicemail automatically answers and directs the caller to leave a voice message when the user is unable to answer the call.

**Notification Delivery**: The voicemail system will notify users via SMS, email or phone when there are new messages

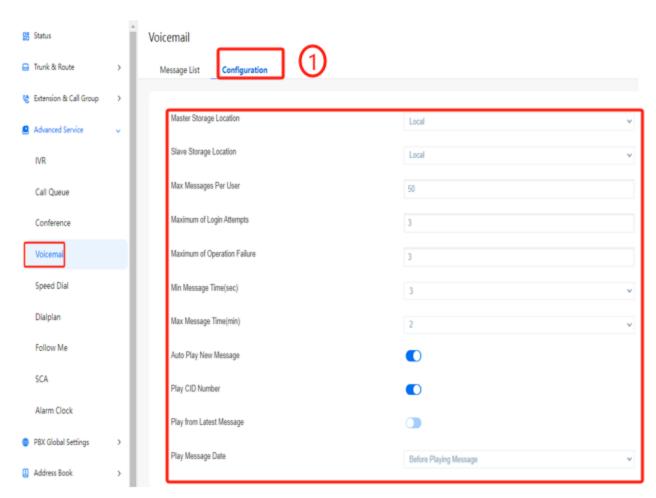




• Configure voicemail rule

1.Click Advanced Service > Voicemail > Configuration

2. Configure storage location、duration of a voicemail and other parameters as needed

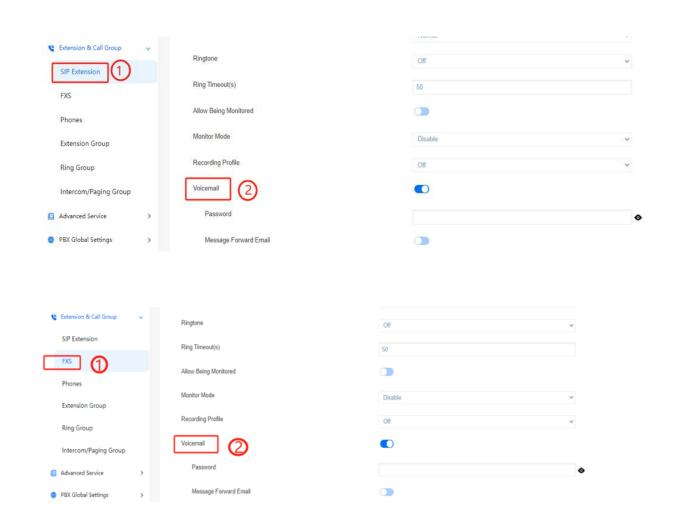




• Enable Voicemail On Extension

1.Click Extension & Call Group > SIP Extension/ FXS

2. enable the Voicemail configuration





### • Configure Email

#### 1.Click on **System > Email>Configuration**

2.set the parameters for sending emails

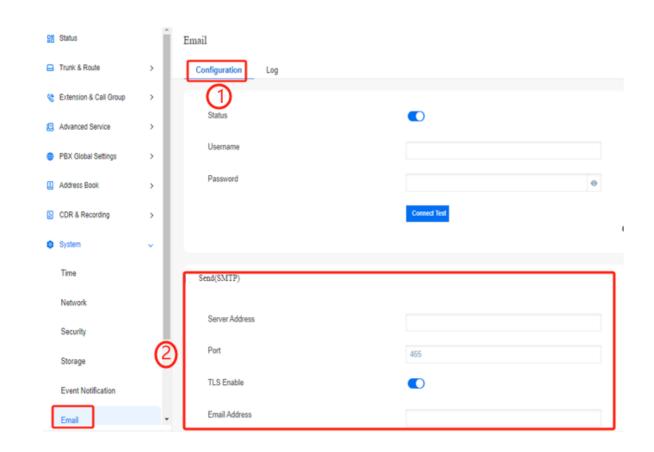
3.After the setting, first "connect test", if the result is "connect successful". You can save the mail configuration.

#### Note:

The email password is usually the client password.

The following link is set way of tencent enterprise email: https://open.work.weixin.qq.com/help2/pc/19902?p erson\_id=1

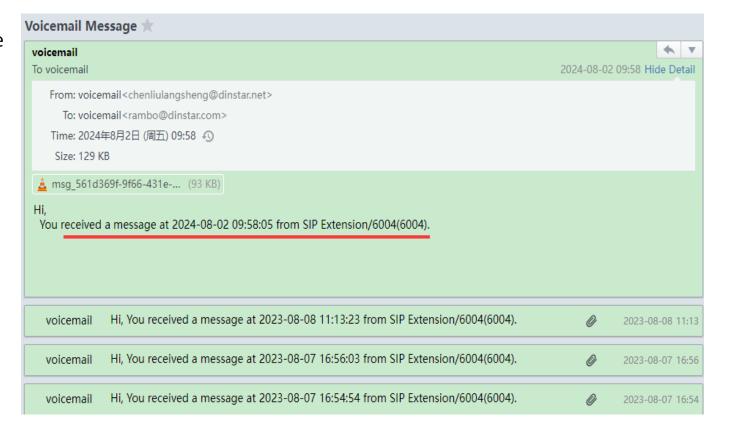
The client password setting method varies depending on different email address.





### Email of Voicemail Message

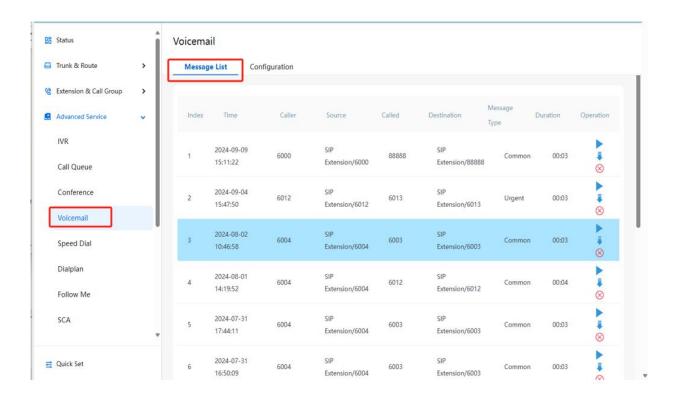
After the extension sets the message forward email, the personal mailbox will receive the **voice message file** as attachment and display the **message time**, **source number**. Users can directly download the message file to listen to the message





Voicemail message management

On the **Advanced Service > Voicemail> Message List** interface, the detailed message information is displayed. Users can manage (play, download, and delete) voice message files





#### Main Scenarios

The user has multiple workstations

Users cannot receive calls from workstations during business trips or outings

#### Main Function

Unify various commonly used communication numbers (mobile phone, office phone, residential phone) into an extension number, so that anyone can find the user by simply dialing this number

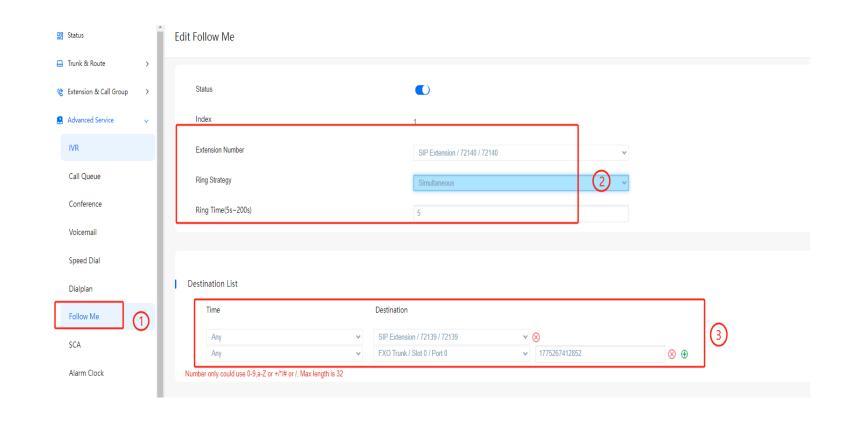




### • Configure Follow Me

# 1.Click Advanced Service->Follow Me

- 2. Select extension number、ring strategy and ring time
- 3.Configure destination, It can be an extension number or a mobile phone number

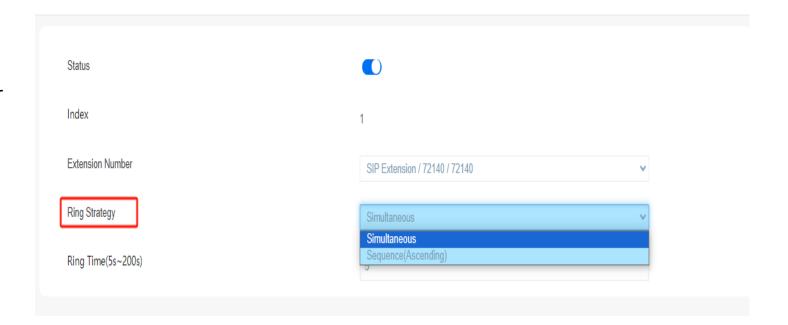




• Parameters-Ring Strategy

**simultaneous**: all numbers ring together

**sequence (ascending )**: starts from the extension and rings from top to bottom

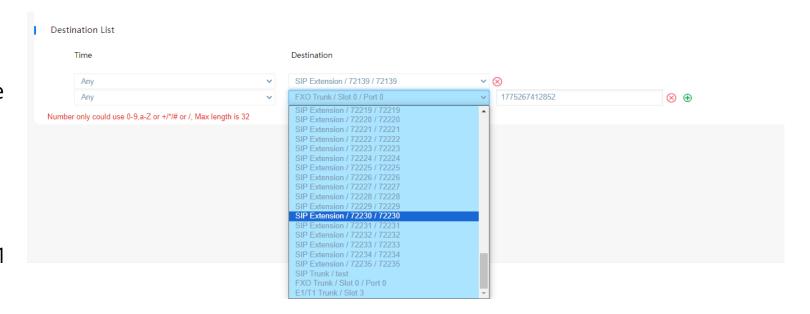




#### • Parameters-Destination List

**Time**: Make calls within the specified time period after setting up

**Destination:** users can choose SIP extension, SIP Trunk, FXO Trunk, and E1/T1 Trunk. When selecting a Trunk, users can configure a phone number



SCA



#### Main Scenarios

Managers need to focus on more important tasks and delegate daily phone conversations to their secretaries, which improves their work efficiency

#### • Main Function

All calls from the call manager will be forwarded to the secretary extension, filtered by the secretary extension, and then forwarded to the manager extension

Support multiple secretary extensions



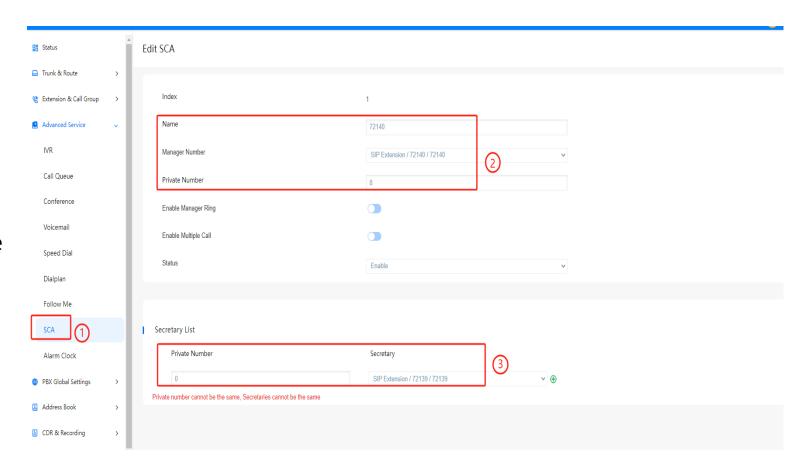




### Configure SCA

#### 1.Click Advanced Service- >SCA

- 2. Select Manager Number and customize manager private number
- 3. Select Secretary Number and customize Secretary private number

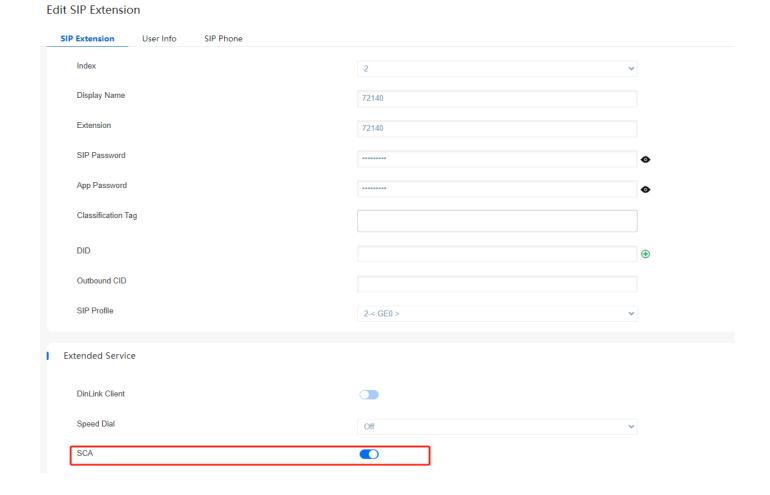






### • Parameters-Manager Number

The manager number must be a SIP extension number that enables the SCA function

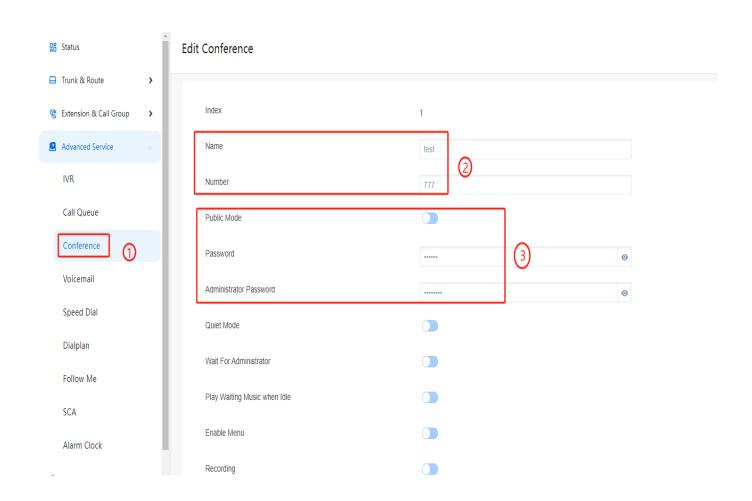


# Conference



### • Configure Conference

- 1.Click Advanced Service->Conference
- 2. Custom Name and conference Number
- 3. No password required in public mode, if not , Password and administrator password need to be set



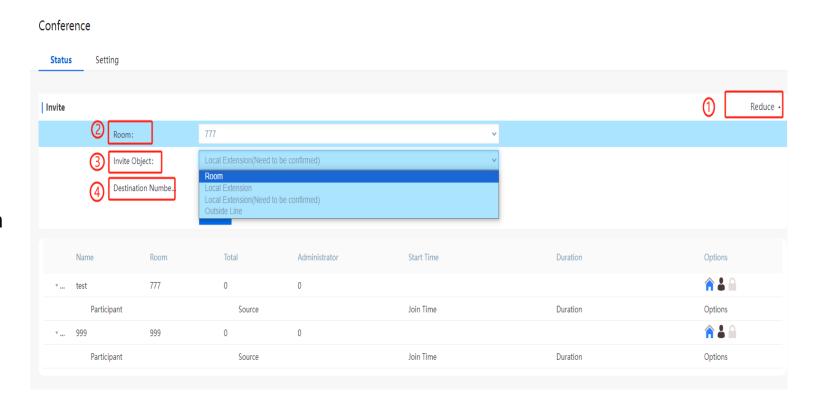
# Conference



#### Invite Members Method 1

#### 1.Click **Expand**

- 2. Select conference room that needs to invite people
- 3. Set invite object: room、local extension and outside line can be select
- 4.Set Destination Number



## Conference



#### Invite Members Method 2

Invite members through DTMF operations

- 1. Dial the conference number and password to join
- 2. Press 1, and after hearing the prompt sound for entering the number, enter the extension number that needs to be invited
- 3. The extension that was invited to join rings

DTMF⋳	Description =	Notes₽
1←	Invite members∉	Non-administrators need to enable
		configuration⊲
2↩	<u>Invite</u> members, need to be	Non-administrators need to enable
	confirmed by the invite⊲	configuration∈
3←	Initiate a conference⊧	Non-administrators need to enable
		configuration∈
4€	Decrease the volume of the	<i> </i> ←
	handset↩	,-
6∈	Increase the volume of the	<b>/</b> ←
	handset₽	
7↩	Decrease the volume of the	<i> </i> ←
	microphone⊲	
9∈	Increase the volume of the	/e
	microphone⊲	
*~	Mute∈	<b>/</b> ←
0€	All non-administrators are	<u>Administrator</u> permissions∈
	muted⊲	
<b>#</b> ←³	Exit all non-administrators	<u>Administrator</u> permissions∈
	from the conference∈	

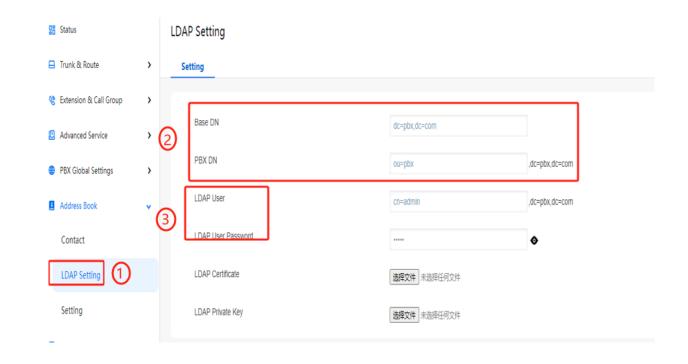
### **LDAP**



### • Configure LDAP-IPPBX

#### 1.Click Address Book->LDAP Setting

- 2. The default configuration generally does not need to be changed
- 3. Record user, password and other parameters for convenient IP phone configuration



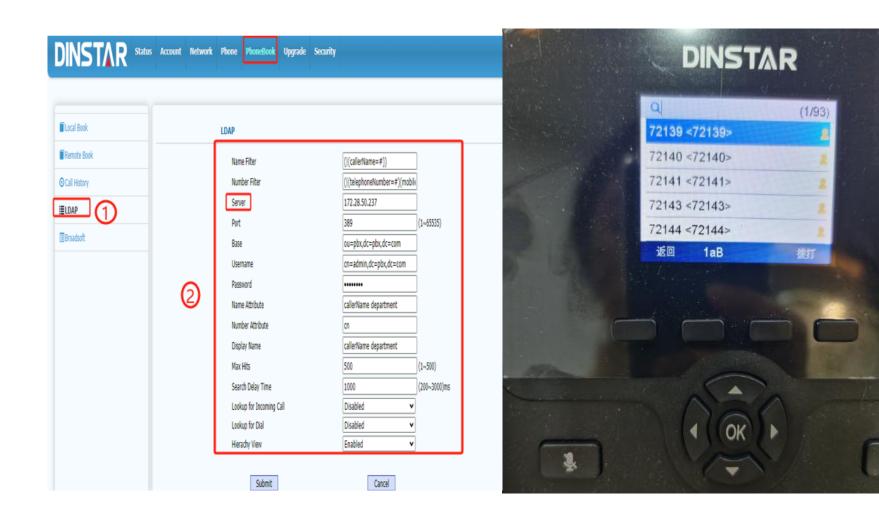
### **LDAP**

#### **DINSTAR**

Configure LDAP-IP Phone(Dinstar)

#### 1.Click Phone Book->LDAP

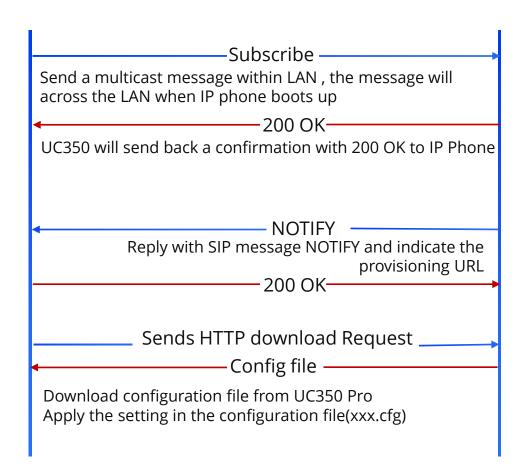
- 2. Change the server address to the corresponding IPPBX address, refer to the image for other configurations
- 3. The LDAP of the phone can display the extension number of IPPBX





### Logic







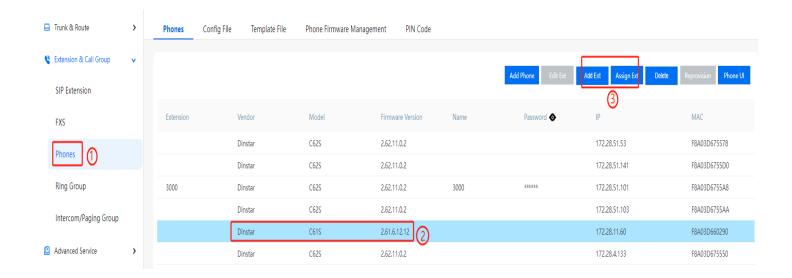
**IPPBX** 

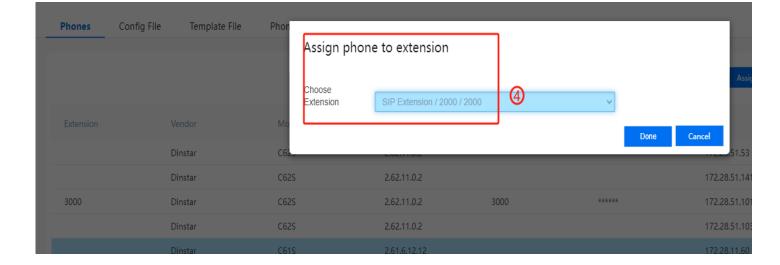


Assigning an extension to phone

IP Phone connected to the network

- 1. Click **Extension & Call Group-> Phones,** view the online IP Phones who send subscribe to UC
- 2. Select the IP Phone
- 3.Choose add Ext or Assign Ext
- 4. Create an extension number or select an existing extension number

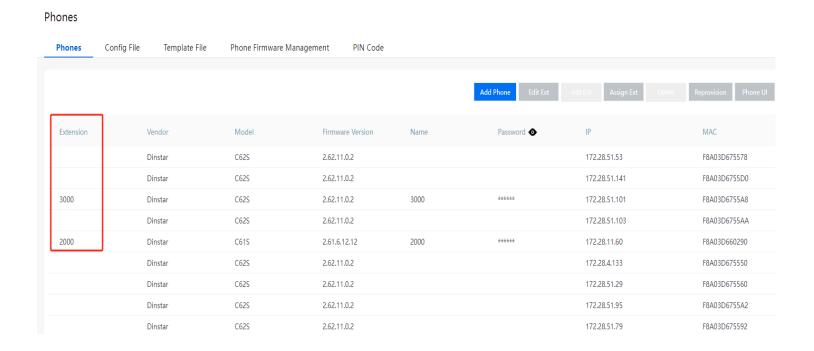






Assigning an extension to phone

Click Extension & Call Group->
Phones, users can view the binding
of the phone to the extension

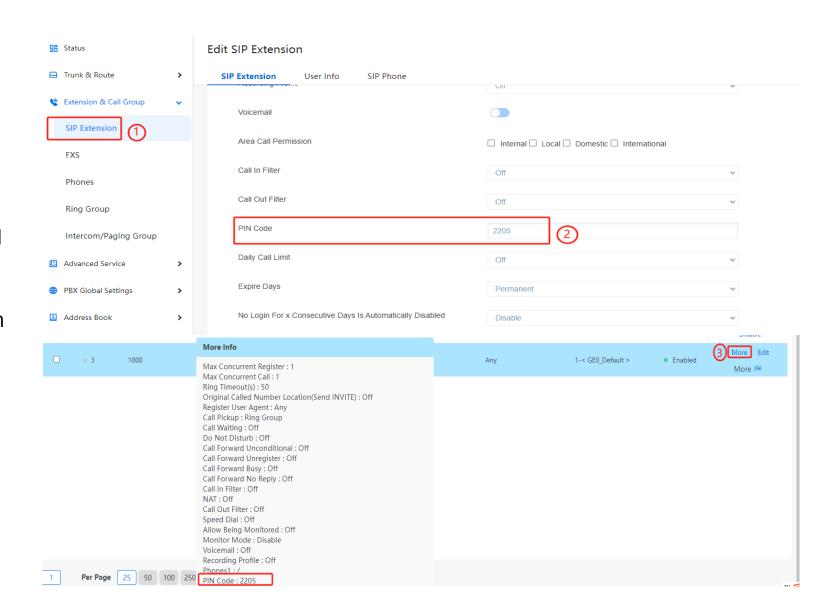




 Assigning an extension to phone(PIN Code)

# 1.Click Extension & Call Group-> SIP Extension

- 2. Select an extension and Configure PIN code
- 3. After saving, check if the extension pin code has been successfully configured

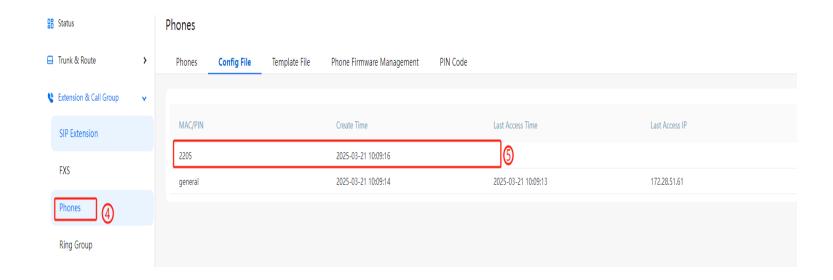


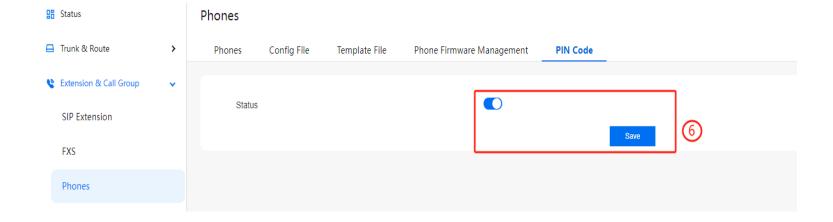


 Assigning an extension to phone(PIN Code)

4.Click Extension & Call Group->
Phones

- 5. View Config File
- 6. Activate pin code



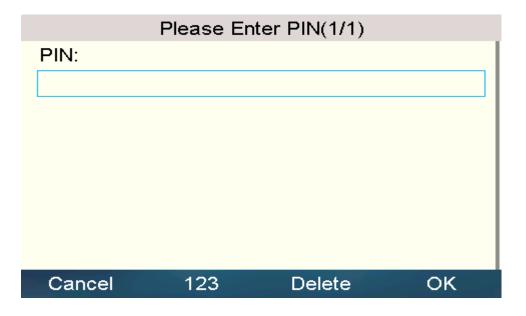




 Assigning an extension to phone(PIN Code)

When the phone prompts "Please Enter PIN", enter the file name(For example 2205)

The file will be downloaded from the UC, and the configuration will take effect in the phone



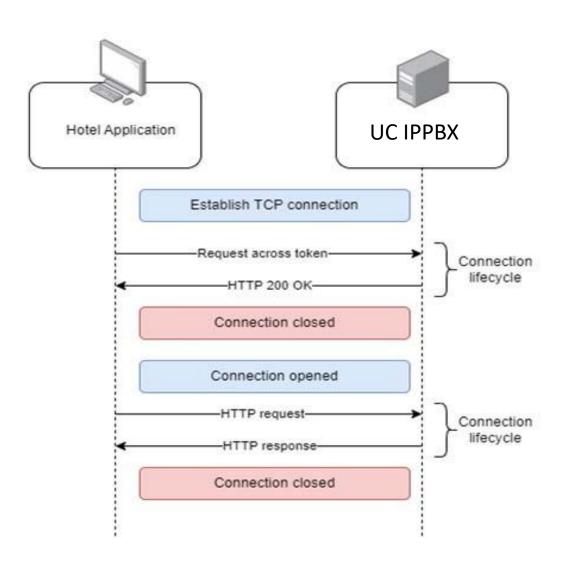


## **API**



#### API Workflow

- 1. Client sends a request to the API endpoint with the necessary parameters.
- 2. The API server receives the request and validates the parameters.
- 3. If the parameters are valid, the API server processes the request and retrieves the ne cessary data.
- 4. The API server formats the data into the requested response format (ex. JSON, XML or other format).
- 5. The API server sends the response back to the client.
- 6. The client receives the response and processes the data as needed.



**API** 



### Key Features

**Check:** check device information/

parameter/setting/statue

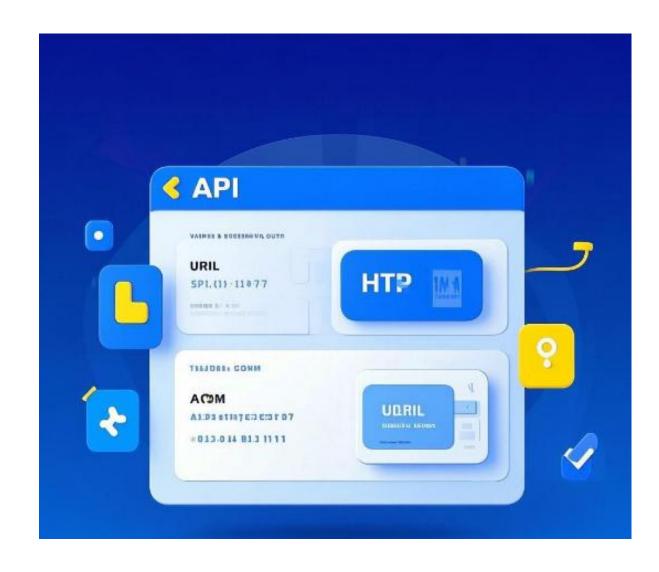
**Change:** change setting

Add: create new setting

**Delete:** delete exit setting

**Call:** generate call

**Report:** report CDR/call status/sip information/recording



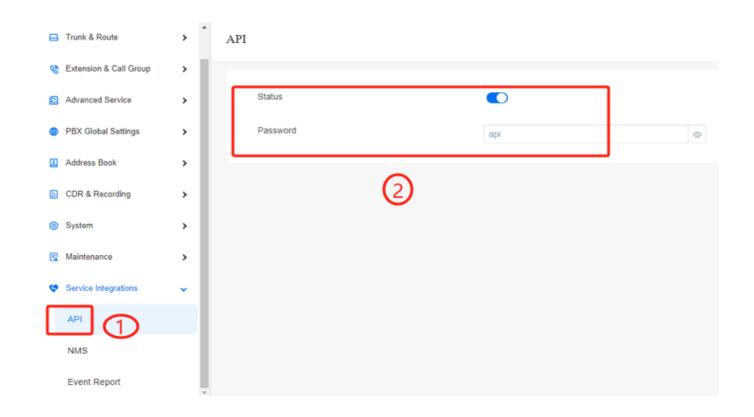




### Configure API-Enable API

1.Click on **Service->Integrations->API** 

**2.** enable API interface and change the password. the default password is api

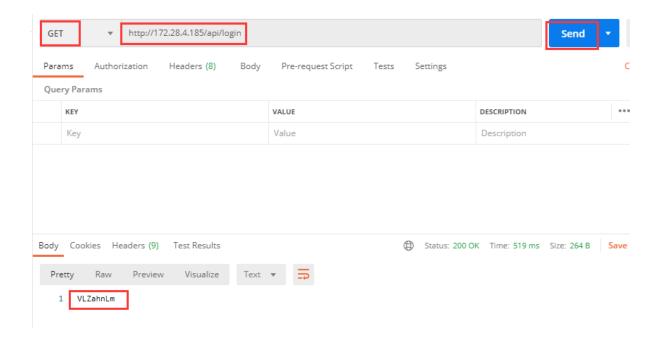






Configure API-Get A MD5 Crypt Salt

Send http request **http://UC\_ip/api/login** to get a md5 crypt salt from http server





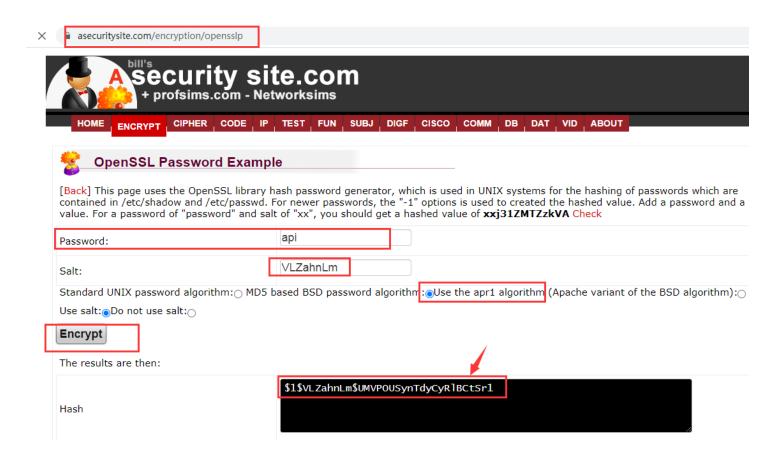


# Configure API-Calculation Password With The Salt

user can calculation password with salt on some encrypt websit

There is a free website which can calc password with salt .

https://asecuritysite.com/encryption/op ensslp



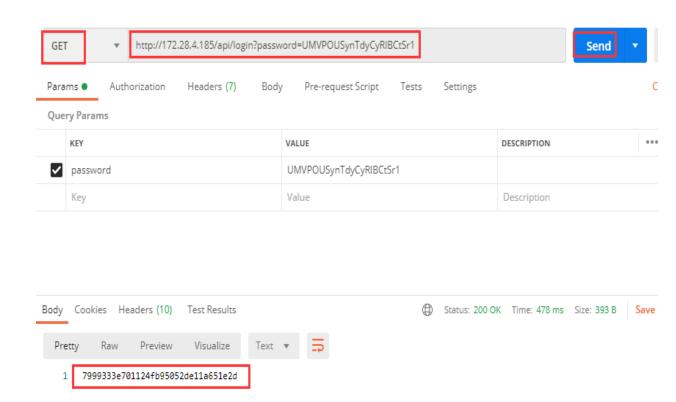




• Configure API-Get a token

users can use the URL

http://UC\_ip/api/login?password=caculataion
password to get the token

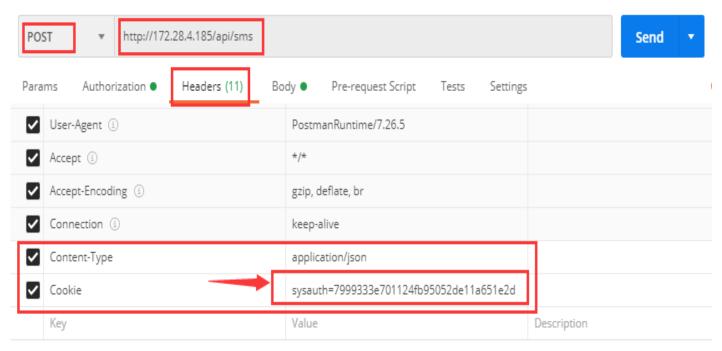






### Configure API-Send Http Request

Users can send http request to UC to control device by API interface.



Response

### Contents



1 Advanced Features

2 Call Configuration

3 New Features

# Call Configuration

02

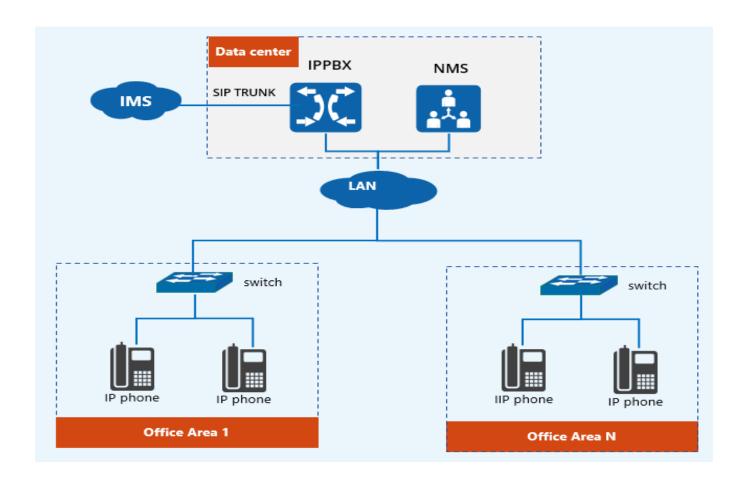
- 1. IMS Configuration
- 2. Sip Trunk to MTG
- 3. Sip Trunk to IAD/FXO
- 4. Number Matching
- 5. Manipulation

DINSTAR

#### **DINSTAR**

#### Main Scenarios

- When the operator opens IMS accounts, UC needs to register the IMS account successfully before making calls
- The IP phone is registered with the SIP extension of UC, which enables internal extension calling and outbound calling through IMS

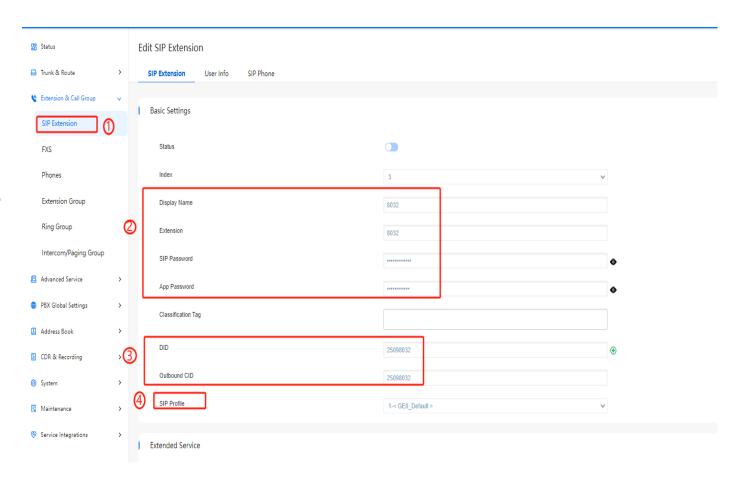




**SIP Extension Setting** 

SIP Trunk Setting

- Click Extension & Call Group->SIP
   Extension
- 2. Create a new SIP extension, configure the extension number and password
- 3. Fill in the DID and Outbound CID as the operator's number
- 4. Select network port

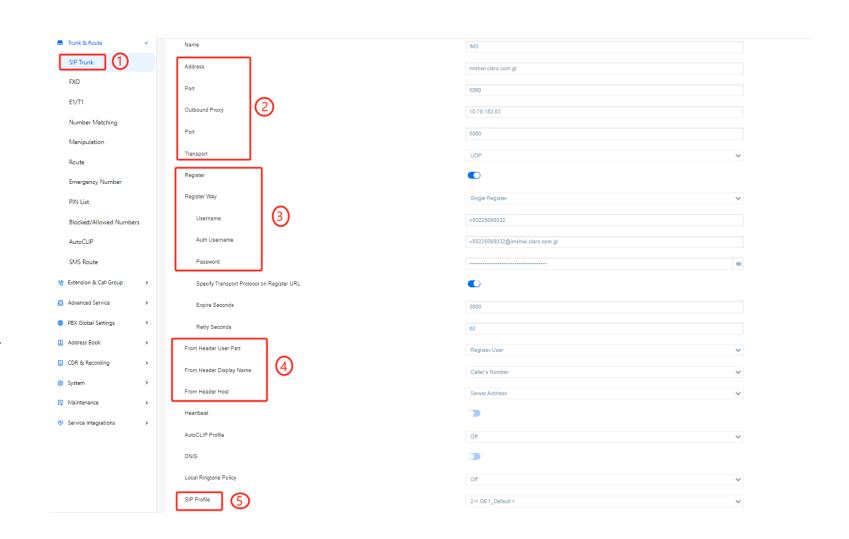




#### **SIP Extension Setting**

#### **SIP Trunk Setting**

- 1. Click Trunk & Route->SIP Trunk
- 2. Fill in the IMS domain name and IP address, select the protocol
- Open register. There are many IMS accounts that can be registered using account group, but only one account can select single register and filling in account information
- Modify the From header User Part \
  Display Name and Host
- Select a network port that is compatible with the IMS proxy address network



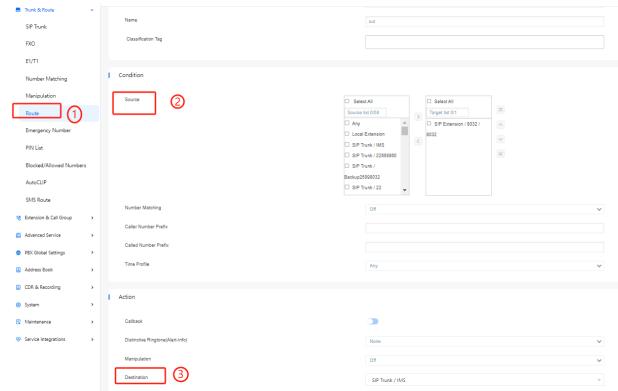


SIP Extension Setting

SIP Trunk Setting

- 1.Click Trunk & Route->Route
- 2. Select call source/SIP extension
- 3. Select call Destination/IMS

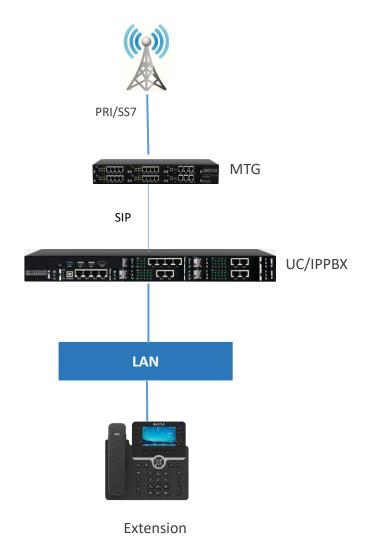






#### • Main Scenarios

- The operator side uses PRI/SS7 signaling, and the E1 line is connected to the E1 port of MTG
- 2. UC and MTG use SIP protocol, and the two networks can communicate with each other
- Register the SIP extension of UC on the IP phone to enable internal extension calls and use E1 lines for outgoing calls

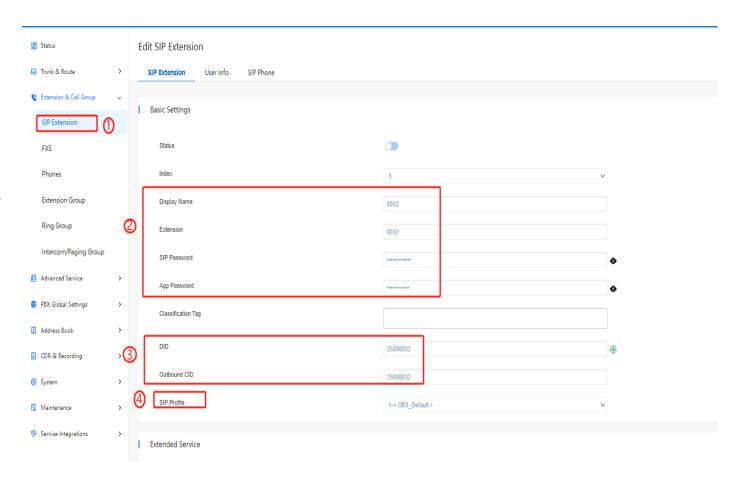




SIP Extension Setting

SIP Trunk Setting

- Click Extension & Call Group->SIP Extension
- 2. Create a new SIP extension, configure the extension number and password
- 3. Fill in the DID and Outbound CID as the operator's number
- 4. Select network port





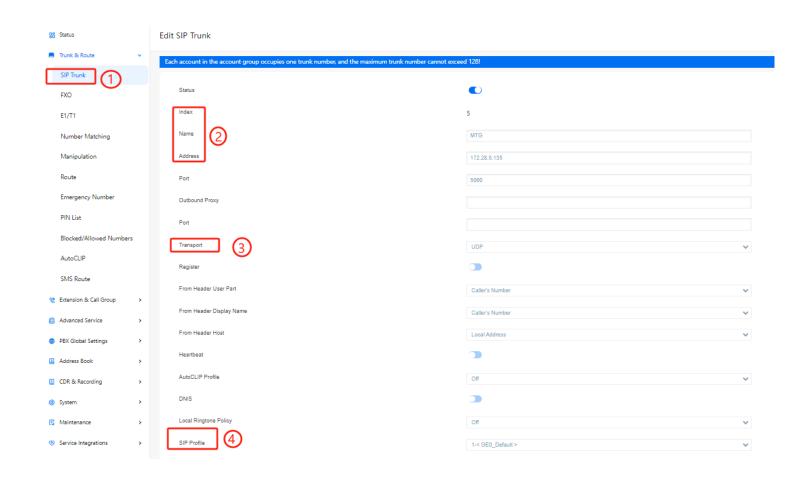
**SIP Extension Setting** 

**SIP Trunk Setting** 

#### **Route Setting**

### SIP Trunk Setting

- 1. Click Trunk & Route->SIP Trunk
- 2. Fill in the IP and port of MTG
- 3. Select Protocol
- 4. Select network port





SIP Extension Setting

SIP Trunk Setting

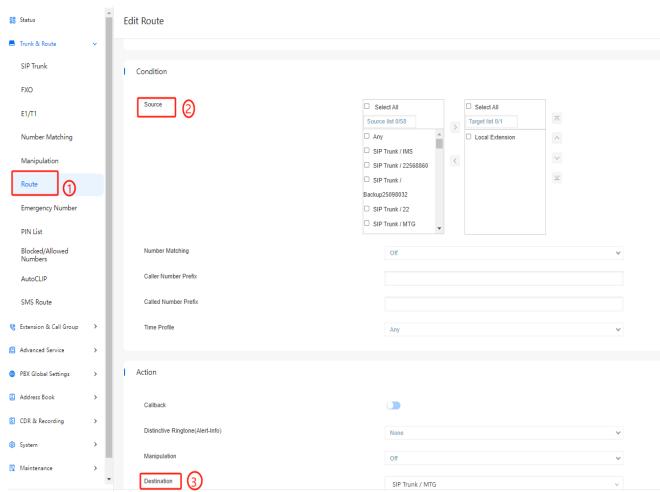
**Route Setting** 

Route Setting

Route

- 1.Click Trunk & Route->Route
- 2. Select call source/SIP extension
- 3. Select call Destination/MTG

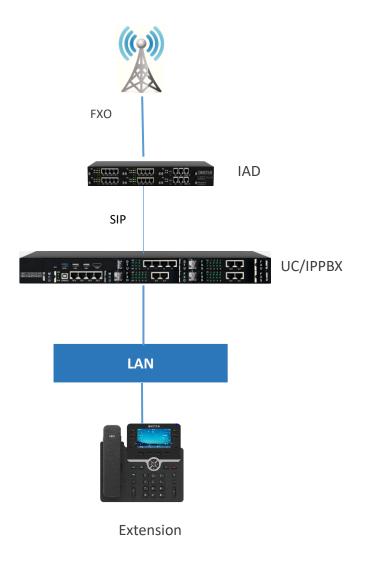




#### DINSTAR

#### Main Scenarios

- The operator uses an analog telephone line connected to the IAD-FXO port
- 2. UC and IAD use SIP protocol, and the two networks can communicate with each other
- 3. SIP extension uses FXO line for outgoing calls

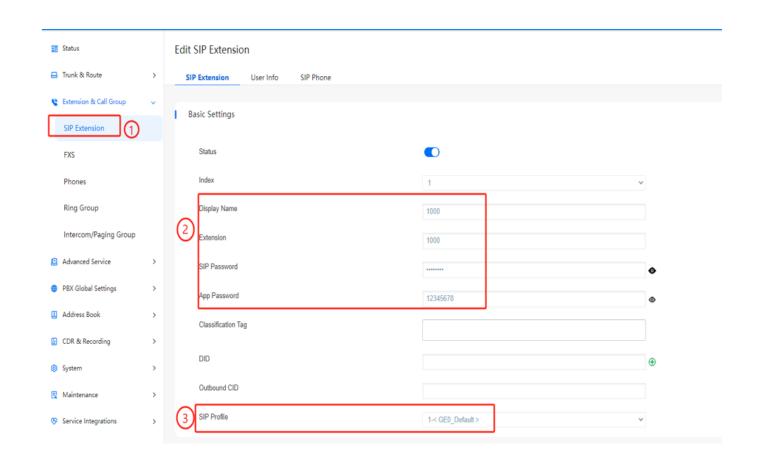




**SIP Extension Setting** 

SIP Trunk Setting

- 1. Click Extension & Call Group->SIP Extension
- 2. Create a new SIP extension, configure the extension number and password
- 3. Select network port





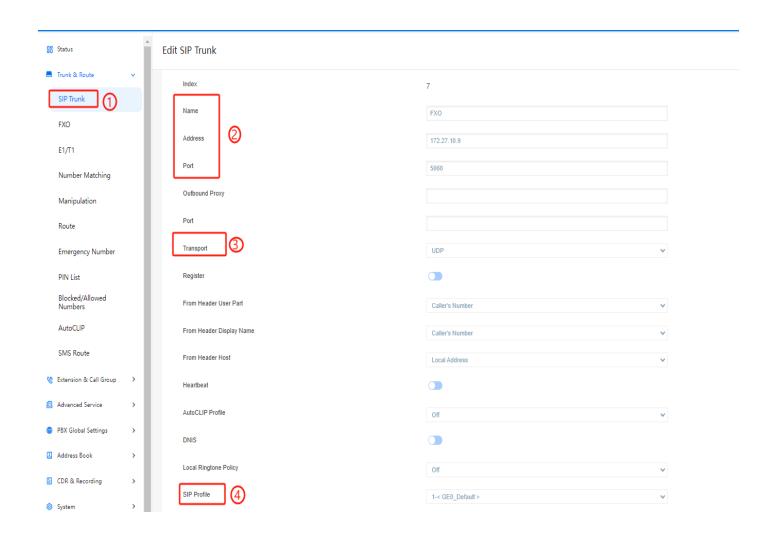
SIP Extension Setting

SIP Trunk Setting

#### **Route Setting**

### SIP Trunk Setting

- 1. Click Trunk & Route->SIP Trunk
- 2. Fill in the IP and port of IAD/FXO
- Select Protocol
- 4. Select network port

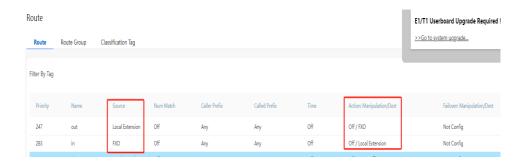


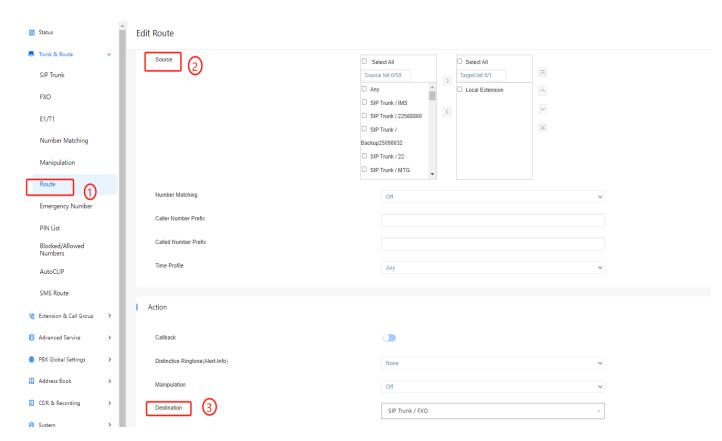


SIP Extension Setting

SIP Trunk Setting

- Route Setting
- 1.Click Trunk & Route->Route
- 2. Select call source/SIP extension
- 3. Select call Destination/FXO





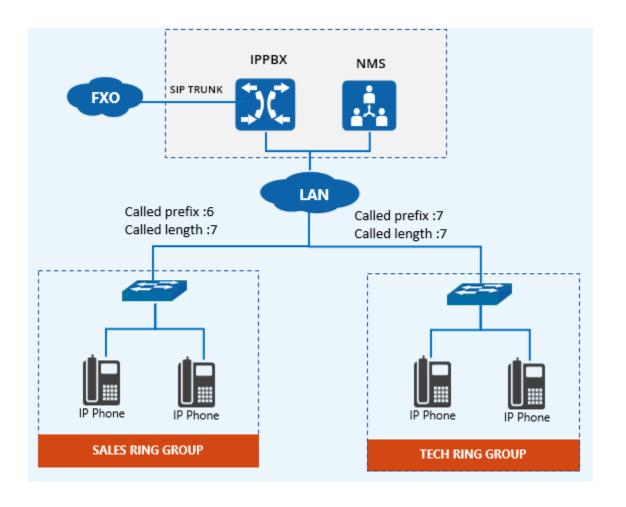
### **Number Matching**

#### **DINSTAR**

#### Main Scenarios

Users can set a prefix for calling numbers or called numbers.

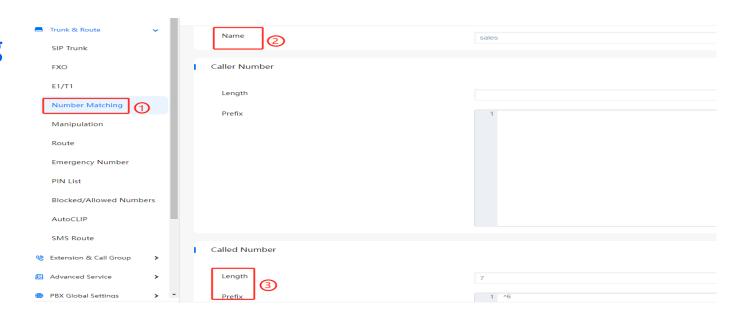
When the prefix of a calling number or a called number matches the set prefix, the call will be passed to choose a route



### **Number Matching**



- Configure Number Matching
- 1.Click **Trunk &Route->Number Matching**
- 2. Custom name
- 3. Configure number prefix and length



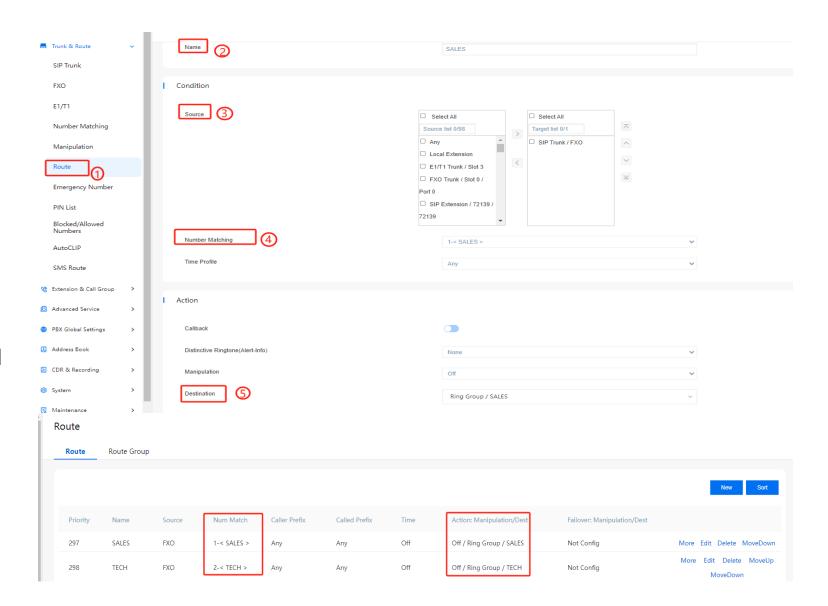
Index	Name	Caller Prefix	Caller Length	Called Prefix	Called Length	
1	sales	*	*	^6	7	Edit Delete
2	TECH	*	*	^7	7	Edit Delete

### **Number Matching**

#### DINSTAR

#### Configure Route

- 1.Click Trunk &Route->Route
- 2. Custom name
- 3. Select call source: SIP Trunk /FXO
- 4. Select number matching: SALES
- 5. Select destination: Ring Group/SALES
- 6. Configure tech using the same method

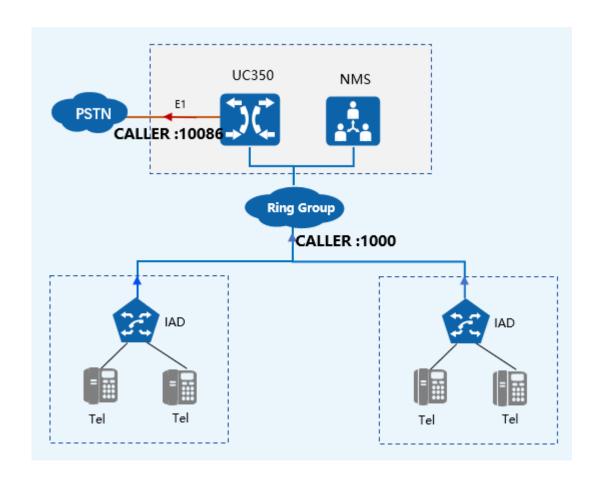


# Manipulation



#### • Main Scenarios

The called number or caller number will be changed during calling process when the called number or the caller number matches the preset rules.

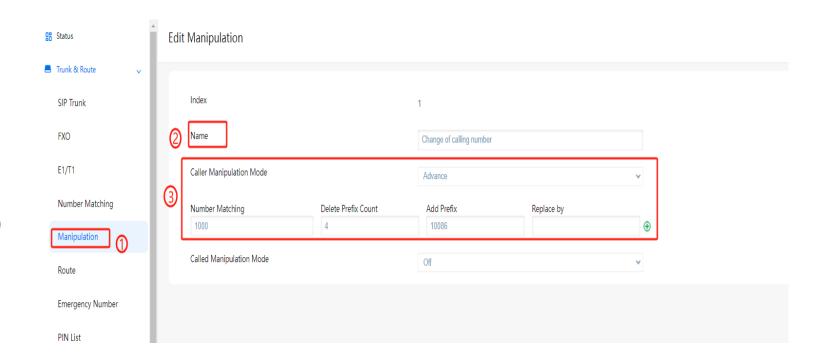


# Manipulation



### Configure Manipulation

- 1.Click Trunk & Route->Manipulation
- 2. Custom name
- 3.Select caller manipulation mode and Configure to convert the caller from 1000 to 10086

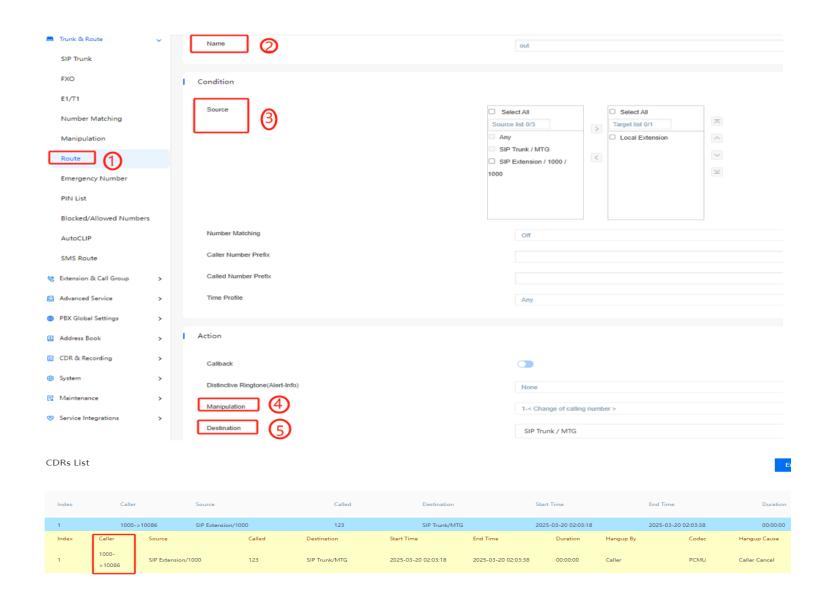


# Manipulation



### • Configure Route

- 1.Click Trunk &Route->Route
- 2. Custom name
- 3. Select call source: Local Extension
- 4. Select Manipulation
- 5. Select destination: SIP Trunk/MTG
- 6. After making a call, users can check on the CDRs if the change was successful



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3 New Features

# New Features

03

- 1. Dinlink
- 2. PMS
- 3. Attendant Console
- 4. Nat Public Proxy Server

DINSTAR

### What is Dinlink



#### Dinlink

Dinlink APP is a softphone that can be used together with IPPBX

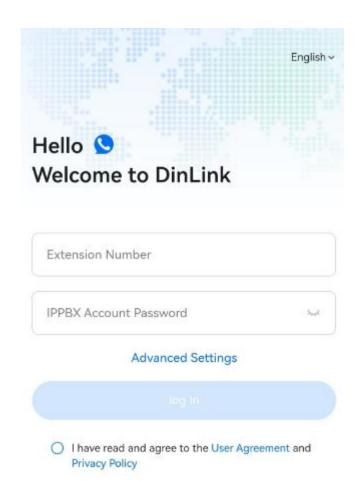
#### • Main function

Make and receive phone calls:

Address book

recording

voice messaging





login uc web page

SIP Stack setting

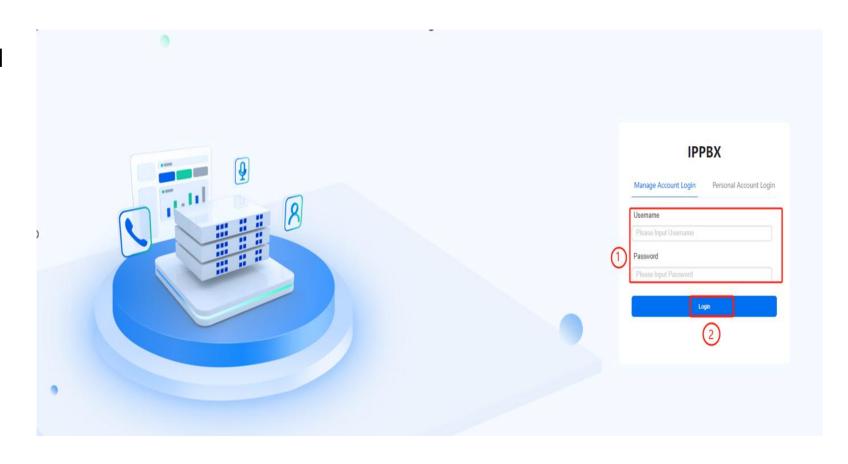
SIP Extension setting

Dinlink configuration

registration status

1.Type username and password default is admin/admin@123#

2.Click Login





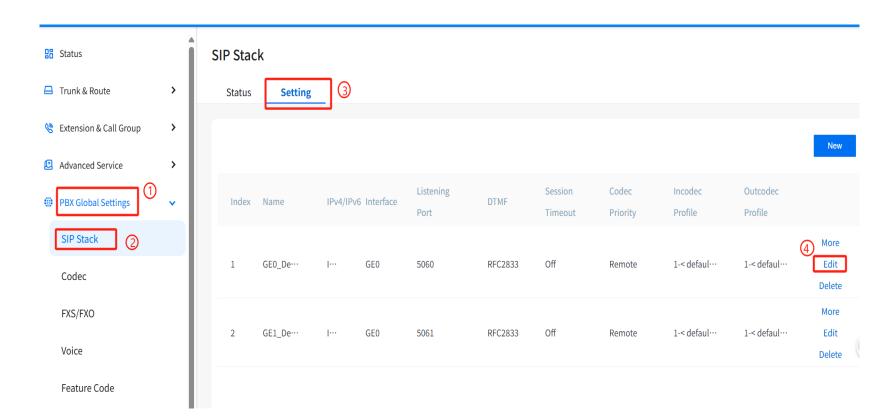
login uc web page

SIP Stack setting

SIP Extension setting

Dinlink configuration

- 1.Click **PBX Global Setting**
- 2. Click SIP Stack
- 3. Setting
- 4. Choose interface and then Edit





login uc web page

SIP Stack setting

SIP Extension setting

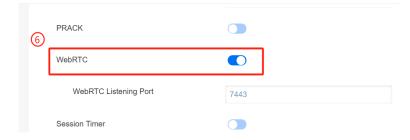
Dinlink configuration

registration status

5. NAT: select Public Proxy, input the server ip, port and select protocol as shown figure

6. Enable WebRTC

**Edit SIP Profile** 



88	Status		
	Trunk & Route	>	
By.	Extension & Call Group	>	
▣	Advanced Service	>	
	PBX Global Settings	<b>~</b>	
	SIP Stack		
	Codec		
	FXS/FXO		
	Voice		
	Feature Code		
1	Address Book	>	
N	CDR & Recording	>	
<b>©</b>	System	>	

Edit SIP Profile						
l B	Basic Settings					
	Index	1				
	SIP Stack Name	GE0_Default				
	IP Version Used By SIP	IPv4 ~				
	SIP Listening Interface	GE0(172.27.10.33)				
(5)	SIP Listening Port	5060				
	NAT	Public Proxy ~				
	Public Proxy Server Address	41 6				
	Public Proxy Server Port	1060				
	Proxy Options	☑ TCP ☑ TLS ☑ WebRTC				



login uc web

SIP Stack setting

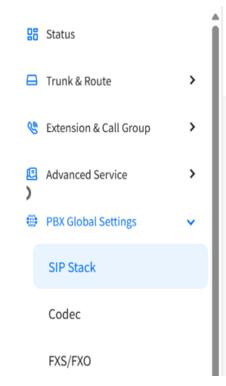
SIP Extension setting

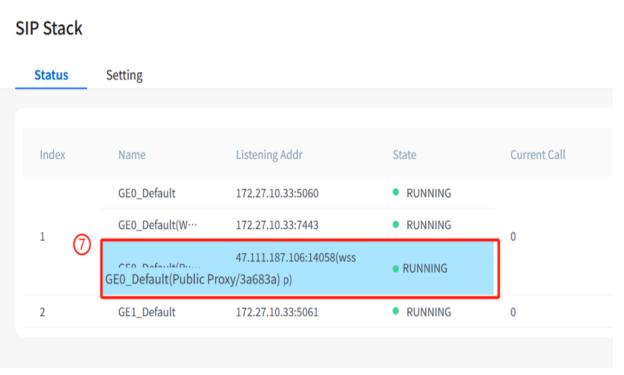
Dinlink configuration

registration status

7. Public Proxy alias is show '3a683a' and State Is 'RUNNING'. It means Public Proxy works.

Note: each UC has unique alias







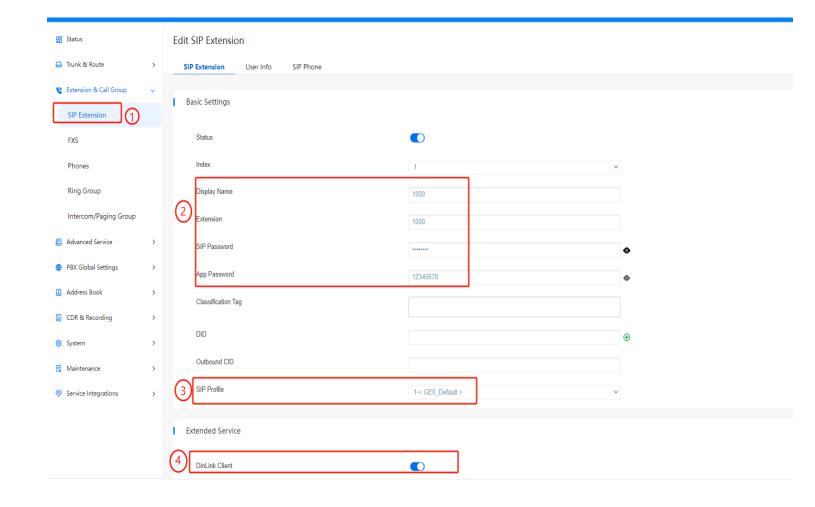
login uc web page

SIP Stack setting

SIP Extension setting

Dinlink configuration

- Click Extension & Call Group->SIP Extension
- 2. Create a new SIP extension, configure the extension number and password
- 3. Select network port
- 4. Enable the dinlink client and save





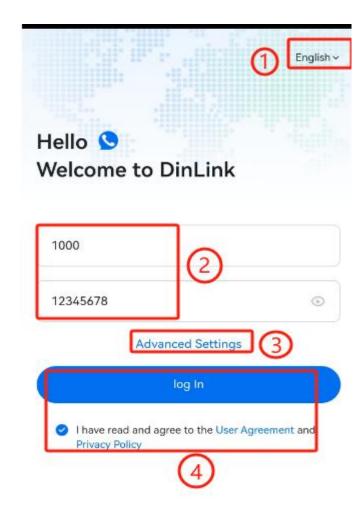
login uc web page

SIP Stack setting

SIP Extension setting

Dinlink configuration

- 1. Choose language
- Configure extension number and password
- Select advanced configuration and configure Public Proxy information (UC SN or Alias, IP, Port)
- 4. Click log in



<	Advanced Settings					
Push Met	hod					
OS Bran	d	3	× )			
Login Me	thod					
Public P	roxy		~			
	N/Device Alias 0830-50 -0016					
Hostnam						
Port 1060						



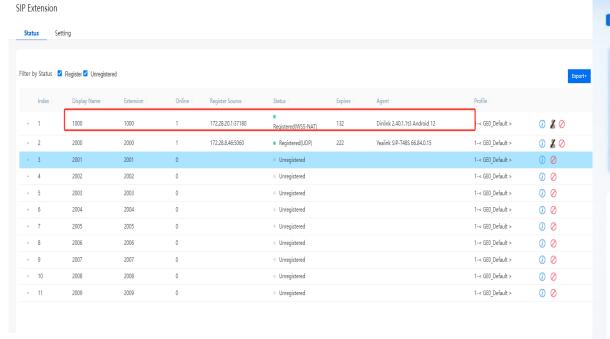
login uc web

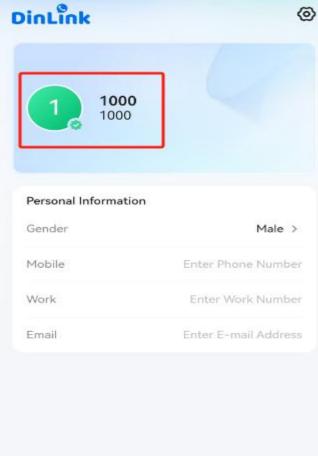
SIP Stack setting

SIP Extension setting

Dinlink configuration

- 1. IPPBX:SIP Extension-Status can check the registration status
- 2. Dinlink: click Mine can check the registration status





### How To Call Dinlink

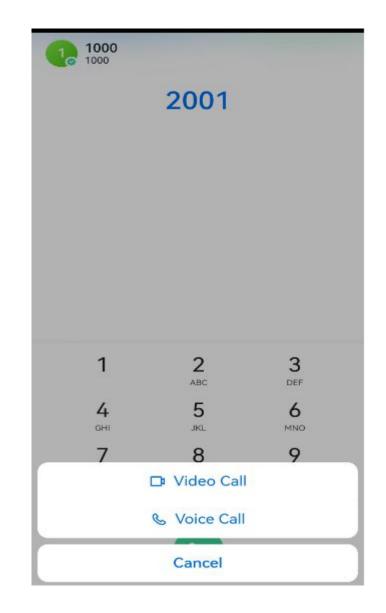


#### **Initiate a call**

- 1. Dial the number
- 2. choose video or voice call

#### **Received call:**

Choose to connect or disconnect









### What is PMS



#### PMS

Property Management System, Focusing on serving the hotel industry

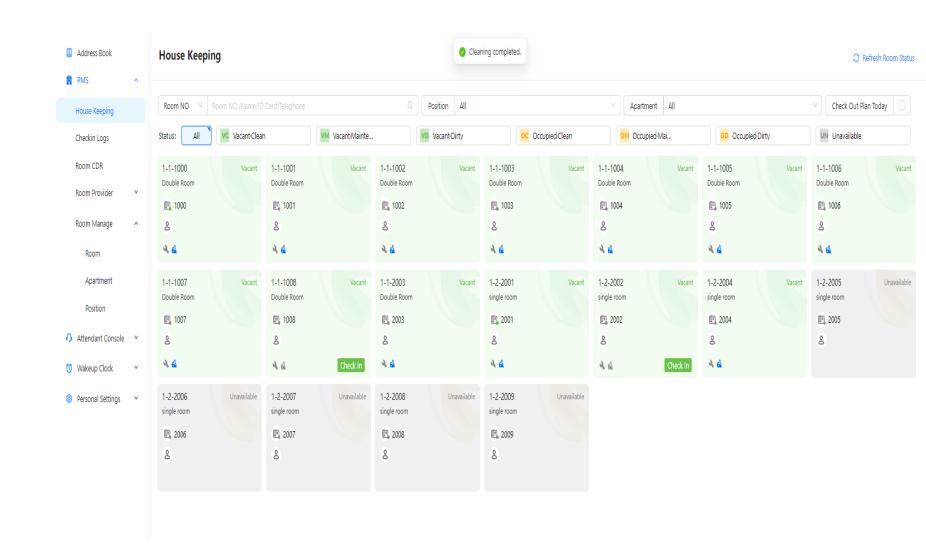
#### Main function

Room Manage

Check-In

Room Call

**Room Service** 





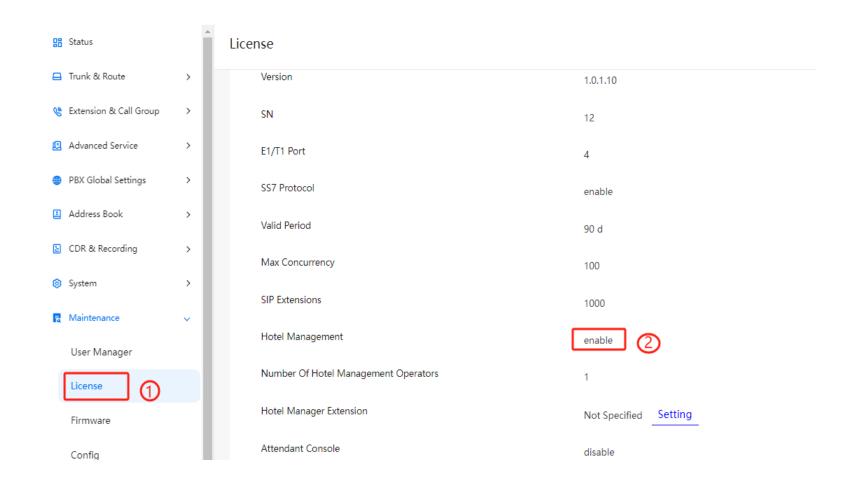
License view

SIP Stack setting

SIP Extension setting

Hotel Manager Extension setting

- 1. Click Maintenance->License
- View Hotel Manager permissions





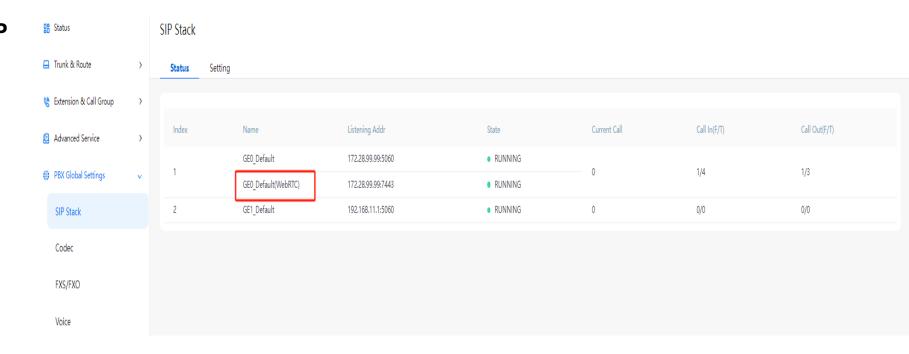
License view

SIP Stack setting

SIP Extension setting

Hotel Manager Extension setting

- Click PBX Global Setting->SIP Stack
- 2. Enable WebRTC option





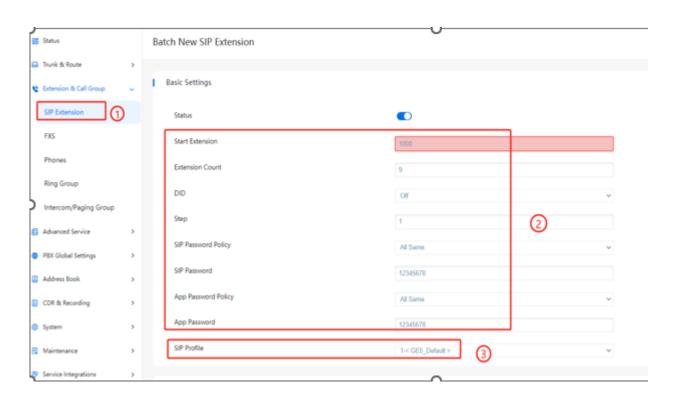
License view

SIP Stack setting

SIP Extension setting

Hotel Manager Extension setting

- 1. Click Extension & Call Group->SIP Extension
- 2. Create Batch new SIP extension, configure the start extension number and password
- 3. Select network port





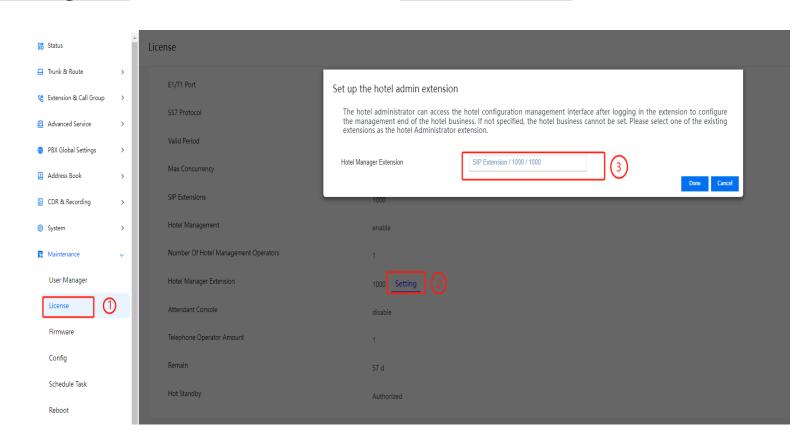
License view

SIP Stack setting

SIP Extension setting

Hotel Manager Extension setting

- 1. Click Maintenance->License
- 2. Click setting Hotel Manager Extension
- Select extension number





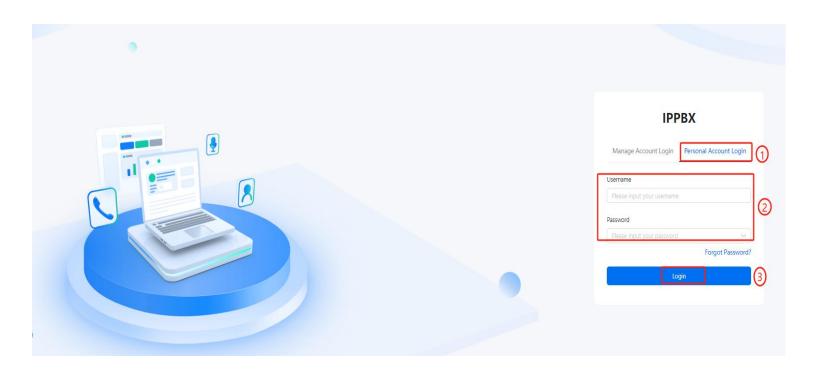
License view

SIP Stack setting

SIP Extension setting

Hotel Manager Extension setting

- 1. Select Personal Account Login
- 2. Fill in the hotel manager extension number and app password
- 3. Click login



### How To Use PMS



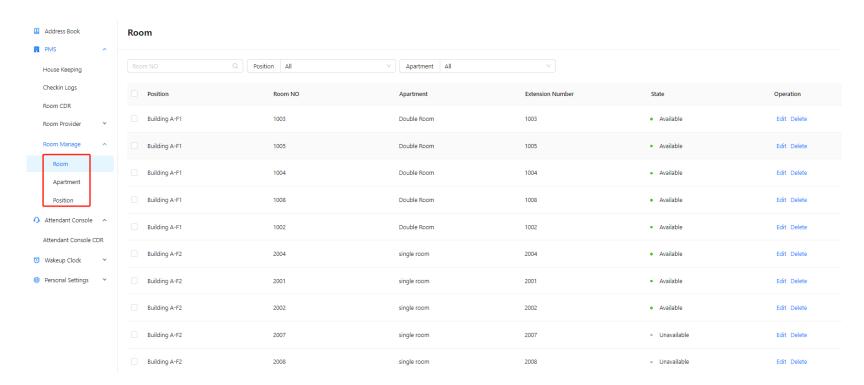
### • Room manage

**Room**: select the location information of the hotel room and bind the room extension and corresponding room number

**Apartment**: Customize room types and number of occupants

**Position:** Set building and floor

information



### How To Use PMS

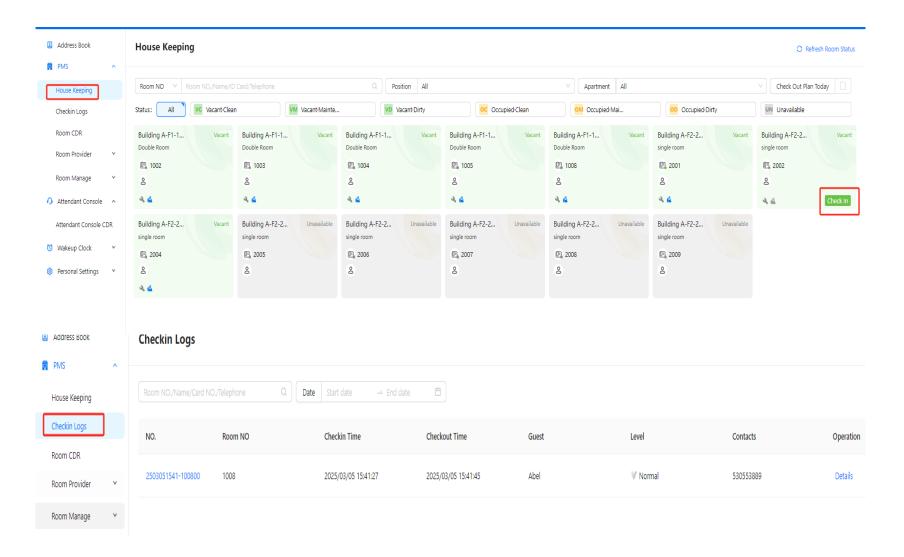


### House keeping

Check the cleaning status of the room, and the cleaned room can be checked in

### Checkin logs

Can be viewed after check-out



### How To Use PMS

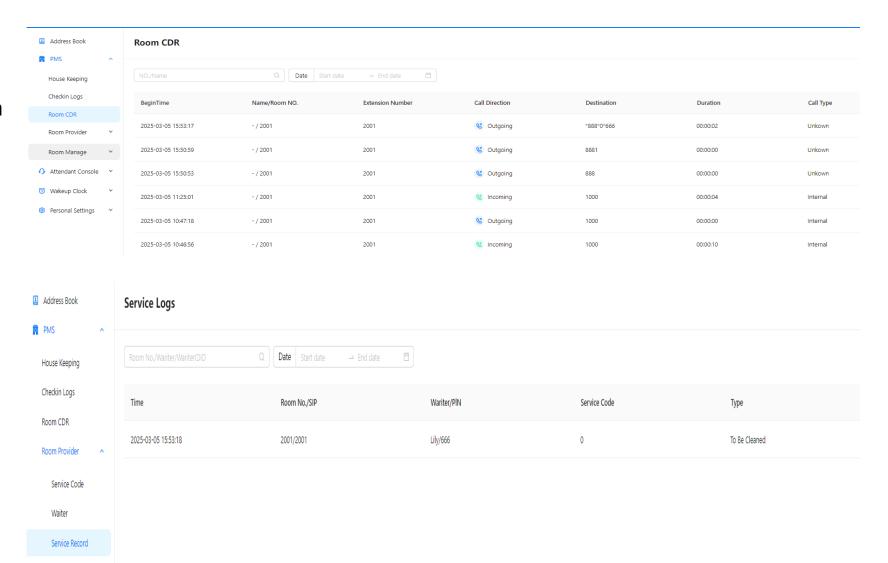


#### Room CDR

Display the call history of the room extension

#### Room Provider

Customizable access number, service code, and waiter for calling corresponding services at room extensions



### What is Attendant Console



#### Attendant Console

Providing an efficient and convenient operating platform for operators

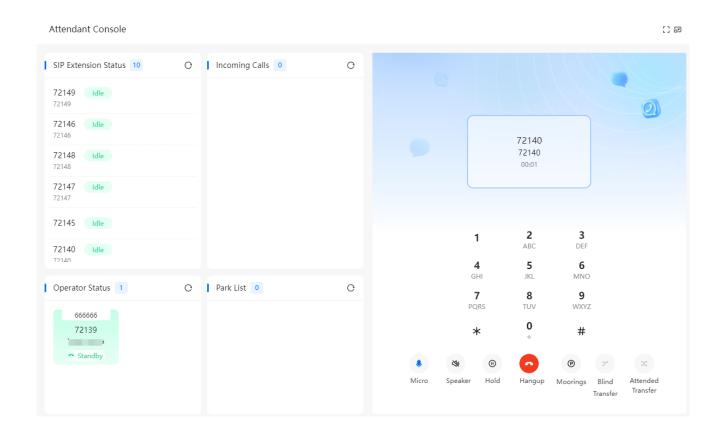
#### Main function

Call answering and hanging up

Call hold/call moorings/ call forwarding

DND

Alarm



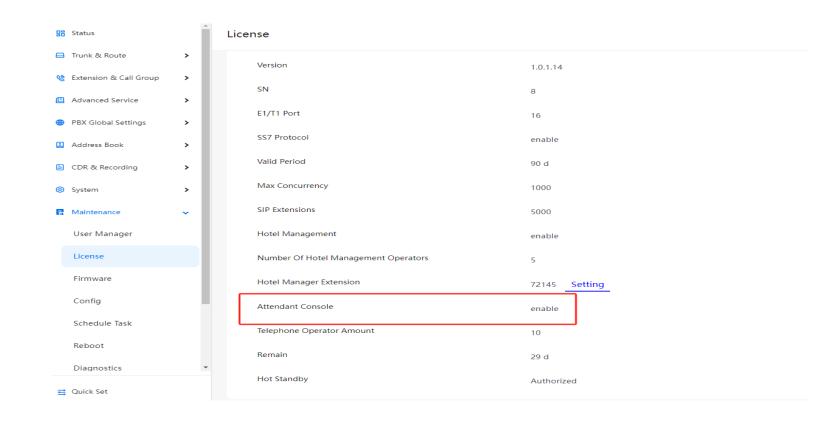


License view

Call Queue setting

Attendant Console login

- 1. Click Maintenance->License
- 2. View Attendant Console permissions



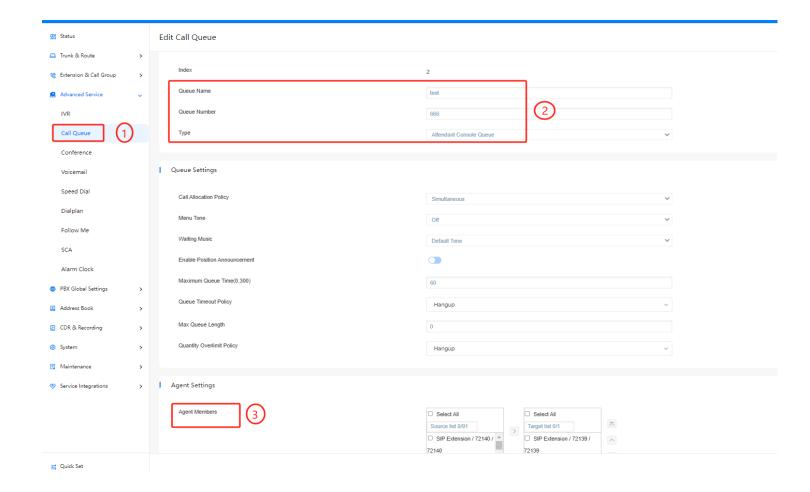


License view

Call Queue setting

Attendant Console login

- 1.Click Advanced Service->Call Queue
- 2. Set queue number and type
- 3.Set Agent Members



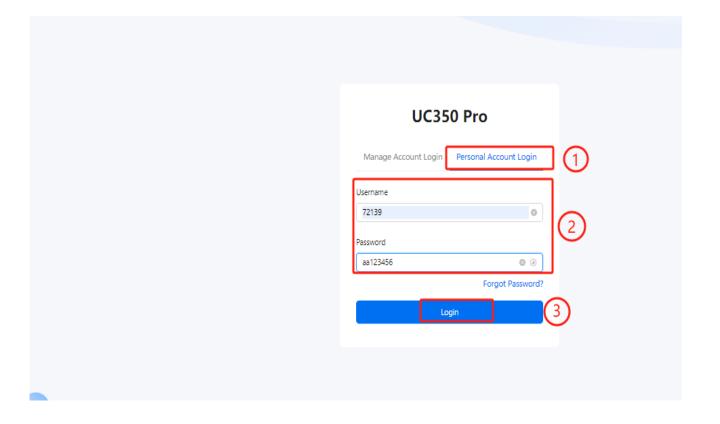


License view

Call Queue setting

Attendant Console login

- Select Personal Account Login
- 2. Fill in the Agent Members extension number and app password
- 3. Click login



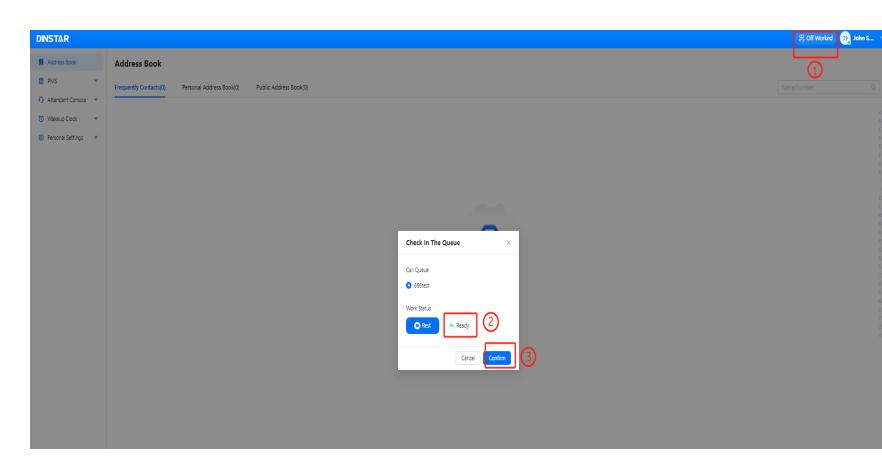


License view

Call Queue setting

Attendant Console login

- 1. Click Check in The Queue
- 2. Click ready
- 3. Click Confirm



### How To Use Attendant Console

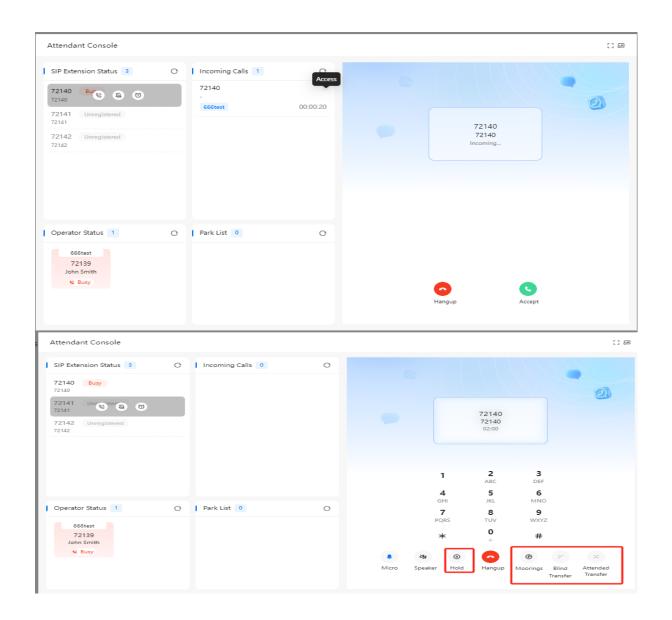


Call answering and hanging up

When receiving a call, users can choose accept or hangup

Call hold/call moorings/ call forwarding

After answering the phone, users can choose call hold, mooring or transfer



### How To Use Attendant Console

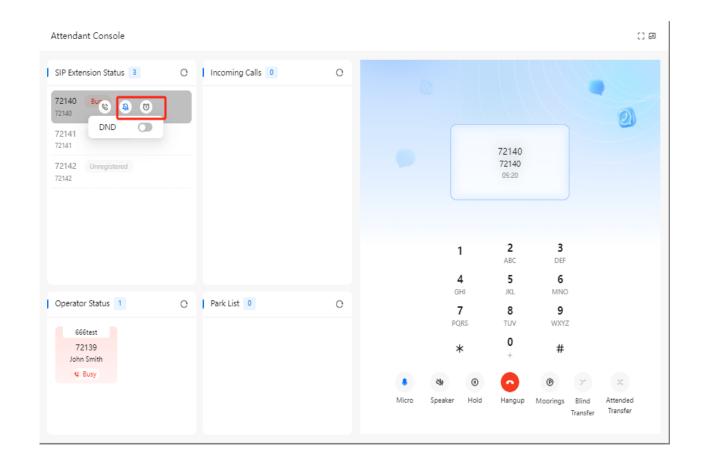


#### DND

After enabling DND, calling the extension will fail

#### Alarm

After setting the alarm task, the extension will ring at the designated time and play alarm music upon answering



# Why Need Nat Public Proxy Server



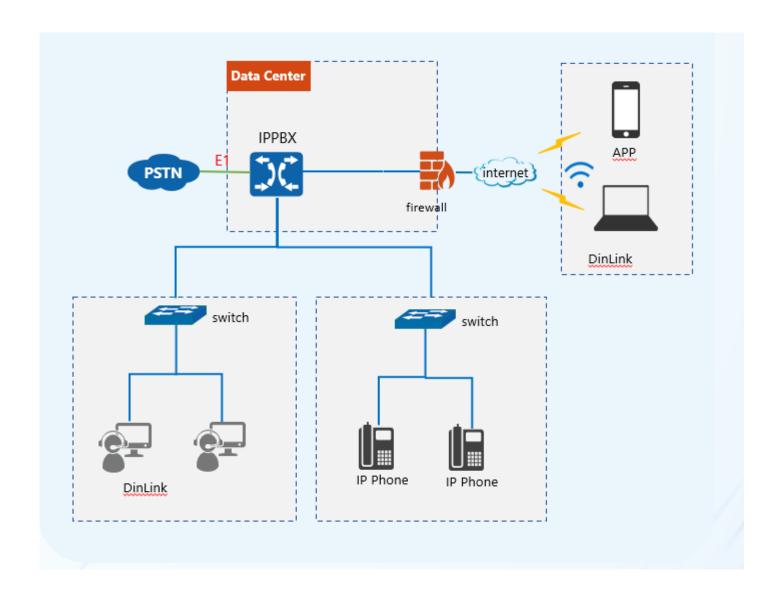
#### Main Scenarios

- 1.UC deployment on enterprise intranet
- 2. Outward personnel may need to use a mobile app to access UC through external public WIFI or mobile data

#### Main function

Signal penetration, using frpc and frps for forwarding

Media penetration, using Stun server to penetrate through ice



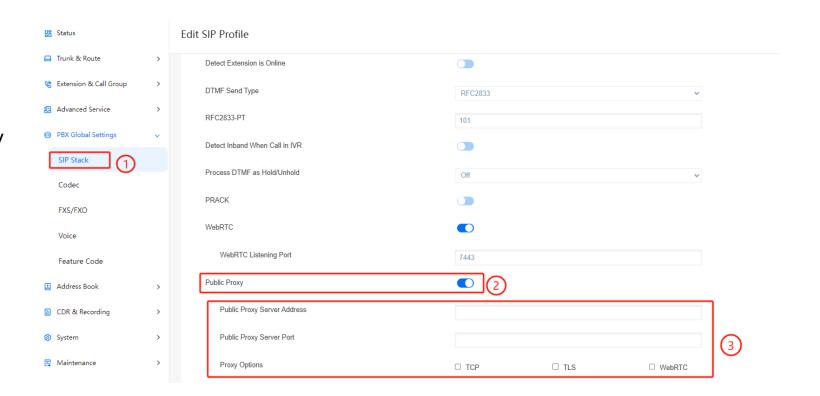
# How To Configure Nat Public Proxy Server



1.Click PBX Global Settings->SIP stack

2. Edit SIP Profile and enable public proxy function

3. Set the address and port of the public proxy server, and select the proxy options



















+86 755 6191 9966