UC350 /UC350 Pro IPPBX User Manual

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Preface

Welcome

Thanks for choosing the Dinstar's Product! We hope users will make full use of this rich-feature product. Contact us if users need any technical support: 0755-61919966.

About This Manual

This manual provides information about the introduction of the device, and about how to install, configure or use the device. Please read this document carefully before install the device.

Intended Audience

This manual is aimed primarily at the following people:

- Users
- Engineers who install, configure, and maintain the device.

Document Information

Document Name	UC350 /UC350 Pro IPPBX User Manual
Document Version	V1.0

Conventions

Gateway or device mentioned in this document refers to the UC350 series IPPBX. "Note" marked in the document is what users need to pay attention to.

Security Statement

●Password Configuration and Modification Statement

To ensure the security of UC350 series devices, it is strongly recommended that you change the default password upon your first login and regularly update your password periodically.

● **Personal Data Statement**

During the operation, fault diagnosis, or log auditing processes of your purchased products, services, or features, certain user personal data (such as end-user MAC addresses, end-user IPs, etc.) may be collected or stored. Therefore, it is your responsibility to develop necessary user privacy policies and implement sufficient measures to ensure that user personal data is adequately protected, in accordance with applicable national laws. Any log or diagnostic data that needs to be transmitted out of the customer's network must receive authorization from the customer. Furthermore, any personal data included in the data to be transmitted must be anonymized to ensure that personal data cannot be restored in any way.

Contents

1 Product Introduction

1.1 Overview

Dinstar UC350 series IPPBX is a high-performance office phone system designed for medium to large enterprises. It facilitates the creation of efficient IP phone systems for business and industry clients. The UC350 series features a 19-inch, 1U rack-mounted design with four user board interfaces, supporting FXS, FXO, and E1 interfaces.

There are two models: UC350 and UC350 Pro. The UC350 supports up to 1,000 registered extensions and 120 concurrent calls, while the UC350 Pro supports up to 5,000 registered extensions and 500 concurrent calls. Both models enable remote work via SIP terminals and can integrate with other IPPBX or traditional PBX systems to meet diverse user needs.

The UC350 series IPPBX adopts multiple encryption and security strategies to ensure system safety. It is suitable for small to medium call centers and remote branch deployments, enhancing communication efficiency, reducing costs, and promoting cost-effective digital transformation.

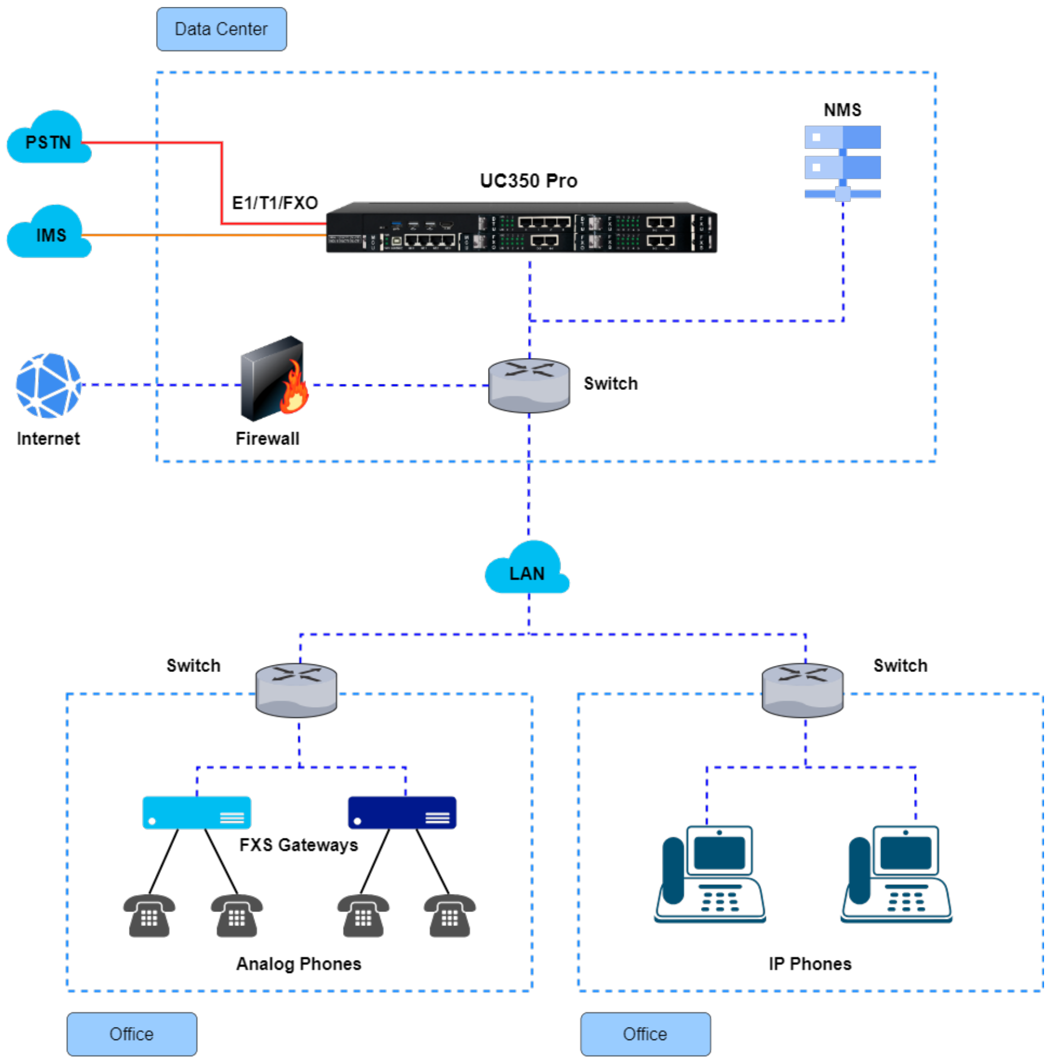
1.2 Main specification

specifications	UC350	UC350 Pro
SIP Users	1000	5000
Concurrent Calls	120	500
Concurrent Recording	60	100

Participants Per Session	30	30
MCU Board Slots	1	1
Gigabit Ethernet Ports	2	4
USB2.0	1	2
USB3.0	—	1
Console Port	1*RJ45	1*USB-B
User board Slots	4	4
FXS Board (8 FXS Ports)	2*RJ45	2*RJ45
FXO Board (8 FXO Ports)	2*RJ45	2*RJ45
FXU Board (4 FXS and 4 FXO Ports)	2*RJ45	2*RJ45
DTU Board (4 E1/T1 Ports)	4*RJ45	4*RJ45
1+1 Power Supply (100-240 VAC, 50/60 Hz)	✓	✓
Dimensions (W/D/H)	437*345*49 mm	437*345*49 mm
Power Consumption	50W	55W
Weight	5.7kg	5.6kg
Operating Temperature	0 ℃ ~ 45 ℃	0 ℃ ~ 45 ℃
Storage Temperature	-20 ℃~80 ℃	-20 ℃ ~80 ℃
Humidity	10%~90% Non-Condensing	10%~90% Non-Condensing

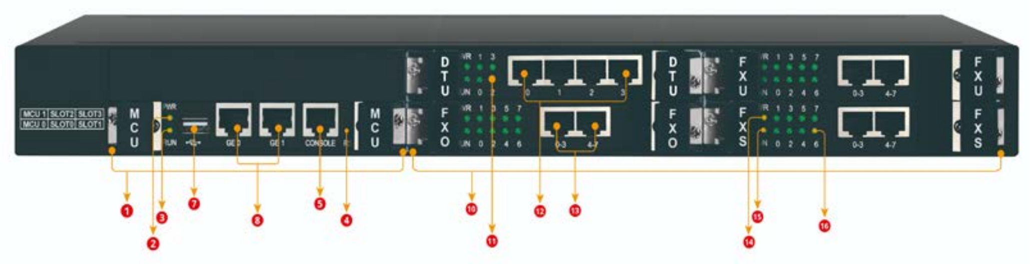
1.3 Typical Application Scenario

The typical application scenario of UC350 series IPPBX is shown as the follow(the scenario is based on the UC350 Pro):

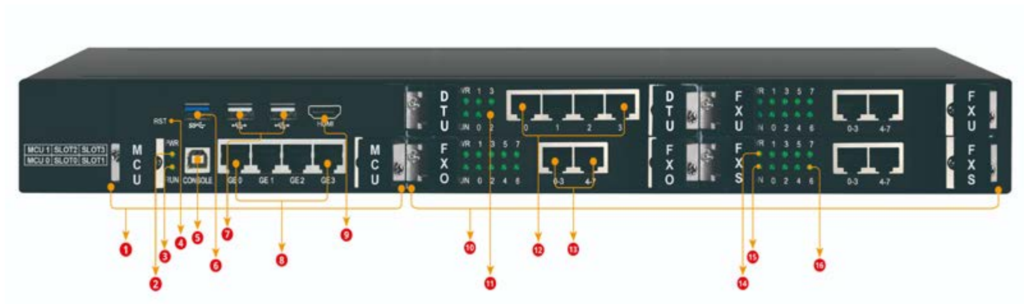


1.4 Product Appearance

Front View(UC350):



Front View(UC350 Pro):



Back View:

Note:	The device supports four different types of user boards (FXS/FXO/FXU/DTU). The FXS user board supports 8 FXS ports, the FXO user board supports 8 FXO ports, the FXU user board supports 4 FXS and 4 FXO ports, and the DTU user board supports 4 E1/T1 ports. If installed with E1 user boards, the RJ48 ports are connected to the E1 lines. If installed with FXS user boards, the RJ11

ports are connected to the FXS telephone or Fax. If installed with FXO user boards, the RJ11 ports are connected to the PSTN cable.

No.	Definition	Description
1	MCU Board Slot	For installing the MCU Board
2	Power Indicator	show the device power status
3	Running Indicator	show device system running status
4	RST Button	To restart the device.
5	Console Port	Console port for debugging and configuring the device
6	USB3.0 Interface	USB 3.0 interface and it is compatible with USB 2.0
7	USB2.0 Interface	The USB interface supports 2.5-inch external hard drives
8	Network Port	Connect to IP network via DSL modem or LAN switch or router
9	HDMI Interface	Reserved interface for connecting a monitor
10	User Board Slot	For installing the User Board
11	E1/T1 Indicator	Indicate E1/T1 line status
12	E1/T1 Interface	RJ48 interface, for connecting E1/T1 lines
13	FXS/FXO Interface	RJ45 interface, can use RJ45-RJ11 adapter to connect analog phone or PSTN line

14	User Board Power Indicator	Indicate user board power status
15	User Board Run Indicator	Indicate user board running status
16	FXS/FXO Indicator	Indicate FXS/FXO port occupancy status
17	Grounding Lug	To connect to grounding wire
18	Power Switch	Power switch button
19	Power Jack	Power interface: 100~240VAC, 50/60HZ

1.5 Description of All Boards

Type	Indicator	Definition	Status	Description
MCU Board	PWR	Power Indicator	Off	There is no power supply or power supply is abnormal
			On	The device is powered on
	RUN	Running Indicator	Flashing	The device is initialized successfully and running normally
			On	The system is initializing
			Off	The device is not running normally

FXS/FXO/FXU User Board	PWR	Power Indicator	On	The power supply is normal
			Off	The power supply is not normal
	RUN	Running Indicator	Off	The system is starting up
			Fast Flashing	Part of the port registered successfully
			Slow Flashing	All ports are registered
	FXS/FXO	FXS/FXO Indicator	On	The FXS port is in off-hook (in-use) status
			Off	The FXS port is in on-hook status.
DTU User Board	PWR	Power Indicator	Off	There is no power supply or power supply is abnormal
			On	The device is powered on
	RUN	Running Indicator	Slow Flashing	The device is initialized successfully and running normally
			On	The system is initializing

			Off	The device is not running normally
	E1/T1	E1/T1 Indicator	On	E1/T1 line is connected
			Off	E1/T1 line is disconnected

1.6 Features & Functions

1.6.1 Key Features

- FXU board supports power failure lifeline.
- Supports E1/T1, FXO and FXS ports with the flexible and alternative capability.
- Distributed multi-core CPU, greatly improves the call processing capacity.
- Flexible dial rules based on time, number or source IP etc.
- Supports Multi-level IVR, helps to build personalized voice navigation for enterprise.
- Supports voicemail/ Voice recording.
- User-friendly web interface, classification of web user's access permission.

1.6.2 FXS/FXO

- Connector: RJ45
- Supports 8 FXS or 8 FXO or 8 mixed FXS/FXO ports
- Caller ID: Bellcore Type 1&2, ETSI, BT, NTT and DTMF
- Answer and Disconnect Signaling: Answer, Disconnect, Busy Tone, Polarity Reversal, Hook Flash
- Caller ID Detection: FSK, DTMF
- Delayed Answer Lifter Busy Tone Detection
- No Current Hang-up Detection

1.6.3 E1/T1

- Interface: RJ48 (120Ohm)
- Supports E1 line to traditional PSTN network
- R2 MFC
- ISDN PRI: 23B+D (T1), 30B+D (E1), NT or TE can configure ITU-T Q.921, ITU-T Q.931, Q.Sig
- Signal 7/SS7: ITU-T, ANSI, ITU-CHINA, MTP1/MTP2/MTP3, TUP/ISUP
- E1 Frame Format: DF, CRC-4, CRC_ITU
- T1 Frame Format:
- 2-Frame Multi-frame(F12, D3/4), Extended Super-frame(F24, ESF)
- Line Code: HDB3(E1), B8ZS(T1)
- Clock Source: remote/local Clock Source, Each DTU can be configured independently

1.6.4 Voice Capabilities

- VoIP Protocols: SIP over UDP/TCP/TLS, SDP, RTP/RTCP/SRTP/ZRTP
- Audio Codecs: PCMU, PCMA, G.723, G.729, G.722, OPUS, G.726-16, G.726-24, G.726-32, G.726-40
- Video Codecs: VP8/H.264/H.263/H.263-1998/H.263-2000/H.261
- Silence Suppression
- Voice Interrupt Protection
- Comfort Noise Generator (CNG)
- Voice Activity Detection (VAD)
- Echo Cancellation: G.168 with up to 128ms
- Dynamic Jitter Buffer
- Adjustable Gain Control
- Automatic Gain Control (AGC)

- Call Progress Tones: Dial Tone, Ring Back Tone, Busy Tone
- FAX: T.38 and Pass-through
- NAT: STUN, Rport, DDNS, Static IP
- DTMF: RFC2833/Signal/Inband
- FAX: T.38 and Pass-through

1.6.5 PBX Services

- 3-Way Conference
- Voicemail to Email
- CDRs, Multi-level IVR
- Auto-attendant Function
- Voicemaill, Voice Recording
- Event Report, Email Client
- Zero configuration of the phone
- Dial Rules, Failover Routing
- Routing Groups, Ring Group, Call Queue
- Paging/Intercom, Hotline, Do-not-disturb
- Routing Based on Time Period
- Routing Based on Source Trunks
- Routing Based on Caller/Called Prefixes
- Caller/Called Number Manipulation
- Call Forward (Unconditional/No Answer/Busy)
- Call Waiting/Call Holding/Call Transfer
- Surpport Dinlink (APP)
- Attendant Console

- PMS (Property Management System)

1.6.6 Maintenance

- Web GUI Configuration
- Configuration Restore/Backup
- Multiple Languages Supported
- HTTP Firmware Upgrade
- Syslog, Ping/Nslookup/Traceroute
- Auto Provision
- Traffic Statistics: TCP, UDP, RTP
- Network Capture
- NTP, FTP Server
- Classification of Web Users' Access permission
- HTTP&HTTPS/NATS API
- Schedule Task, Event Report
- Remote management via cloud services
- Firewall, Hosts

2 Installation

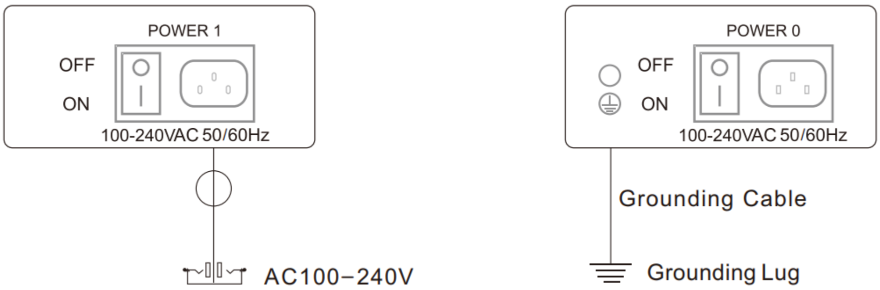
2.1 Installation Attentions

To avoid unexpected accident or device damage, please read the following instructions before users install the UC350 series gateway.

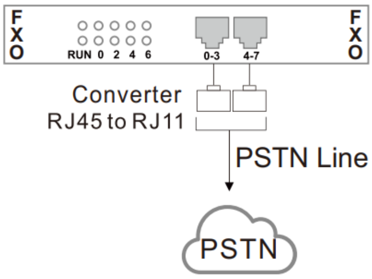
- The adapter of the gateway accepts DC220V 10A dual power input. Please ensure it is stable and safe power supply.
- To reduce the interference to telephone calls, please separate power cables from telephone lines.
- To guarantee stable running of the gateway, please make sure that there is enough network bandwidth.
- For better heat dissipation, please place the gateway on a level surface and do not pile up with other devices.

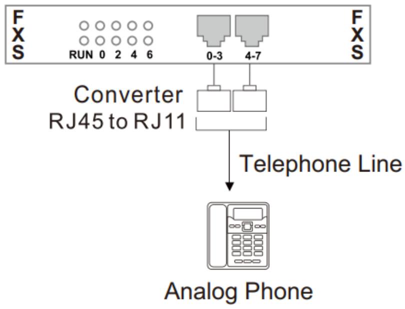
2.2 Installation Instruction

1) Connect with Power Input and Grounding Lug.

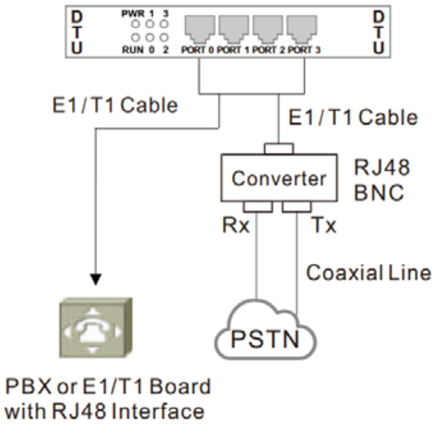


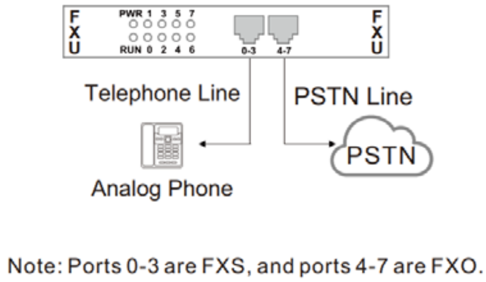
2) Connect telephone line to the FXS port and connect PSTN line to the FXO port.



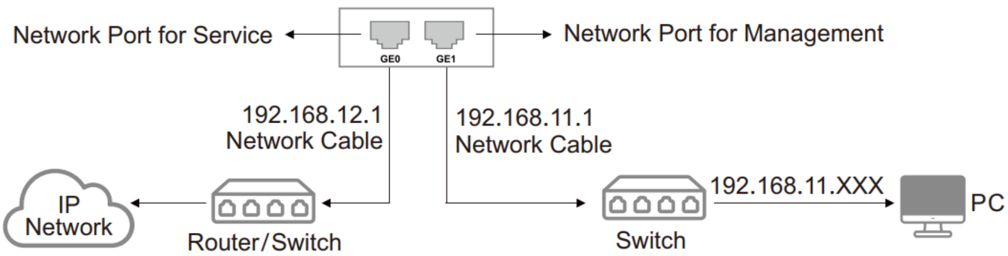


3) Connect E1/T1 cable to E1/T1 ports; connect telephone line and PSTN line to the FXS port and FXO port (FXU Board)





4) Connect network cable to the GE0 port for service, and the GE3 or GE1 port is connected to the PC for management.

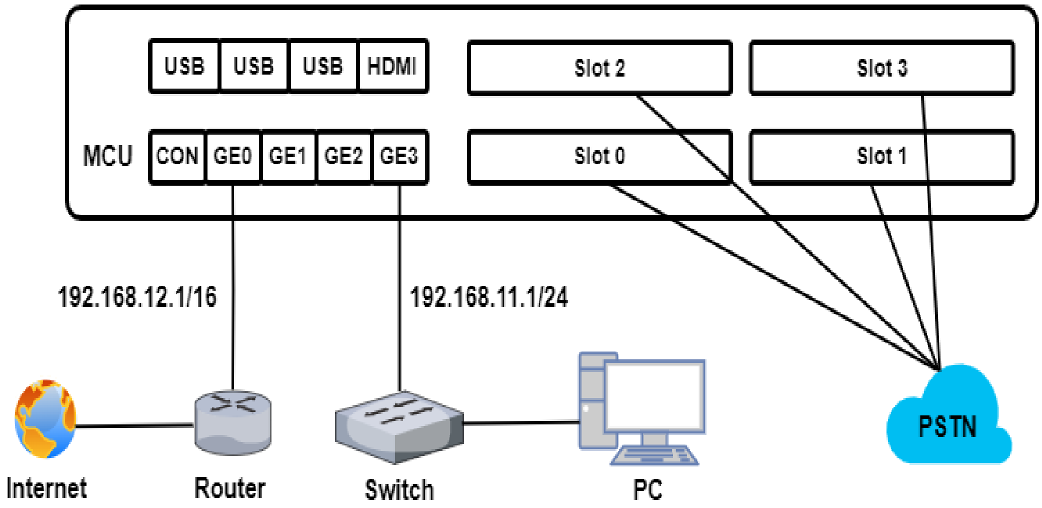


Note: The UC350's default network port for management is the GE1 port, and the UC350 Pro's default network port for management is the GE3 port.

2.3 Network Connection

The UC350 series IPPBX support Gigabit Ethernet port. Among them, UC350 provides 2 RJ45 interfaces and UC350 Pro provides 4 RJ45 interfaces. The default IP address of the management port of UC350 series devices is 192.168.11.1. Users can modify the IP addresses of other service ports for accessing the external network, and the service ports need to be configured with static IPv4 addresses in the same network segment as the uplink, as shown in the following figure:

Figure-Network Connection(using UC350 Pro as an example)



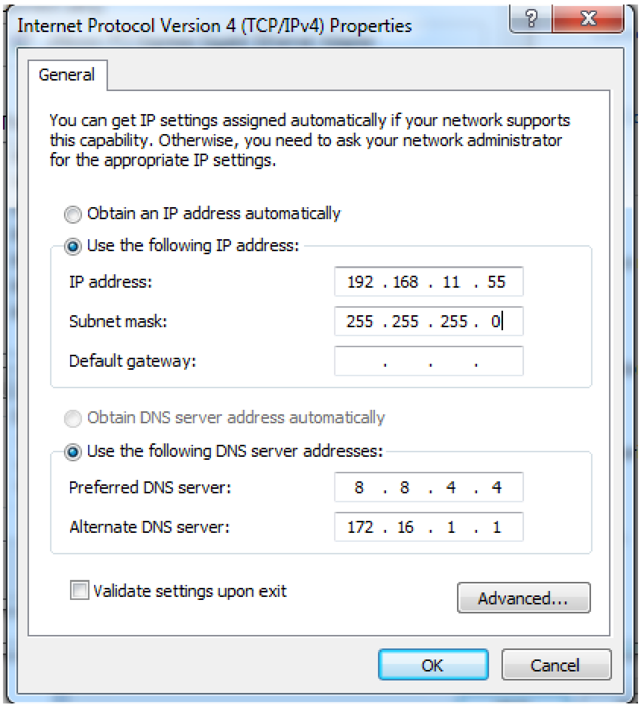
2.4 Connect Gateway to Network

2.4.1 Preparations for Login

Since the default IP address of management port is 192.168.11.1, user has to modify the IP address of the PC to make it at the same network segment with the UC350 Series IPPBX.

Take the Windows 7 operating system as an example, edit the IP address of the local computer and the gateway to be in the same network segment.

Figure-Modify the local computer address



Check the connectivity between the PC and the UC350 Series IPPBX. Click **Start Run** of PC and enter **cmd** to execute 'ping 192.168.11.1' to make sure the IP address of management port is pinable.

```
C:\Users\DEEP>ping 192.168.11.1

Pinging 192.168.11.1 with 32 bytes of data:
Reply from 192.168.11.1: bytes=32 time=1ms TTL=64
Reply from 192.168.11.1: bytes=32 time=1ms TTL=64
Reply from 192.168.11.1: bytes=32 time<1ms TTL=64
Reply from 192.168.11.1: bytes=32 time<1ms TTL=64

Ping statistics for 192.168.11.1:
    Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
Approximate round trip times in milli-seconds:
    Minimum = 0ms, Maximum = 1ms, Average = 0ms
```

2.4.2 Log In Web

Open a browser and enter the IP address of management port (the default IP is 192.168.11.1). Then the login GUI will be displayed.

Users also can enter the IP address of service port, but it's required to modify the IP address of PC to make it at the same network segment with that of service port.

It is suggested that users should modify the username and password for security.

Figure- Login GUI of UC350 Series IPPBX

Note: When logging into the device's system, you need to use the HTTPS protocol!

By default, the username is **admin**, while the password is **admin@123#**. After entering username and password, click **Login** to enter into the web interface.

Under some circumstances, login of the Web will be limited:

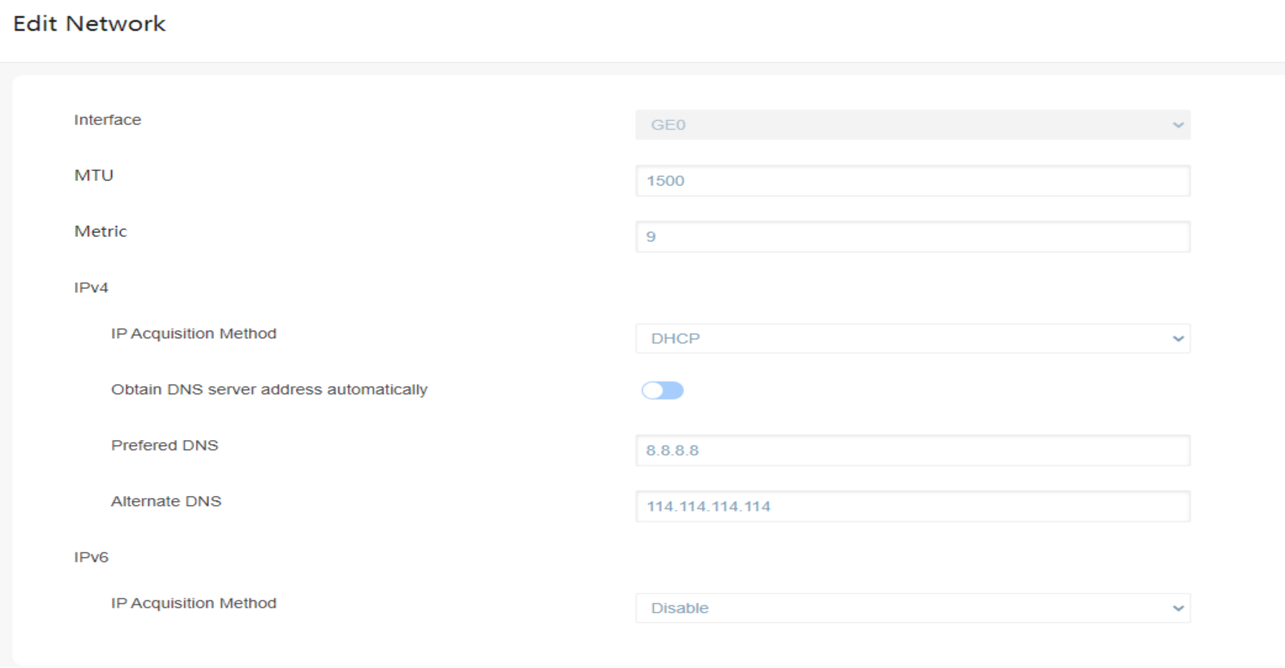
- For three consecutive login failures, users need to slide to validate user account
- Failing to log in the Web for ten times consecutively, the IP address of the UC350 Series IPPBX will be put into the blacklist, and users need to reset a new IP address for the device;
- Successful login or device restart will wipe out login failure records.

2.4.3 Modify the IP address

Connect UC350 Series IPPBX to the network (refer to figure-network connection) before modifying IP address. Users need to modify the IP address of the service ports so that the service ports and the upstream network are in the same network segment.

Please go to **System** > **Network** to modify the IP address.

Figure-Modify IP address via Web



3 Basic Operation

3.1 The Methods of Dialing

There are two methods to dial telephone or extension number:

- Dial the called number and wait for 4 seconds for dialing timeout, or dial the called number directly (the system will judge whether the dialing is completed according to Digitmap and Regular Expression dialplans).
- Press # after dialing the called number to end.

3.2 Call Holding

The current call can be held by pressing the "flash" key on the phone (if available), and then pressing the "flash" key again to resume the held call. If there is no "flash" key, users can use "hook flash" instead.

3.3 Call Waiting

When call waiting is enabled, if users hear the call waiting voice (three beeps of the FXS extension) during a call, it indicates that a new call is incoming. Users can switch between the incoming call and the current call through the Flash key or hook flash.

3.4 Instruction of Hook Flash

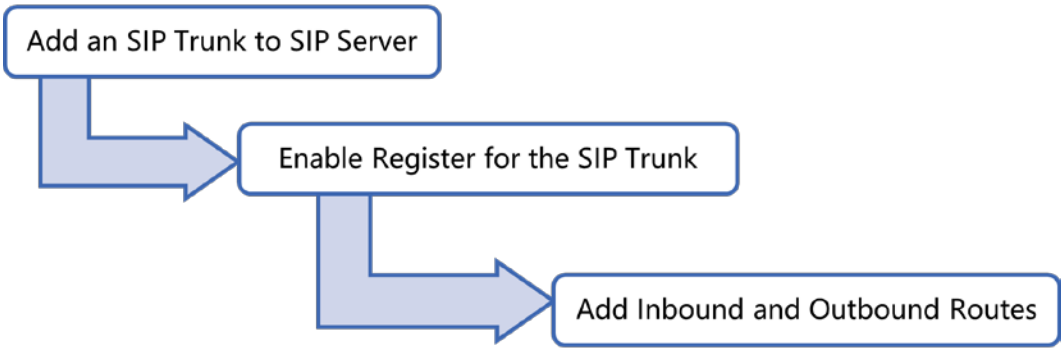
A talks with B, A dials the number of C after pressing the hook flash, A and C talk, and the conversation between A and B is kept (B hears the waiting tone). At this time, A can switch to the call with B by hook flash and pressing the 1 key, switching to the call with C by hook flash and pressing the 2 buttons, and entering the three-way call by hook flash and pressing the 3 buttons.

4 Configuration Wizard

The following are the common ways to configure the UC350 series IPPBX.

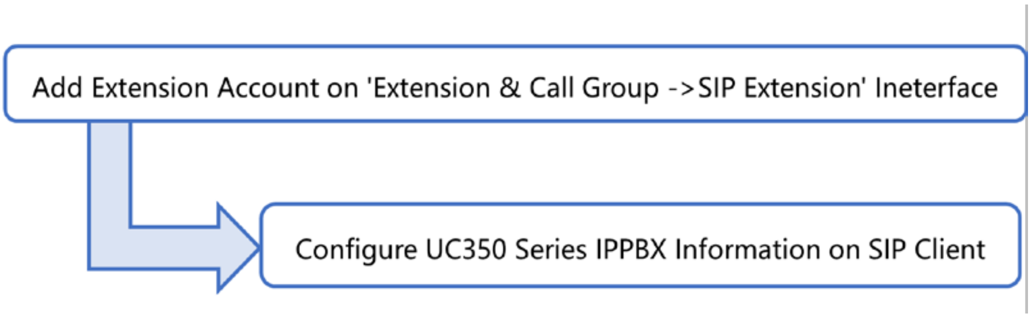
4.1 Regarded as Terminal and Registered to SIP Server

UC350 Series IPPBX Registered to SIP Server

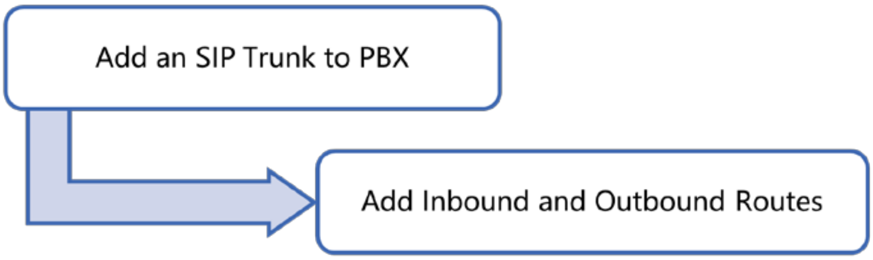


4.2 Other SIP Clients registered to UC350 Series IPPBX

Under this mode, UC350 Series IPPBX is regarded as an SIP Server. Create an extension account first on the **Extension & Call Group > SIP Extension** interface, and configure listening port on the **PBX Global Settings > SIP Stack** interface. Then, configure the server and account on SIP clients.



4.3 Connected to PBX through Trunking



5 Web Platform

5.1 Status

Modify the IP address of PC to make it at the same network segment with that of management port of the UC350 Series IPPBX (the default IP of management port is 192.168.11.1).

Open a web browser on the PC and then enter the IP address of management port. Click **Login**, and the login GUI is displayed. The default username and password are **admin / admin@123#**.

The displayed login GUI is shown as follows:

Figure-Introduction to login GUI

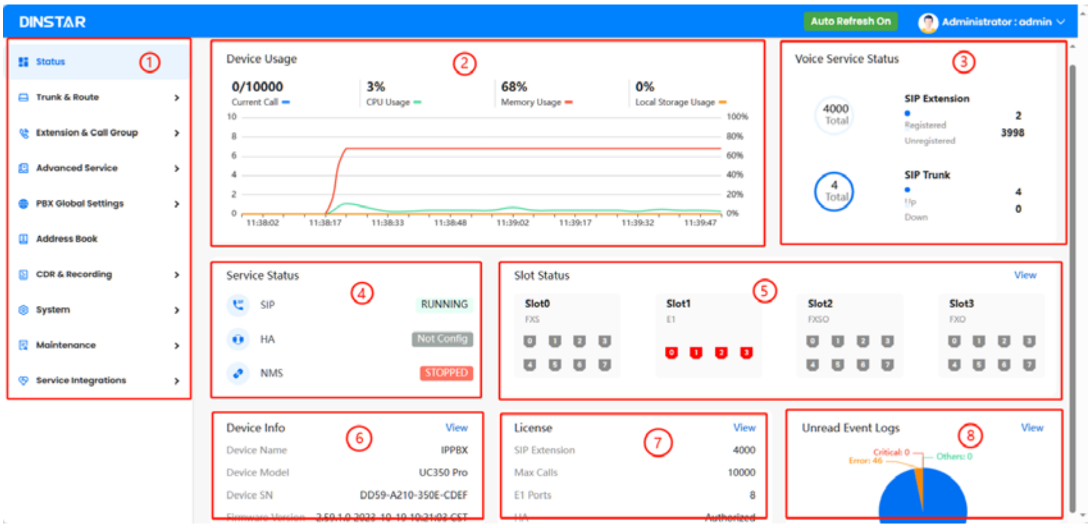


Table-Introduction of Web Interface

Index	Item	Description
1	Menu Bar	The menu bar of UC350 Series IPPBX

2	Device Usage	The status of device usage including Current Call, CPU Usage, Memory Usage and Local Storage Usage
3	Voice Service Status	The status of voice service including status of SIP Extension and status of SIP Trunk
4	Service Status	The status of service including SIP Status, HA Status and NMS Status
5	Slot Status	The status of each slot and each port
6	Device Info	The detailed information of device, including Device Name, Device Model, Device SN, Firmware Version and so on
7	License	The detailed information of License, including SIP Extension, Max Calls, E1 Ports, HA and so on
8	Unread Event Logs	The Event Notification of the device

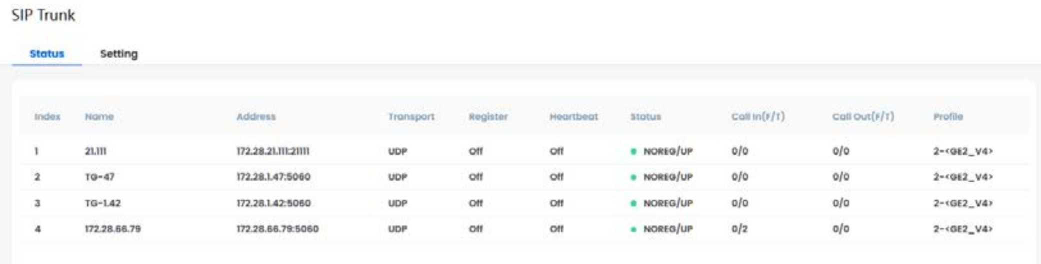
5.2 Trunk & Route

5.2.1 SIP Trunk

A SIP trunk is a trunk group by sip connection that enables users to make outgoing and receive incoming calls from sip/IMS. It empowers businesses to place local or long-distance calls over the internet, without relying on traditional phone lines. And SIP trunk can connect UC350 Series IPPBX with other PBX or SIP servers.

On the **Trunk & Route > SIP Trunk > Status** page, users can view the status of the configured sip trunks. On the **Trunk & Route > SIP Trunk >Settings** page, users can create, delete, edit, or disable SIP trunks.

Figure-Status of SIP Trunk



Index	Name	Address	Transport	Register	Heartbeat	Status	Call In(F/T)	Call Out(F/T)	Profile
1	21.111	172.28.21.111:21111	UDP	Off	Off	NOREG/UP	0/0	0/0	2-<GE2_V4>
2	TG-47	172.28.1.47:5060	UDP	Off	Off	NOREG/UP	0/0	0/0	2-<GE2_V4>
3	TG-1.42	172.28.1.42:5060	UDP	Off	Off	NOREG/UP	0/0	0/0	2-<GE2_V4>
4	172.28.66.79	172.28.66.79:5060	UDP	Off	Off	NOREG/UP	0/2	0/0	2-<GE2_V4>

Figure-Parameters of SIP Trunk

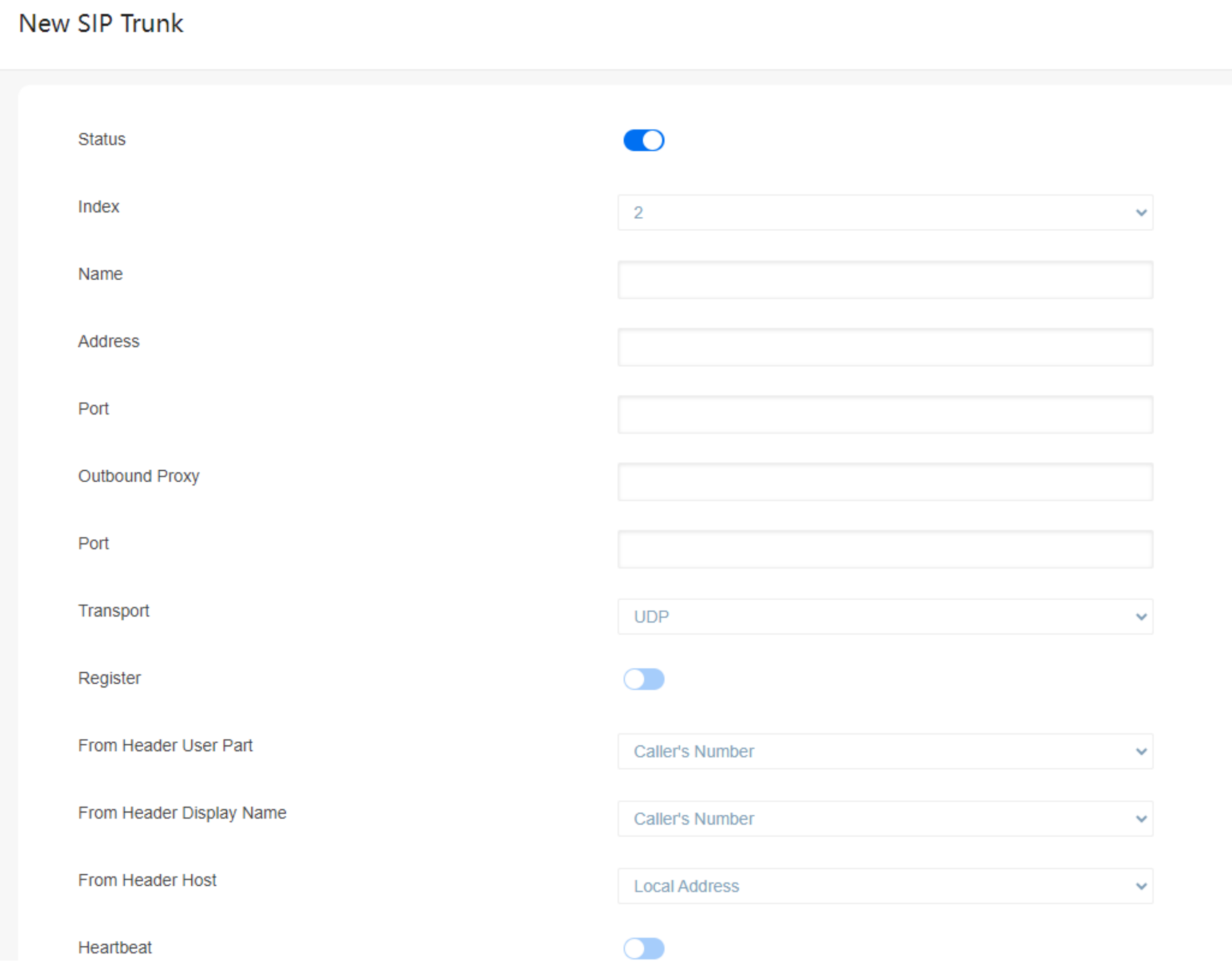


Table-Parameters of SIP Trunk

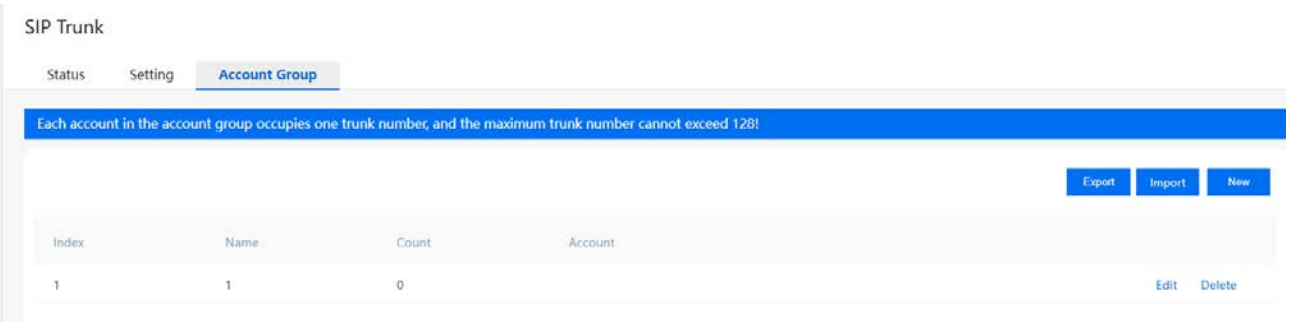
Parameter	Description
Status	Enable or disable SIP Trunk.
Index	Index of SIP Trunk. Range from 1 to 32.
Name	The name of the SIP trunk. The input value is text type and cannot be null. The value is up to 32 characters and cannot contain " ".
Address	The IP address or domain name of the peer SIP devices or servers.
Port	The SIP listening port of the peer SIP devices or servers. 5060 is the default port.
Outbound Proxy	If outbound proxy is used, enter the IP address or domain name of the proxy server.
Port	If outbound proxy is used, enter the listening port of the proxy server.
Transport	Transport protocol: TCP or UDP.
Register	If it is on, the SIP trunk will send register request to the peer device.
Register Way	single register or account group
Username	The username of this SIP trunk, it is generally a phone number.
Auth Username	The username used for register authentication by this SIP trunk.

Password	The password used for register authentication by this SIP trunk.
Specify Transport Protocol on Register URL	When enabled, it specifies the current transport protocol on the Register URL
Expire Seconds	The validity period after the SIP trunk is registered successfully. When the time expires, the SIP trunk will send register request to the server. Default value is 1800s.
Retry Seconds	Time interval to re-initiate registration if registration fails. Default is 60s.
From Header User Part	Caller's Number, Caller's Display Name, and Custom can be selected. The default is Caller's Number. When selecting Custom, users need to enter text, which cannot be null, up to 32 characters and cannot include " ".
From Header Display Name	Choose the registered username or the true caller ID for the 'from header' of the invite message when a call goes out.
From Header Host	Local Address, Server Address, Custom can be selected. The default is Local Address. When selecting Custom, users need to enter text, which cannot be null, up to 32 characters and cannot include " ".

Heartbeat	If heartbeat in on, heartbeat (options) messages will be sent to examine the connection with servers. The default value is 'Off' .
Heartbeat Period(s)	The interval between each heartbeat message OPTION.
AutoCLIP Profile	AutoCLIP is mainly used to SIP trunks/FXO trunks and it helps record the outgoing and incoming calls of a trunk.
DNIS	When the SIP trunk calls in, the called number matches the DNIS, then the from display name of the invite should be the display name of the called number.
Called Number / Display Name	The called number and display name of DNIS.
SIP Profile	The SIP profile of the SIP Trunk, make reference to Profile SIP section.
Outbound Codec Profile	Select the Outbound Codec Profile to use or click on New to create it.
Manipulation for Call In	Select the configuration for “Trunk&Route-manipulation”
Manipulation for Call Out	Select the configuration for “Trunk&Route-manipulation”

Extra Param	It is developer-configurable feature. It allows users to configure customized extra parameters supported by the system.
Area Code	Fill in the area code corresponding to the configured trunk. Corresponding to international/domestic. To be used in conjunction with the Area Call Permission of the SIP extension
Inbound/Outbound Concurrency	Set the concurrency number of inbound or outbound calls for the sip trunk, and the calls will not be established if the concurrency number is exceeded. Default is 9999.
Total Concurrency	Set the total number of concurrent calls, and the default is 9999. The number of inbound or outbound concurrent calls cannot be more than the total number of concurrent calls.

Figure- Account Group



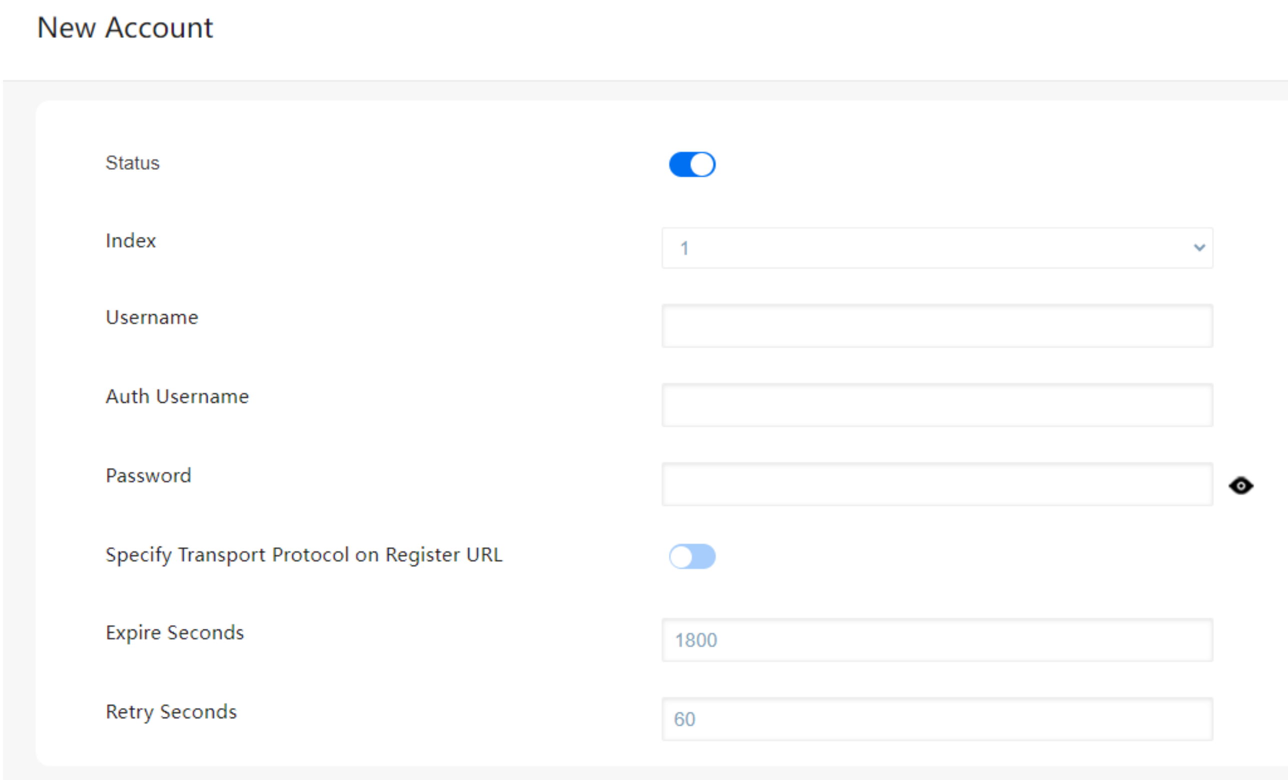


Table-Parameters of Account Group

Parameter	Description
Status	Enable or disable SIP Trunk.
Index	Index of SIP Trunk. Range from 1 to 32.
Name	The name of the SIP trunk. The input value is text type and cannot be null. The value is up to 32 characters and cannot contain " ".
Username	The username of this SIP trunk, it is generally a phone number.
Auth Username	The username used for register authentication by this SIP trunk.

Password	The password used for register authentication by this SIP trunk.
Specify Transport Protocol on Register URL	When enabled, it specifies the current transport protocol on the Register URL
Expire Seconds	The validity period after the SIP trunk is registered successfully. When the time expires, the SIP trunk will send register request to the server. Default value is 1800s.
Retry Seconds	Time interval to re-initiate registration if registration fails. Default is 60s.

5.2.2 FXO

FXO trunk can connect UC350 Series IPPBX and Public Switched Telephone Network (PSTN) or a traditional PBX.

The MCU board of UC350 Series IPPBX does not be configured with FXO port by default. When the FXO trunk configuration is done, the device will send the configuration information to the SLOT user board which configure FXO Port.

Note: The UC350 Series IPPBX supports 4 types of user boards: FXS, FXO, FXU and DTU. Among them, the FXU user board supports hybrid FXS and FXO ports. For the UC350 Series IPPBX, only FXO and FXU user boards can be configured with the FXO trunk.

Figure-Status of FXO Trunk

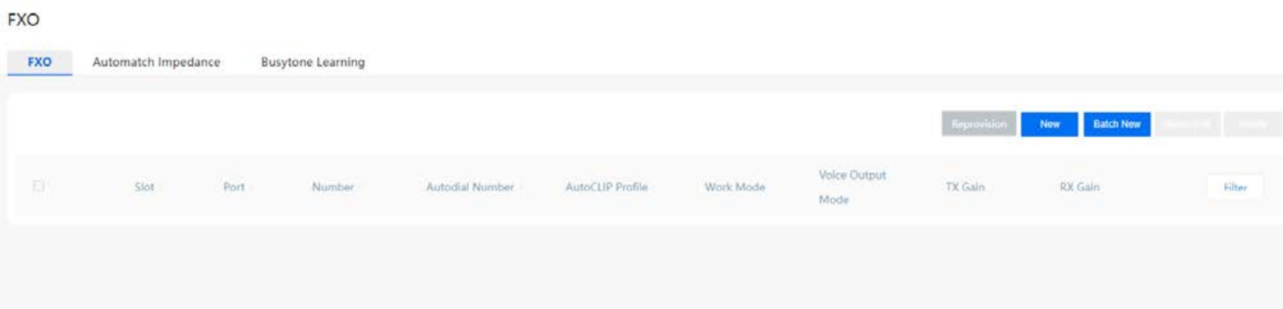


Figure-Parameters of FXO Trunk

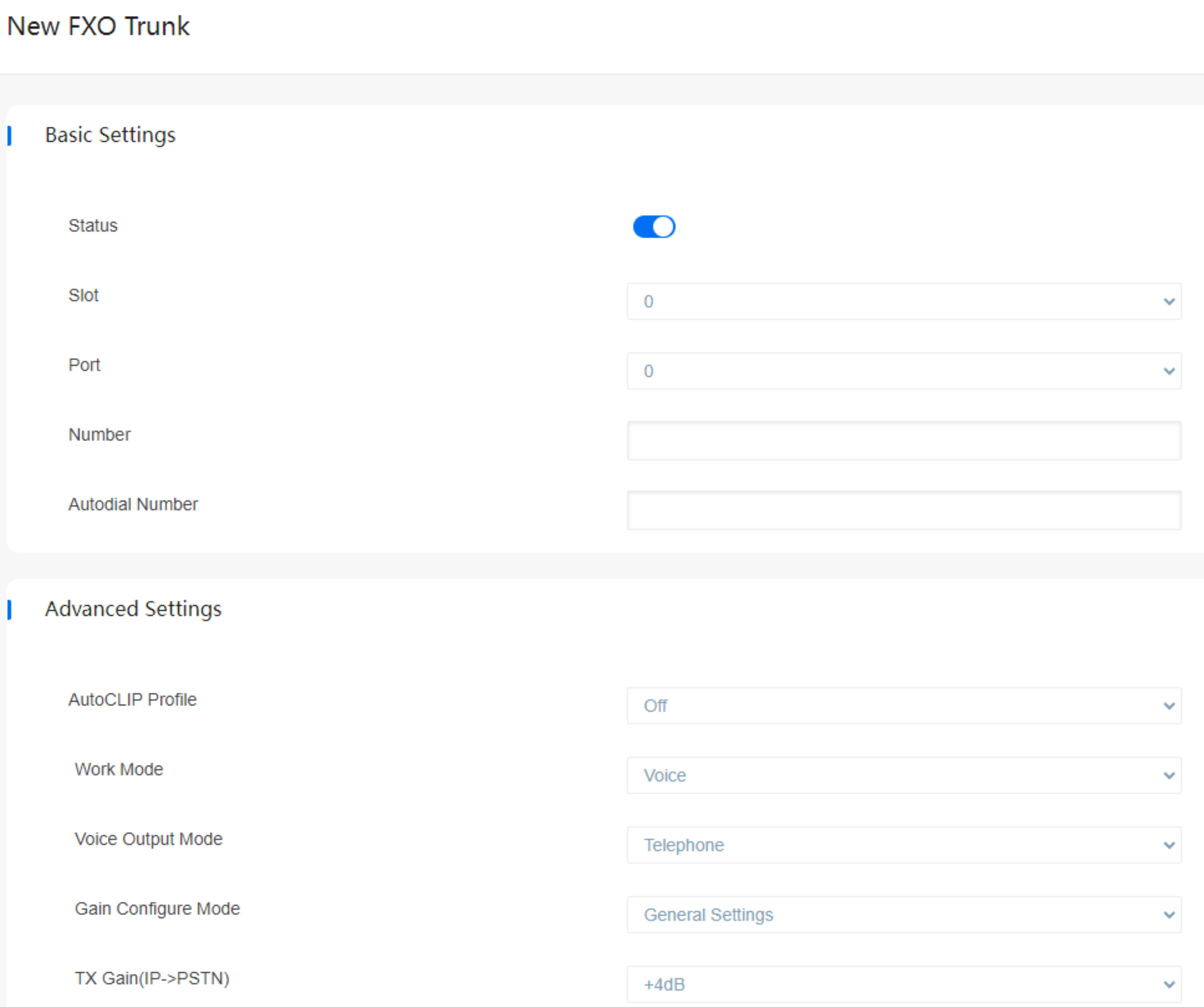


Table-Parameters of FXO

Parameter	Description
Status	Enable or disable FXO Trunk.
Port	The FXO port number.
Name	The name of the FXO
Number	Configuring Trunk Number.
Autodial Number	The autodial number of the FXO port when there are incoming calls.
AutoCLIP Profile	AutoCLIP is mainly used to SIP trunks, FXO trunks and VoLTE trunks and it helps record the outgoing and incoming calls of a trunk.
Work Mode	Configure the working mode of the FXO port.
Voice Output Mode	Configure the voice output mode of the FXO port.
Gain Configure Mode	Select the gain configuration mode of the FXO port (general settings and advanced settings), TX gain and RX gain are newly added to advanced settings than general settings.
TX Gain(IP->PSTN)	The volume level of the remote end during a call, that is, adjusting the "TX gain" will affect the volume of the sound heard by the remote end.
RX Gain(PSTN->IP)	The volume level of the user during the call, that is, adjusting the "RX gain" will affect the level of sound users hear.

Impedance	Set the value of impedance(SLIC) for the remote FXS port.
Hybrid	Set the value of hybrid for the remote FXS port.
Area Code	Fill in the area code corresponding to the configured trunk. Corresponding to international/domestic. To be used in conjunction with the Area Call Permission of the SIP extension

Automatch Impedance

The FXO Trunk interconnects the PSTN. When the value of impedance at both ends does not match, the automatch impedance function can be used to automatically adapt the impedance of the remote FXS port to ensure stable communication.

Conditions: Only the ports in the online can be tested.

Steps:

1. Please go to **Trunk & Route > FXO > Automatch Impedance**.
2. Configure the mode and parameters of automatch impedance , and click **Start**.
3. After the test is completed, the Impedance and Hybrid values are displayed.

Figure-Automatch Impedance

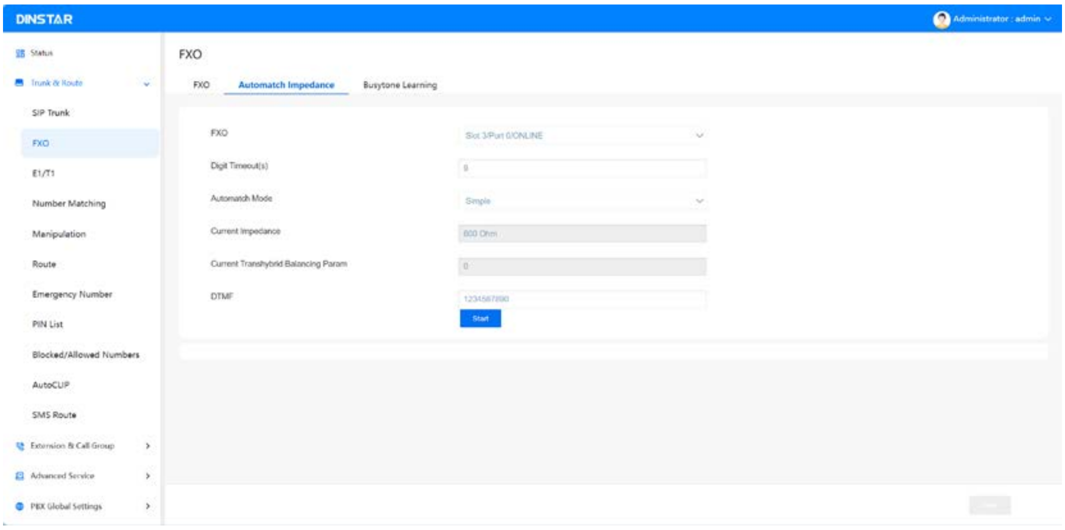


Table-Parameters of Automatch Impedance

Parameter	Description
FXO	Select FXO port.
Digit Timeout(s)	Set a time for dial timeout.
Automatch Mode	Automatch mode: simple/standard/accurate (The higher the mode, the higher the accuracy and the longer it takes).
Current Impedance	Display the current impedance value of the FXO port (just display, cannot be modified).
Current Transhybrid Balancing Param	Display the current hybrid of the FXO port (just display, cannot be modified).
DTMF	When automatch impedance, the DTMF value sent by the local FXO port is 1234567890 by default, which can be modified.

Busytone

Learning

The FXO Trunk interconnects the PSTN. When the busy tone at both ends does not match, it may cause problems such as incorrect or unsuccessful detection of the busy tone. At this time, users can use the busy tone learning function to automatically adapt to the busy tone of the remote FXS port ensures that the busy tone can be correctly identified.

Conditions: Only the ports in the online can be tested.

Steps:

1. Please go to **Trunk & Route** > **FXO** > **Busytone Learning**.
2. After selecting an online port, filling in the destination number, click **Start**.
3. Busy tone learning takes about 30-60s. After the learning is completed, the progress bar displays 100%.
4. Learning is completed, the Optimal cadence will be displayed. It will only take effect after saving.

Figure-Busytone Learning

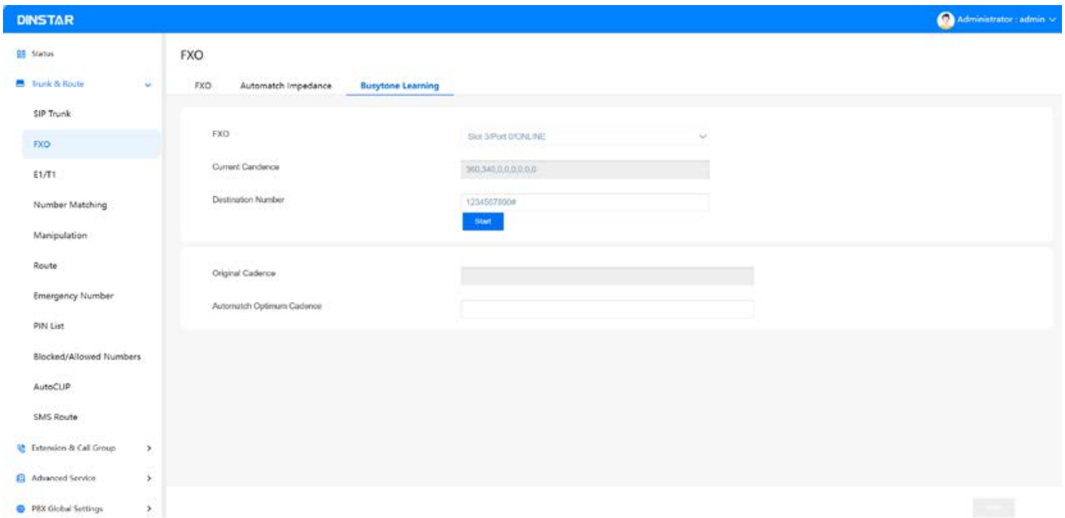


Table-Parameters of Busytone Learning

Parameter	Description
FXO	Select FXO port.
Current Candence	Display the current FXO trunk busy tone cadence.
Destination Number	The destination number during busy tone learning.
Original Cadence	Previous busy tone cadence data.
Automatch Optimum Cadence	Optimal cadence after FXO trunk busy tone learning.

5.2.3 E1/T1

There are two types of E1 trunk: SS7 and PRI. The two different trunks are as follows.

SS7 Trunk

SS7 is a global telecom protocol standard that governs how network elements in a public switched telephone network (PSTN) communicate and control signals. Nodes in an SS7 network are referred to as signaling points.

Go to **Trunk & Route** > **E1/T1**, select SS7 in the type bar.

Figure-Parameters of SS7 Trunk

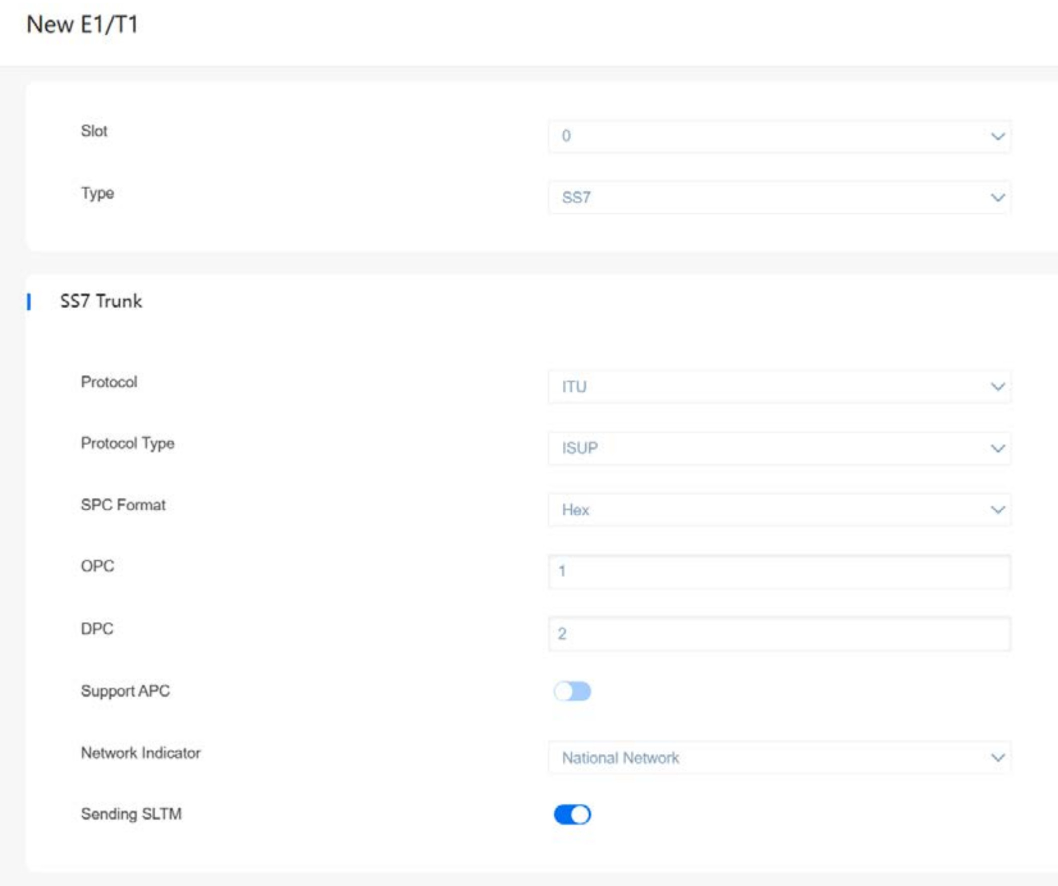


Table-Parameters of SS7 Trunk

Parameter	Description
Slot	Select the Slot that needs to be configured.
Type	Type: SS7 and PRI.
Protocol	Protocol standard SPC types: ITU (14 bit), ANSI (24 bit), ITUCHINA (24 bit).
Protocol Type	SS7 service types: ISUP (ISDN user side) and TUP (Telephone side).
SPC Format	SPC format: Hexadecimal system and 14Select the Slot that needs to be configured bit (3-8-3).
OPC	OPC: Original Point Code, usually uniformly assigned by the operator.

DPC	DPC: Destination Point Code, usually uniformly assigned by the operator.
Support APC	APC is required when it is enabled, and the format is the same as the SPC format. Enter the STP code provided by the operator.
Network Indicator	Display the network indicator of SS7, including: domestic network, domestic network backup, international network, and international network backup. the default is domestic network(mainly used in China, the United States and Japan), and "international network" is usually used exchange in the office, others are according to the environment.
Sending SLTM	SLTM: Signaling Link Test Message.
E1/T1 No	Set E1/T1 No from 0 to 3.
Channel No	The channel for establishing link 7, usually channel No.16 or No.1, and the default channel No.16.
Start CIC No	The start CIC No of E1 port.
Caller Type	Configure the caller number type (not configured/ international/ domestic/ user).
Callee Type	Configure the callee number type (not configured/ international/ domestic/ user).
OrgCallee Type	Configure the original callee number type (not configured/ international/ domestic/ user).

Numbering Plan	Configure the number plan (ISDN/ data/ telegram/ special).
Calling Presentation	Calling presentation (allowed /limited/ invalid/ not configured).
Screening indicator	Configure screening indicator (user-provided/ network-provided).
Called Stop sending	After enabled, the called number with the suffix F.
Calling Stop sending	After enabled, the calling number with the suffix F.
AutoCLIP Profile	AutoCLIP is mainly used to SIP trunks, FXO trunks, VoLTE trunks and it helps record the outgoing and incoming calls of a trunk.
RFC2833-PT	The default value is 101.
DTMF First Priority	The default value is RFC2833. Users can select the SIP INFO or Inband.
DTMF Second Priority	The default value is SIP INFO. Users can select the RFC2833 or Inband.
DTMF Third Priority	The default value is Inband. Users can select the RFC2833 or SIP INFO.
Overlap Receiving	To receive the same number repeatedly. The default is disabled.

Ringback Tone to PSTN Originated from	The default is Adaptive. Users can select local or IP. If it's set to local, the device will play it. If it's set to adaptive, it will play by using the PEM header of the callee. If it's set to IP, it will play by the callee.
Ringback Tone to IP Originated from	The default is PSTN, and local is optional. If it is set to Local, the ringback tone is played from the device. If set to PSTN, ringback tone will be played by the called side.
Area Code	Fill in the area code corresponding to the configured trunk. Corresponding to international/domestic. To be used in conjunction with the Area Call Permission of the SIP extension
DNIS	When the trunk calls in, the called number matches the DNIS, then the from display name of the invite should be the display name of the called number.
Called Number / Display Name	The called number and display name of DNIS.

PRI Trunk

PRI, which stands for Primary Rate Interface, is an older technology that utilizes a physical, wired connection to transmit calls, messages, and data. It became popular among businesses as it provided a higher capacity form of TDM (time-division multiplexing) connection. This technology was an improvement over traditional plain old telecom (POTS) systems, allowing businesses to handle up to 23 or 31 concurrent communication channels. PRI systems depend on physical circuits to route voice and data calls through the service provider, typically the telephone line.

Go to **Trunk & Route** > **E1/T1**, select SS7 in the type bar.

Figure-Parameters of PRI Trunk

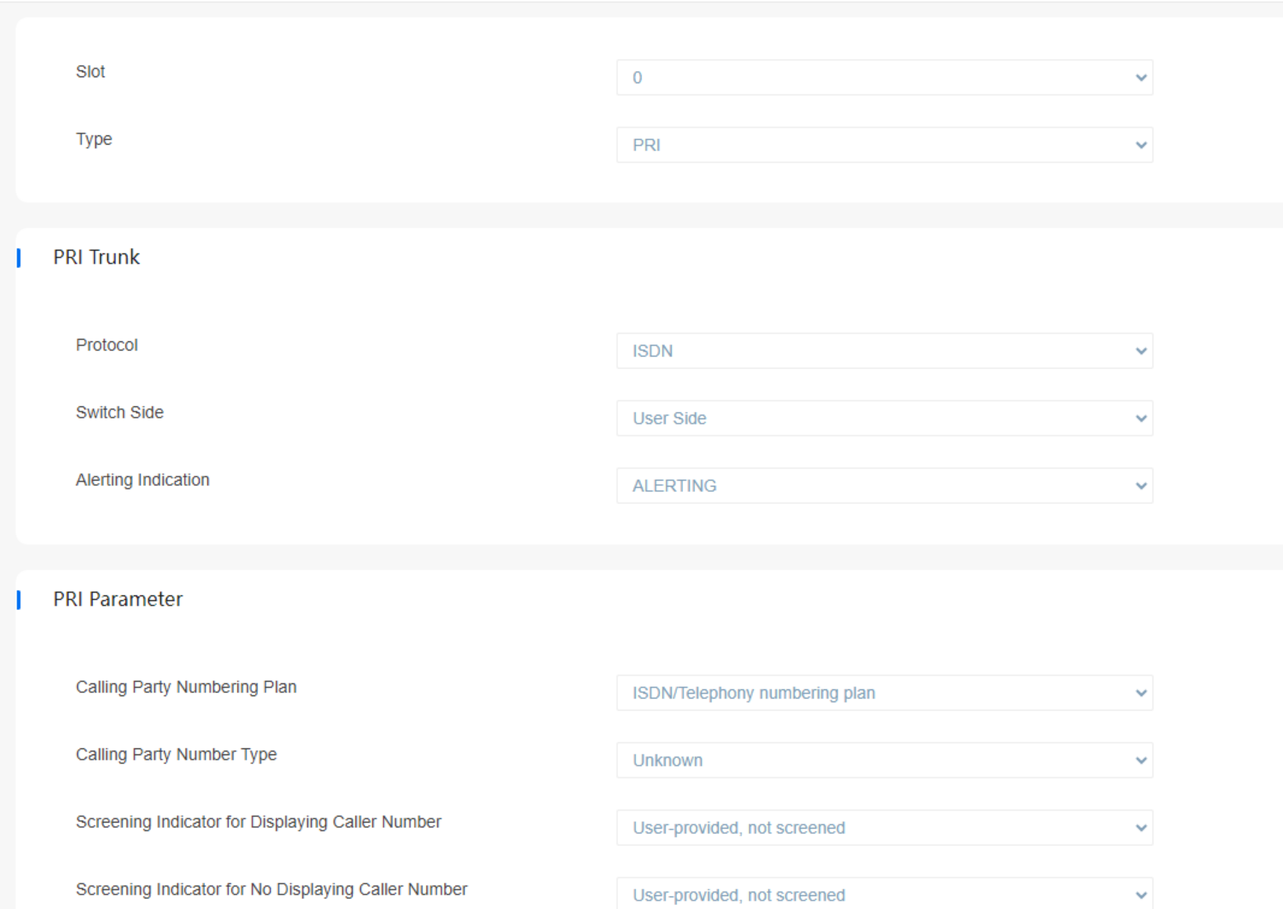


Table-Parameters of PRI Trunk

Parameter	Description
Slot	Select the Slot that needs to be configured.
Type	Type: SS7 and PRI.
Protocol	There are two types of PRI protocol: ISDN and QSIG.

Switch Side	"User side" and "Network side" can be chosen. When implementing a PRI circuit, the nature of E1 in the network must be different on the receiving and sending sides.
Alerting Indication	Configure alerting indication (Alerting and progressing).
Calling Party Numbering Plan	6 options, the default is "ISDN/telephone numbering plan".
Calling Party Number Type	6 types of calling party numbers can be selected.
Screening Indicator for Displaying Caller Number	4 options, the default is "user-provided, not screened".
Screening Indicator for No Displaying Caller Number	4 options, the default is "user-provided, not screened".
Called Party Numbering Plan	6 options, the default is "ISDN/telephone numbering plan".
Called Party Number Type	6 options, the default is "unknown".
Information Transfer Capability	Support voice and 3.1khz voice.
AutoCLIP Profile	AutoCLIP is mainly used to SIP trunks, FXO trunks, VoLTE trunks and it helps record the outgoing and incoming calls of a trunk.
RFC2833-PT	The default value is 101.

DTMF First Priority	The default value is RFC2833. Users can select the SIP INFO or Inband.
DTMF Second Priority	The default value is SIP INFO. Users can select the RFC2833 or Inband.
DTMF Third Priority	The default value is Inband. Users can select the RFC2833 or SIP INFO.
Overlap Receiving	To receive the same number repeatedly. The default is disabled.
Ringback Tone to PSTN Originated from	The default is Adaptive. Users can select local or IP. If it's set to local, the device will play it. If it's set to adaptive, it will play by using the PEM header of the callee. If it's set to IP, it will play by the callee.
Ringback Tone to IP Originated from	The default is PSTN, and local is optional. If it is set to Local, the ringback tone is played from the device. If set to PSTN, ringback tone will be played by the called side.
Area Code	Fill in the area code corresponding to the configured trunk. Corresponding to international/domestic. To be used in conjunction with the Area Call Permission of the SIP extension

DNIS	When the trunk calls in, the called number matches the DNIS, then the from display name of the invite should be the display name of the called number.
Called Number / Display Name	The called number and display name of DNIS.

Parameter Modification

Figure-Parameters of E1/T1

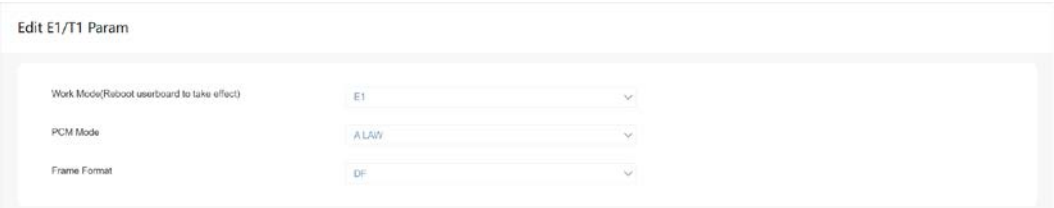


Table-Parameters of E1/T1

Parameter	Description
Work Mode (Reboot userboard to take effect)	Support E1 or T1 work mode.
PCM Mode	PCM (Pulse Code Modulation) Mode: ALAW or Mu LAW.
Frame Format	The frame format of the E1 port is: DF, MF-CRC4, MF, the default is DF.

5.2.4 Number Matching

On the **Trunk & Route > Number Matching** interface, 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.

Figure-Parameters of Number Matching

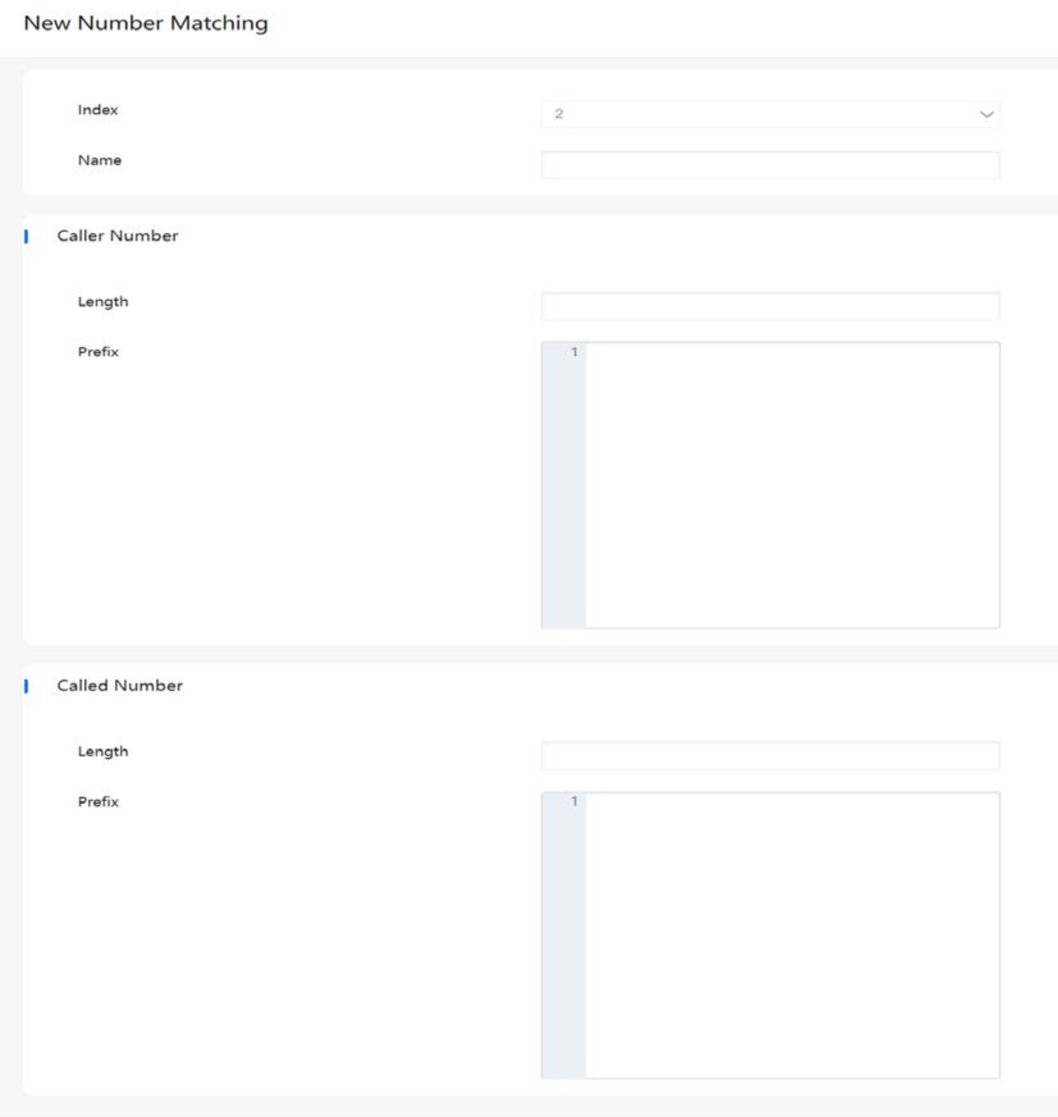


Table-Parameters of Number Matching

<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td>Index</td><td>The index of number matching rule. Range from 1 to 32.</td></tr>
<tr><td>Name</td><td>The name of the number profile.</td></tr>
<tr><td>Length</td><td>The length of the calling number or called number. For example, ： 4|6|7 means the calling number or called number must be 4 digits, 6 digits or 7 digits except the prefix.</td></tr>
<tr><td>Prefix of Caller Number</td><td>The prefix of the calling number. It supports multiple prefixes, multiple rules for "or" relationships. It supports regular expression.</td></tr>
<tr><td>Prefix of Called Number</td><td>The prefix of the called number. It supports regular expression. It supports multiple prefixes, multiple rules for "or" relationships.</td></tr>
</table>

Regex(Regular Expression)Syntax

<table>
<tr><td>^</td><td>Matches the starting position in a number string. For example, ^134 matches the numbers starting with 134.</td></tr>
<tr><td>$</td><td>Matches the ending position of a string. For example, 2$ matches the numbers ending with 2.</td></tr>
<tr><td>|</td><td>Separates alternate possibilities. For example, 2|3|4 means 2,3 or 4.</td></tr>
</table>

\	Marks the next character as a special character, a literal, a backreference, or an octal escape.
[ ]	Matches a single character that is contained within the bracket. For example, [123] matches 1, 2, or 3. [0-9] matches any digit from "0" to "9".
[^]	Matches any one character except those enclosed in [ ]. For example, [^9] matches any character except 9.
.	Matches any single character except the newline character. For example, 3.4 matches 314, 324, 334, 344.
?	Indicates there is zero or one of the preceding element. For example, colou?r matches both color and colour.
*	Indicates there is zero or more of the preceding elements. For example, ab*c matches ac, abc, abbc, abbbc, and so on.
+	Indicates there is one or more of the preceding element. For example, ab+c matches abc, abbc, abbbc, and so on, but not ac.
/d	Marks any digit, equal to [0-9].
\D	Marks any character that is not a digit, equal to [^0-9].
\s	Marks any blank character such as a space or a tab.
\S	Marks any character that is not a blank character.

Examples of Regex Syntax:

^0755	Matches the phone numbers with starting digits of 0755.
^0755|^8899|^0110	Matches the phone numbers with starting digits of 0755, 8899 or 0110.
^[1][358][0-9]{9}$	Matches the phone numbers with the first digit as 1, the second digit as 3, 5 or 8, the left nine digits as any of 0 to 9.

Note:	The matching of number prefix also supports some digits that are not conform to the format of regular expression. For example, 0755 matches the numbers starting with 0755, and 0755|8899|0110 matches the numbers starting with 0755, 8899 or 0110.

5.2.5 Manipulation

Number manipulation refers to the change of a called number or a caller number during calling process when the called number or the caller number matches the preset rules.

Click the **New** button, and users will see the following interface:

Figure-Parameters of Manipulation

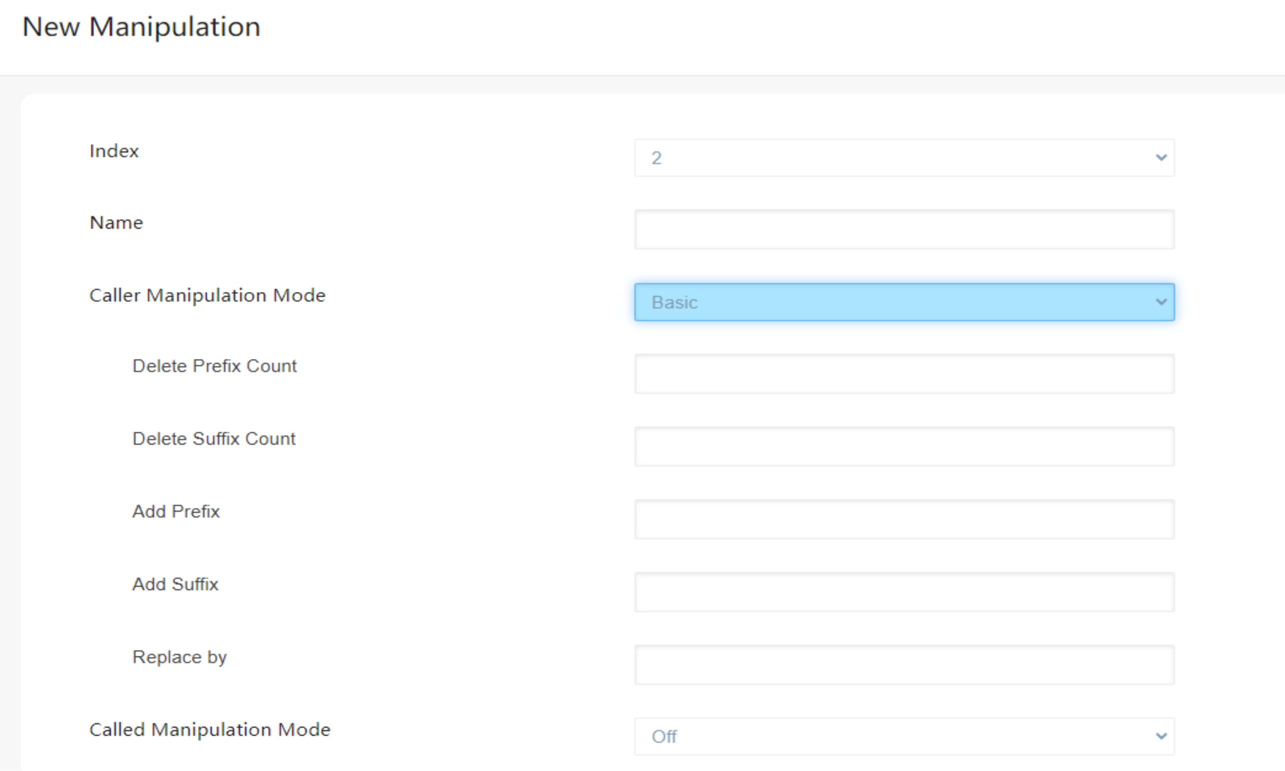


Table-Parameters of Manipulation

Parameter	Description
Index	The index of number manipulation rule. Range from 1 to 32.
Name	The name of this manipulation profile.
Delete Prefix Count	The number of digits that are deleted from the left of the caller number or calling number.
Delete Suffix Count	The number of digits that are deleted from the right of the caller number or calling number.
Add Prefix	The prefix added to the caller number or the calling number.

Add Suffix	The suffix added to the caller number or the calling number.
Replace by	The number which replaces the caller number or the calling number.
Batch Replace/Replace Rules	Batch Replacement - Replacement Rules Caller/Callee Replacement Rules Syntax: original number, new number; numbers can only use numbers, letters or +/*/#, maximum length 32; Example: 1000,8001000 1001,8001001
Advance	Caller/Callee Number Matching: Fill in the caller and called number for configuration in routing. When initiating a call, the corresponding caller and called number will take effect. This advanced number change configuration Delete prefix digits, add prefix, and replace with the same usage as before

5.2.6 Route

This section is to configure routes or route groups for incoming and outgoing calls through UC350 Series IPPBX.

Route

On the **Trunk & Route** > **Route** > **Route** interface, users can configure routes for incoming calls and outgoing calls.

Figure-Parameters of Route

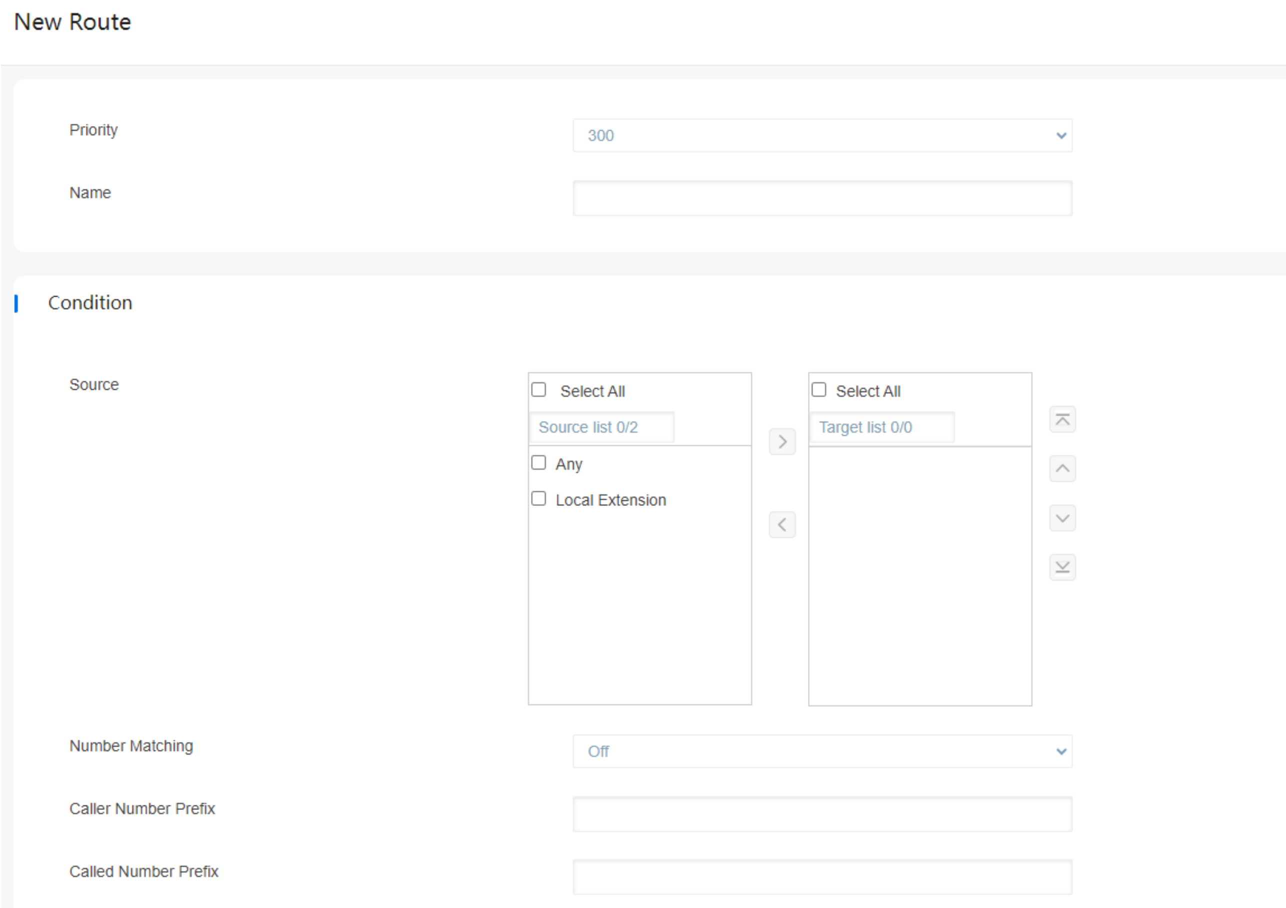


Table-Parameters of Route

Parameter	Description
Priority	The priority for choosing the route. the higher value, the lower priority.
Name	The name of the route.
Condition	The condition under which the route will be used.

Source	The source of the call. It can be the FXS extension, SIP extension, FXO trunk, GSM trunk, a customized source or any.
Number Matching	The calling and calling numbers match **Note**: it cannot be simultaneously used with the following parameters of 'caller number prefix' and 'called number prefix'.
Caller Number Prefix	The prefix of caller number. It supports regular expression.
Called Number Prefix	The prefix of called number. It supports regular expression.
Time Profile	The profile of time during which the route can be used. Please make reference to the **System > Time** section.
Action	Include manipulating number and sending call to destination.
Callback	After enabling, the caller who configures this route will directly hang up after the incoming call, and then initiate a call to the called after the waiting time expires, and then initiate a call to the caller after the called picks up.
Delay before Callback(s)	Set Delay before Callback(s)

Distinctive Ringtone(Alert-Info)	After it is configured, the INVITE header field will contain the Alert-Info.
Manipulation	If it is on, the caller number or called number of the route will be manipulated. Please make reference to the **Trunk & Route> Manipulation** section.
Destination	The destination of the route.
Password Type	When enabled, users need to enter password to match the route. The default is disabled, and the password type can be either a single password or a list of PIN codes.
Recording Profile	Record according to the configured rules.
Failover Action	The processing when a call through this route fails.
Condition	Reasons for failed calls: Busy, Timeout, or Unavailable. If neither is checked, all failed calls are processed. If only some of the options are checked, only calls that satisfy the checked conditions are processed.

Other Condition Code	The conditions for failed calls. Only Busy, Timeout and Unavailable can be checked. When users need to extend other conditions, users can fill in the codes of other conditions. If there are more than one other condition code values, please separate them with ",".

Route Group

On the **Trunk & Route > Route > Route Group** interface, users can group SIP trunks, SIP extensions, FXS extensions and FXO trunks together according to user's needs and set strategy for choosing which trunk or extension as the destination route under a route group.

Figure-Parameters of Route Group

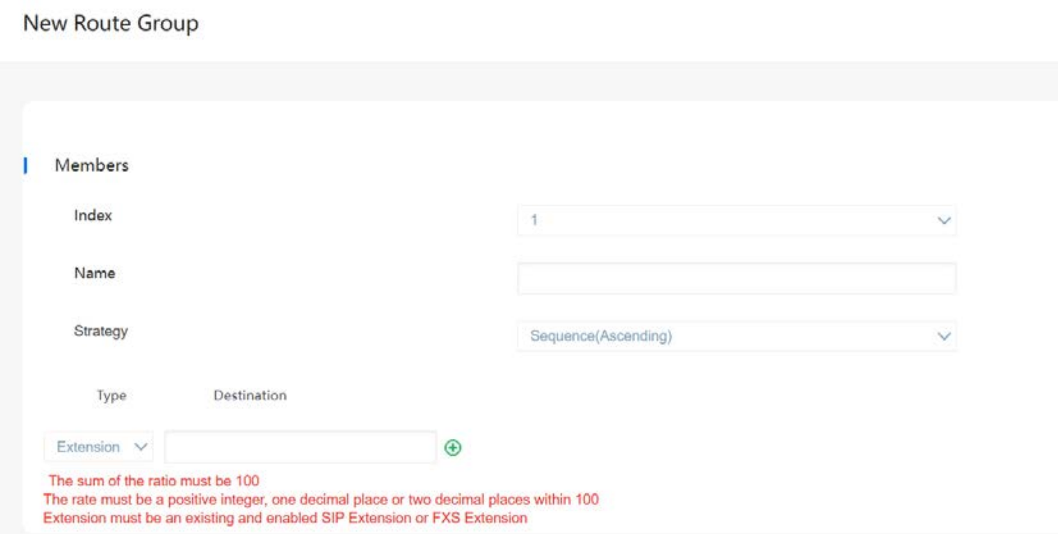


Table-Parameters of Route Group

Parameter	Description
Index	The index of the route group.
Name	The name of the route group.

Members	Select FXS extension, SIP extension, SIP trunk, FXO trunk or GSM trunk.
Strategy	The strategies for choosing which route under the route group as the destination route, including Sequence (Ascending), Sequence (Cyclic Ascending), Simultaneous and Random.

5.2.7 Emergency Number

Emergency numbers are used for more urgent call scenarios, such as: 120, 119, 110, 911 in UAS etc. The Emergency Numbers get priority over any other settings. Emergency numbers will be dialed directly based on the configured routes. And the emergency number must be answered at any PBX extension regardless of the extension status or other PBX settings.

On the **Trunk & Route > Emergency Number** interface, users can specify the emergency call number and bind the corresponding trunk, so that in the emergency call conditions, it will directly match the trunk to ensure the validity of the call.

Note: *The call priority of emergency number is higher than route, and an emergency number can be bound to multiple outbound trunks.*

Figure-Parameters of Emergency Number

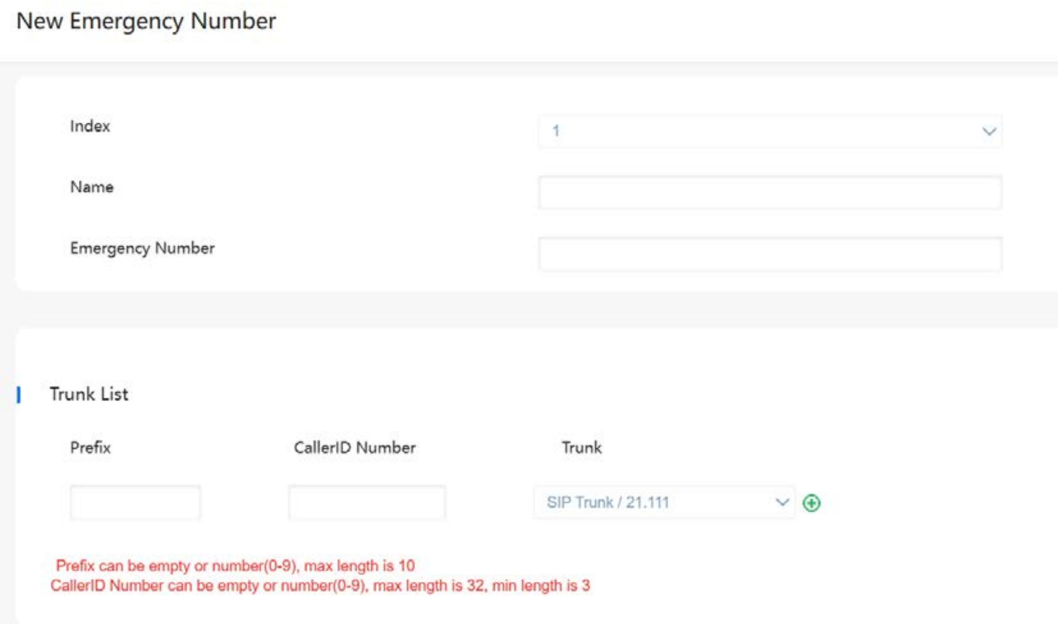


Table-Parameters of Emergency Number

Parameter	Description
Index	The index of the emergency call number rule.
Name	The name of the emergency call number rule.
Emergency Number	Specify the emergency call number and match it when the call is made.
Prefix	Matching Caller Number Prefix which is used to limit the SIP end points using this feature.
CallerID Number	When using the Emergency Call feature, the original caller is replaced, and the configured number will be carried for outgoing calls.
Trunk	Specify the outbound trunk. Users can select SIP trunk, E1/T1 trunk or FXO trunk.

5.2.8 PIN List

On the **Trunk & Route > PIN List** interface, users can configure the password for outgoing calls, which can be used to restrict outgoing calls. This configuration takes effect in the route configuration. After the configuration takes effect, when the SIP terminals match the routes and make outgoing calls, users need to enter the corresponding PIN code to make the calls.

Note: When multiple passwords are configured in a PIN code list, user can enter any one of the passwords when making outgoing calls.

Figure-Parameters of PIN List

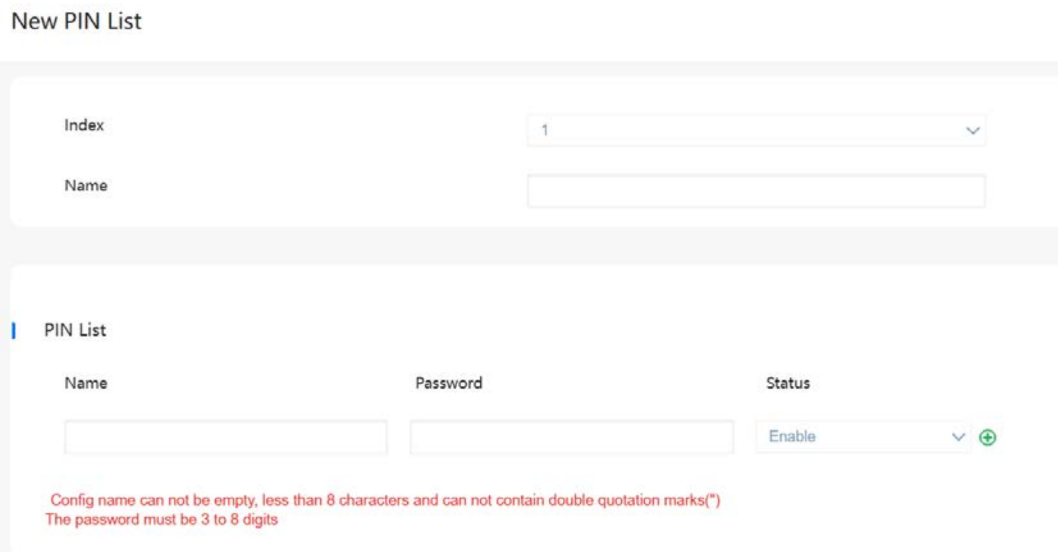


Table-Parameters of PIN List

Parameter	Description
Index	The index of the PIN List.
Name	The name of the PIN List.
Password	Specify the password that needs to be entered for outgoing calls from SIP terminals.
Status	Enable or disable the PIN List.

5.2.9 Blocked/Allowed Numbers

On the **Trunk & Route > Blocked/Allowed Numbers** interface, users can configure the overall blocked/allowed call numbers according to the actual needs, and can select the blocked/allowed call type such as inbound, outbound, or inbound & outbound.

Blocked Numbers

Figure-Parameters of Blocked Numbers

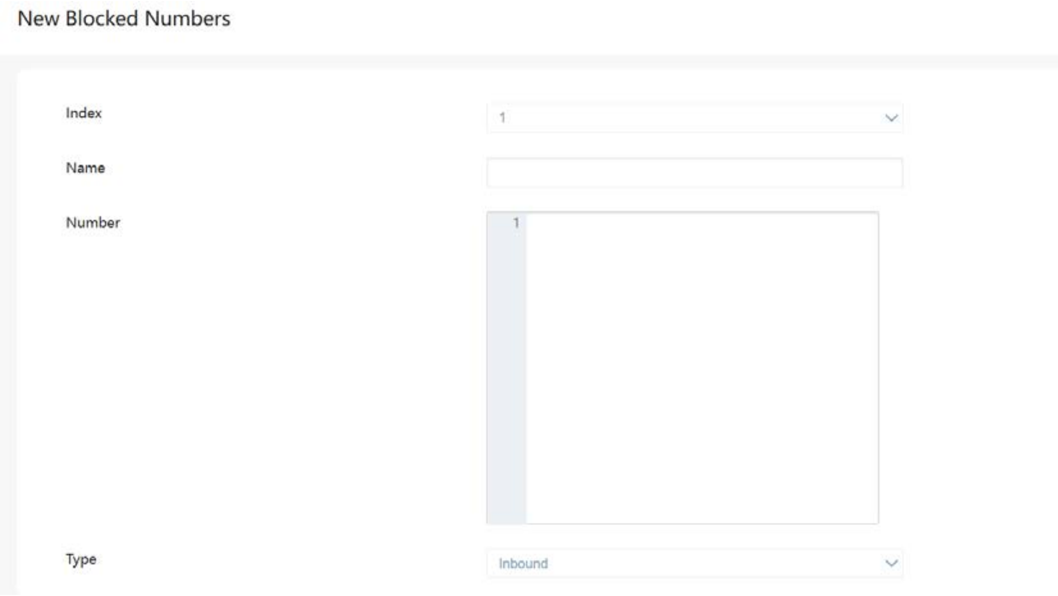


Table-Parameters of Blocked Numbers

Parameter	Description
Index	The index of the blocked number list.
Name	The number of the blocked number list.
Number	Configure the blocked call numbers.
Type	Configure the blocked call type: inbound, outbound, or inbound & outbound.

Allowed Numbers

Figure-Parameters of Allowed Numbers

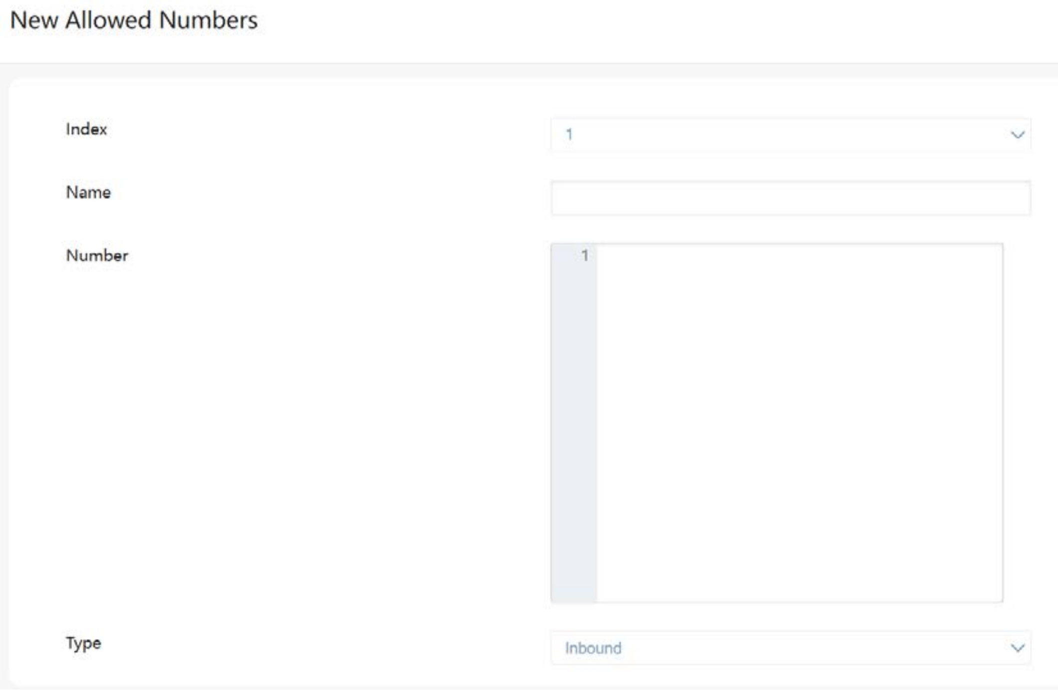


Table-Parameters of Allowed Numbers

Parameter	Description
Index	The index of the allowed number list.
Name	The name of the allowed number list.
Number	Configure the allowed call numbers.
Type	Configure the allowed call type: inbound, outbound, or inbound & outbound.

5.2.10 AutoCLIP

AutoCLIP is mainly used to SIP trunks, FXO trunks and VoLTE trunks and it helps record down the outgoing and incoming calls of a trunk.

Figure-Parameters of AutoCLIP

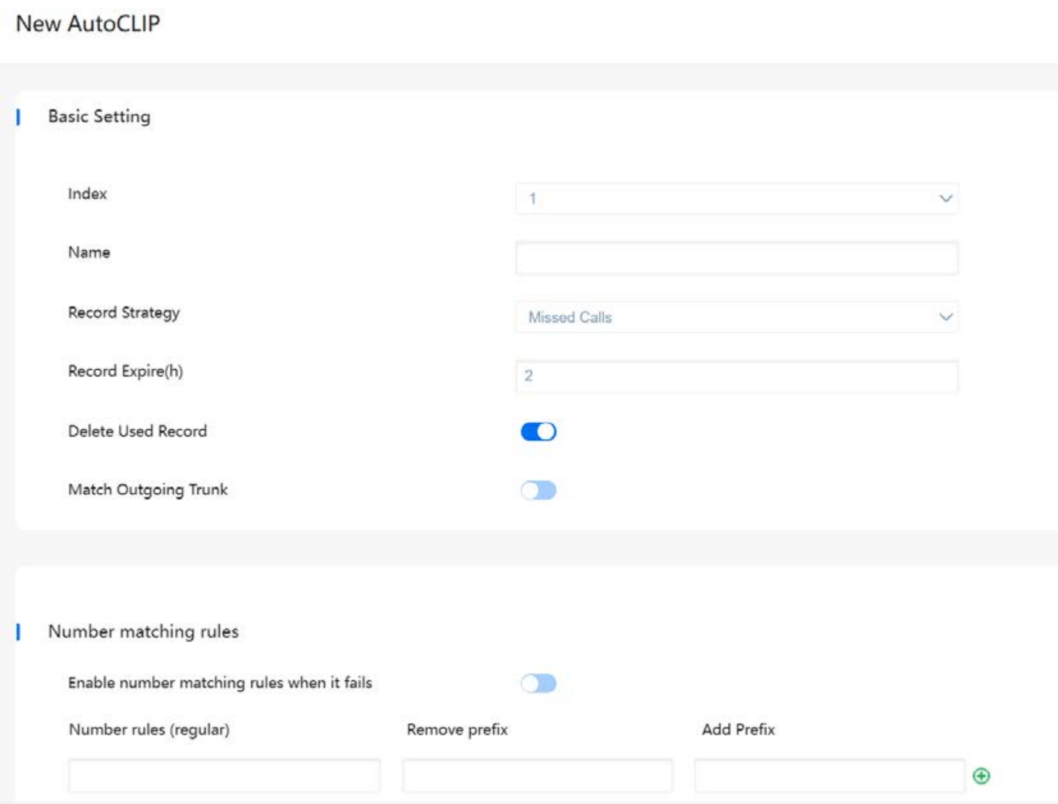


Table-Parameters of AutoCLIP

Parameter	Description
Index	The index of AutoCLIP profile.
Name	The name of AutoCLIP profile.
Record Strategy	Users can choose missed calls or all calls. If missed calls are been selected, the device will record the missed calls of the trunk. If all calls are been selected, all the calls going through the trunk would be recorded.

Record Expire(h)	The validity period of a record. For example, if this parameter is set as 2 hours, the record will be valid in 2 hours since the record is generated. During the validity period, if there is coming call for the extension number contained in the record, the call will directly led to the extension without routing.
Delete Used Record	By default, this parameter is disabled.
Match Outgoing Trunk	By default, this parameter is enabled. If this parameter is enabled, those calls going through the trunks in the record can coming in without routing.
Enable number matching rules when it fail	Enable number matching rules

5.2.11 SMS Route

UC350 Series IPPBX allows SMS to be sent between SIP clients, On the **Trunk & Route > SMS Route** interface, users can establish route for these SMS.

Figure-Parameters of SMS Route

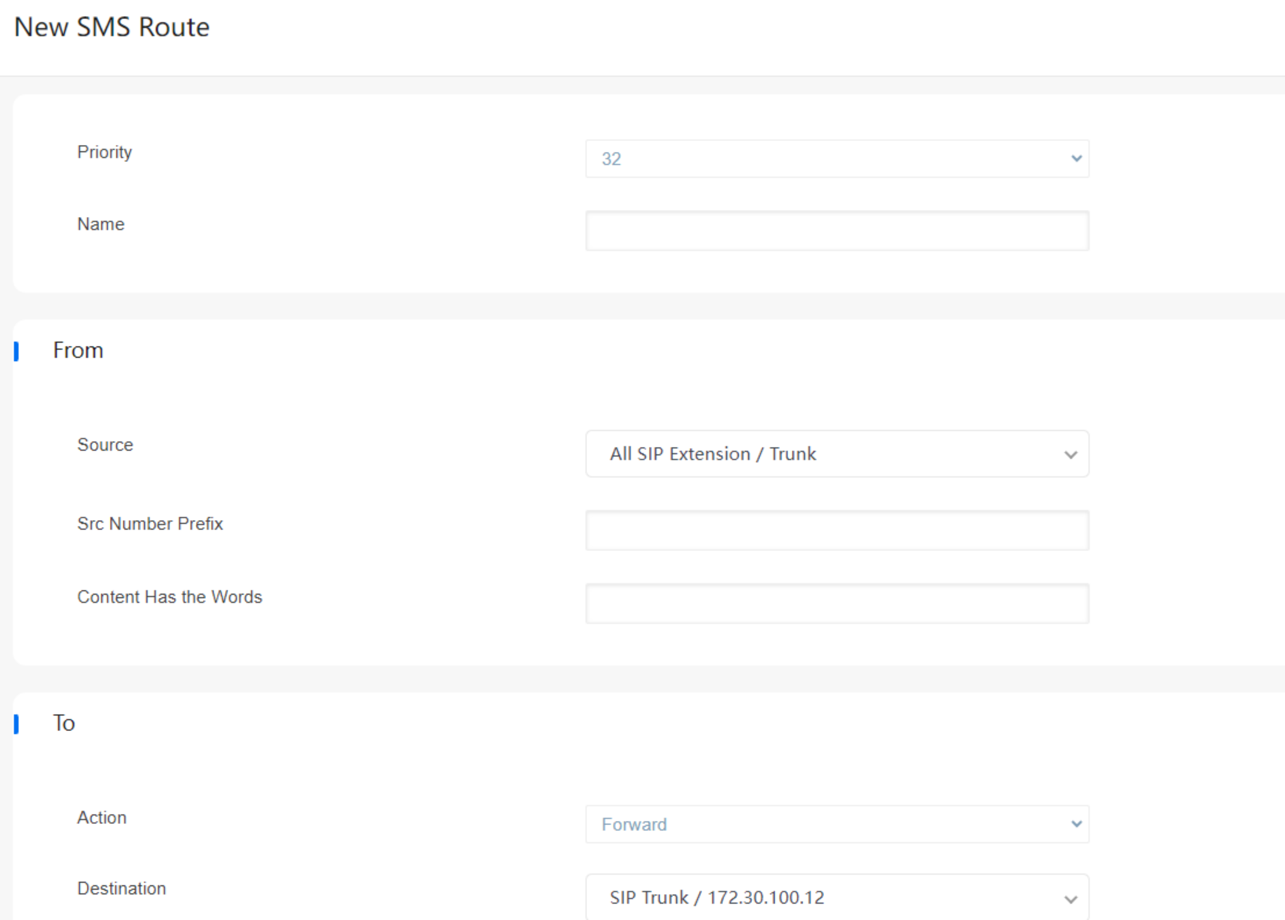


Table-Parameters of SMS Route

Parameter	Description
Priority	The priority for the SMS route. The higher value, the lower priority.
Name	The name of the SMS route.
Source	The source of the SMS route. It can be a trunk or an extension. It also can be a LTE SMS and USSD.

Src Number Prefix	Set source number prefix
Content Has the Words	Match key words in text message content.
Action	The text message action can choose whether to forward or reply.
Destination	The destination of the SMS route. It can be a trunk or an extension.
Dest Number Src	Users can choose to custom, get from the to header field, get from the content, get from the subject
Dest Number	The destination number is set when you select Custom. The number can only use numbers, letters or +/*/#, and the maximum length is 32
Add Prefix in Content	The prefix of the SMS content. It is generally 'none', which means there is no prefix to be matched.
Add Suffix in Content	The suffix of the SMS content. It is generally 'none', which means there is no suffix to be matched.

5.3 Extension & Call Group

5.3.1 SIP Extension

On the **Extension & Call Group > SIP Extension** interface, user can configure the SIP accounts registered in the UC350 Series IPPBX by SIP clients (hereby UC350 Series IPPBX is regarded and act as a SIP server).

Figure-Status of SIP Extension

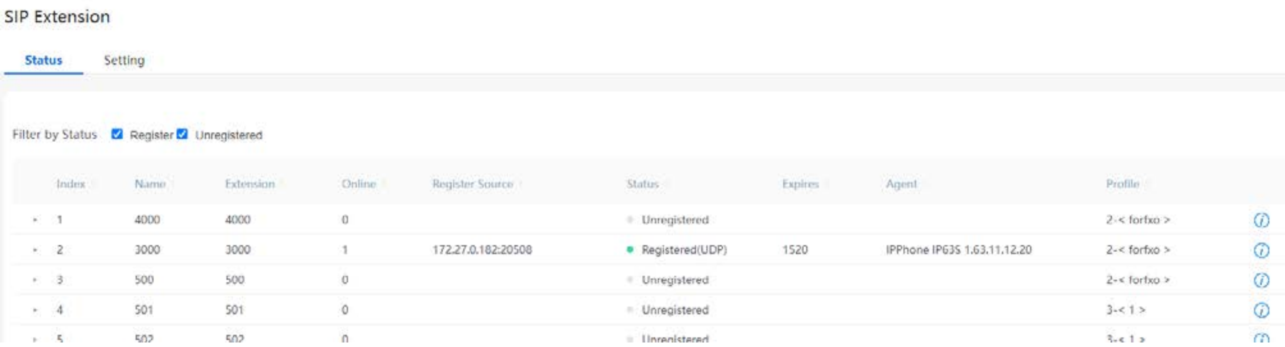


Figure-Parameters of SIP Extension configuration

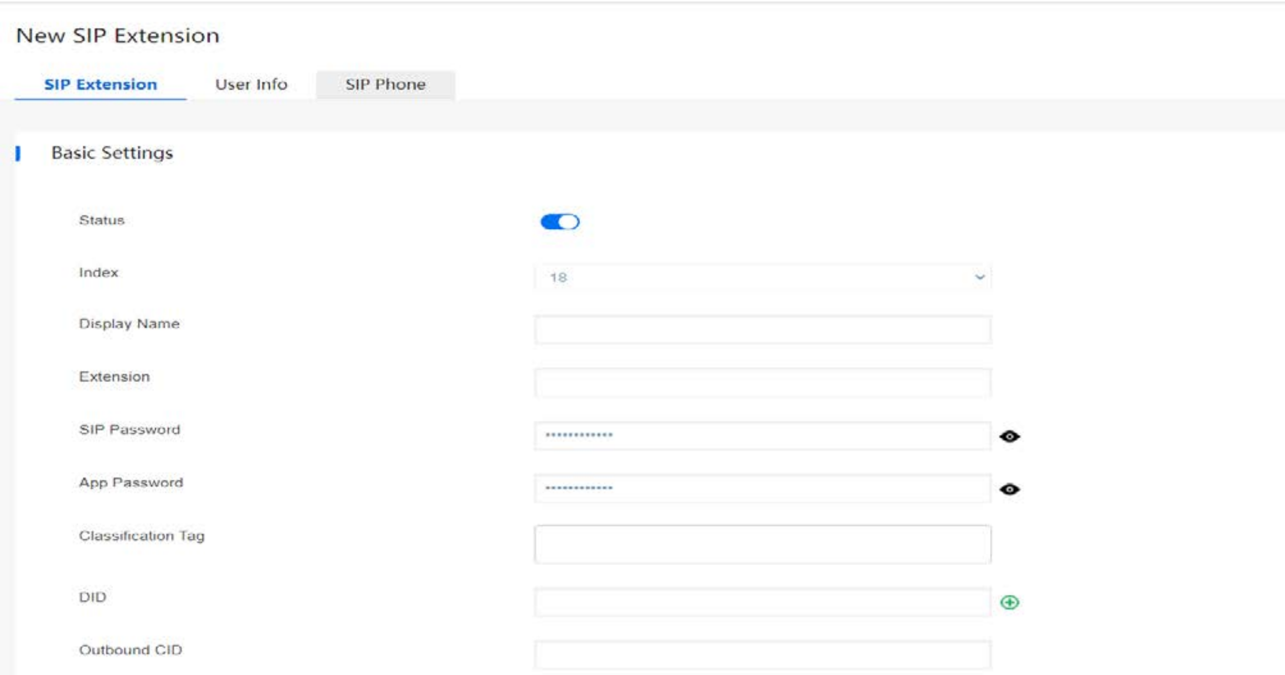


Table-Parameters of New SIP Extension

Parameter	Description
Status	Enable or disable SIP extension.
Index	The index of SIP extension.
Display Name	The name of this SIP extension.
Extension	The SIP account of the extension registered in UC350 a SIP client.
Password	The password of the SIP account registered in UC350 a SIP client.
SIP Password	The password for the new SIP extension is a password randomly generated by the device by default and is used as the client authentication password. Click the "eye" to display it in plain text; text input, 8-32 characters
App Password	Create a new SIP extension APP password. The default is a password randomly generated by the device and used as the APP authentication password. Click "eye" to display it in plain text; text input, 8-32 characters
Classification Tag	Labels for extension classification.
DID	Direct Inward Dialing. If the called number is same with DID, the call will be directly forwarded to the extension, rather than choosing a route. Users can set multiple DID.

Outbound CID	After the outgoing caller number is configured, the caller number dialed from the SIP extension is replaced with the number configured here.
SIP Profile	The SIP profile that is selected for the extension.
DinLink Client	After enabling, the APP can use this extension number to register
Speed Dial	Configuration for Speed dial.
SCA	When enabled, it can be configured in **Advanced Service** > **SCA** interface.
Do Not Disturb	If 'Do Not Disturb' feature is enabled, calls cannot reach the called party.
Call Waiting	If a calling party places a call to a called party which is otherwise engaged, and the called party has the call waiting feature enabled, the calling party will hear an IVR voice.
Call Pickup	After configuration, the designated call can be picked up (ring group/local extension, the default is the ring group).

Call Forward Unconditional	If 'Call Forward Unconditional' feature is enabled, all coming calls will be forwarded to a preset number. **Designated Forward Unconditional**: if value is empty or null, busy call forwarding will be activated for all incoming numbers, unconditional forwarding will apply to all incoming numbers; if a specific number is set, only calls from that number will be forwarded. For example, if you enter the number 13200010002, only calls from 13200010002 will be forwarded, while calls from other numbers will be answered normally.
Call Forward Unregister	When the SIP extension is not registered, users can transfer all the calls to the set number.

Call Forward Busy	If 'Call Forward Busy' feature is enabled, new coming call will be forwarded when the corresponding local port is busy. **Designated Forward Busy**: If value is empty or null, busy call forwarding will be activated for all incoming numbers. If a specific number is set, only calls from that number will activate busy call forwarding when busy. For example: if you enter the number 13200010002, then only calls from 13200010002 will be forwarded when busy. Calls from other numbers will receive the standard busy alert tone.

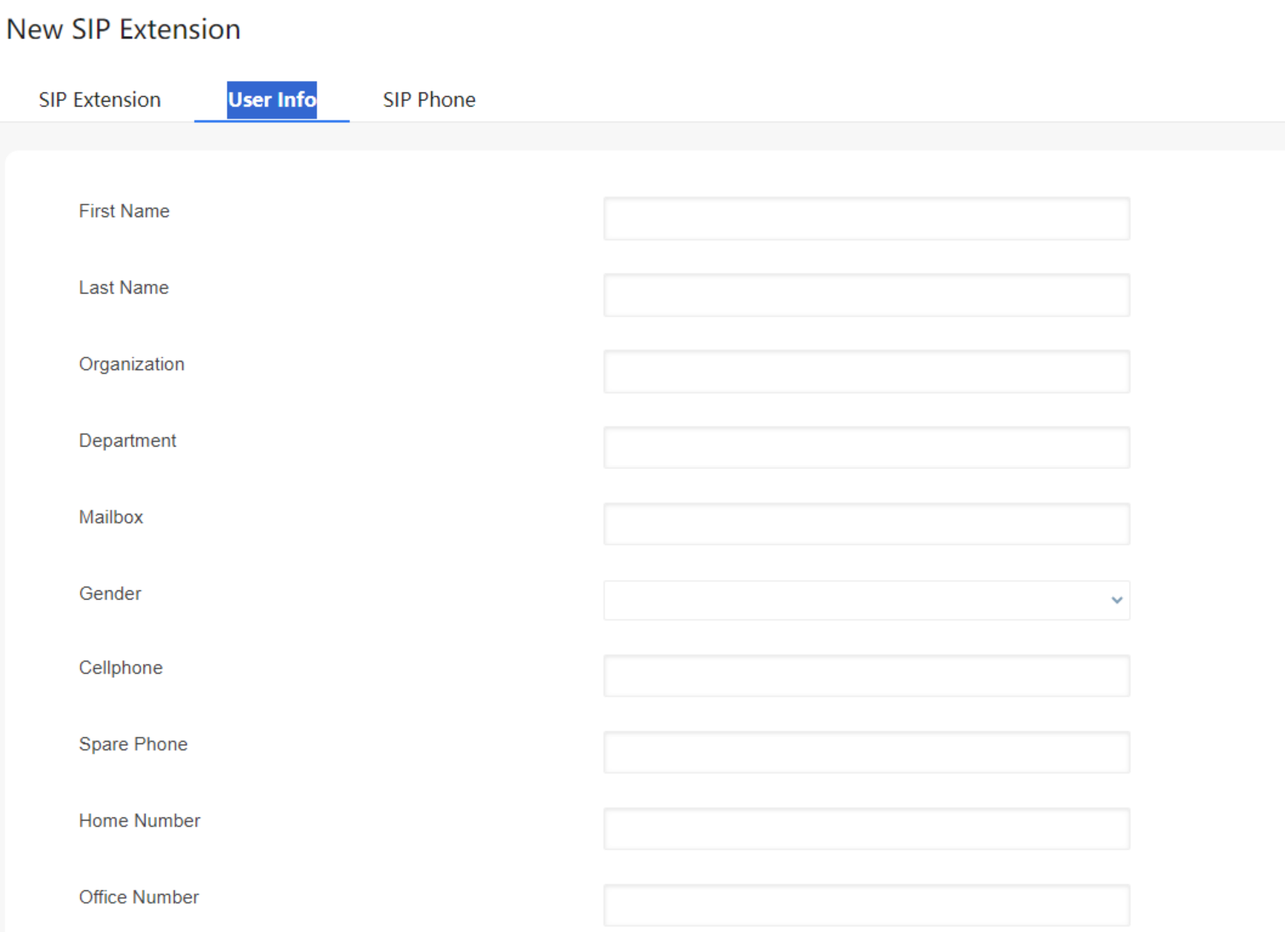
Call Forward No Reply	If 'Call Forward No Reply' feature is enabled, calls will be forwarded when nobody answer the calls during a specified period. **Designated Forward No Reply**: If value is empty or null, busy call forwarding will be activated for all incoming numbers, no reply call forwarding will be activated for all incoming numbers. If a specific number is set, only calls from that number will activate no reply call forwarding when no reply. For example: if you enter the number 13200010002, then only calls from 13200010002 will be forwarded when no reply. Calls from other numbers will hang up after timeout.
Call Back When Dest Ext Busy	After enabled, when Extension A dials Extension B which is busy, the system will detect the status of Extension B and will call back when Extension B is idle.
Priority	Normal or high
Ringtone	When enabled, the configured ringtone will be played during a call to this extension.

Ring Timeout(s)	The ringing timeout period for incoming calls to this extension, the default value is 50. If the extension does not go off-hook within 50s after ringing, the device will initiate disconnection.
Allow Being Monitored	Enable to allow being monitored.
Monitor Mode	Configure Monitor Mode of extension.
Recording Profile	When recording is enabled, calls will be recorded according to the recording rules.
Voicemail	Choose to on or off the voice mail.
Password	Configure the password for logging in to the extension's voice mail.
Message Forward Email	Configure the e-mail address for voice mail messages, and make sure that the e-mail is normally.
Area Call Permission	Internal, local, domestic and international permissions can be selected. If it is not checked by default, SIP extensions can make calls in the above areas; if checked, it corresponds to different regional permissions
Call In Filter	When users breathe in to SIP, users match the relevant filter conditions.
Call Out Filter	When the SIP is called out, The filter conditions are matched.

PIN Code	When configured, it can be used for phone auto-provision.
Daily Call Limit	Once enabled, you can customize the daily frequency of internal, external and international calls
Expire Days	Set the validity period of the extension number after registration
No Login For x Consecutive Days Is Automatically Disabled	Customize whether the extension number is disabled after x days of inactivity
Register Source	If 'Any' is chosen, all SIP clients are allowed to register the SIP account of this extension. If 'Specified' is chosen, only the SIP client with the specified IP address or network segment is allowed to register the SIP account of this extension. For example, 172.16.0.0/16 means the register source is 172.16
Max Concurrent Register	Number of clients that can register online at the same time.
Register User Agent	Filter the user agent field in the register message during registration.
Max Concurrent Call	The number of concurrent calls that can be made at the same time.

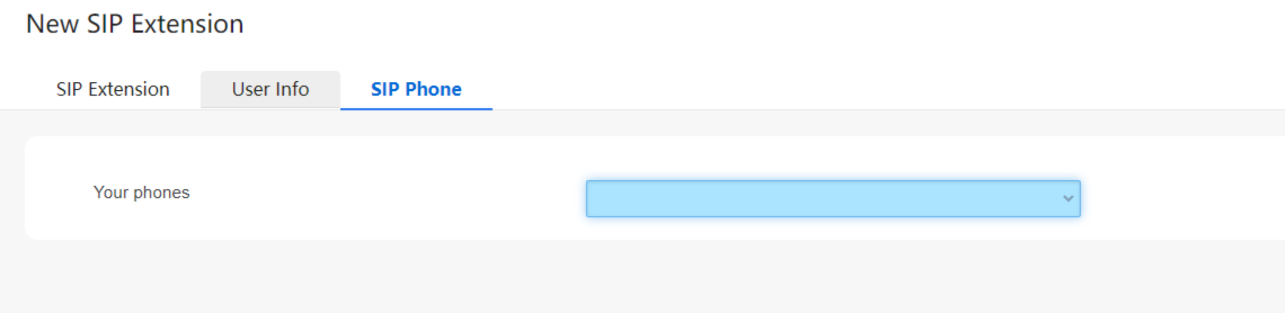
Max Call Duration(s)	Limit the duration of each phone call, unit: seconds
Original Called Number Location(Send INVITE)	When sending INVITE, configure the location of the original called number.
NAT	If NAT is enabled, the IP address of SIP extension in LAN will be bound into the outbound IP address of public network, thus making NAT traversal possible.

Figure-Parameters of User Info



For parameter details, please refer to 5.6 Address Book – Contacts

Figure-Parameters of SIP Phone



SIP phones are used to send configurations to IP phones

5.3.2 FXS

On the **Extension & Call Group > FXS** interface, users can configure the parameters of the FXS extension.

Figure-Parameters of FXS Extension Configuration

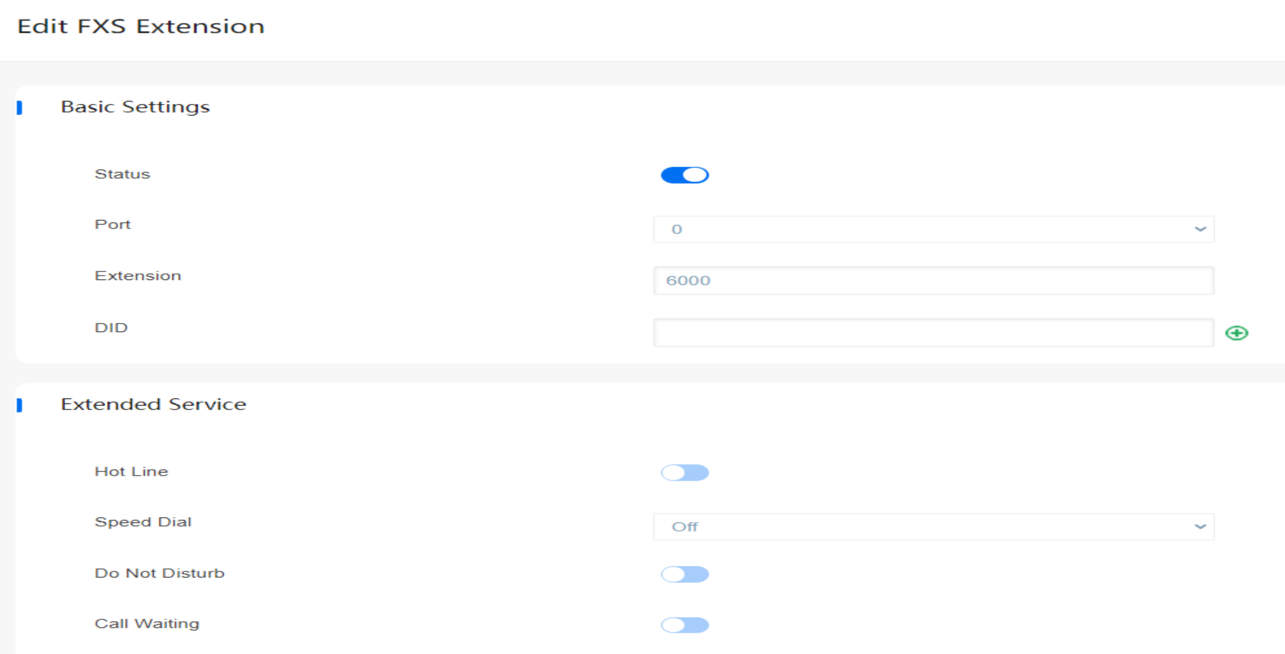


Table-Parameters of FXS Extension Configuration

Parameter	Description
Status	Enable or disable FXS extension.
Port	Select the port that needs to be configured.
Extension	The extension account of FXS port, which is used to register.
DID	Direct Inward Dialing. If the called number is same with DID, the call will be directly forwarded to the extension, rather than choosing a route.
Hot Line	If hotline is enabled, calls will directly go to the hotline number.
Speed Dial	Select Speed Dial Configuration; pull down to select, close/the rules in "Configuration > Speed Dial"
Do Not Disturb	If 'Do Not Disturb' feature is enabled, calls cannot reach the called party.
Call Waiting	If a calling party places a call to a called party which is otherwise engaged, and the called party has the call waiting feature enabled, the calling party will hear an IVR voice.
Call Pickup	After configuration, the designated call can be picked up (ring group/local extension, the default is the ring group).

Call Forward Unconditional	If 'Call Forward Unconditional' feature is enabled, all coming calls will be forwarded to a preset number.
Call Forward Busy	If 'Call Forward Busy' feature is enabled, new coming call will be forwarded when the corresponding local port is busy.
Call Forward No Reply	If 'Call Forward No Reply' feature is enabled, calls will be forwarded when nobody answer the calls during a specified period.
Priority	Configure priority of extension.
Ringtone	When enabled, the configured ringtone will be played during a call to this extension.
Ring Timeout(s)	The ringing timeout period for incoming calls to this extension, the default value is 50. If the extension does not go off-hook within 50s after ringing, the device will initiate disconnection.
Allow Being Monitored	Enable to allow being monitored.
Monitor Mode	Configure monitoring mode. For detailed function description, refer to the feature code "*164*"; pull down to select disable/listening mode/whisper mode/force insertion mode
Recording Profile	When recording is enabled, FXS calls will be recorded according to the recording rules.
Voicemail	Choose to on or off the voice mail.

Call In Filter	When a call is given to the FXS port of UC350, the call will not be connected to the FXO port if it is in the blacklist.
Call Out Filter	When a call goes out from the FXS port of UC350, the call cannot go out if it is in the blacklist.
Work Mode	Configure the working mode of the FXS port.
Voice Output Mode	Configure the voice output mode of the FXS port.
Gain Configure Mode	Select the gain configuration mode of the FXS port (general settings and advanced settings), TX gain and RX gain are newly added to advanced settings than general settings.
TX Gain(IP->PSTN)	The volume level of the remote end during a call, that is, adjusting the "TX gain" will affect the volume of the sound heard by the remote end.
RX Gain(PSTN->IP)	The volume level of the user during the call, that is, adjusting the "RX gain" will affect the level of sound users hear.
CID Send Timing	Set the phone caller ID display before ringing or after ringing.

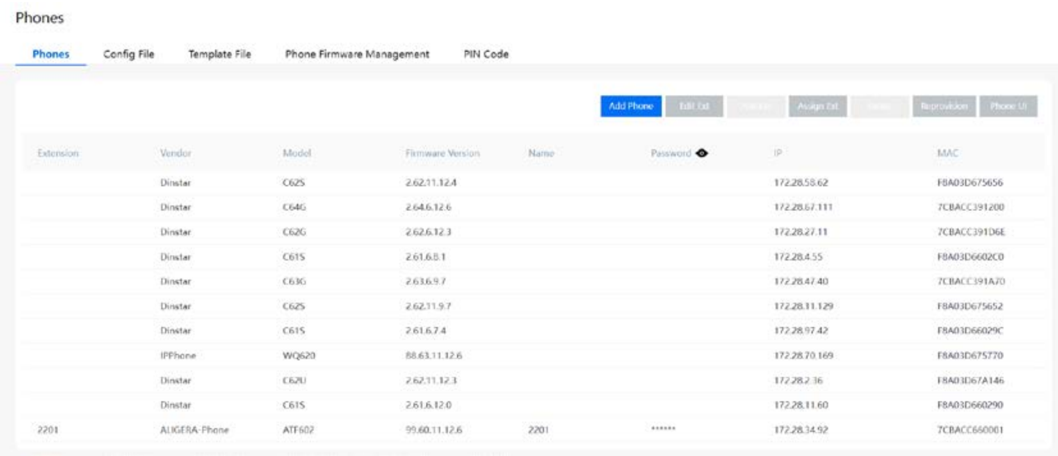
Delay Timeout After Ring(ms)	CID sending timing is configured when sending after ringing. Set how long the phone will ring before sending CID. When "CID send timing" is configured to "send after ring", users need to configure the delay timeout, that is, how long to ring before sending the CID.

5.3.3 Phones

On the **Extension & Call Group > Phones** interface, the user can configure the configuration file to the phone according to the template file.

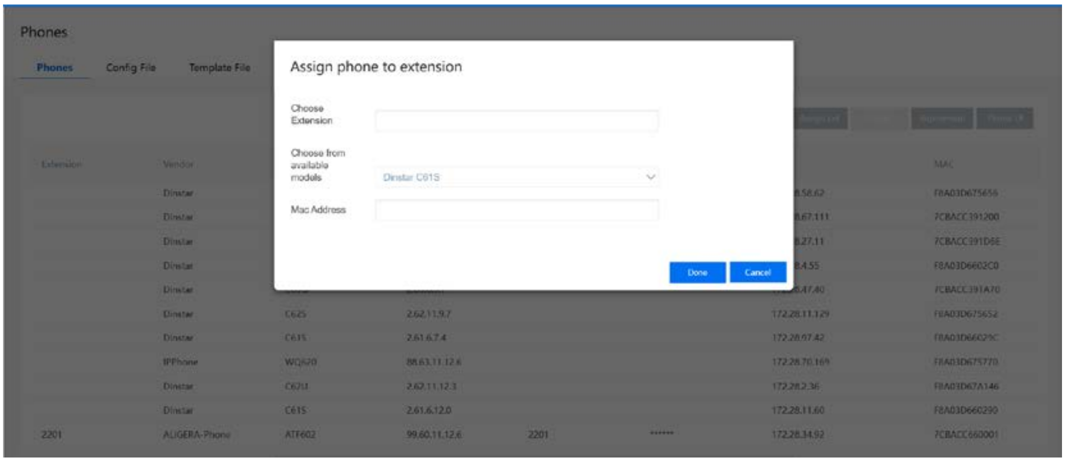
After enabling PNP, the phone will periodically send a Subscribe message to the multicast address. If the PBX receives the multicast message, it will list the phone models in the PBX configuration list.

Figure-Parameters of Phones



If the phone is in the configuration list of UC, after selecting, user can assign the phone to an existing extension, or user can create a new extension through "Add phone".

Figure-Add new phones



At this time, UC will automatically generate a configuration file suitable for the phone model, and send a SIP NOTIFY message to the phone, carrying the download address of the configuration file in the body, and notify the phone to download. After the phone receives it, users can use the assigned extension to register.

5.3.4 Ring Group

On the **Extension & Call Group > Ring Group** interface, users can group FXS extension and SIP extension(s) together and set strategy for choosing the FXS extension and which SIP extension to ring under a ring group. The ring group function is widely used in call centers.

Figure-Parameters of Ring Group

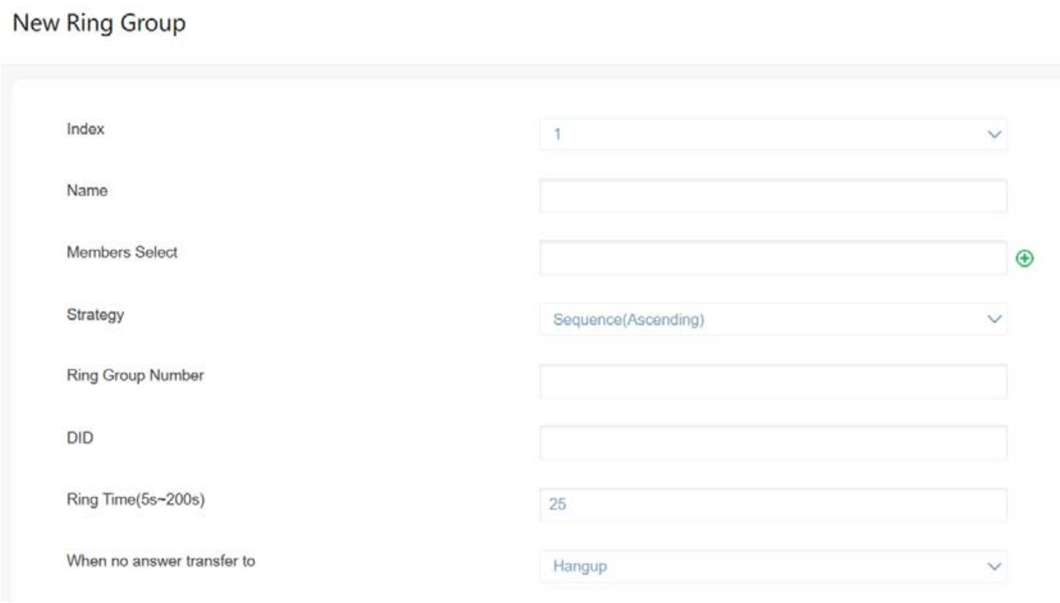


Table-Parameters of Ring Group

Parameter	Description
Index	The index of Ring Group.
Name	The name of this ring group.
Members Select	Select the FXS extension and an SIP extension or several SIP extensions.
Strategy	The strategies for choosing which SIP extension to ring, including Sequence (Ascending), Sequence (Cyclic Ascending), Simultaneous and Random.
Ring Group Number	The number of the ring group. It is generally the same with DID.
DID	Same with Ring Group Number. It is optional to fill in.

Ring Time(5s~200s)	The duration of ring when there is a coming call. Range: 5s to 60s.
When no answer transfer to	When none of the members in the ring group answer, users can transfer the call to a specified extension or hang up.

NOTE: If ring group function has been set, the call forwarding function is unavailable.

5.3.5 Intercom/ Paging Group

On the **Extension & Call Group >Intercom/ Paging Group** interface, users can group SIP extensions into a Intercom/ paging group and then if there are calls given from FXS/FXO/SIP to the Intercom/paging group, the calls will be led to one extension of the Intercom/paging group according to the preset strategy.

Figure-Parameters of Intercom/Paging Group

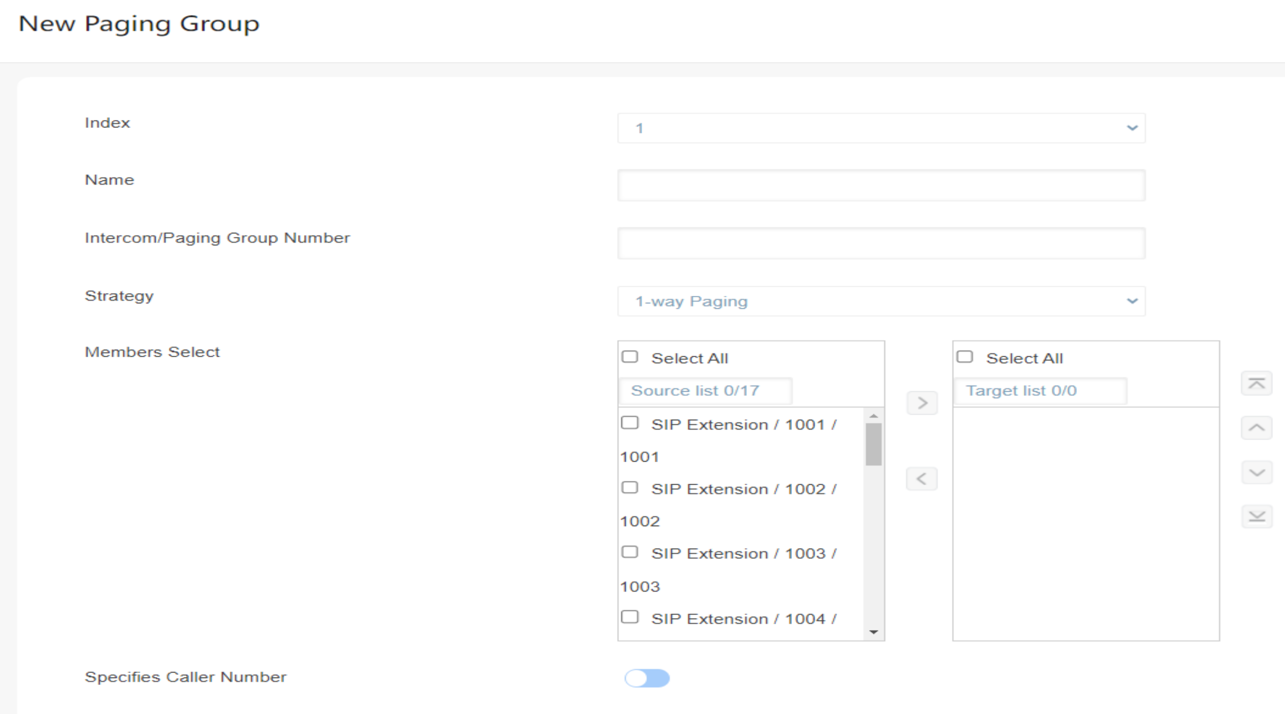


Table-Parameters of Paging Group

Parameter	Description
Index	The index of this paging group.
Name	The name of this paging group.
Intercom/Paging Group Number	The number of the Intercom/paging group. When there calls given from FXS/FXO/SIP to this number, the calls will be led to one extension of the Intercom/paging group according to the preset strategy.
Strategy	Include one-way paging and two-way intercom. one-way paging：members of the paging group only can listen to the voice of presenter and cannot answer the call. two-way intercom：members of the paging group can have conversation with the presenter, but members cannot talk to each other.
Members Select	Select the SIP extensions that are added into the paging group. An SIP extension cannot exist in two paging groups at the same time.
Specifies Caller Number	After opening, users an specify a number, and only this number can be the caller to make an incoming call
Verify PIN Code	After opening, users an set the password to be entered when calling into the intercom/paging group.

Media Play	Users an choose to play the media of the called party when calling the intercom/paging group, customize the number of times to play, and choose the corresponding actions of the caller and the called party after the playback.
Timing Trigger	Users an customize the SIP extension to call the intercom/broadcast group at a specific time

5.4 Advanced Service

5.4.1 IVR

On the **Advanced Service** > **IVR** interface, users can carry out specific configurations for the IVR which has been uploaded from the **PBX Global Settings** > **Voice** interface. IVR is often used for voice prompts in call centers.

Figure-Parameters of IVR

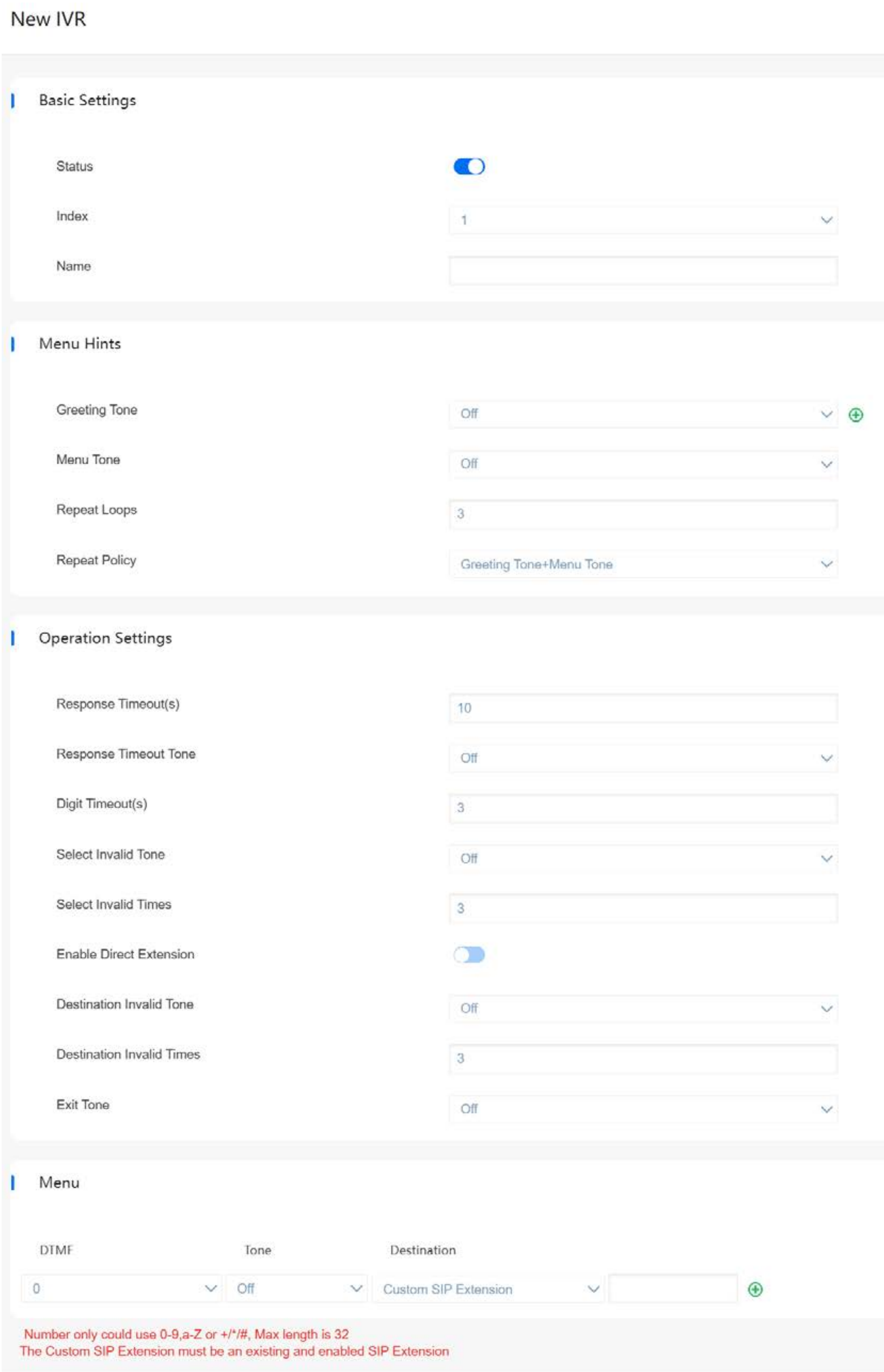

New IVR
Basic Settings
Status
Index
1
Name
Menu Hints
Greeting Tone
Off
Menu Tone
Off
Repeat Loops
3
Repeat Policy
Greeting Tone+Menu Tone
Operation Settings
Response Timeout(s)
10
Response Timeout Tone
Off
Digit Timeout(s)
3
Select Invalid Tone
Off
Select Invalid Times
3
Enable Direct Extension
Destination Invalid Tone
Off
Destination Invalid Times
3
Exit Tone
Off
Menu
DTMF
Tone
Destination
0
Off
Custom SIP Extension
Number only could use 0-9,a-Z or +/*/#, Max length is 32
The Custom SIP Extension must be an existing and enabled SIP Extension

Table-Parameters of IVR

Parameter	Description
Status	Enable or disable IVR.
Index	The index of the IVR.
Name	The name of the IVR.
Greeting Tone	The default is disabled, and users can use the upload tone. When a call comes to the IVR, play the greeting tone first and then the Menu tone.
Menu Tone	When a call comes to the IVR, the menu tone heard.
Repeat Loops	If it is set as '3', the call will be hung up after the IVR has been repeated for three times during timeout.
Repeat Policy	It can be configured with "Greeting Tone + Menu Tone" or "Menu Tone".
Response Timeout(s)	When a call comes to the IVR, according to the voice prompt, the second dial is not received within the set time, the response is timed out, and the timeout tone is played.
Response Timeout Tone	When the second dialing timeout, the timeout will be played after being enabled.
Digit Timeout(s)	The timeout for dialing DTMF.

Select Invalid Tone	When an invalid dial is received, an invalid tone will be played.
Select Invalid Times	When a call comes to the IVR, according to the voice prompt, if users receive two dialings that do not match the DTMF, then the dialing is invalid, and the invalid prompt tone is played. When the invalid times is exceeded, the voice prompt: Goodbye.
Enable Direct Extension	Whether to allow direct dialing of extensions during the playing of IVR.
Destination Invalid Tone	When receiving an invalid destination dial, the invalid tone will be played.
Destination Invalid Times	It takes effect when the direct extension is enabled. When users call into the IVR, and the entered number does not exist, the destination invalid prompt will be played. When the time of entries exceeds the set value, the voice prompt: Goodbye.
Exit Tone	When exiting IVR, the exit tone will be played.

Table-Parameters of IVR Menu

Parameter	Description
DTMF	DTMF number, select the number of the destination.
Others	IVR destination when the dialed DTMF is not in the selected number list.
Timeout	Destination of IVR when DTMF is not dialed for a set period of time.
DTMF as Destination Numb	Destination of IVR when DTMF is used as a destination.
Tone	The tone that is played before the callee rings.
Destination	Destination type for IVR, which can be: Custom SIP Extension, Extension, Trunk, Call Queue, IVR, Previous Menu, Exit, and Repeat.

5.4.2 Call Queue

On the **Advanced Service > Call Queue** interface, the user can add the local extension to a queue. When calling into the call queue, the system will transfer the call to the queue member/agent to answer the call according to the strategy.

For example, when a large number of customers call in at the same time, and the customer service staff is limited, queue the incoming and play a voice waiting tone or custom music file. At the same time, the agent can answer the call according to the preset call queue strategy.

Figure-Parameters of Call Queue

Table-Parameters of Call Queue

Parameter	Description
Index	The index of the call queue.
Queue Name	The name of the call queue.
Queue Number	The number of the call queue can be called into the queue.
Type	Common queue: corresponds to the current call queue function, suitable for scenarios where the phone is used as an agent terminal Attendant console queue: used to support the console service. To use the Web console, you must first create this type of call queue

Call Allocation Policy	Calls into the queue, the agents ring according to the strategy. Simultaneous: The agents ring together. Linear: When there is no incoming call, a new user calls in, each time it will ring sequentially from the first agent). Random: one is randomly selected for ringing. Memory round robin: When there is no incoming call, a new user calls in, and the ringing starts from the next agent who hangs up last before. Least recent, namely the time from the end of the agent's last call to the present, ringing in the order from longest to shortest time. Fewest calls: The ringing starts from the least to the most according to the times of calls.
Menu Tone	The menu prompt tone that the user hears first when calling in
Waiting Music	The remote end waits for the agent to answer the waiting tone after calling in.
Enable Position Announcement	Timely notify the user of the waiting position in the queue, the first one does not notify.

Maximum Queue Time(0,300)	The longest time the caller waits. The caller will exit after this time. 0 means no limit, but it should be noted that this time is not necessary. For example, an agent is ringing and the caller has reached the timeout. The caller will wait until the agent answers or hang up after the timeout.
Queue Timeout Policy	If the caller times out, other actions can be configured.
Max Queue Length	How many users are waiting, those connected are not counted, 0 means no limit, hang up if the maximum number of queues is exceeded.
Quantity Overlimit Policy	After the maximum number of queues is exceeded, new call transfer configuration, optional hang up/extension/play
Agent Members	Agent members can only belong to one queue Common queue: select FXS extension and SIP extension, move left to add a member, move right to delete a member Attendant console queue: members can only add SIP extensions, not FXS extensions
Break Time After The Call(5,300)	The interval between the next ring after the agent hangs up the phone; text input, valid value range: 5-300s

Agent Ring Time(5s~300s)	If the ringing exceeds the time, it will call to the next agent.
Max No Answer	If the times that the agent does not answer is exceeded, it will enter On-Break state, in this state, it will not be ringing again until the agent answer.
Strategy for Agents Offline	When the queue is empty, users can select: Continue to wait, Hangup, Play Music, Custom SIP Extension, Call Queue, IVR, FXO Trunk, SIP Trunk
Extension List Members	When the call queue type is set to "Common Queue", you can add extension list members. The added extension will be displayed in the "Extension List" of the Web Console client, which can show the status of the extension;

Dynamic Agent Login Setting

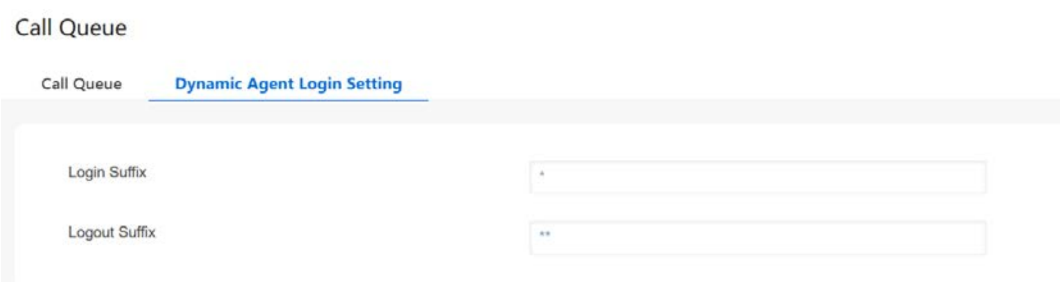


Table-Parameters of Dynamic Agent Login Setting

Parameter	Description
Login Suffix	Extensions dial "Call Queue Number" + login suffix, log into the specified queue, and register as an available member of the queue.
Logout Suffix	Extensions dial "Call Queue Number" + logout suffix to exit from the specified queue and stop receiving calls assigned to the queue.

Note: The seats added in this way are dynamic seats and are only displayed on the status page

5.4.3 Conference

On the **Advanced Service > Conference** interface, users can create a conference room, and the caller can create a multi-party conference by dialing the number of the conference room.

Figure-Parameters of Conference

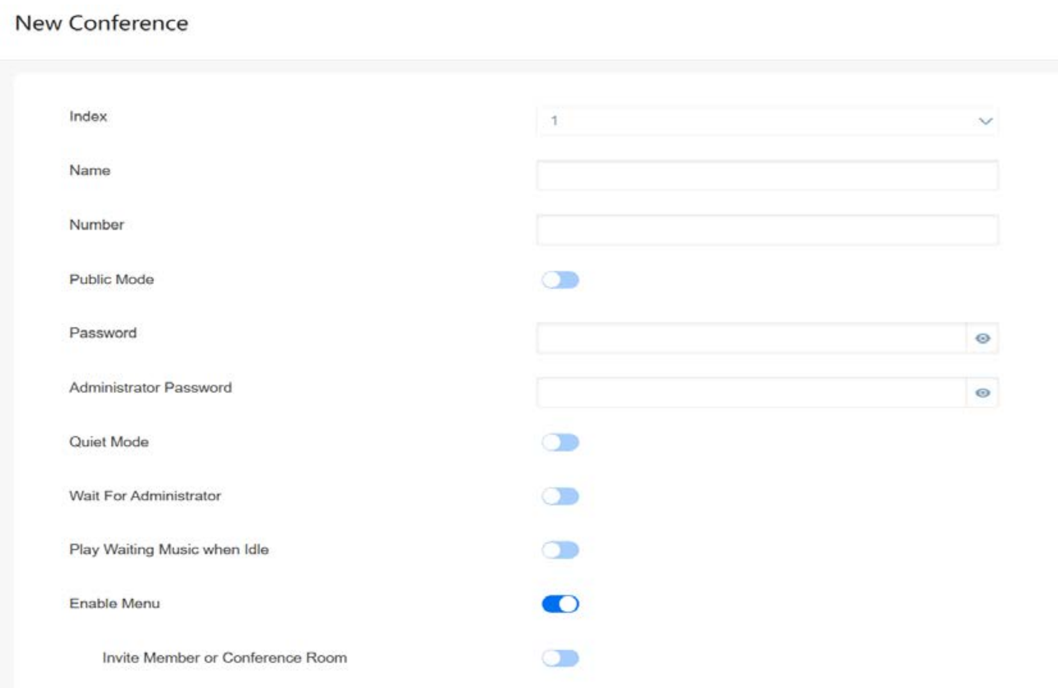


Table-Parameters of Conference

Parameter	Description
Index	The index of the conference room.
Name	The name of the conference room.
Number	Conference room number, the extension can join the conference by dialing this number.
Public Mode	No password is required to join the conference in public mode.
Password	The password for users to join the conference in non-public mode.
Administrator Password	Administrator password: the password for the administrator to join the conference in non-public mode. A conference can have multiple administrators, and the administrator password cannot be blank.
Quiet Mode	When the quiet mode is enabled, the conference will not hear any voice.
Wait For Administrator	Once enabled, the conference will only start after the administrator join the conference, otherwise it will be idle.
Play Waiting Music when Idle	When the conference is in idle, the waiting tone will be played after being enabled.
Enable Menu	After activation, everyone can use the menu through DTMF.

Invite Member or Conference Room	After enabling the menu, non-administrators can invite members to join the conference room.
Recording	After turning it on, the conference will be recorded

Table-Parameters of Conference Menu

DTMF	Description	Notes
1	Invite members	Non-administrators need to enable configuration
2	Invite members, need to be confirmed by the invite	Non-administrators need to enable configuration
3	Initiate a conference	Non-administrators need to enable configuration
4	Decrease the volume of the handset	/
6	Increase the volume of the handset	/
7	Decrease the volume of the microphone	/
9	Increase the volume of the microphone	/
*	Mute	/

0	All non-administrators are muted	Administrator permissions
#	Exit all non-administrators from the conference	Administrator permissions

Menu instructions:

Invite members: Invite multiple SIP extensions

1) After pressing 1, it will prompt to enter the number and the extension number.
2) The extension rings.
3) After the extension is connected, join the conference as a non-administrator.

Invite members (requires confirmation):

1) After pressing 2, it will prompt to enter the number and the extension number.
2) The extension rings.
3) After the extension is connected, users hear the prompt that users will join the conference, press 1 to join the conference as a non-administrator, press 2 or other to hang up.

Invite a conference: The conference room is activated

1) After pressing 3, it will prompt to enter the conference room number.
2) If there is a password, users will be prompted to enter the conference room password.
3) Connect to the meeting.

5.4.4 Voicemail

On the **Advanced Service > Voicemail interface**, users can configure the location, number and duration of a voicemail.

How to use voicemail:

Navigate to **Extension & Call Group > SIP Extension** interface, enable the voicemail function, and the voicemail will be activated when the call times out.

Figure-Parameters of Voicemail Configuration

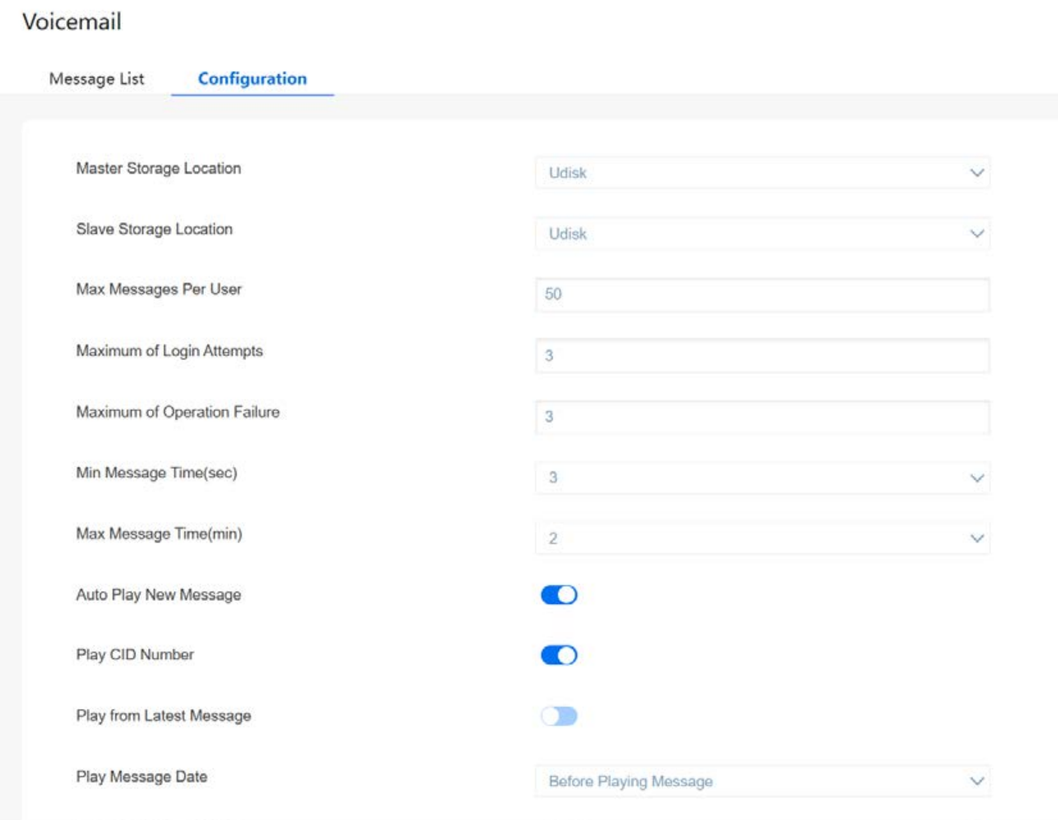


Table-Parameters of Voicemail Configuration

Parameter	Description
Master/Slave Storage Location	Select local or Udisk to store voice files.
Max Messages Per User	If this maximum number of messages is reached, a prompt voice "the mail box is full" will be played.
Maximum of Login Attempts	If this maximum number of attempts (by dialing *170*2 to log in the voice mailbox) is reached, the call will hang up.

Maximum of Operation Failure	When a call enters into the voice mailbox and the caller dial inexistent DTMF repeatedly, the caller will be forced to log out the voice mailbox after the repetition times exceed this value.
Min Message Time(sec)	The minimum duration of a voice mail.
Max Message Time(min)	The maximum duration of a voice mail.
Auto Play New Message	If this parameter is on, new messages will be played automatically. If it is off, a prompt voice "please dial 1 to listen to new message" will be given.
Play CID Number	If this parameter is on, the caller number will be played together with messages.
Play from Latest Message	If this parameter is on, the latest messages will be played first.
Play Message Date	When to play message date. User can choose 'Before Playing Message', 'After Playing Message' and 'Never'.

Figure-Parameters of Voicemail Message List

5.4.5 Speed Dial

On the **Advanced Service > Speed Dial** interface, users can configure the correspondence between short and long numbers. For example, if the short number (speed dial number) is set as 1, the long number is set as 8000, and this speed dial profile is applied to an SIP extension, the SIP extension only needs to dial 1 and the call will be directed to the extension number of 8000.

Figure-Parameters of Speed Dial

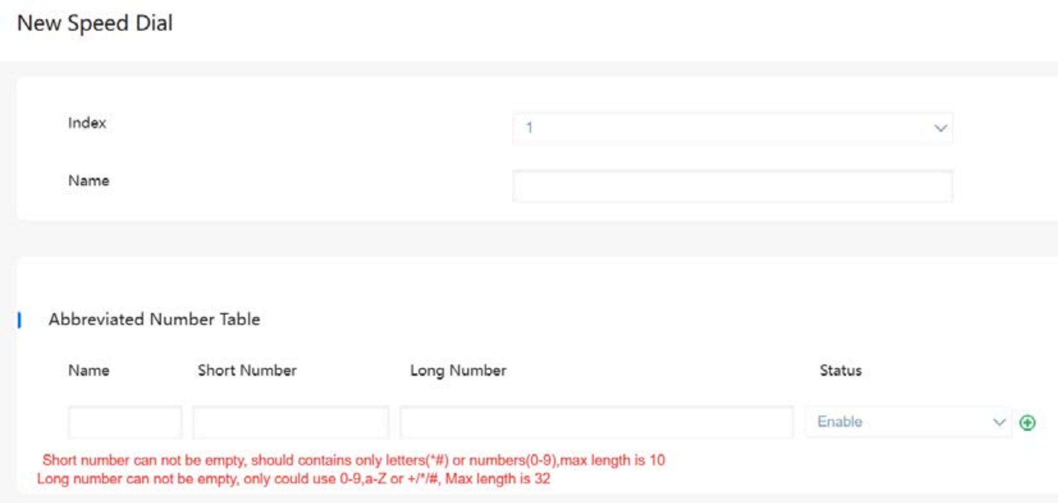


Table-Parameters of Speed Dial

Parameter	Description
Index	Numbering of speed dial rules, drop-down selection, 1-32.
Name	Name of speed dial rule, text input cannot be empty, less than 32 characters.
Abbreviated Number Table	Short numbers and long numbers correspond to the abbreviated number table, can add more than one, the maximum add 104.

Name	Name of the abbreviated number table, text input can be empty, less than 32 characters.
Short Number	Short number configuration, text input, support numbers 0-9/*/#, maximum support 2 characters.
Long Number	Short numbers corresponding to long numbers, text input, only numbers, less than 32 characters.

5.4.6 Dial plan

Dialing rules are used for dialing settings when an extension call occurs. It supports Regular Expression (Regex) and DigitMap.

Figure-Parameters of Dial plan

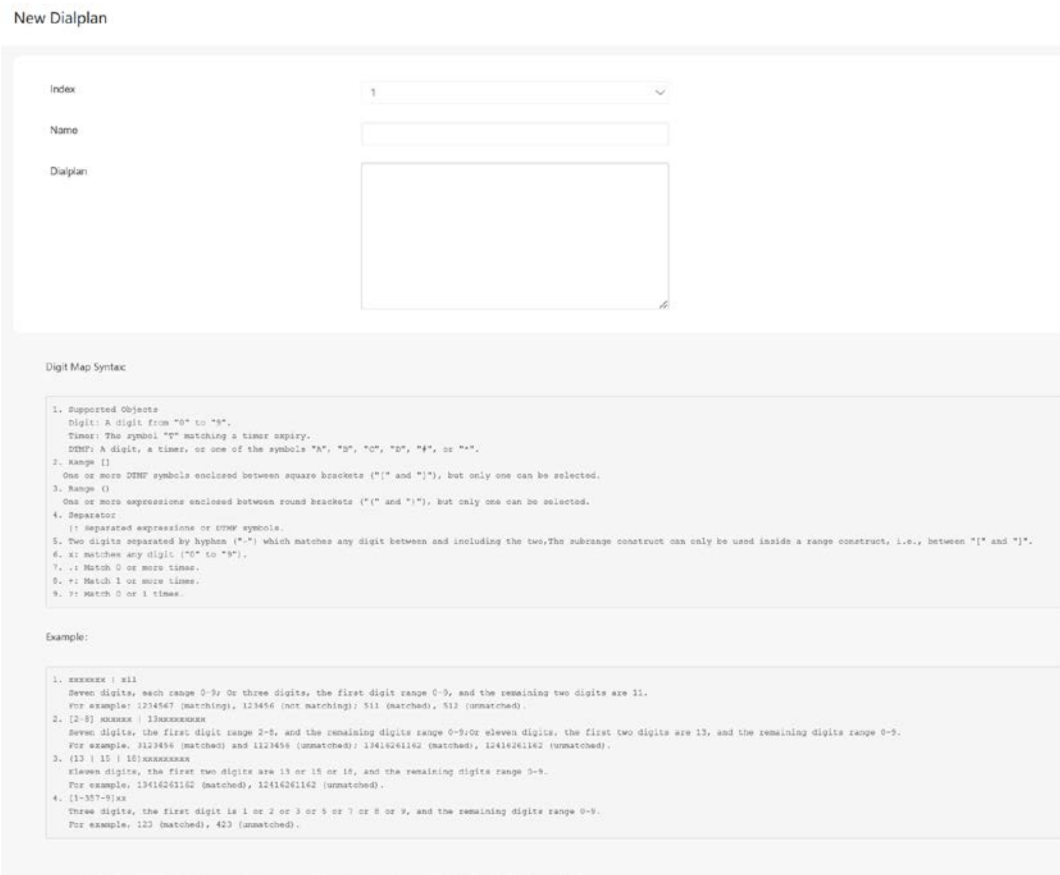


Table-Parameters of Dial plan

Parameter	Description
Index	The index of the Dialplan.
Name	The name of the Dialplan.
Dialplan	Set Dialing rules.

Table-Regex (Regular Expression) Syntax

<table>
<tr><td rowspan="3">Supported Objects</td><td>Digit</td><td>0-9</td></tr>
<tr><td>T</td><td>Timer</td></tr>
<tr><td>DTMF</td><td>A digit, a timer, or one of the symbols of A, B, C, D, #, or *</td></tr>
<tr><td>Range</td><td>[ ]</td><td>One or more DTMF symbols enclosed in the [], but only one DTMF symbol can be selected</td></tr>
<tr><td>Range</td><td>()</td><td>One or more expressions enclosed the (), but only one can be selected</td></tr>
<tr><td>Separator</td><td>|</td><td>Separate expressions or DTMF symbols</td></tr>
<tr><td>Subrange</td><td>-</td><td>Two digits separated by hyphen (-) which matches any digit between and including the two digits</td></tr>
<tr><td>Wildcard</td><td>x</td><td>Matches any digit of 0 to 9</td></tr>
<tr><td>Modifiers</td><td>.</td><td>Matches 0 or more times of the preceding element</td></tr>
<tr><td>Modifiers</td><td>?</td><td>Matches 0 or 1 times the preceding element</td></tr>
</table>

Table-Examples of DigitMap Syntax

(13 |15 | 18)xxxxxxxxx	Matches the phone numbers with stating digits as 13, 15 or 18 and the left nine digits as any of 0 to 9.
[2-8]xxxxxx |13xxxxxxxxx	Matches the phone numbers starting with any digit of 2 to 8 and the left six digits as any of 0 to 9, or matches the phone numbers starting with 13 and the left nine digits as any of 0 to 9.

5.4.7 Follow Me

After the operator enables Follow Me, users can unify their common various communication numbers (cell phone, pager, office phone, voice mail, residential phone) into a new phone number, so that anyone can simply dial this phone number to find the user in the future.

An extension can be tied to a string of extensions and trunks, so that when no one answers a call to that extension number, it will go ring its list.

Operation steps:

1. On the **Advanced Service >Follow Me** interface, click New.
2. Save the application.
3. Any number dialing the extension number such as 100, will ring based on the corresponding ringing strategy. If it is sequential (incremental), it will ring from the extension 100, and after the timeout, it will ring the next number in turn according to the order of the extension following list. If it is resonant, the extension 100 will ring together with other destination numbers until it is connected or timeout.

Note:

- Extension call forward is not valid for Follow Me.
- The same SIP extension cannot be used for both SCA and Follow Me.

Figure-Parameters of Follow Me

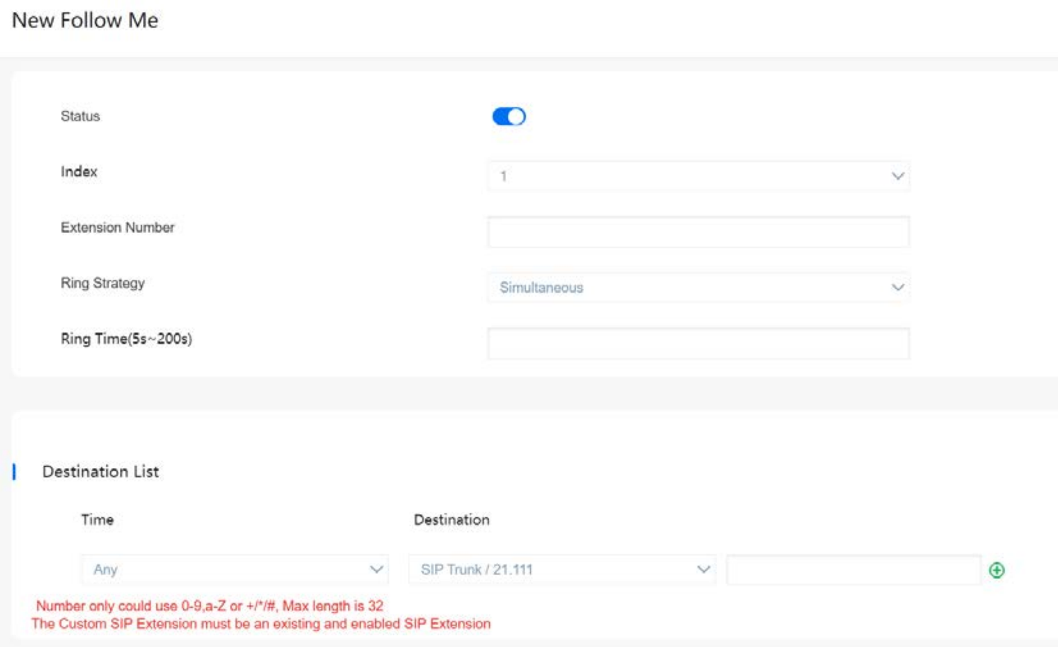


Table-Parameters of Follow Me

Parameter	Description
Status	Enable or disable follow me feature.
Index	The index of the follow me.
Extension Number	Select the extension to enable this feature, users cannot select the SIP extension with the SCA enabled and the SIP extension as secretary.
Ring Strategy	Support simultaneous and sequence (ascending). simultaneous is all numbers ring together, and the sequence starts from the extension and rings from top to bottom.
Ring Time(5s~200s)	Ringing time per number.

Time	Any is unlimited. If users choose to set the time period as shown above1-<Time>, they will only be called during this time period.
Destination	Other numbers for this extension, users can select SIP extension, SIP trunk Relay, fill in the extension number to be called when selecting the relay.

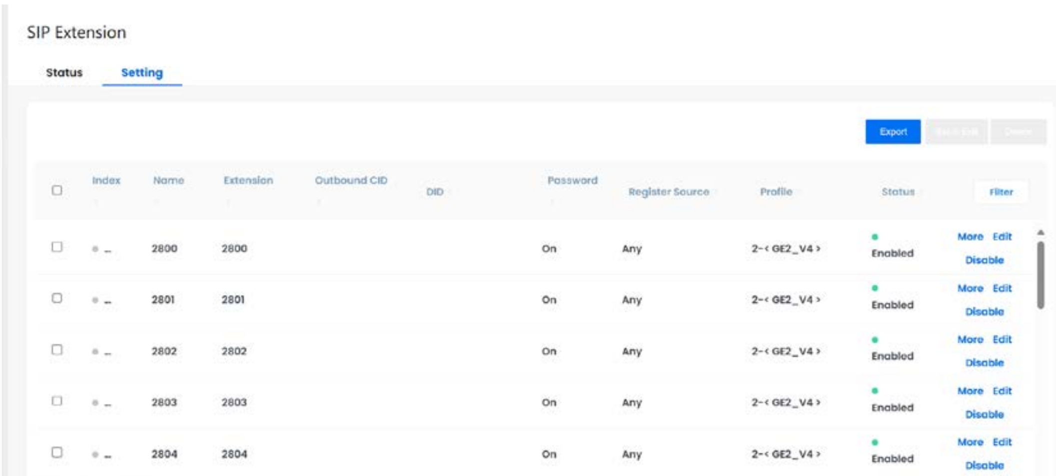
5.4.8 SCA

When someone calls a company manager, the secretary will receive the call first and determine whether to forward the person's call to the manager. Sometimes the manager wants to answer the call directly, so a switch is used to control whether the manager can receive the call directly. Managers and secretaries can also call each other.

Operation steps:

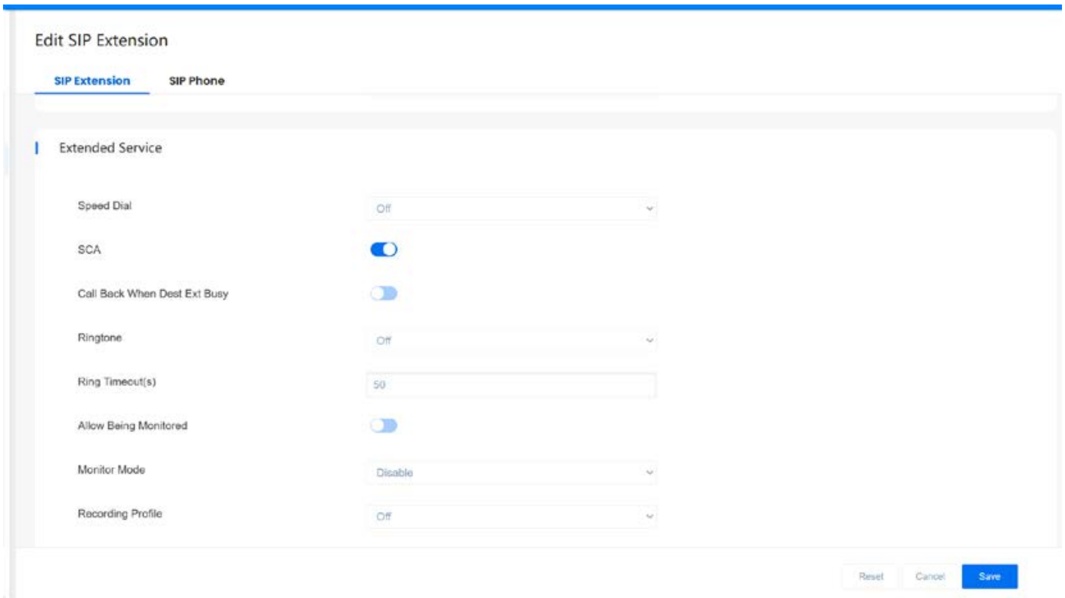
On the **Extension & Call Group > SIP Extension** interface, users can select the extension where they want to open the SCA.

Figure-Select the extension



Edit the SIP extension to be on and open the SCA.

Figure-Open the SCA



On the **Advanced Service >SCA** interface, users click on new, and select each option as follow.

Figure-Parameters of SCA

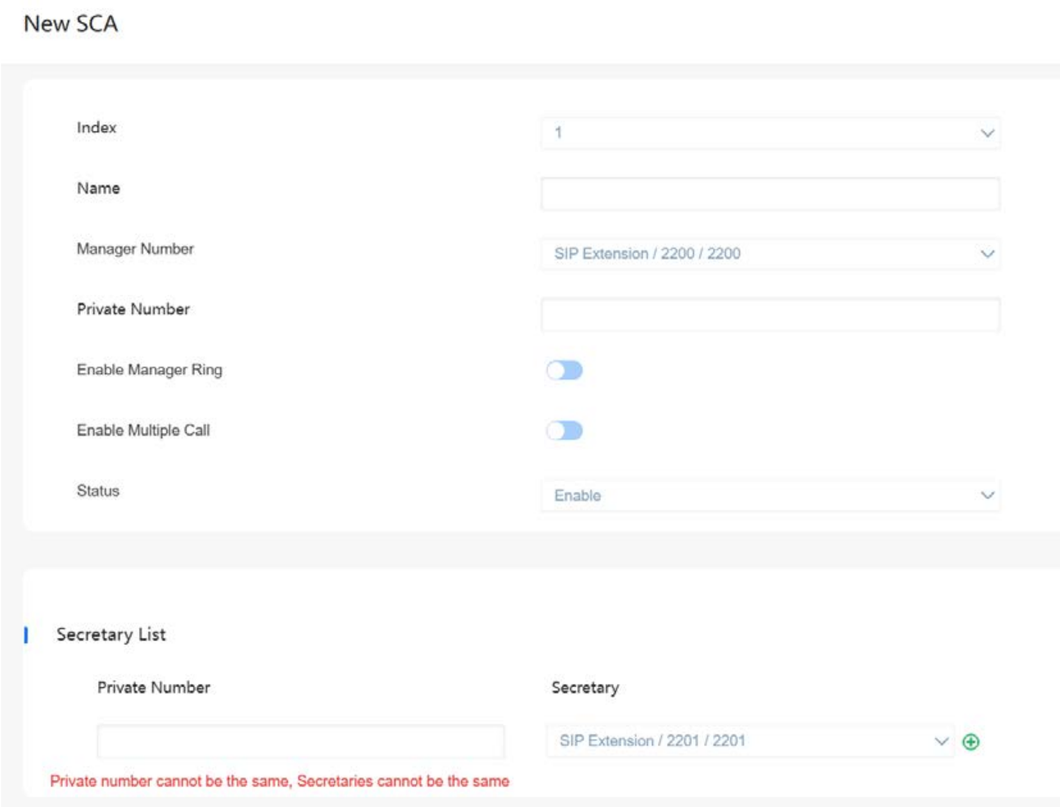


Table-Parameters of SCA

Parameter	Description
Index	The index of the SCA.
Name	The configuration name cannot be empty, up to 32 characters and cannot contain English double quotes.
Manager Number	Only SIP extensions with SCA enabled can be selected.
Private Number	Manager's private number cannot be duplicated with other numbers, it can only be used for calls between managers and secretaries in the same business.

Enable Manager Ring	If turned on, it will ring with the secretary and users can answer the call directly to the call manager.
Enable Multiple Call	If turned on, users can have multiple calls coming in at the same time. Allowed the maximum number of incoming calls is the number of secretaries.
Status	Enable or disable SCA feature.
Secretary List - Private Number	Secretary's private number cannot be duplicated with other numbers, it can only be used for calls between managers and secretaries in the same business.
Secretary List - Secretary	Select the appropriate SIP extension as the manager's secretary, a manager can have multiple secretaries.

5.4.9 Alarm Clock

The alarm clock rings up the destination number at the time when the system has been pre-configured with Alarm Clock. The system matches the time user set before, then the system will automatically ring an extension selected.

Operation steps:

1. On the **Advanced Service** > **Alarm Clock** interface, click New.

Figure-Parameters of Alarm Clock

New Alarm Clock

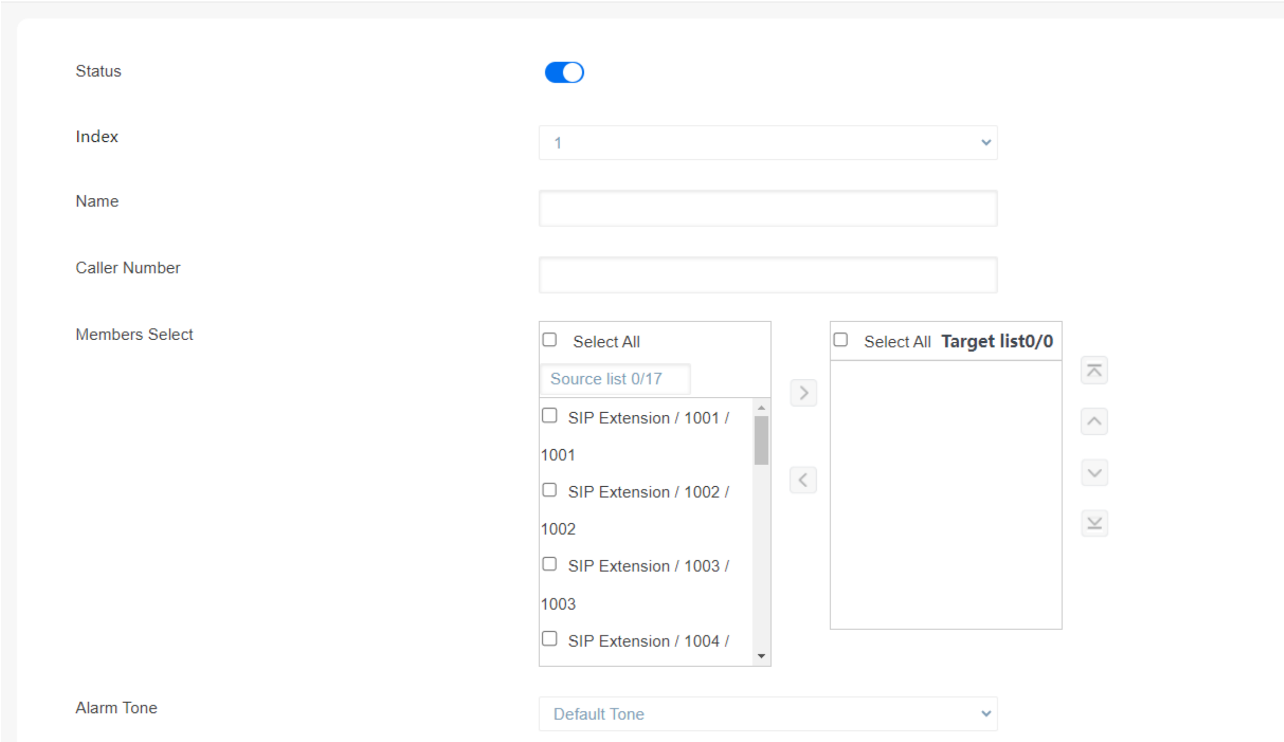


Table-Parameters of Alarm Clock

Parameter	Description
Status	Enable or disable Alarm Clock.
Index	The index of the Alarm Clock.
Name	Custom alarm name; the configuration name cannot be empty, has a maximum of 32 characters and cannot contain double quotes
Caller Number	Customize the caller number; the number can only contain numbers, letters or +/*/#, and the maximum length is 32

Members Select	Configure the extension number. Move right to add an extension and move left to delete an extension. The added extension cannot be the same as the calling number.
Alarm Tone	Users can choose to customize the uploaded waiting music or use the default music. The phone rings when the set alarm time is reached. Music will play automatically when off-hook.
Alarm Time	Customized alarm clock ringing time.
Ring Time(5s~200s)	Alarm clock to the set time phone ringing time.
Strategy	Select the alarm activation strategy: Once/Daily

2. After saving the application, check whether the extension number is ringing at the configured time.

Use scenario: Hotel wake-up call service, timed phone ringing, automatic play of wake-up call service music after taking off the phone.

5.5 PBX Global Settings

5.5.1 SIP Stack

On the **PBX Global Settings > SIP Stack** interface, users can set SIP information such as listening port, which will be used in extension and trunk. Up to eight SIP profiles can be configured for one UC350 Series IPPBX device, so users can choose different SIP profiles according to different requirements.

Figure-Parameters of SIP Profile

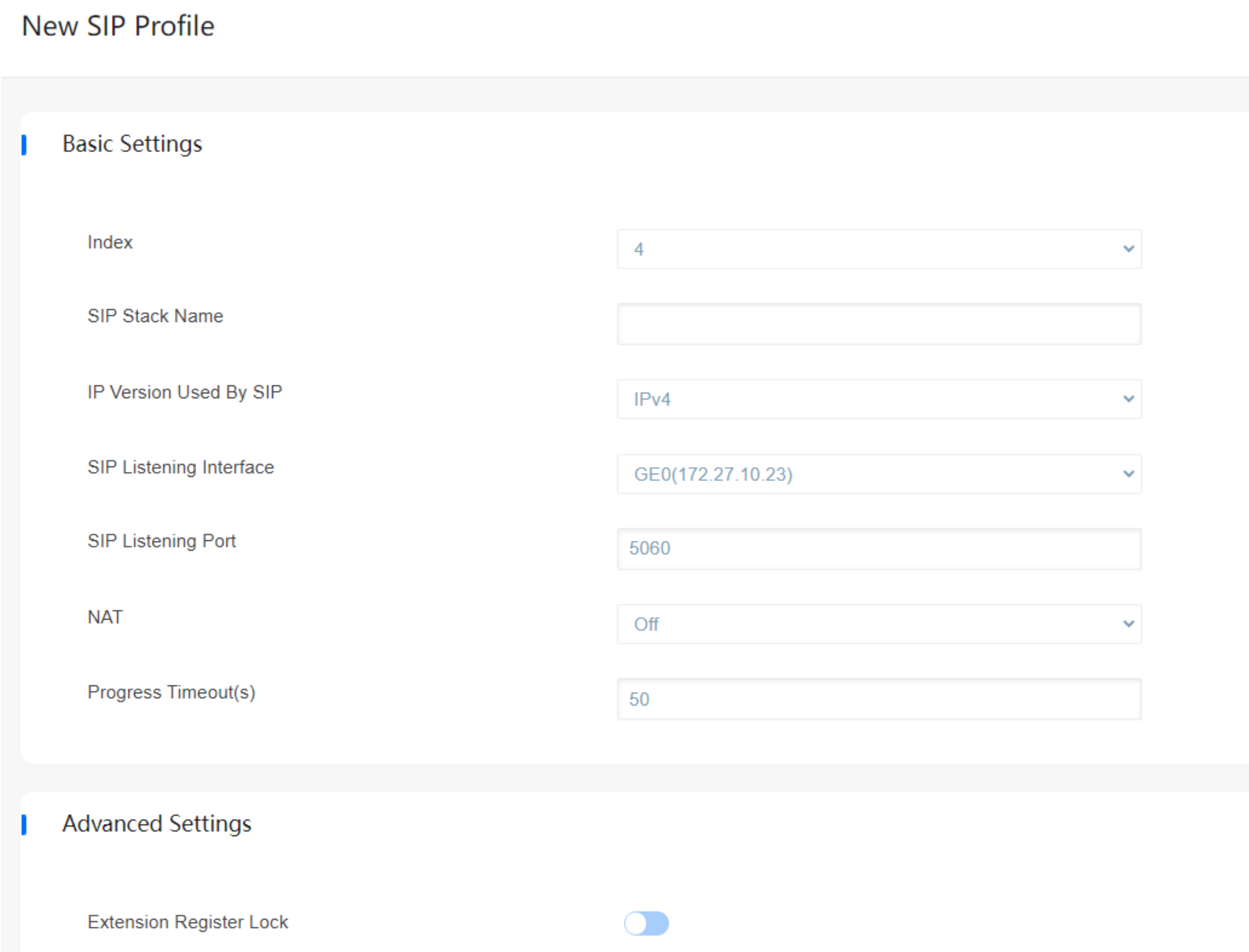


Table-Parameters of SIP Profile

Parameter	Description
Index	The index of the SIP profile.
SIP Stack Name	The name of the SIP profile.
IP Version Used By SIP	Select network mode, IPv4 or IPv6.
SIP Listening Interface	The local listening interface of this SIP profile. Display the floating IP address when the active and standby function is enabled.

Local Listening Port	The local listening port of this SIP profile. If the SIP profile is used by a SIP trunk, the port filled in here is the listening port for the SIP trunk.
NAT	NAT configuration of SIP messages, optional IP address, stun, dynamic domain name, rport, off, used to solve the problem of voice calls in NAT environment. This configuration should be configured by professionals.
Progress Timeout(s)	If the parameter is set as 50 seconds, it means that the call will be considered as timeout in case that no one answers the call during 50 seconds.
Extension Register Lock	When enabled, only the first successfully registered client is allowed to register.
Detect Extension is Online	The device sends an OPTION message to the SIP client to detect the online status of the client within the detection period. Receiving 200 OK means that the client is online, and vice versa.
Detect Period(s)	Set the interval of sending OPTION message by the device. The range is 5-99999.
DTMF Send Type	DTMF is short for Dual Tone Multi Frequency. There are three DTMF modes, including SIP Info, INBAND, RFC2833.
RFC2833-PT	RFC2833 payload coding.

Detect Inband When Call in IVR	After enabling, the DTMF sent by the caller inband is supported in the IVR.
Process DTMF as Hold/Unhold	By default, this parameter is off. When it is set as on, DTMF will be addressed as call hold/unhold.
PRACK	Provisional Response ACKnowledgement.
WebRTC	After enabling, the SIP protocol stack supports WebRTC mode, and the default port is 7443 for registration.
Public Proxy	After enabling, configure the correct public network address parameters, and register to UC through the public network proxy
Session Timer	**Session Expires:** The validity period of a SIP session. When a SIP session times out, an invite message needs to be sent to refresh the session, otherwise, the session ends. It is 1800 seconds by default. **Min Session Expires:** the minimum validity period to respond to a SIP session. **Session Refresh Method:** re-INVITE or UPDATE
Trunk Reg Num to the Same Addr per Second	When multiple trunks are registered at the same address, please set the interval for sending register messages during registration.

Caller Number Source	**From: User Part**: to obtain the caller number from the user part contained in the 'From' field. **From: Display Name**: to obtain the caller number from the display name contained in the 'From' field. **To: User Part**: to obtain the caller number from the user part contained in the 'To' field. **Contact: User Part**: to obtain the caller number from the user part contained in the 'Contact' field.
Transfer Caller Source	Users can select the original caller or the Transfer originator for controlling the display of third-party caller numbers.
Diversion Indication Header Number Src	Used to control the diversion parameter. Users can choose to close, original caller or original called party.

Called Number Source	**From: User Part**: to obtain the called number from the user part contained in the 'From' field. **From: Display Name**: to obtain the called number from the display name contained in the 'From' field. **To: User Part**: to obtain the called number from the user part contained in the 'To' field. **Contact: User Part**: to obtain the called number from the user part contained in the 'Contact' field.
Inbound Codec Negotiation Priority	To take the remote device or the local device as priority for inbound codec negotiation. Assume local device supports PCMA, PCMU, G.729 and G.723, while the remote device supports G.723 and G.729. If remote device is taken as codec negotiation priority, G.723 will be the codec mode, since the remote device supports G.723 and G.729 and G.723 is prior to G.729.
Inbound Codec Profile	The codec supported by SIP for inbound calls.
Outbound Codec Profile	The codec supported by SIP for outbound calls.

CNG(Comfort Noise Generator)	This function is used to generate background noise for the call when there is a short silence during the call, which sounds very comfortable.
Blind Transfer Callback	After it is enabled, if the extension is busy during a blind transfer call, the call will be called back when the extension is free.
Bypass Media(SIP to SIP)	Whether to allow inter SIP calls media to communicate directly, bypassing the server.
Proxy Media(SIP to SIP)	Whether to allow inter SIP calls to be communicated by profile proxy media addresses.
Ignore ACK	After enabling, the gateway will not retransmit 200 OK if the remote end does not send an ACK, otherwise it will retransmit at intervals.
BLF	After enabling, users can monitor the working status of other extension through the preset indicator lights on a specified extension. The indicator lights will show different states according to the status of the monitored number.
CID Header	Add the CID header to the invite message sent by the gateway.
PickUp Caller Refresh Method	The default is disabled. Users can select re-INVITE or UPDATE.

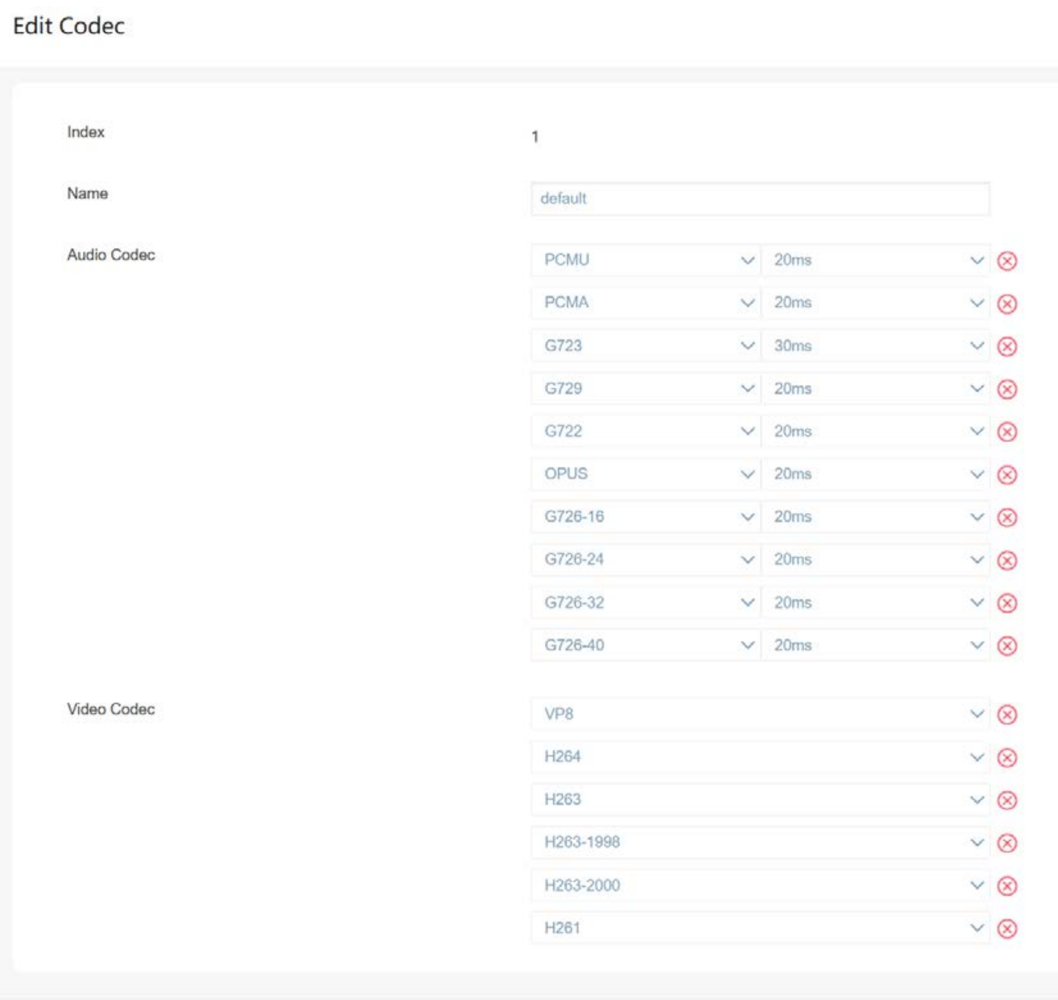
QoS	Whether to enable QoS. QoS is a technology used to solve network delay or congestion.
User Agent	Then content of the 'user agent' field in SIP packets.
Timer T1(ms)	The value of timer T1 in SIP protocol. Default value is 500ms.
Timer T2(ms)	The value of timer T2 in SIP protocol. Default value is 4000ms.
Timer T4(ms)	The value of timer T4 in SIP protocol. Default value is 5000ms.
Timer T1X64(ms)	The value of timer T1X64 in SIP protocol. Default value is 32000ms.
Signal Encryption	After enabling, the gateway will transmit signaling via TLS.
TLS SIP Port	The listening port for TLS SIP, which ranges from 1 to 65535. It can't conflict with existing ports and cannot be NULL.
RTP Encryption	Select encrypted SRTP for RTP stream transmission. SRTP is a secure real-time transmission protocol to ensure the security of voice communication.

Allow Unknown Call	If this function is enabled, incoming calls from unknown sources are allowed. Unknown sources are those IP addresses that do not fall into the source range configured for SIP trunks or SIP extensions.
Inbound Source Filter	The source of inbound calls, which is allowed. It can be an IP address or a network segment. If it is a network segment, the format is 172.16.0.0/16 or 172.16.0.0/255.255.0.0, which means calls from the network segment of 172.16 is allowed to come in. 0.0.0.0 means calls of any source is allowed to come in.

5.5.2 Codec

At present, UC350 Series IPPBX supports audio codec and video codec, and all voice codecs and video codecs are enabled in the default configuration. Users can also group and prioritize any of the 16 codecs according to their requirements.

Figure-Parameters of Codec



5.5.3 FXS/FXO

On the **PBX Global Settings** > **FXS/FXO** interface, users can configure the driving parameters of FXS port and FXO port, including tone standard, dial timeout, ring timeout, hook-flash detection, DTMF parameters, CID-related parameters, impedance, dialplan and so on.

Figure-Parameters of FXS Profile

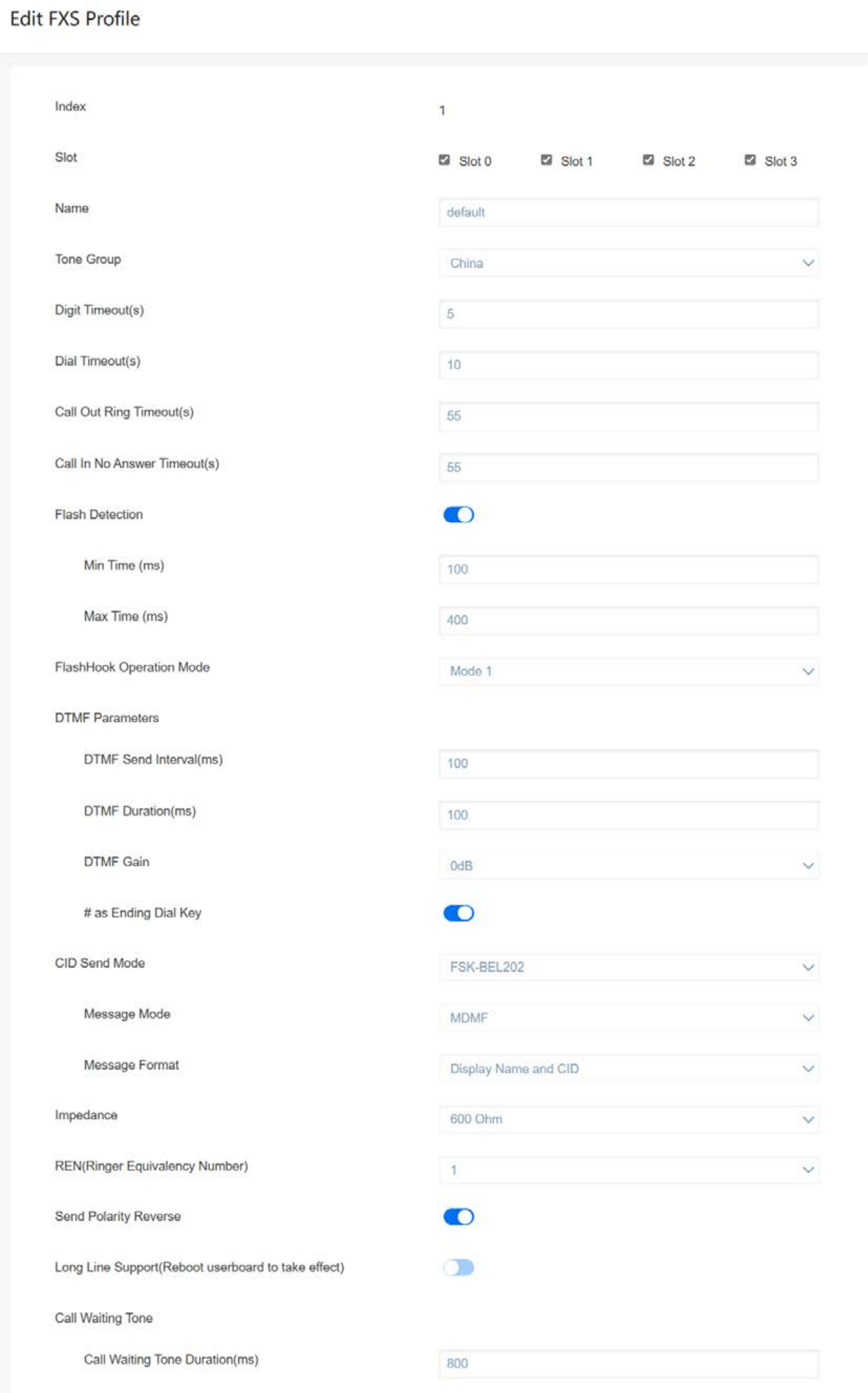


Table-Parameters of FXS Profile

Parameter	Description
Index	The index of the FXS profile.
Slot	The name of the FXS profile.
Name	The name of this FXS profile.
Tone Group	The national standard of dialing tone, busy tone and ring tone. The default value is China.
Digit Timeout(s)	The timeout value for dialing a digit of a telephone number. When the time of dialing a digit exceeds this value, the system will think the dialing has completed. Default value is 4 seconds.
Dial Timeout(s)	The timeout value for dialing the first telephone number after offhook. Default value is 10 seconds.
Call Out Ring Timeout(s)	The timeout value for the ringing of the analog phones of the FXS port when when calling out.
Call In No Answer Timeout(s)	The timeout value for ending a call which goes out through the FXS port, when nobody answers the call.
Flash Detection	Whether to enable flash-hook detection. If flash detection is not enabled, the press on flash-hook will be ignored and won't be processed.

Min Time (ms)/Max Time (ms)	Min Time: when flash-hook detection is enabled, if the time of the press on the flash-hook is less than this minimum time, the press will be ignored and won't be processed. Max Time: when flash-hook detection is enabled, if the time of the press on the flash-hook is longer than this maximum time, the phone will be hanged up.
FlashHook Operation Mode	Choose Mode one, Mode two or Mode three.
DTMF Send Interval(ms)	The minimum interval between the sending of two DTMF tone. DTMF: Dual Tone Multi Frequency
DTMF Duration(ms)	The minimum duration of a DTMF tone.
DTMF Gain	Default value is 0 DB.
# as Ending Dial Key	If this parameter is enabled, '#' is used as the end mark for dialing.
CID Send Mode	There are three CID send modes, namely FSK-BEL202, FSK-V.23 and DTMF.
Message Mode	There are two call display types including SDMF and MDMF.

Message Format	The call display format in analog phone. It can be "Display Name and CID", "Only CID", or "OnlyDisplay Name". The default value is "Display Name and CID".
Impedance	The impedance (SLIC) matched with analog phones.
REN(Ringer Equivalency Number)	The equivalent number of ringing phones. It is used to determine how many devices can be connected by telephone lines. Range:1to 4
Send Polarity Reverse	If polarity reverse is on, call tolls will be calculated based on the changes in voltage. If polarity reverse is off, users need to set the time for offhook detect and call tolls will be calculated starting from the set time.
Long Line Support(Reboot userboard to take effect)	The UC350 supports up to 8km wiring length. When the length of the telephone line is less than 1km, the long line mode cannot be enabled.
Call Waiting Tone	Configure the duration, gap and repeat count for the call waiting tone.
Auto Gain Control	Automatically adjust the gain after enabling.
Dialplan	The rules for dialing. The UC350 device supports regular expression. Please make reference to **Advanced Service > Dianplan** section.

Fax Mode	There are three fax modes: T.38, T.30(Pass through), and Adaptive.
Include 'a=X-fax' Attribute	If this parameter is enabled, "a=X-fax" attribute will be carried in SDP
Include 'a=fax' Attribute	If this parameter is enabled, "a=fax" attribute will be carried in SDP
Include 'a=X-modem' Attribute	If this parameter is enabled, "a=X-modem" attribute will be carried in SDP
Include 'a=modem' Attribute	If this parameter is enabled, "a=modem" attribute will be carried in SDP
Include 'vbd' Parameter	If this parameter is enabled, "a=gpmd:0 vbd=yes" attribute will be carried in SDP
Include 'silenceSupp' Parameter	If this parameter is enabled, "a=silenceSupp:off" attribute will be carried in SDP
ECM	Whether to enable 'Error Correction Mode' (ECM).
Rate	The rate of sending or receiving fax, default value is 14400bps.
Tone Detection by	Fax sound is detected by caller, callee or automatically.
Switch into Fax Mode When Detected CNG or CED	If this parameter is enabled, the system will switch into fax mode when CNG or CED is detected.

Figure-Edit FXO Profile

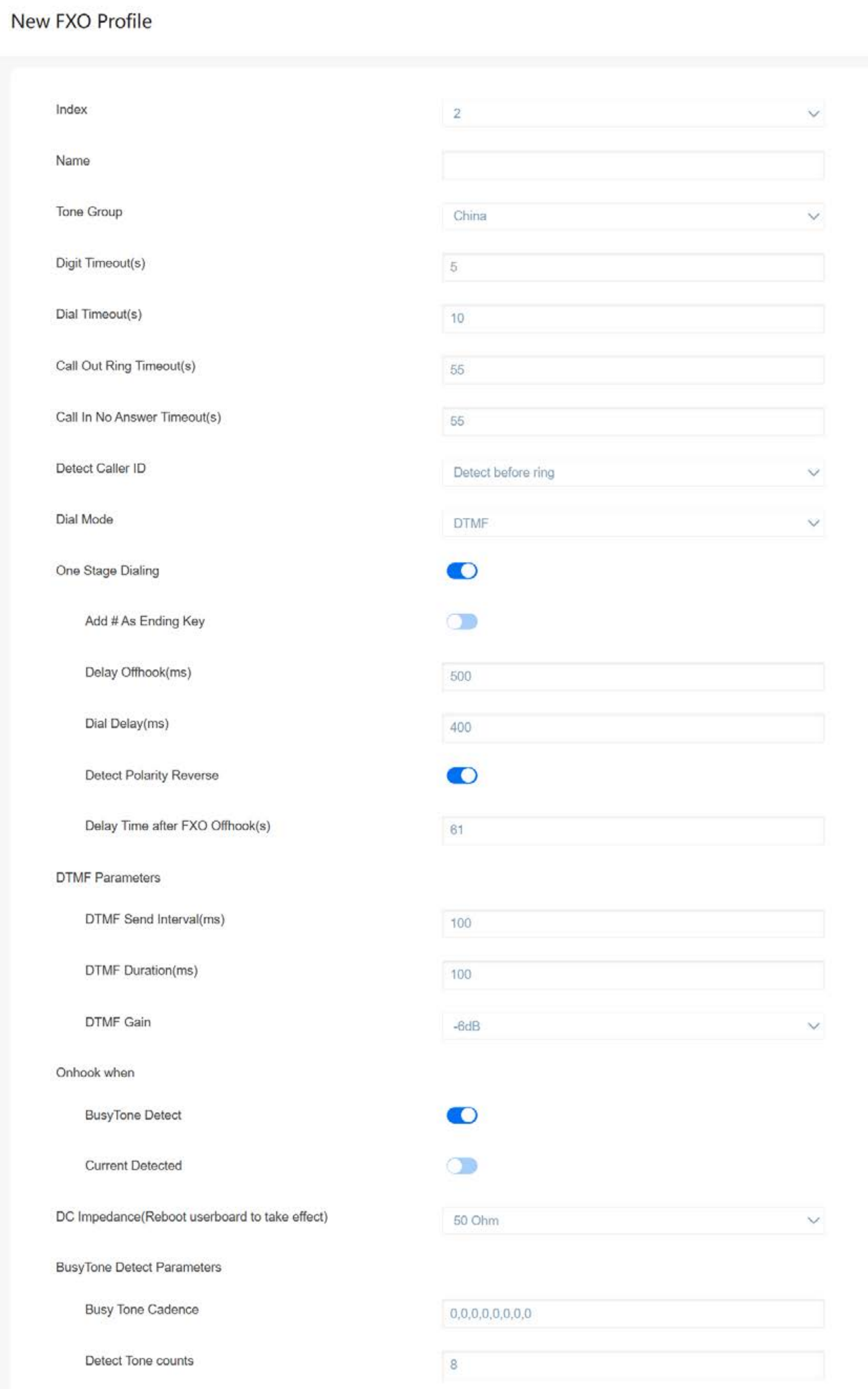


Table-Configure FXO Profile

Parameter	Description
Index	The index of the FXO profile.
Name	The name of the FXO profile.
Tone Group	The national standard of dialing tone, busy tone and ring tone. The default value is China.
Digit Timeout(s)	The timeout value for dialing a digit of a telephone number. When the time of dialing a digit exceeds this value, the system will think the dialing has completed. Default value is 4 seconds.
Dial Timeout(s)	The timeout value for dialing the first telephone number after offhook. Default value is 10 seconds.
Call Out Ring Timeout(s)	The timeout value for the ringing of the analog phones of the FXS port when calling out.
Call In No Answer Timeout(s)	The timeout value for ending a call which goes out through the FXS port, when nobody answers the call.
Detect Caller ID	Detect before ring: the CID will be shown before ringing. Otherwise, CID will be displayed after ringing. Detect after ring: the CID will be shown after ringing. Otherwise, CID will be displayed before ringing Off: the CID will not be shown

Dial Mode	The dialing mode when FXO port calls the PSTN side (supports 3 dialing modes)
One Stage Dialing	The FXO call mode means that when the FXO port makes an outgoing call, the called number in the SIP message is sent to the PSTN side digit by digit at a time.
Add # As Ending Key	After it is turned on, the FXO port makes an outgoing call, it will automatically add # after the original number as a ending key.
Delay Offhook(ms)	Set the delay dial time, the default is 400ms.
Dial Delay(ms)	The delay time of dialing. Default value is 400ms.
Detect Polarity Reverse	Whether to enable 'detect polarity reverse'. If 'detect polarity reverse' is on, call tolls will be calculated based on the changes in voltage. If 'detect polarity reverse' is off, users need to set the time for offhook delay and call tolls will be calculated starting from the set time.
Delay Time after FXO Offhook(s)	When the FXO port calls the PSTN side, the delay time from the port is on-hook to the port is off-hook (default 1000ms).
DTMF Send Interval(ms)	The minimum interval between the sending of two DTMF tone. DTMF: Dual Tone Multi Frequency.
DTMF Duration(ms)	The minimum duration of a DTMF tone.

DTMF Gain	Signal gain of DTMF.
Onhook when	When FXO calls the PSTN side, the conditions for the FXO port to on-hook: detect busy tone and detect current. Busy Tone Detect: After enabling, if FXO detects a busy tone, FXO hangs up. Current Detected: After enabling, if FXO detects that there is no current, FXO will hang up.
BusyTone Detect	Enable or disable BusyTone detection.
Current Detected	Enable or disable Current detection.
Current Disconnect Threshold	This current threshold is used to determine whether a phone is onhook.
DC Impedance (Reboot userboard to take effect)	Matching impedance value when FXO and PBX or PSTN are interconnected.
Busy Tone Cadence	The busy tone detection cadence needs to be set according to the busy tone system of the PSTN. If users do not know the busy tone standard, users can use the busy tone detection function to detect the busy tone cadence.
Detect Tone counts	Set the number of busy notes to check.
Detect Tone Delta(ms)	Set the error size to check the busy tone.
On->Off Energy Threshold	Busy tone signal On→Off energy threshold.

Off->On Energy Threshold	Busy tone signal Off→On energy threshold.

5.5.4 Voice

On the **PBX Global Settings > Voice interface**, users can upload an IVR file according to their requirement. At present, only a wav audio file is allowed. The format of the wav audio file uploaded must be: monaural, 8000hz, 16bit, and size of no more than 3M.

Figure-Settings of Voice

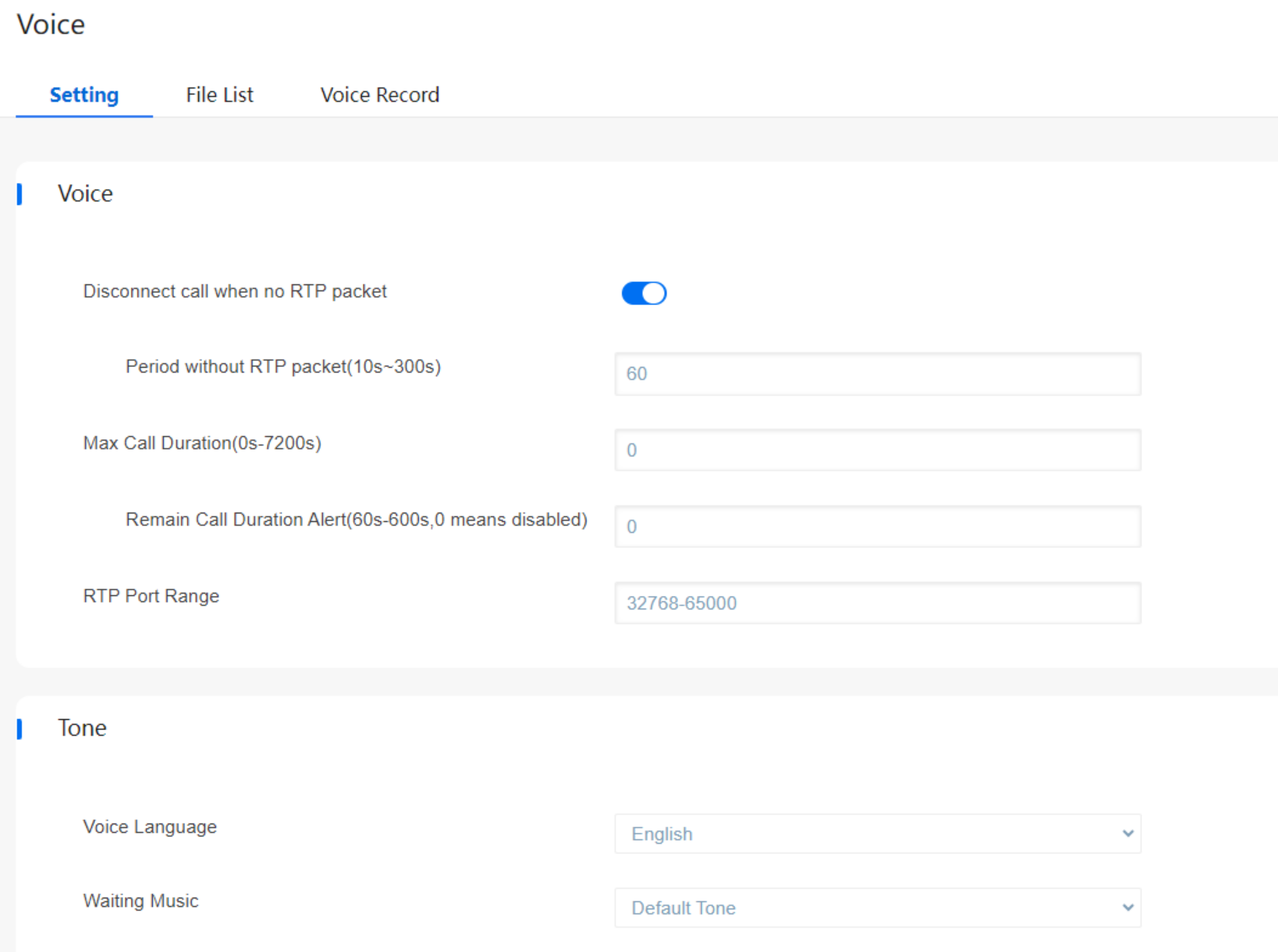


Figure-File List of Voice

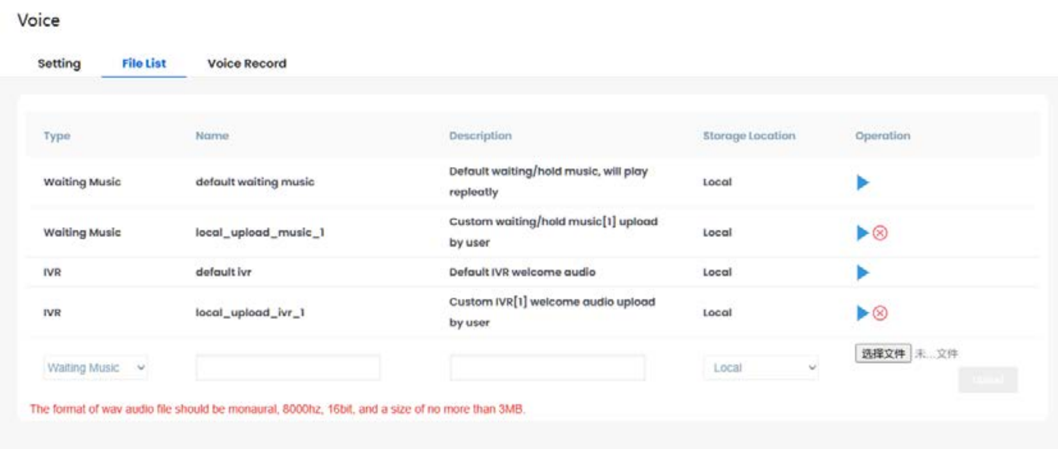


Figure-Voice Record of Voice

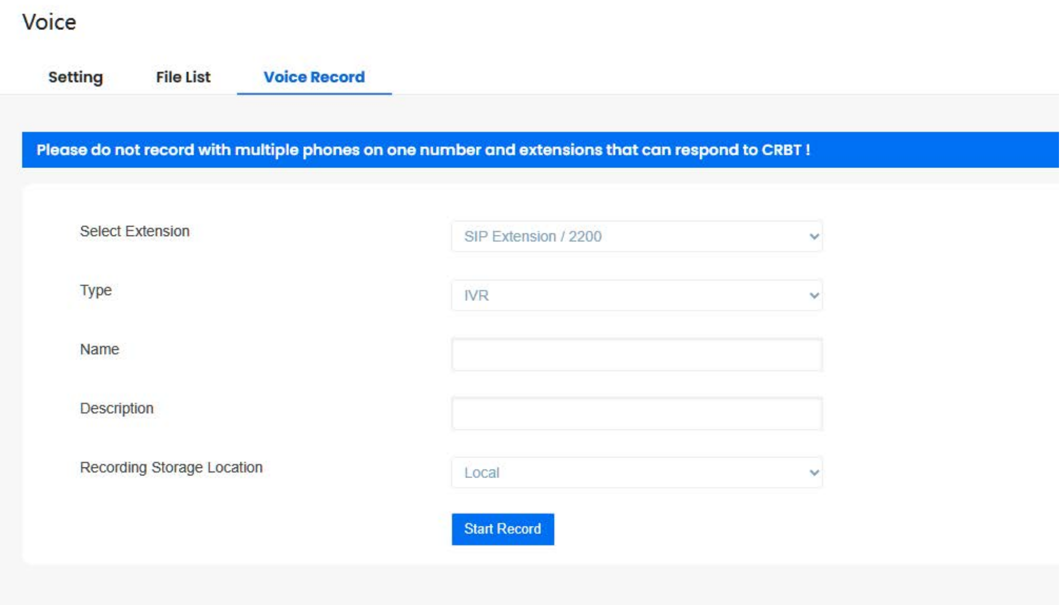


Table-parameter of voice setting

Parameter	Description
Disconnect call when no RTP packet	If it is enabled, and no RTP packets are received within the preset time, calls will be disconnected.
Period without RTP packet(10s~300s)	The default is 60s, and the range is 10s-300s.

Max Call Duration(0s-7200s)	0-7200 seconds, 0 means disabled; when enabled, the call will be disconnected when the maximum call duration is reached
Remain Call Duration Alert(60s-600s,0 means disabled)	60-600 seconds, 0 means disabled; when enabled, there will be a voice prompt to indicate the remaining call time
RTP Port Range	The default is 32768-65000.
Voice Language	Users can select Chinese, English, Portuguese or Spanish as Voice Language.
Waiting Music	Select the waiting music.
Timeout Tone	Set the waiting timeout reminder tone, which is off by default. You can upload the waiting music in the file list for application
Busy Tone	Set the busy tone, which is off by default. You can upload waiting music in the file list for application
Offline Tone	Set the offline reminder tone, which is off by default. You can upload waiting music in the file list for application
Call Waiting Tone	Set the call waiting tone, which is usually the default tone. You can upload the waiting music in the file list for application.
Number Invalid Tone	Set the unavailable number reminder tone, which is off by default. You can upload waiting music in the file list for application

Reject Tone	Set the rejection reminder tone, which is off by default. You can upload the waiting music in the file list for application
NotAuth Tone	Set the unauthorized use reminder tone, which is off by default. You can upload waiting music in the file list for application
Recording Prompt Tone	Set the recording start reminder sound, which is off by default. You can upload waiting music in the file list for application
Area Call Auth	It is disabled by default. If enabled, you need to set regional call permissions at the extension
Local extension call	The default is enabled. When disabled, local extensions need to be configured with routing to make calls.

5.5.5 Feature Code

UC350 provides convenient telephone functions. Connect a telephone to the FXS port and dial a specific feature code, and users can query corresponding information.

The following is the corresponding function of each feature code:

Figure-Feature Code

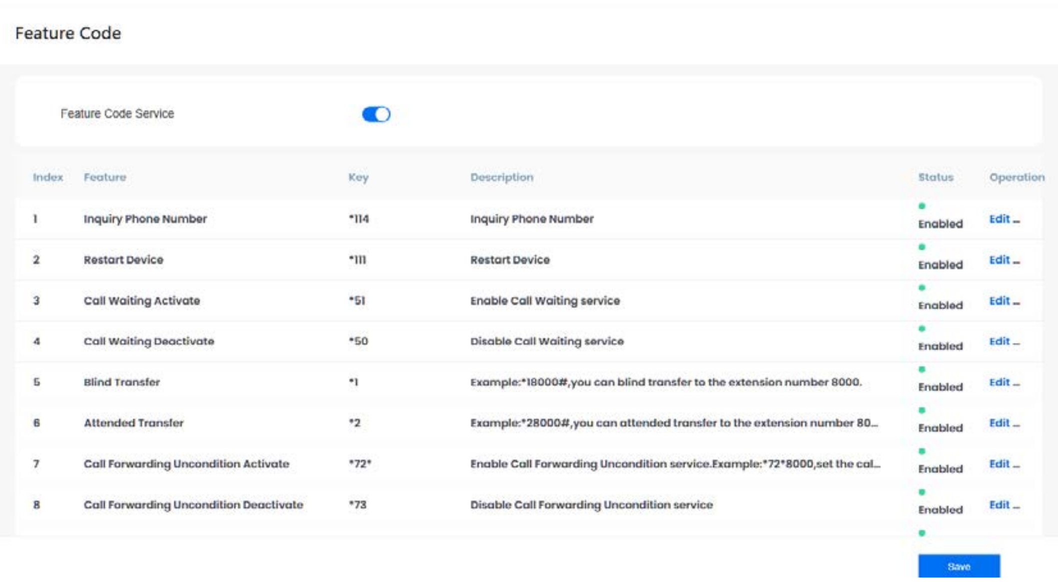


Table-Feature Code

Key	Description
*114	Inquiry Phone Number
*111	Restart Device
*51	Enable Call Waiting service
*50	Disable Call Waiting service
*1	Blind Transfer. Example: *18000#, users can blind transfer to the extension number 8000.
*2	Attended Transfer. Example: *28000#, users can attend transfer to the extension number 8000.
72	Enable Call Forwarding Unconditional service. Example: *72*8000, set the call forwarding number to 8000.
*73	Disable Call Forwarding Unconditional service

90	Enable Call Forwarding Busy service. Example: *90*8000, set the call forwarding number to 8000.
*91	Disable Call Forwarding Busy service.
92	Enable Call Forwarding No Reply service. Example: *92*8000, set the call forwarding number to 8000.
*93	Disable Call Forwarding No Reply service.
*78	Enable Do Not Disturb service.
*79	Disable Do Not Disturb service.
**	Pick up the ringing extension Example: **8000, pick up the extension (8000)
170	*170*1# - Leave messages *170*2# - Play messages
*163	Callback the last received call
*3	Start or stop recording when manual recording.
*4	Call Park. Example: *4, users can park another part during the call. *4100, users pick up the number 100 from parking lot.
164	*164*1 - Listen Mode, *164*2 - Whisper Mode, *164*3 - Barge-in Mode. Example: *164*28000, users can monitor extension 8000 in whisper mode.
*5	Make an intercom with a specific extension user. Example: dial *51000, then the extension 1000 will be automatically picked up.

*162	Redial the last dialed number.

Note: *Each feature code can be customized and edited.*

5.6 Address Book

5.6.1 Contact

The public address book is an address book maintained by the UC administrator, which provides a unified contact query service to the SIP extensions within the enterprise. SIP extension users cannot edit the public address book, but the UC administrator user can manage the public address book according to the account permissions. The public address book is initially empty and does not contain any department or contact information. To use the public address book, the UC administrator needs to enable the public address book and set a company name for the address book. After setting the company name and enabling the public address book, the public address book is divided into two first-level groups:

1. Groups named after the company name are used to store internal contact information of the company;

2. A group with a fixed name of "External Contacts" is used to store contact information outside the enterprise;

Figure- Public Address Book

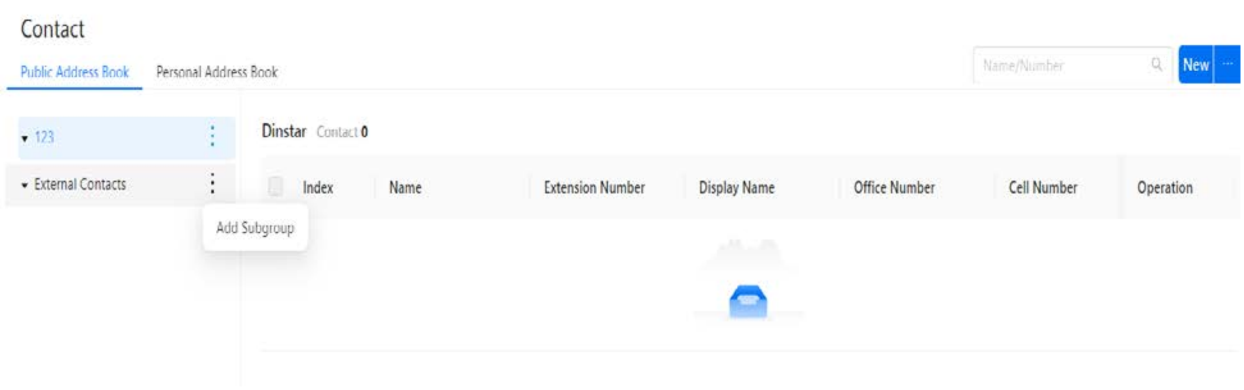
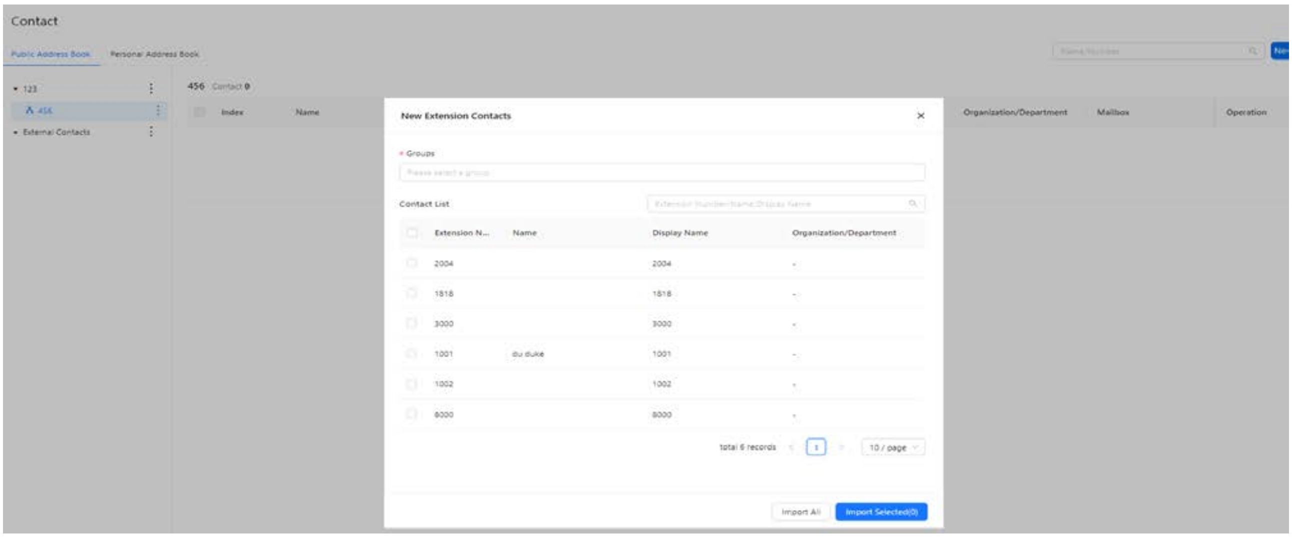


Table- Public Address Book

Parameter	Description
Edit Group/ Add Subgroup	Groups named after the company name can be edited and sub-groups can be added Groups named after " External Contacts" can have sub-groups added
Search Box	Fuzzy search for contact names and numbers
New	Extension contacts and common contacts can be added.
New ···	Contacts can be imported and exported, and LDAP can be selected for contact import

Note: Extension contacts can only be added to a group named after the company name, and this type of contact corresponds to a SIP extension number in UC. Before creating this contact, you should create a SIP extension first; ordinary contacts can only be added to a group named "External Contacts"

Figure- New Contacts



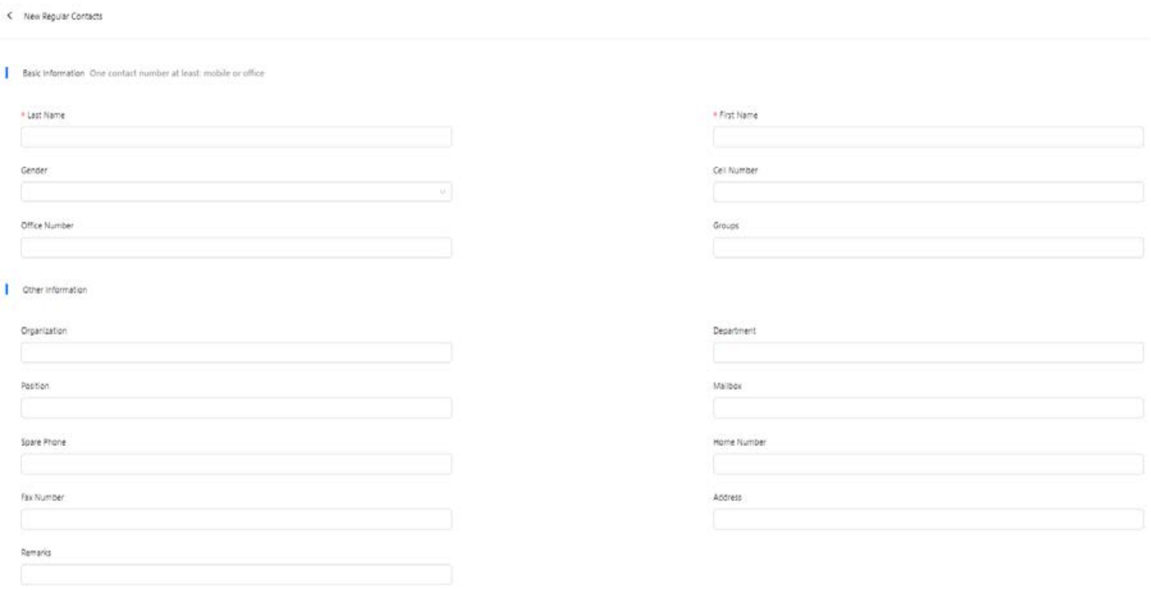


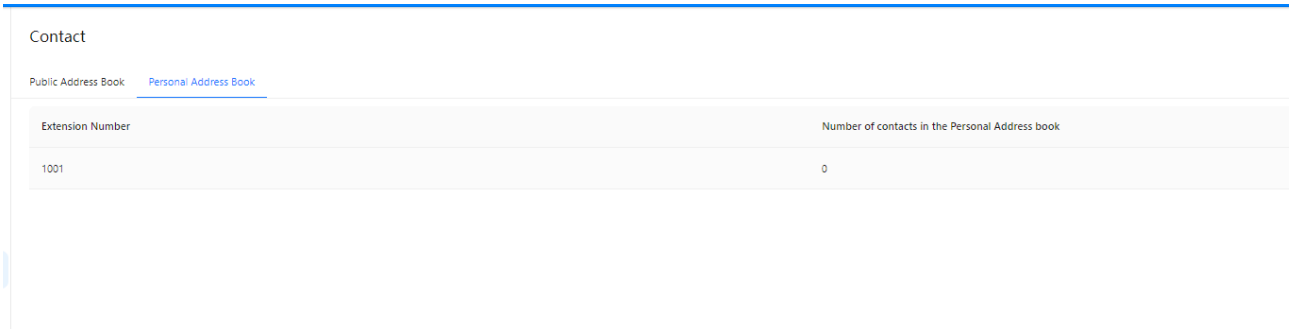
Table- New Regular Contacts

Parameter	Description
Last Name	The contact's last name can only be entered in Chinese or English. The length cannot exceed 32 characters and cannot be left blank.
First Name	The contact's first name can only be entered in Chinese or English. The length cannot exceed 32 characters and cannot be left blank.
Gender	There are three options for users to choose from: "Male, Female, Unknown"
Cell Number	The contact's mobile phone number. Only 0-9 can be used. The maximum length is 32.
Office Number	The work number of the contact can only contain 0-9 and the maximum length is 32. At least one of the work number and the mobile number must be

Groups	Select the contact group
Organization	The organization information of the contact person. This is optional. Symbols <> cannot be entered. The maximum length is 32 characters.
Department	The department information of the contact person. This is optional. Symbols <> cannot be entered. The maximum length is 32 characters.
Position	The position information of the contact person. This is optional. Symbols <> cannot be entered. The maximum length is 32 characters.
Mailbox	Contact person's email address, optional, use the format www.xxx@xxx.com
Spare Phone	The contact's backup phone number, optional, no symbols <> allowed, maximum length 32
Home Number	Contact's home number, optional, no symbols <> allowed, maximum length 32
Fax Number	Contact's fax number, optional, no symbols <> allowed, maximum length 32
Address	Contact's address, optional, no symbols <> allowed, maximum length 32

The personal address book refers to a unique address book space assigned to each SIP extension. The SIP extension can freely manage the contacts in the personal address book, and the contacts in the personal address book are not visible to other SIP extensions/address book clients. The members of the personal address book can be queried, edited, and maintained through the personal portal and APP. The APP does not support adding personal address book members.

Figure- Personal Address Book



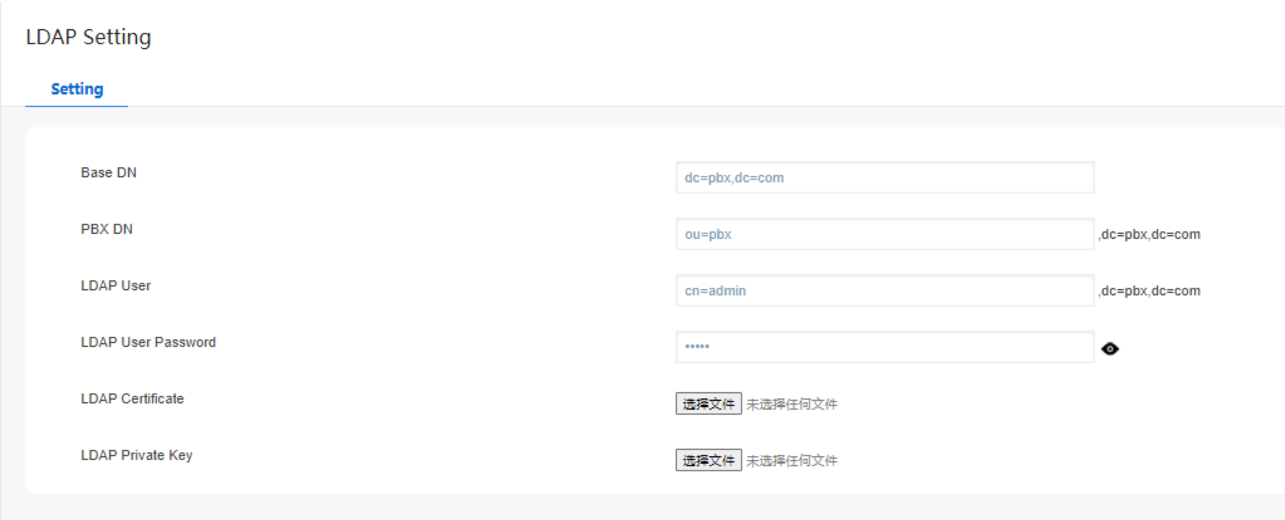
5.6.2 LDAP

UC350 Series IPPBX supports LADP address book function, which can meet the user's needs of managing the device address book. Users can manage the enterprise address book through the "Address Book" page.

The contacts include last name, first name, company/department, email address, phone number, position, address, etc. Meanwhile, it supports LDAP settings, and it can specify LDAP base directory node, PBX directory node, LDAP user, LDAP user password, LDAP certificate and so on, so that the end points can obtain the contents of the enterprise address book.

Figure-Address Book Contact

Figure-LDAP Setting



5.6.3 Setting

The machine code is required for authorization generation (authorization is performed in Maintenance Management-Authorization Information); the UC management terminal provides an interface for importing address book authorization; the current authorization information can be displayed:

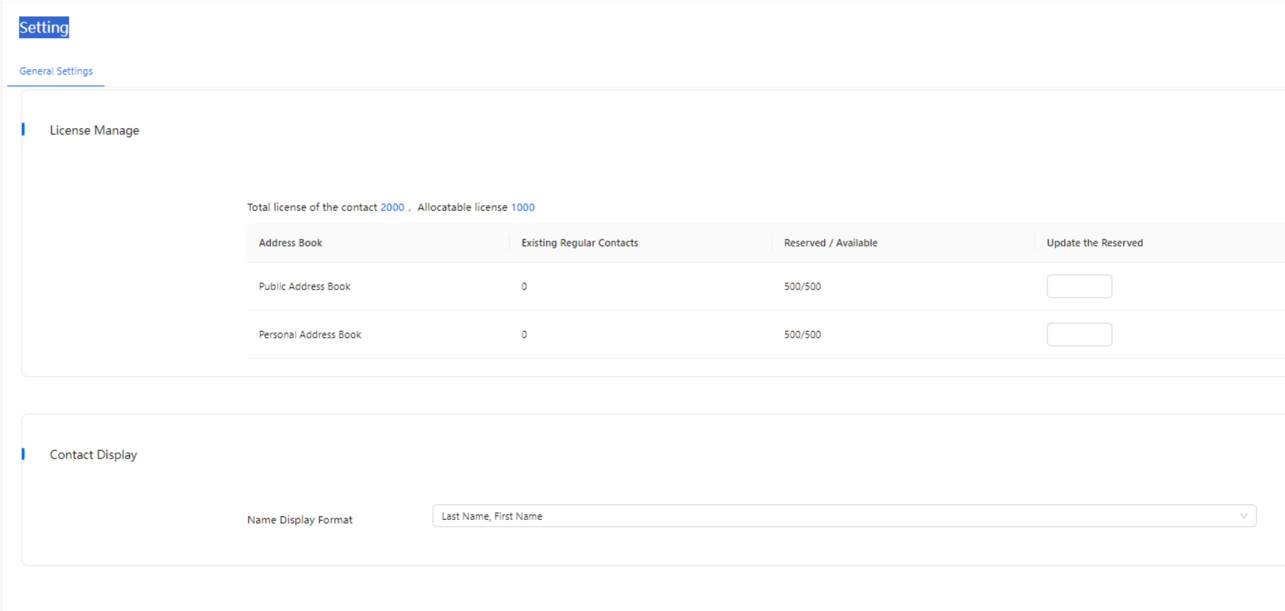
- Total authorization (quantity);
- Allocable authorization (quantity);

Ordinary contacts in the public address book and personal address book contacts are controlled using the same authorization; after importing the authorization, it is allocated as needed

Set the display habit of the name in the address book, supporting 2 display habits:

- Last name first, first name second;
- First name first, last name second;

Figure-Setting

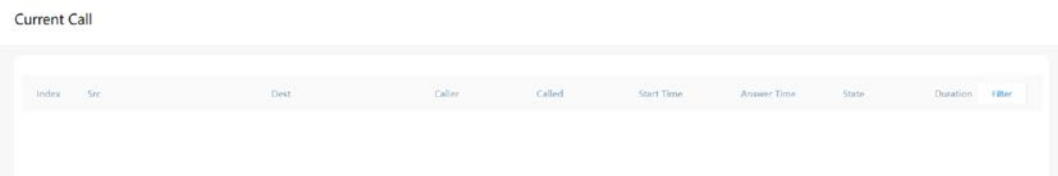


5.7 CDR & Recording

5.7.1 Current Call

On **CDR & Recording** > **Current Call** interface, the source, destination, calling number, called number, start time, answer time, state and duration of the current real-time call are shown. If there is no current call, no information will be empty.

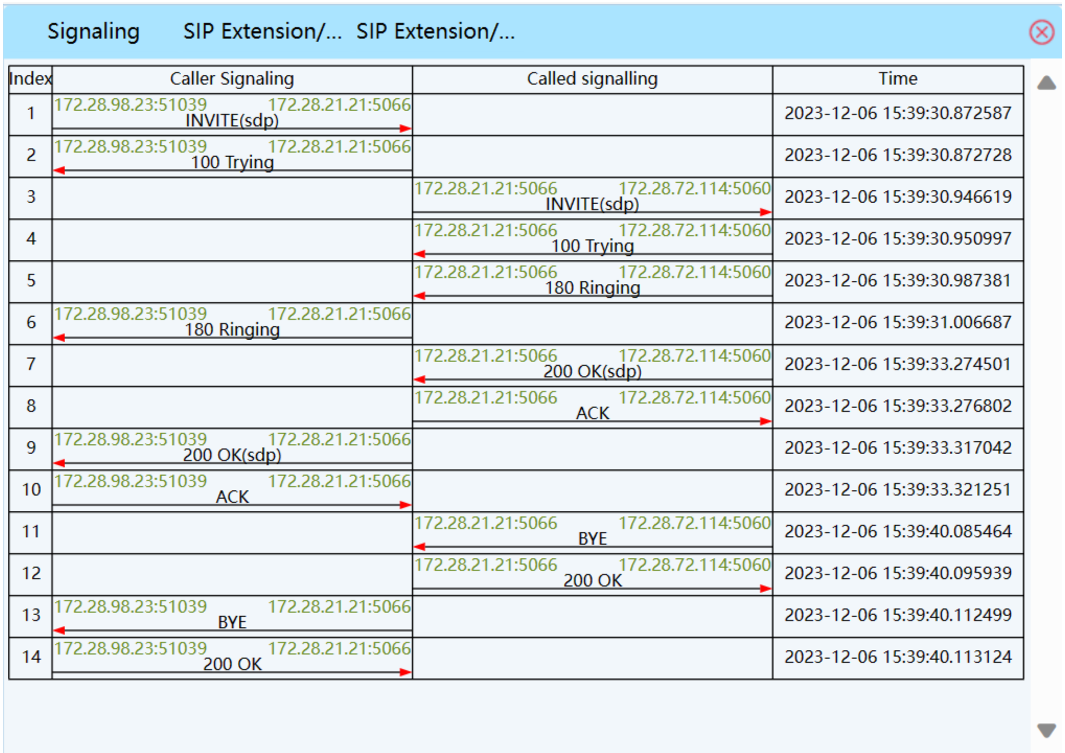
Figure-Parameters of Current Call



5.7.2 CDRs

Click **CDR & Recording** > **CDRs**, and users can set query criteria to query the CDRs (Call Detailed Records) that users want on the displayed interface. Meanwhile, users are allowed to clear CDRs or export CDRs through clicking the Empty or Export button.

Figure-Parameters of CDRs



Signaling SIP Extension/... SIP Extension/...

Index	Caller Signaling	Called signalling	Time
1	172.28.98.23:51039 INVITE(sdp) 172.28.21.21:5066		2023-12-06 15:39:30.872587
2	172.28.98.23:51039 100 Trying 172.28.21.21:5066		2023-12-06 15:39:30.872728
3		172.28.21.21:5066 INVITE(sdp) 172.28.72.114:5060	2023-12-06 15:39:30.946619
4		172.28.21.21:5066 100 Trying 172.28.72.114:5060	2023-12-06 15:39:30.950997
5		172.28.21.21:5066 180 Ringing 172.28.72.114:5060	2023-12-06 15:39:30.987381
6	172.28.98.23:51039 180 Ringing 172.28.21.21:5066		2023-12-06 15:39:31.006687
7		172.28.21.21:5066 200 OK(sdp) 172.28.72.114:5060	2023-12-06 15:39:33.274501
8		172.28.21.21:5066 ACK 172.28.72.114:5060	2023-12-06 15:39:33.276802
9	172.28.98.23:51039 200 OK(sdp) 172.28.21.21:5066		2023-12-06 15:39:33.317042
10	172.28.98.23:51039 ACK 172.28.21.21:5066		2023-12-06 15:39:33.321251
11		172.28.21.21:5066 BYE 172.28.72.114:5060	2023-12-06 15:39:40.085464
12		172.28.21.21:5066 200 OK 172.28.72.114:5060	2023-12-06 15:39:40.095939
13	172.28.98.23:51039 BYE 172.28.21.21:5066		2023-12-06 15:39:40.112499
14	172.28.98.23:51039 200 OK 172.28.21.21:5066		2023-12-06 15:39:40.113124

On the "Call History and Recording -> Recording" page, you can view the recording rules and choose between local recording and streaming recording. For streaming recording, you need to configure the recording server address so that the recording can be transferred to the recording server. For local recording, you can choose the recording storage location. The device supports local and USB storage options.

Figure-Parameters of Recording

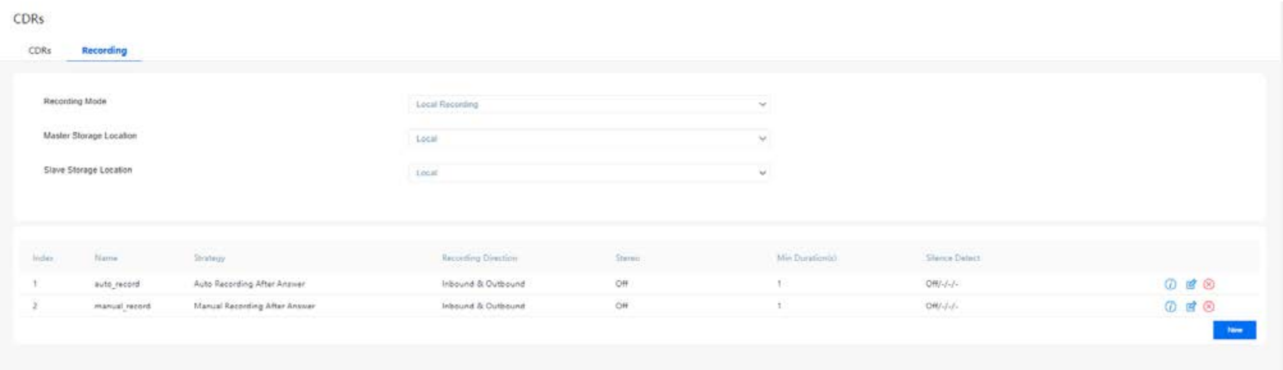


Table-Parameters of Operation

Parameter	Description
	Play the recording files.
	Download the recording files.
	Delete the recording files.

Figure-Parameters of Recording rules

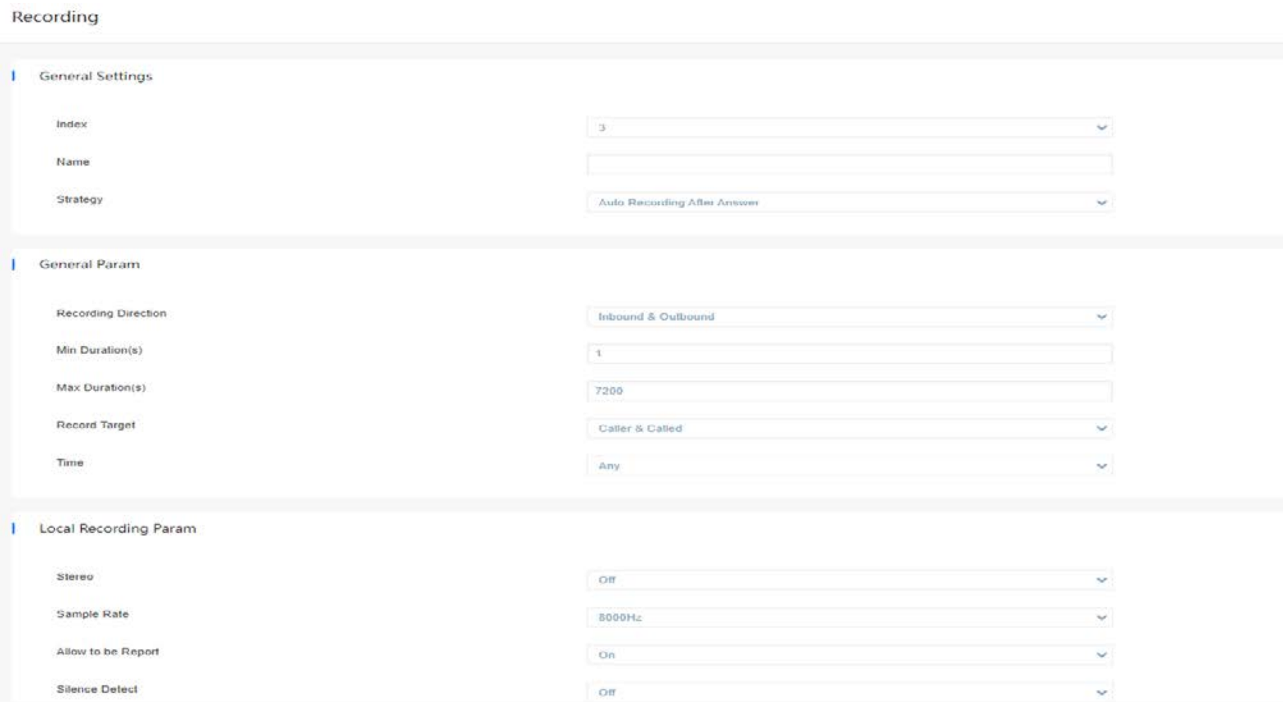


Table-Parameters of Recording rules

Parameter	Description
Index	The index of the recording profile. Range: 1-32.
Name	The name of the recording profile, used to identify the recording profile.
Strategy	Auto Recording after Answer: start recording after the callee pick up the phone. Ban Recording: ether caller or callee enables his function, and then the call in both directions will not be recorded. Manual Recording after Answer: press *3 to start recording after the callee answers the call.
Recording Direction	Inbound & Outbound: If this recording profile is added to FXS/SIP extension, both inbound and outbound calls will be recorded. Inbound: If this recording profile is added to FXS/SIP extension, only inbound calls will be recorded. Outbound: If this recording profile is added to FXS/SIP extension, only outbound calls will be recorded. Note: If this recording profile is added to routing, this parameter is invalid and all calls going through the routing will be recorded.

Min Duration(s)	If the actual recording time is shorter than this value, the recording file will not be saved.
Max Duration(s)	Set the maximum recording time. If the call duration is greater than this time, only the recording before the maximum time will be saved. Valid value range: 600-7200 seconds
Record Target	Select recording object: Caller & Callee / Caller / Callee
Time	You can customize the calls within a certain time period to record
Stereo	At the same call duration, the file size will be twice that of mono; pull down to select on/off
Sample Rate	Set the sampling rate of the recording file
Allow to be Report	If you select Enable, the recording event will be reported (corresponding to Service Connection-Event Reporting, the recording event must be enabled). Note: The recording configuration referenced by both the caller and the called party must be enabled at the same time for the event to be reported; or one party's configuration must be enabled (mandatory)
Silence Detect	When silence is detected, no recording will be done during muting.

Initial Silence Timeout(s)	If the call is muted at the beginning of the call and the duration is out of the set range, the recording file size is around the mute timeout duration.
Final Silence Timeout(s)	If the call is muted after a certain period of time and the duration is out of the set range, the size of the recording file will be smaller than the duration of the call.
Silence Detect Threshold	The voice is judged to be muted below this threshold.

5.8 System

5.8.1 Time

On the **System > Time** interface, users can set a time period for calls to choose routes. If the local time when a call is initiated falls into the set time period, the call will be passed to choose the corresponding route.

Figure-Parameters of Time

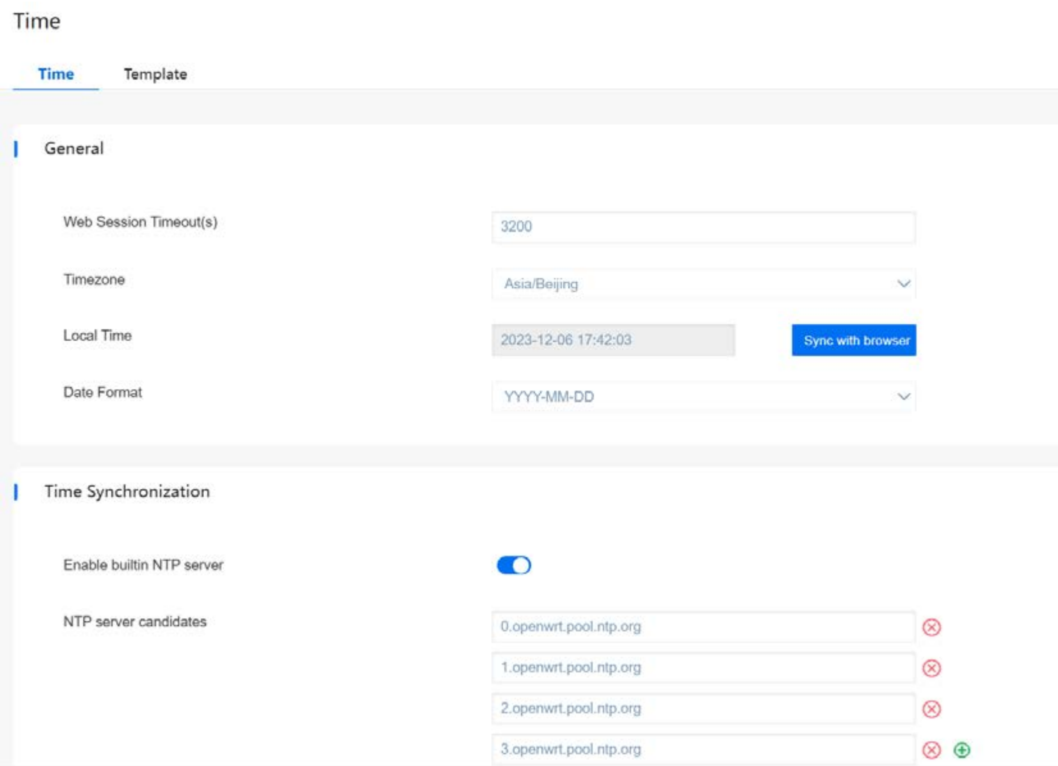


Figure-Parameters of Time Template

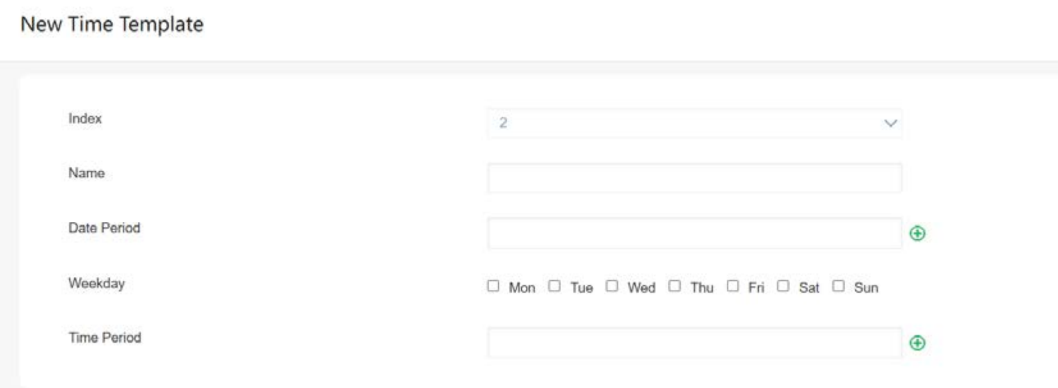


Table-Parameters of Time Template

Parameter	Description
Index	The index of the Time Template.
Name	The name of the number profile
Date Period	Configure the starting date and ending date of a period.

Weekday	Choose a weekday.
Time Period	Choose the starting time and ending time of a day.

5.8.2 Network

The UC350 series IPPBX support 2~4 RJ45 ports, namely GE0, GE1, GE2 and GE3. The default IP address of the management port of UC350 series devices is 192.168.11.1, the management port is used for PC access to management equipment.

Setting

On the "System -> Network -> Settings" page, users can set the IP address of network port

The management port of the device can be configured with a static IP address or obtain an IP address in DHCP mode. The default IP address of the anagement port is 192.168.11.1.

Dynamic IP address: Automatically obtain a dynamic IP through DHCP mode; you can also choose to automatically obtain a DNS server address or a user-defined DNS server address

Static IP Address:

Static IP address is a semi-permanent IP address and remains associated with a single computer over an extended period of time. This differs from a dynamic IP address, which is assigned ad hoc at the start of each session, normally changing from one session to the next. If users choose static IP address, users need to fill in the following information:

IPv4

- IP Address: the IP address of the GE0 port of the device.
- Netmask: the netmask of the router connected the device.
- Default Gateway: the gateway IP address of the router connected the device.
- Use custom DNS server: the IP address of the DNS server.

IPv6

● Mode: Disable or static address can be selected.

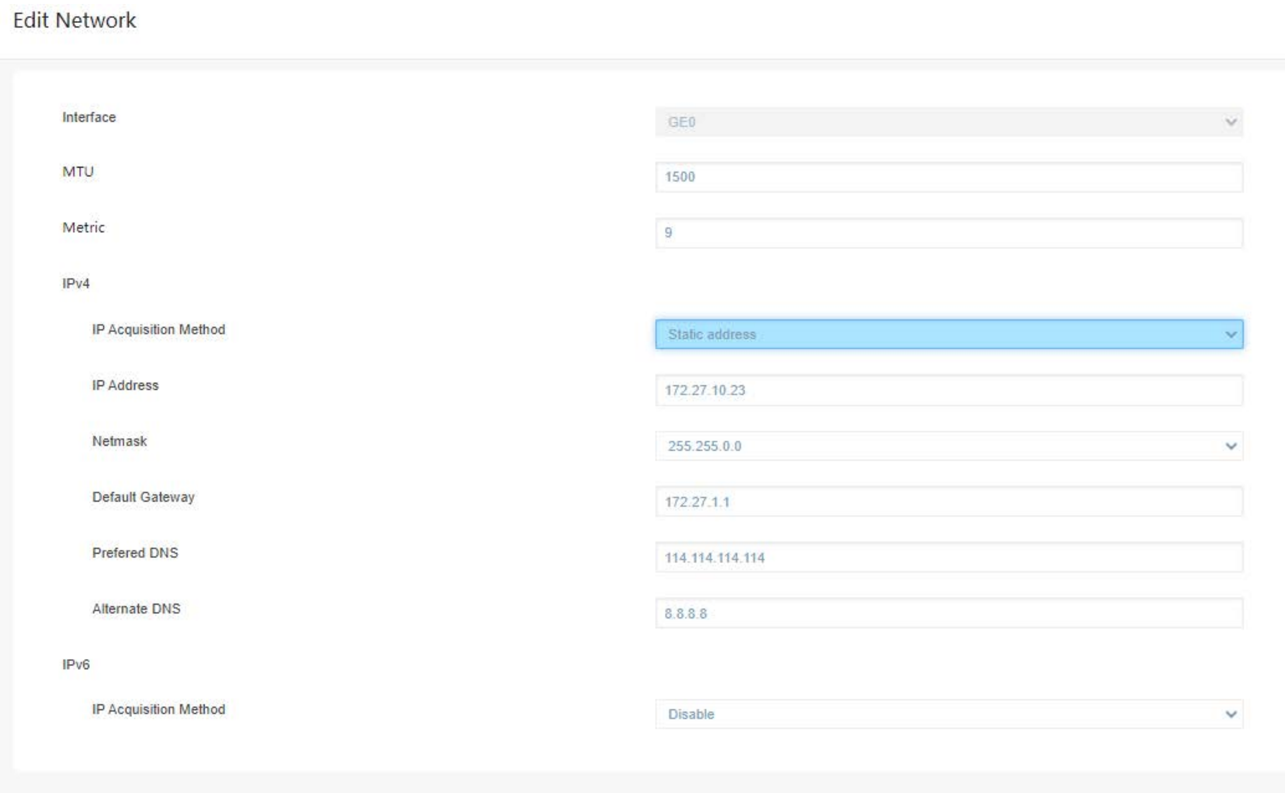
●IP address: Configure the address in IPv6 format, with prefix length; e.g.

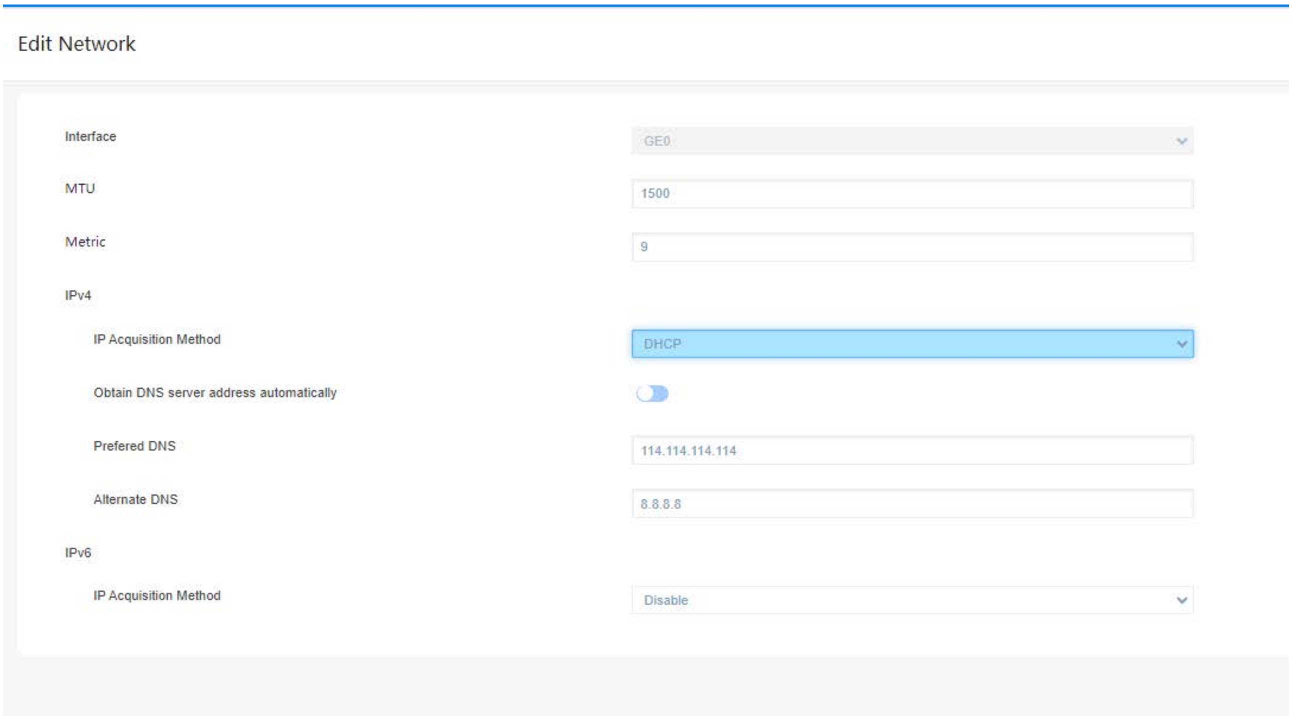
2020::2121/64.

● Default gateway: configure the IPv6 gateway address.

●DNS server: Configure the DNS server IPv6 address.

Figure-Parameters of Network





VLAN Sub Interface

On **System** > **Network** > **VLAN Sub Interface** page, users can set the IP address of the VLAN interface of the device.

To configure the VLAN sub interface of UC350 Series IPPBX, users need to select the corresponding physical interface, specify the VLAN ID and priority, and enter the information below:

●IP address: static IP address assigned to VLAN sub-interface/automatically obtain IP in DHCP mode

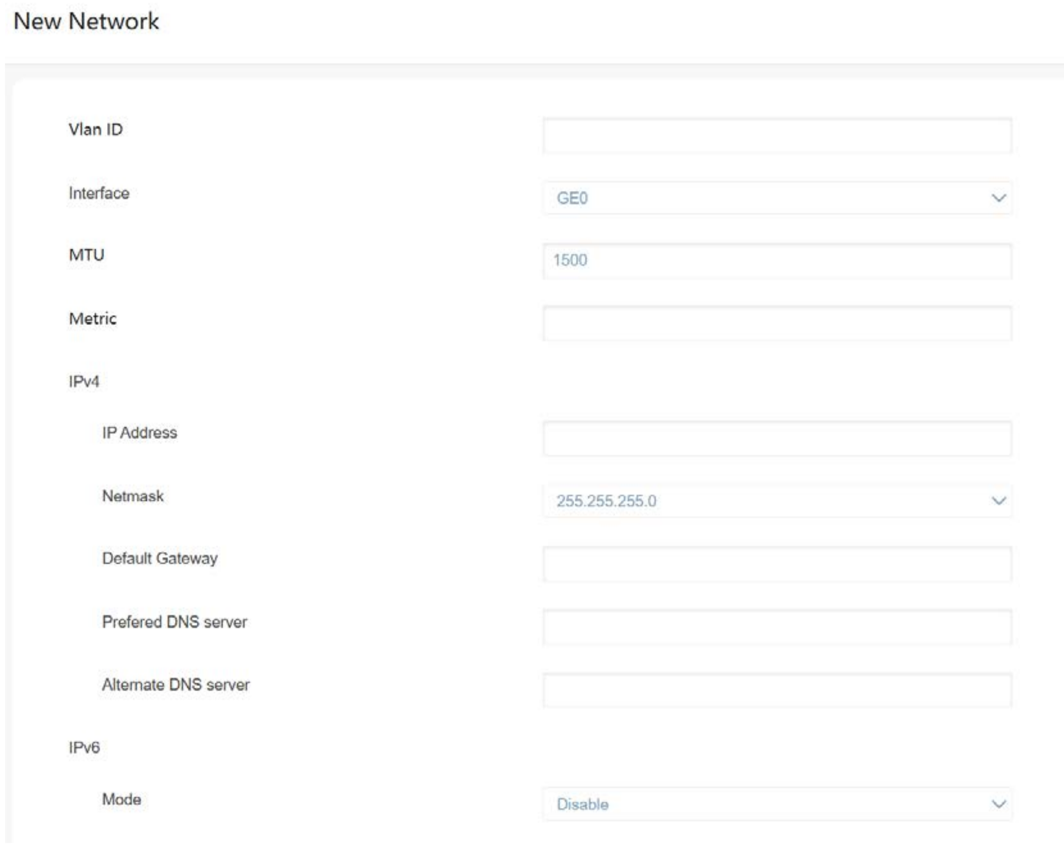
● Subnet Mask: the subnet mask of the router for the VLAN sub interface.

● Default gateway: the gateway IP address of the router for the VLAN sub interface.

● use customized DNS server: the IP address of the DNS server.

● MTU: the default is 1500, and the range is 576-1500.

Figure-Parameters of Network



Static Route

On the **System** > **Network** > **Static Route** interface, users can configure static routes for the network.

Figure-Parameters of Static Route

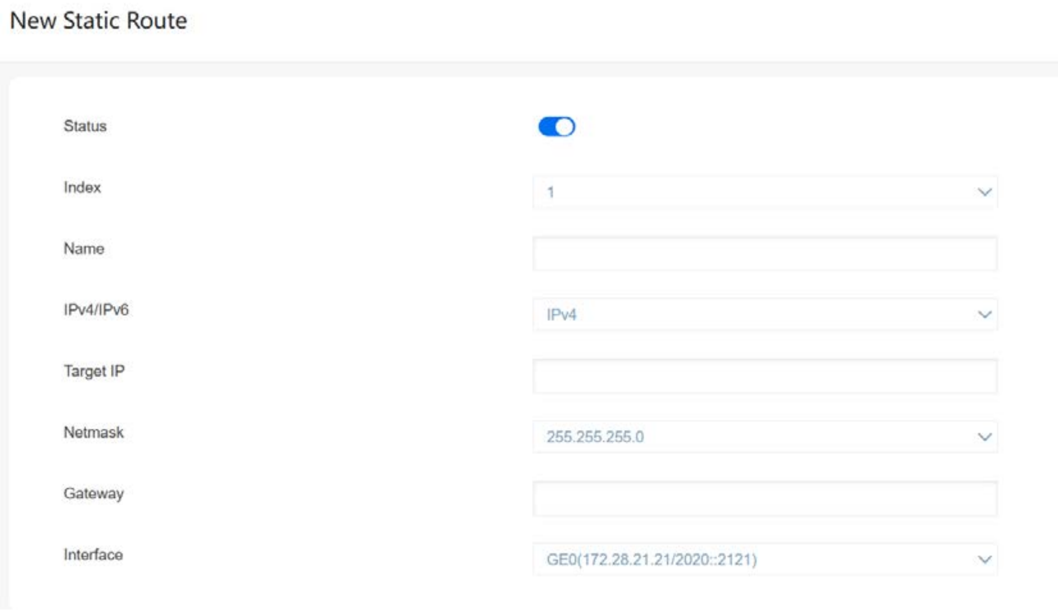


Table-Parameters of Static Route

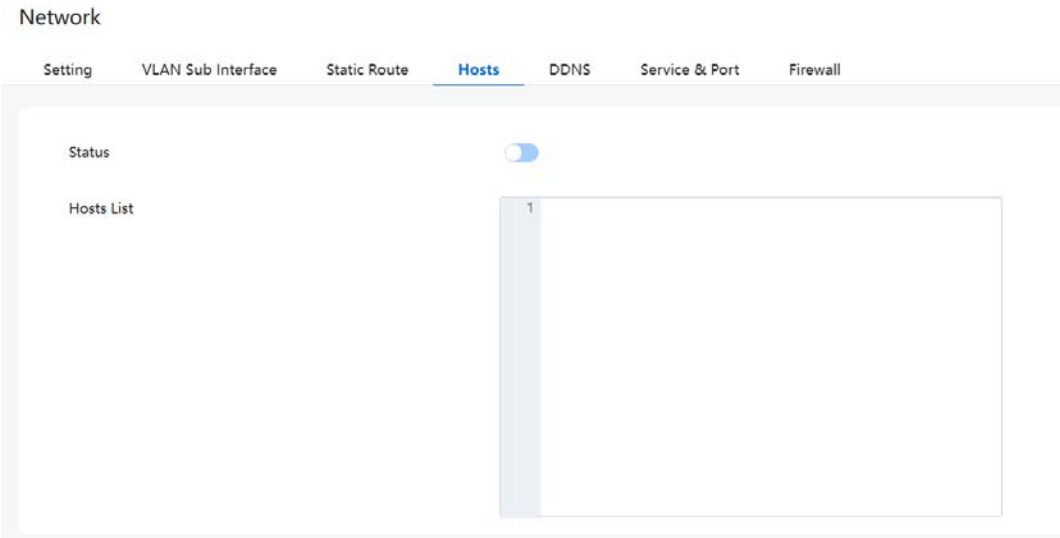
Parameter	Description
Status	Enable or disable static route.
Index	The index of the static route.
Name	The name of the static route.
IPv4/IPv6	Select network mode, IPv4 or IPv6.
Target IP	The destination IP address of the static route.
Netmask	The netmask of the static route, default: 255.255.255.0
Gateway	The IP address of the outbound gateway of the static route.
Interface	The outbound interface of the static route.

Hosts

On the **System** > **Network** > **Hosts** interface, users can add a host file. After being enabled the hosts file, users can visit the corresponding host by entering the alias or domain name of the host. The format of the hosts file is as follows: IP address host alias/domain name.

The hosts file contains the mapping relationship between IP address and host name/alias/domain name. And the mapping relationship allows quick and convenient access to the host.

Figure-Parameters of Hosts



DDNS

On the **System** > **Network** > **DDNS** interface, users can use UC350 Series IPPBX as a dynamic domain name client to map the IP address of the network to the domain name server.

DDNS (Dynamic Domain Name Server) is to map the dynamic IP address to a static domain name server, and the client program will update the currently obtained dynamic IP address to the domain name resolution when the user connects to the network.

Figure-Parameters of DDNS

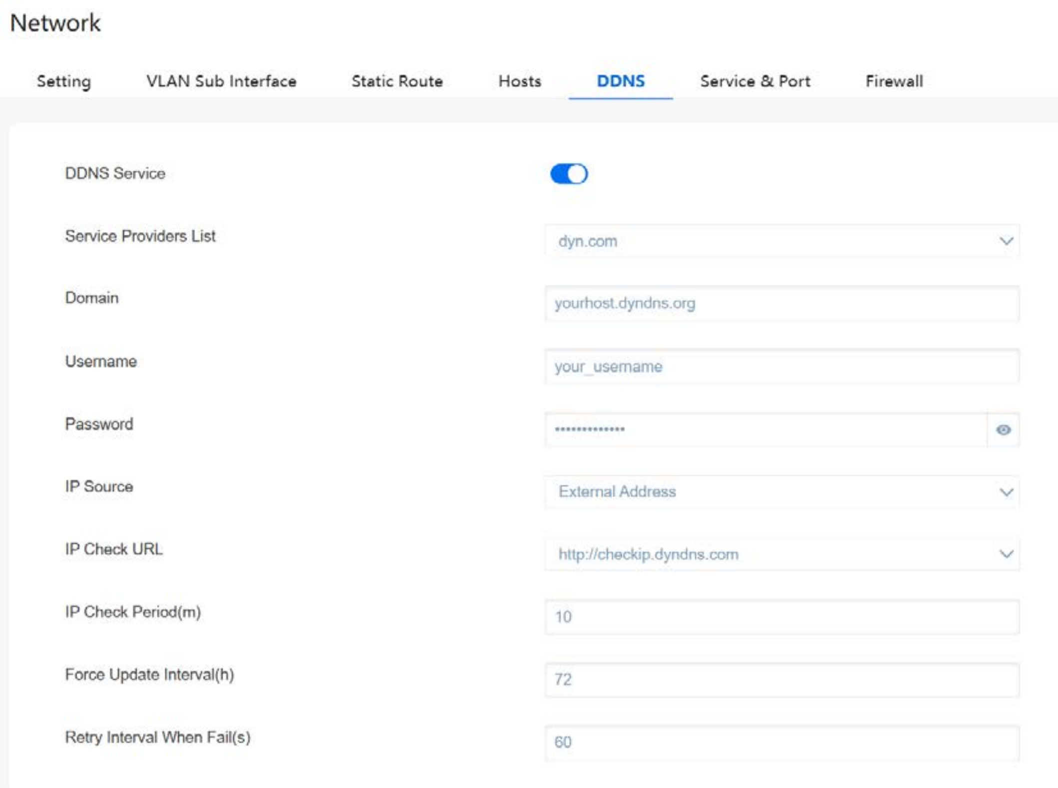


Table-Parameters of DDNS

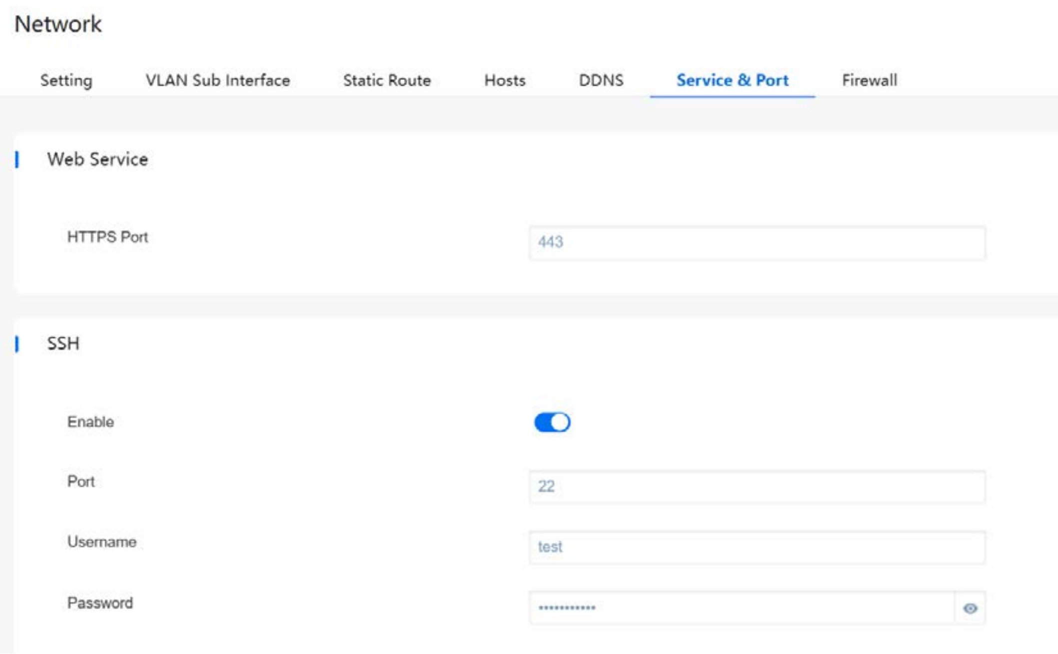
Parameter	Description
Service Providers List	Dynamic domain name service providers.
Domain	Domain name applied for on the service provider website.
Username	The user name when applying for a domain name on the service provider website.
Password	The password when applying for a domain name on the service provider website.

IP Source	The external address and the device address can be selected, the external address is the current network export public network IP address, and the device address is the GE0 port IP address.
IP Check URL	Server address that detects whether the IP address is updated.
IP Check Period(m)	Check whether the IP address has changed detection period.
Force Update Interval(h)	Force update within the configured time interval and report the IP address to the DDNS server.
Retry Interval When Fail(s)	Set the retry interval when updating the IP address fails.

Service&Port

The access ports of Web and SSH, as well as relevant on-off controls, can be configured on the **System > Network > Service&Port** interface.

Figure-Parameters of Service&Port



Firewall

Users can choose to enable the firewall function and adds filtering rules such as protocol/IP address/port/MAC address to accept or reject packets that meet the filtering rules from passing through the firewall.

Figure-Parameters of Firewall Filter Rules

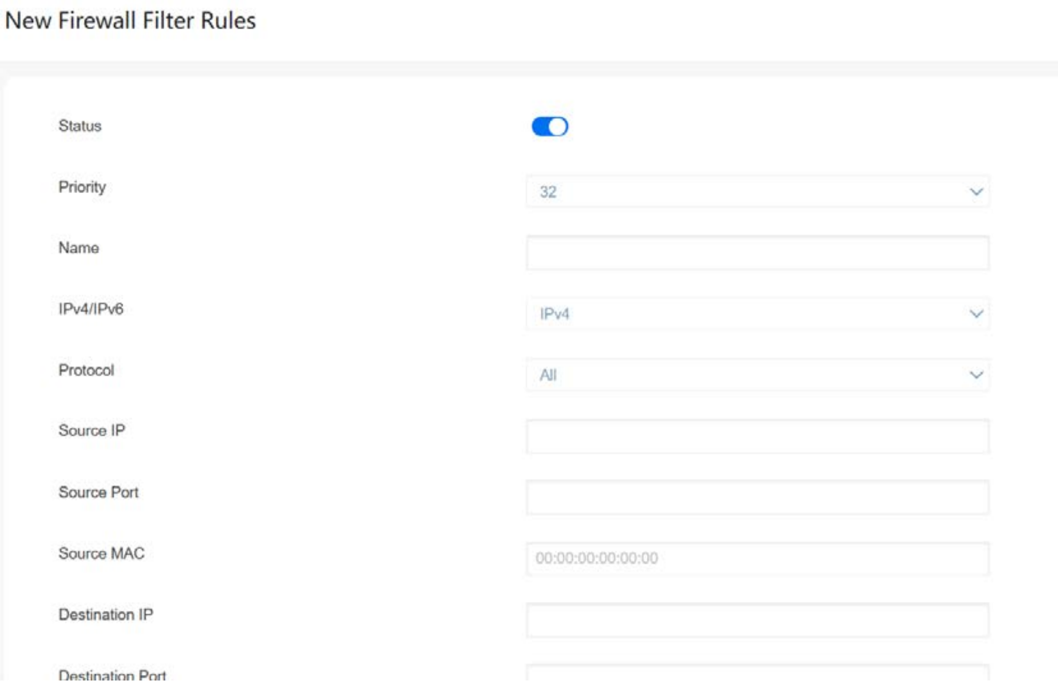


Table-Parameters of Firewall Filter Rules

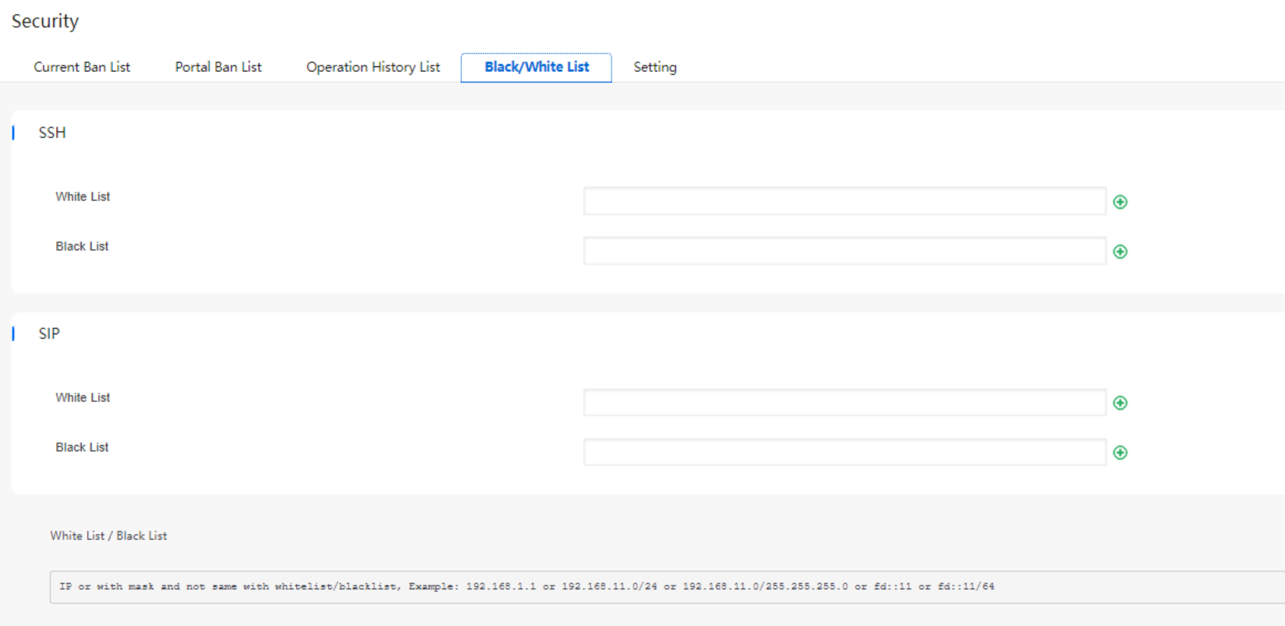
Parameter	Description
Status	Enable or disable firewall filter rules.
Priority	Set the priority of firewall filter rules.
Name	The name of firewall filter rules.
IPv4/IPv6	Select network mode, IPv4 or IPv6.
Protocol	Select Protocol, TCP, UDP or All.
Source IP	The source IP address that users want UC350 Series IPPBX to accept or reject. It is the source IP address of the message. It can also be a string of IP addresses, for example, 172.16.11.1/15.
Source Port	The source port of host which the accepted or rejected IP address belongs to.
Source MAC	The source mac of the host which the accepted or rejected IP address belongs to.
Destination IP	The destination IP address that users want UC350 Series IPPBX to accept or reject. It is the destination IP address of the message. It can also be a string of IP addresses, for example, 152.16.11.11/19.
Destination Port	The destination port of host which the accepted or rejected IP address belongs to.
Action	Choose accept or reject.

5.8.3 Security

In the "current Ban List" and " Portal Ban List", you can view the currently banned devices and unblock them. In the "Operation History list" page, you can view the ban history.

In the "**System->Security->Black / White List**" page, you can set the black and white list of SSH and SIP based on IP address, as shown in the following figure:

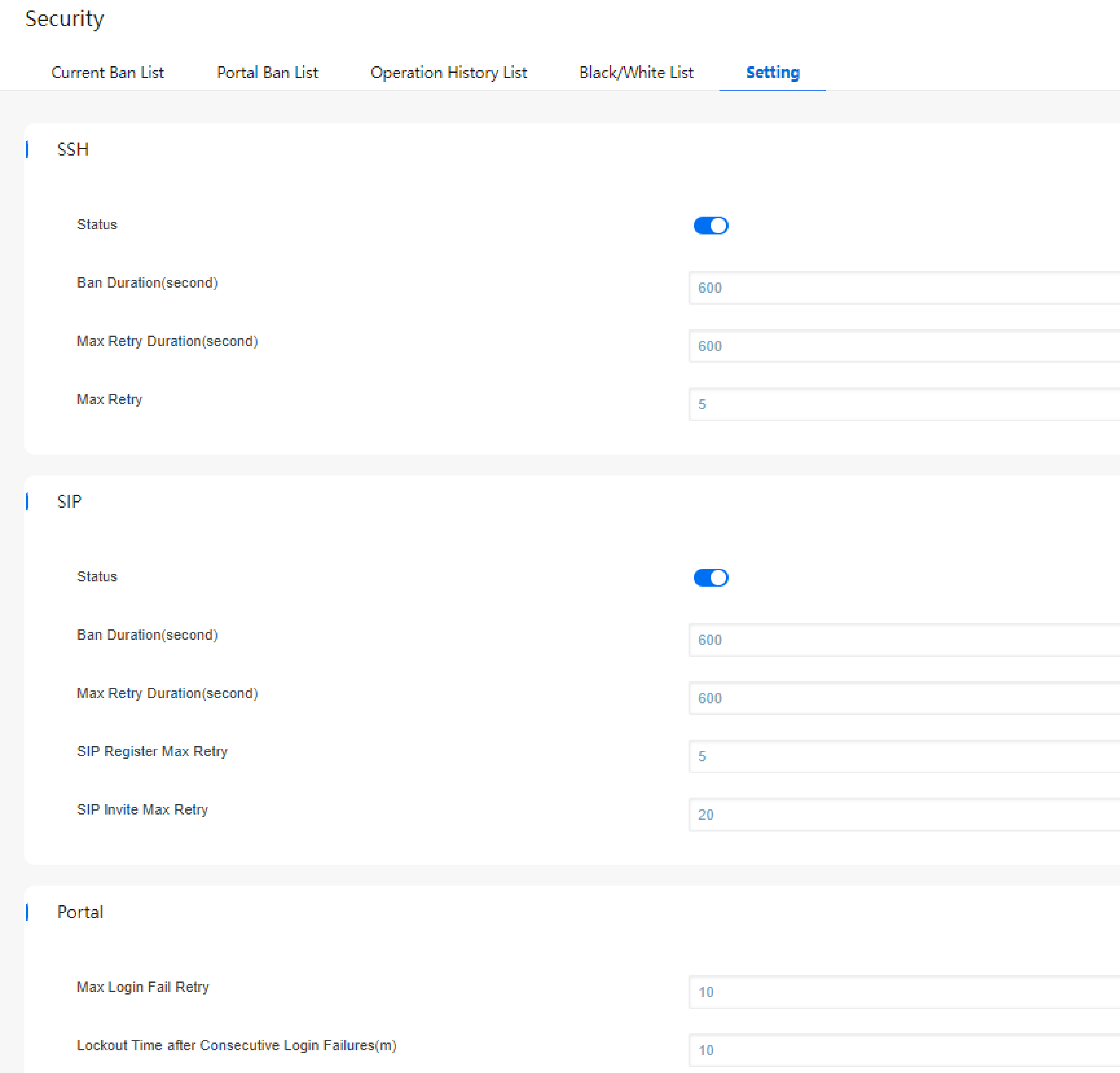
Figure-Parameters of Black/White List



Fail2ban is used to scan system logs and update firewall rules to reject the IP addresses that show malicious signs (for example, too many login failures) for a specified amount of time.

On the **System->Security->Setting** interface, users can configure rules for Fail2ban. Fail2 ban is generally targeted SSH and SIP. The portal is mainly used to set the Max Login Fail Retry

Figure-Parameters of security Setting



5.8.4 Storage

On **System** > **Storage** interface, users can view the storage status of the local storage directory. UC350 Series IPPBX will divide the local storage area into 3 storage zones: recording file storage zone, voicemail storage zone, and other storage zones (by default, the maximum ratio of each storage zone is 50%, 35%, 15%). Users can resize and remove the storage zones on **System** > **Disk Manager** page.

Figure-Parameters of Storage

5.8.5 Hot Standby（UC350 Pro）

In order to ensure that the UC350 Series IPPBX can work normally and stably, the UC350 Pro supports a double-device Hot Standby function.

On **System** > **Hot Standby** page, users can configure the dual hot standby function of the device, the configuration steps are as follows:

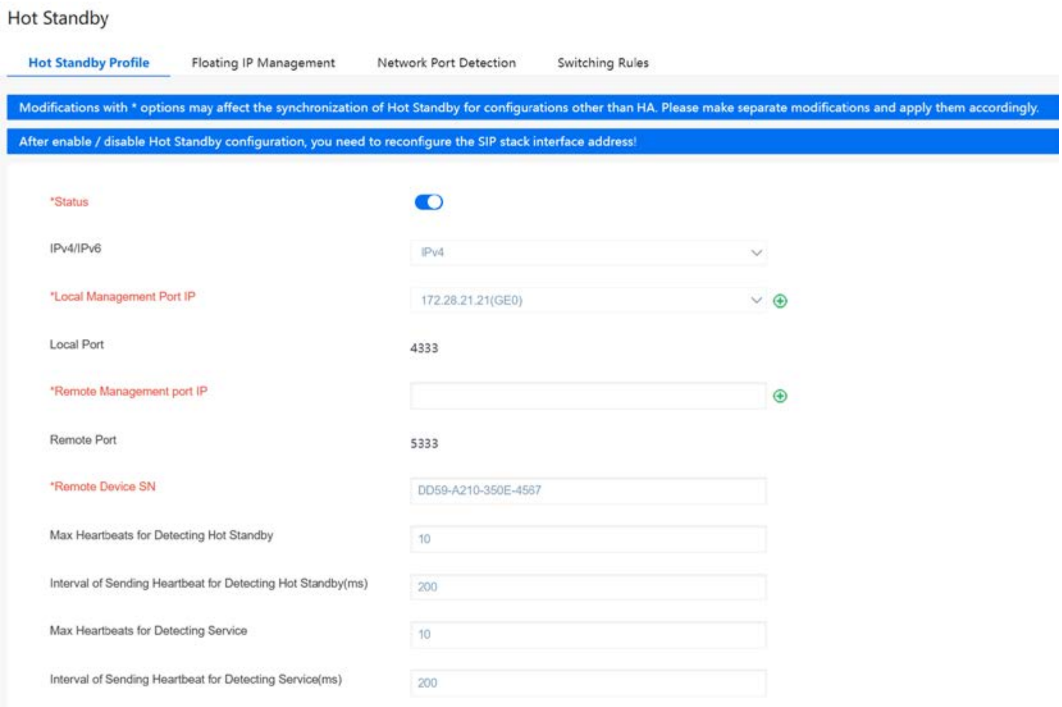
1. Configure the master and backup server information, specify the local management IP address, configure the remote management IP address and the serial number of the standby device, and click **Save** to take effect.

2. Create a new floating IP address for the device and bind the physical interface address for heartbeat detection, and click **Save** to take effect.

3. Configure the network interface detection (which can be distinguished from the local management IP address), click **Save** to take effect:

4. Configure the Switching Rules, configure the weight value of the interface with the local management IP address (the weight ranges from 1 to 10), and click **Save** to take effect.

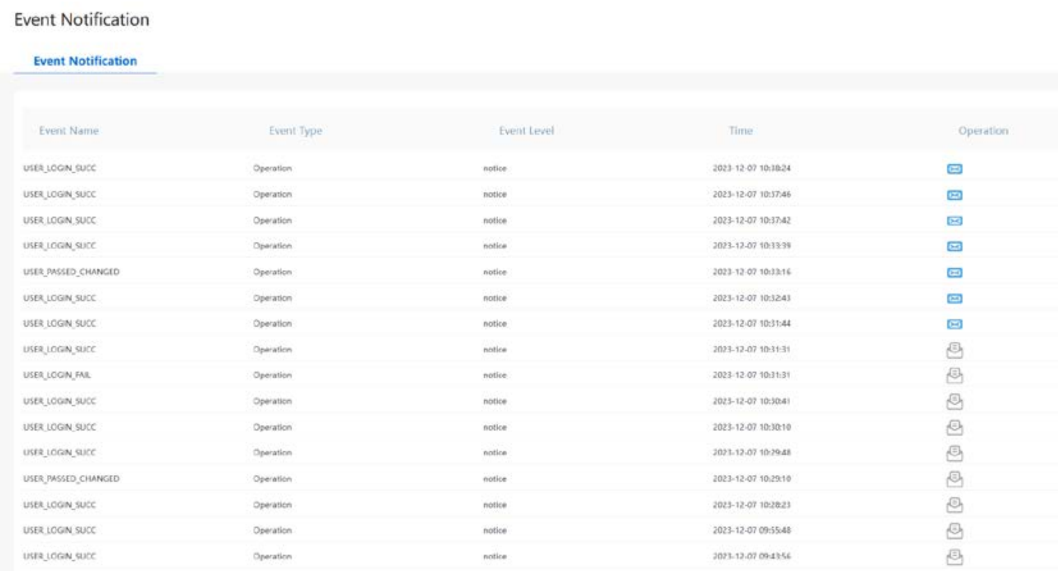
Figure-Parameters of Hot Standby Profile



5.8.6 Event Notification

This page mainly records and displays the events such as login, call service, and warning, etc. Clicking the **Operation** button of the event can view the details, which can be used to troubleshoot and trace the problems.

Figure-Parameters of Event Notification



5.8.7 Email

On the **System** > **Email** > **Configuration** interface, users can configure an email client, and can test the connection for sending mails. But the premise is that the configured email needs to open SMTP, IMAP and POP3 services. With voicemail, it can realize voicemail to email and will generate logs, users can go to **System** > **Log** to check.

Figure-Parameters of Email

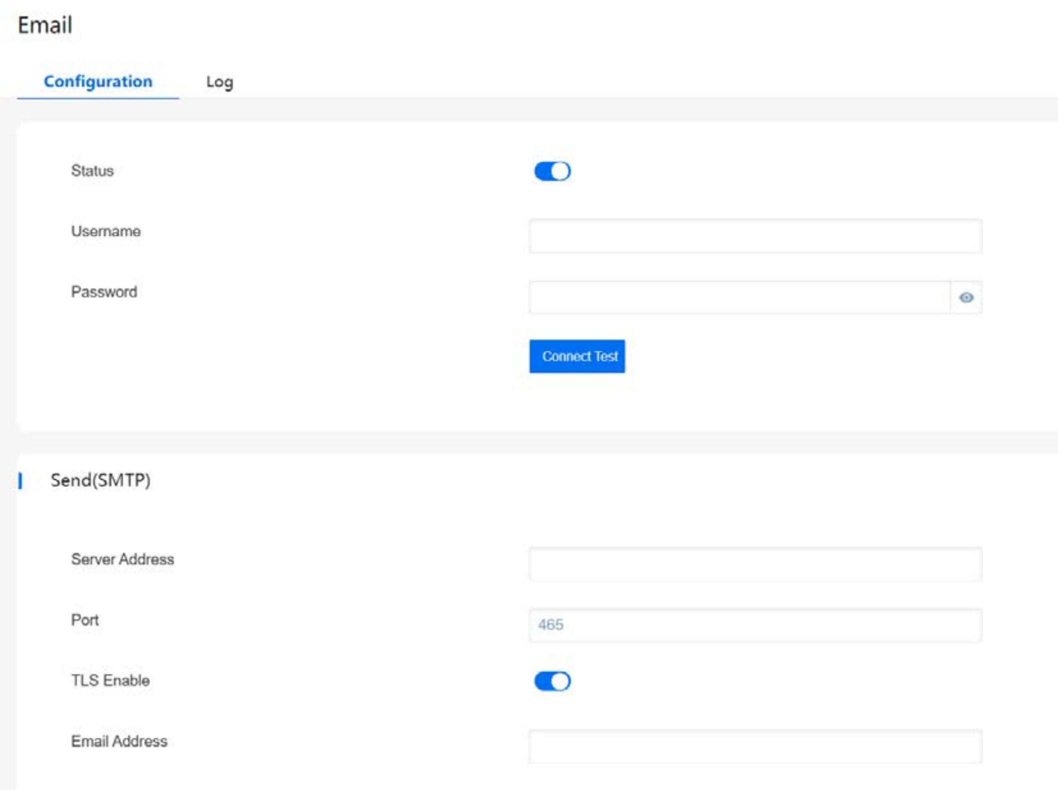


Table-Parameters of Email

Parameter	Description
Status	Enable or disable email client.
Username	Enter the address of email client.
Password	The password or authorization code of the email client.

Server Address	The address of the SMTP server, supported by the email client.
Port	Configure the port of email client.
TLS Enable	Disable or enable TLS.
Email Address	Configure Email Address.

5.8.8 Personalization

On the "System->Personalization Settings" page, users can customize the host name, device name, login portal display, and banner logo display.

Figure-Parameters of Personalization setting

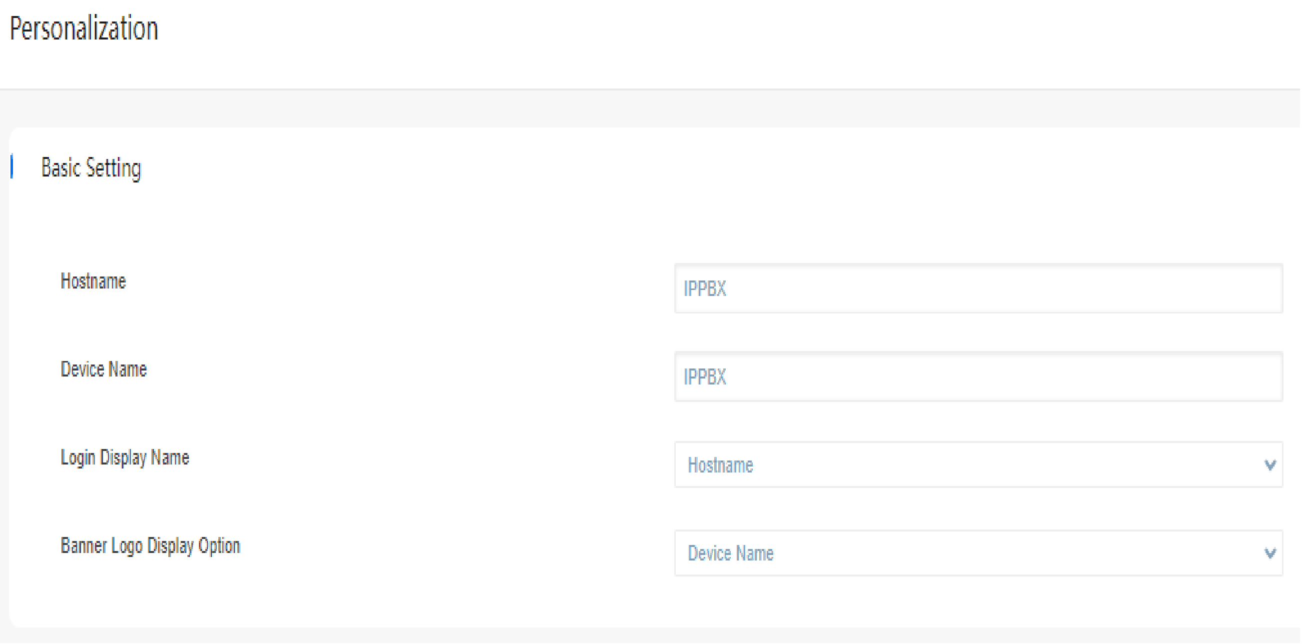


table-Parameters of Personalization

Parameter	Description
Hostname	Customize the host name. A valid host name should start with a letter and can contain numbers, dots or minus signs. The last character cannot be a dot or minus sign. The total length should not exceed 24 characters. After saving, restart the device to take effect
Device Name	Customize the device name. The configuration name cannot be empty, has a maximum of 32 characters and cannot contain double quotes. After saving, restart the device to take effect
Login Display Name	You can select "Host Name" or "Device Name" to define your own style
Banner Logo Display Option	You can choose "Default", "Host Name" or "Device Name" to define your own style. The default option banner logo is: DINSTAR

5.9 Maintenance

5.9.1 User Manager

On the **Maintenance** > **User Manager** interface, and users can set the username, password and manage other users. The default username and password are admin and admin@123#, so it is strongly advised to modify them for security purpose.

The super administrator of the system can add different users to the system and assign different roles for them, like observer, operator, and administrator. Different roles can support different permissions to the functions.

Figure-Parameters of User

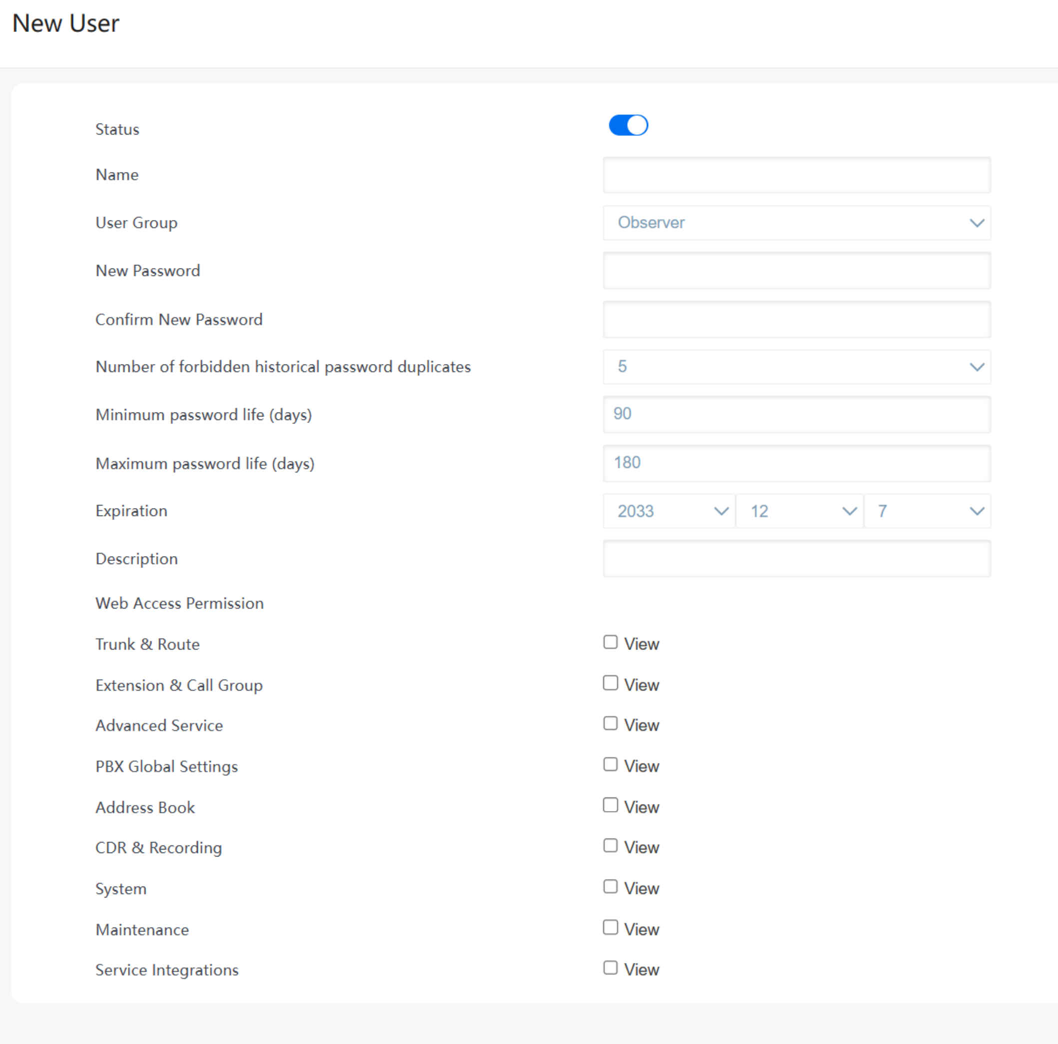


Figure-Parameters of User

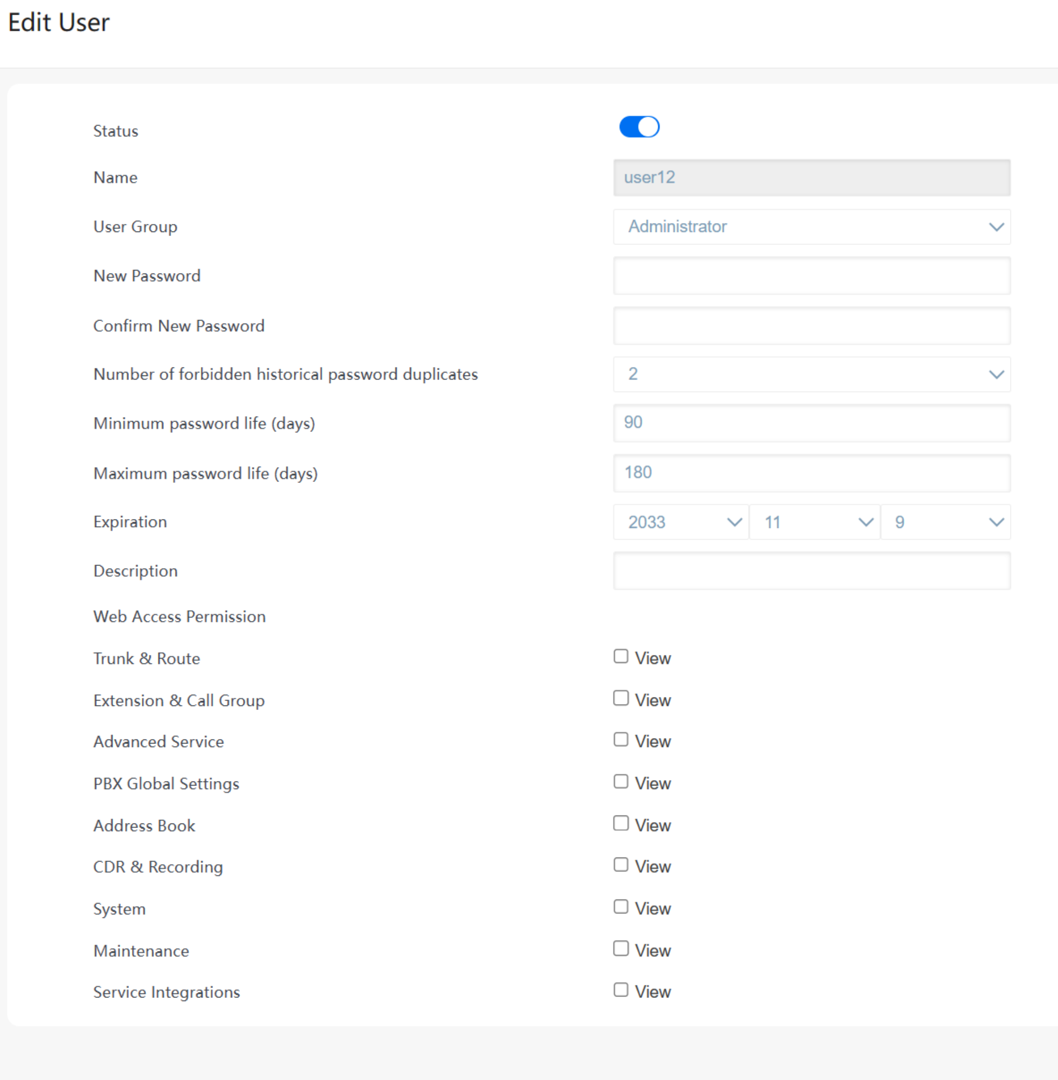


Table-Parameters of User

Parameter	Description
Status	Enable or disable the new user.
Name	The name of the new user. After it is established, the name and the password will be used to log into the web interface of the system.

User Group	Users can choose a role for the new user, such as administrator, operator, and observer. The default value is administrator.
New Password	Setting the login password for the new user. The password needs to consist of 8 to 32 characters.
Confirm New Password	Enter new password to confirm.
Number of forbidden historical password duplicates	Set the number of forbidden historical password duplicates, select from 1-10.
Minimum password life (days)	Set the minimum period of password usage.
Maximum password life (days)	Set the maximum password usage period.
Expiration	The expiration time of this user's login or operation.
Description	The description of the new user.
Web Access Permission	Set the user's access rights.

5.9.2 License

The device features and performance specifications can be controlled through the license. After the user gets the license information, it will be authorized on this page. After the authorization is successful, the license will be taken effect by restarting the device.

Figure-Parameters of License

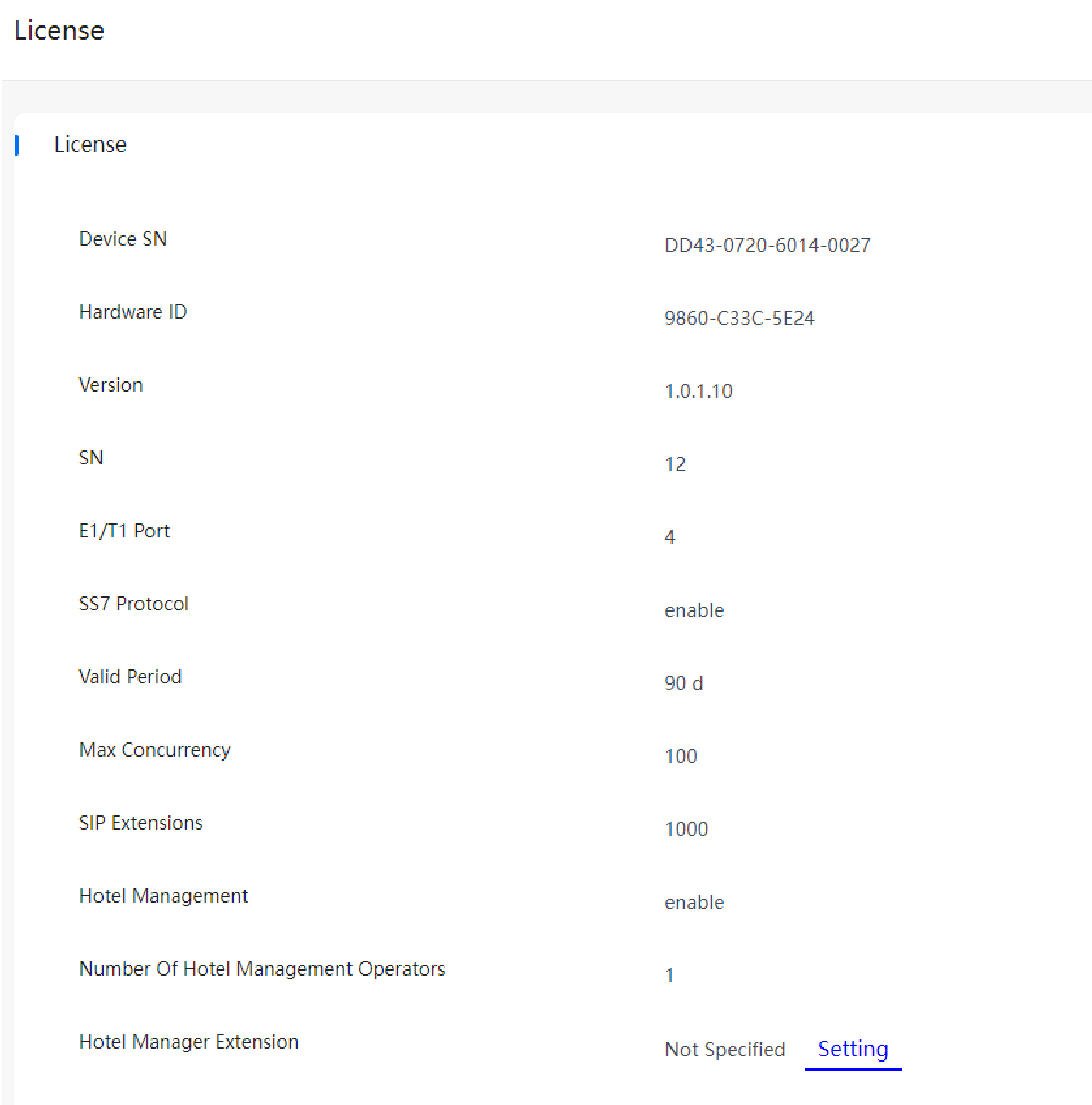


Table-Parameters of License

Parameter	Description
E1/T1 Port	Number of E1/T1 ports authorized by UC
SS7 Protocol	Indicate whether UC has authorized SS7

Valid Period	Represents the validity period of the license. If it expires, you need to reapply.
Max Concurrency	The maximum number of concurrent calls supported by UC
SIP Extensions	The maximum number of SIP extensions that can be created by UC
Hotel Management	Indicates whether UC has authorized the hotel management function (corresponding to the hotel management function of the personal portal)
Number Of Hotel Management Operators	This will take effect only if the console authorization is not performed and only the hotel module authorization is performed. When creating a new console type queue, the number of static seats in all console queues cannot exceed this number
Hotel Manager Extension	Custom hotel administrator extension, used for personal portal hotel management function
Attendant Console	Indicates whether UC has authorized the console function (corresponding to the console function of the personal portal)

Telephone Operator Amount	When creating a new console type queue, the number of static seats in all console queues cannot exceed this number
Remain	Indicates how many days are left for the license to expire
Hot Standby	Indicates whether UC has authorized the dual-machine hot standby function

5.9.3 Firmware

On **Maintenance** > **Firmware** interface, users can upgrade the device version. The upgraded version will take effect after rebooting the device.

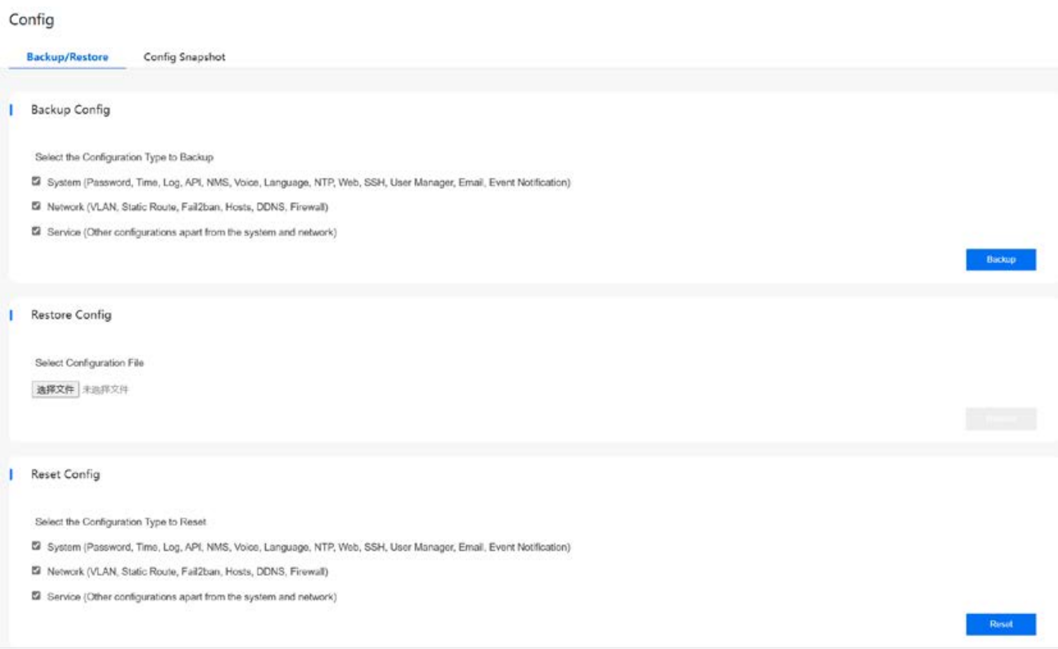
The upgrade types can be: system, patch, user board app, user board image. Users can choose the upgrade type according to different needs for upgrading, and the upgrade files must be provided by the vendor.

Figure-Parameters of Firmware

5.9.4 Config

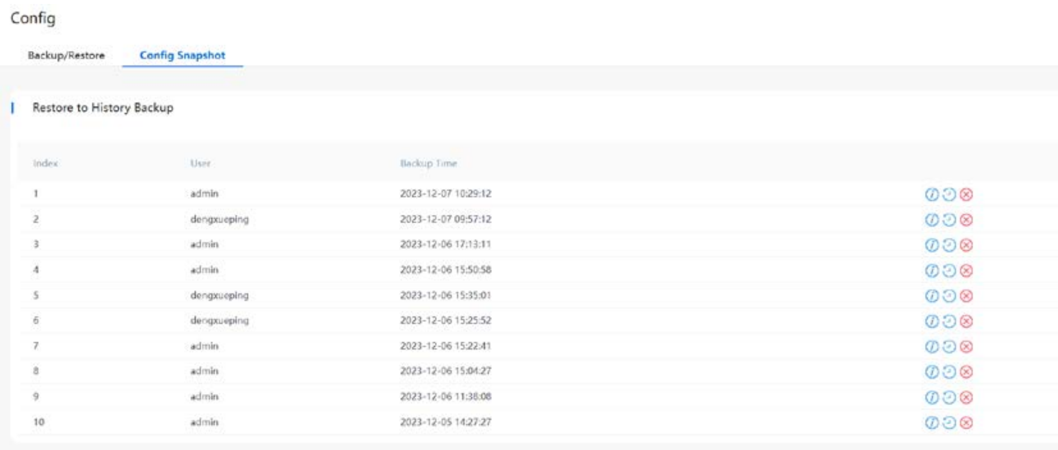
On the **Maintenance** > **Config** interface, users can back up or restore configuration files. But users need to restart the device for the change to take effect after executing restore.

Figure-Parameters of Backup/Restore



The device supports the snapshot function. If users are not sure whether the modified configuration is correct or not, they can restore the historical configuration on **Maintenance > Config > Config Snapshot** interface according to the configuration time.

Figure-Parameters of Config Snapshot



5.9.5 Schedule Task

On the **Maintenance > Config > Schedule Task** interface, users can set a scheduled to restart the UC350 Series IPPBX device, record backup, and back up CDRs, logs or configurations.

Figure-Parameters of Reboot

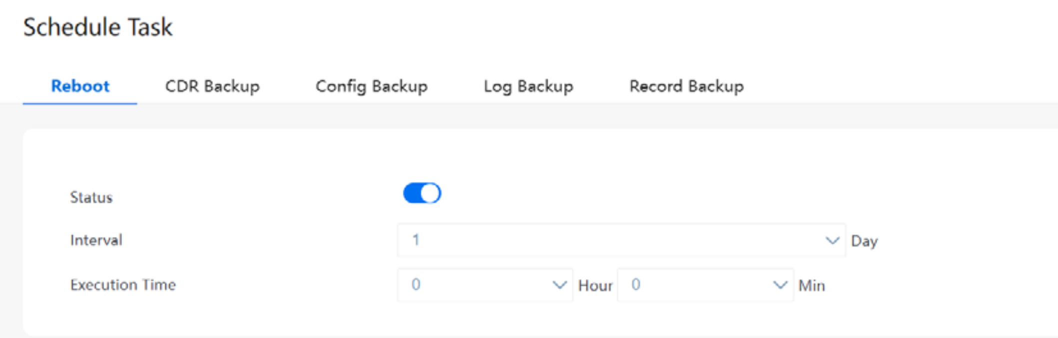


Figure-Parameters of CDR Backup

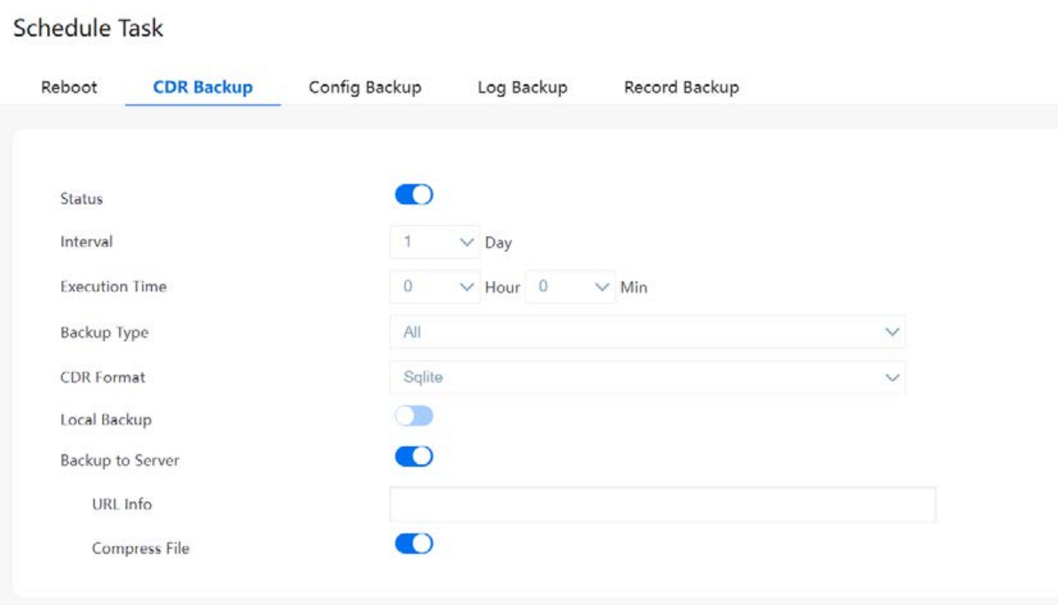


Figure-Parameters of Config Backup

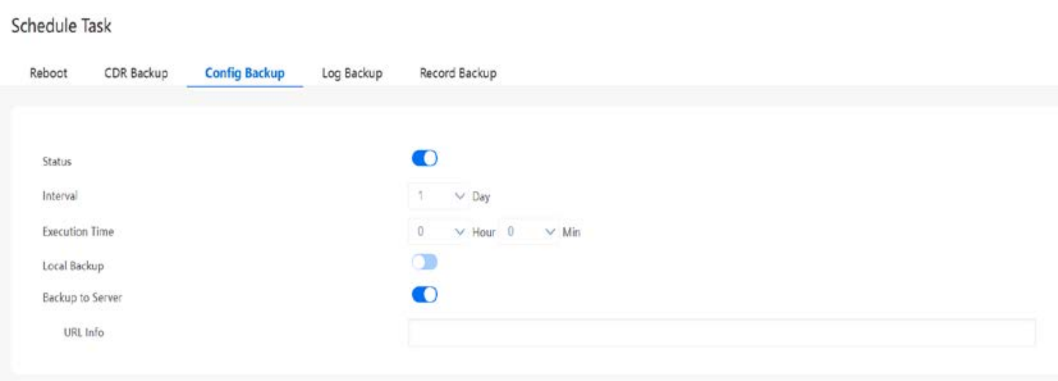


Figure-Parameters of Log Backup

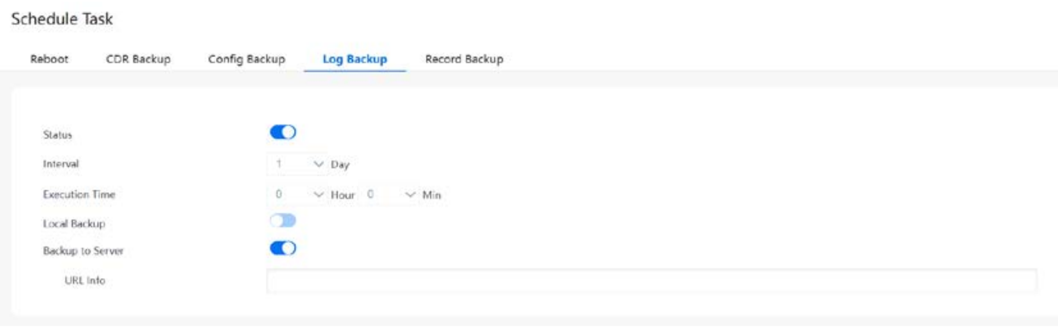
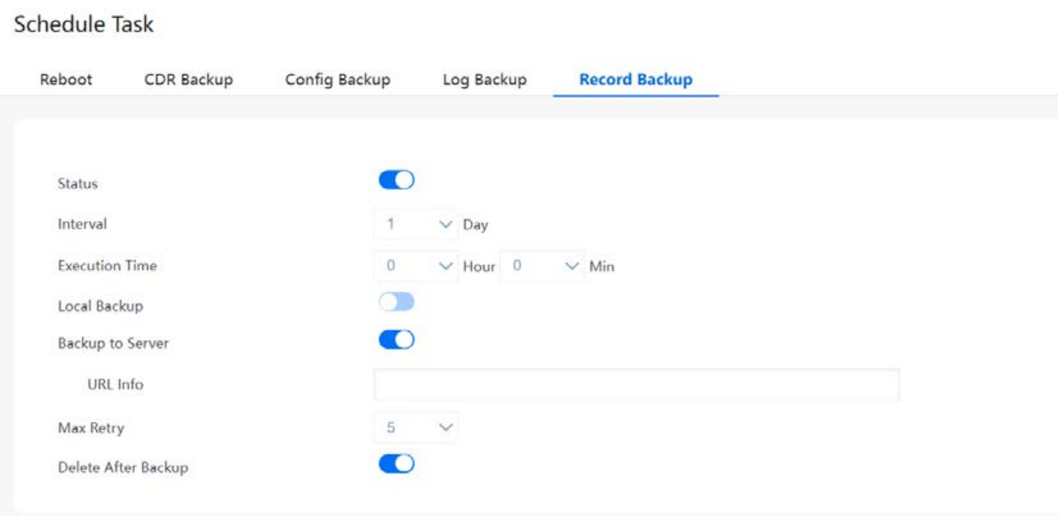


Figure-Parameters of Record Backup

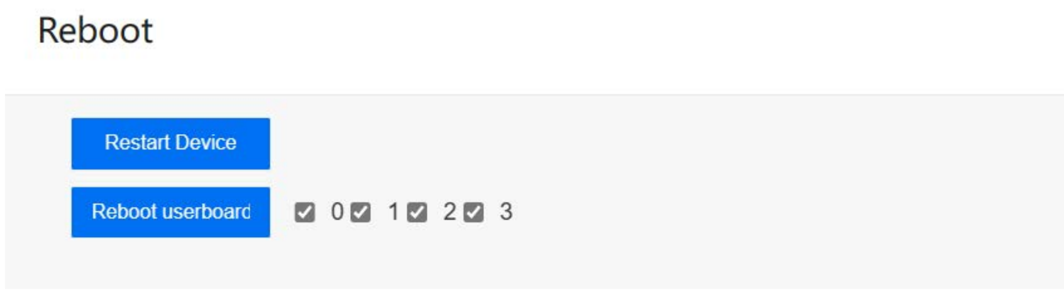


5.9.6 Reboot

On the **Maintenance** > **Reboot** interface, users can click Restart Device to reboot the UC350 Series IPPBX gateway. After the device is rebooted, those configurations that have been saved will remain unchanged.

The device supports userboard reboot operation, select the userboard, click "**Reboot userboard**", the userboard can be rebooted directly, without affecting the normal operation of the device.

Figure-Parameters of Reboot



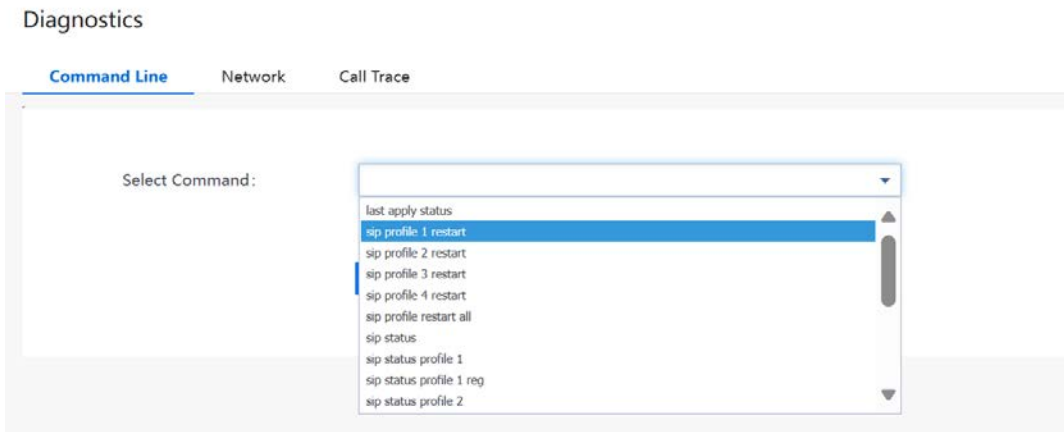
5.9.7 Diagnostics

Command Line

On the **Maintenance** > **Diagnostics** > **Command Line** interface, some commonly-used command lines can be directly selected in the draw-down box, and therefore user has no need to enter command lines on Telnet. In this way, the efficiency of problem diagnostics is greatly improved.

Commonly used command lines include sip status, sip profile and so on.

Figure-Parameters of Command Line



Network

On the **Maintenance** > **Diagnostics** > **Network** interface, users can use three network utilities including Ping, Traceroute and Nslookup to diagnose the network, and can capture data packages of the available network ports.

【Ping】

Ping is used to examine whether a network works normally through sending test packets and calculating response time.

Instructions for using Ping:

1. Enter the IP address or domain name of a network, a website or a device in the input box of Ping, and then click Ping.
2. If related messages are received, it means the network works normally. Otherwise, the network is not connected or is connected faultily.

【Traceroute】

Traceroute is used to determine a route from one IP address to another.

Instruction for using Traceroute:

1. Enter the IP address or domain name of a destination device in the input box of Traceroute, and then click Traceroute.
2. View the route information from the returned message.

【Nslookup】

Nslookup (Name Server Lookup) is a network command-line tool to obtain domain name of internet or to diagnose the problems of DNS.

Instruction for using Nslookup:

1. Enter a domain name and then click Nslookup.
2. View the DNS information from the returned message.

【Network Capture】

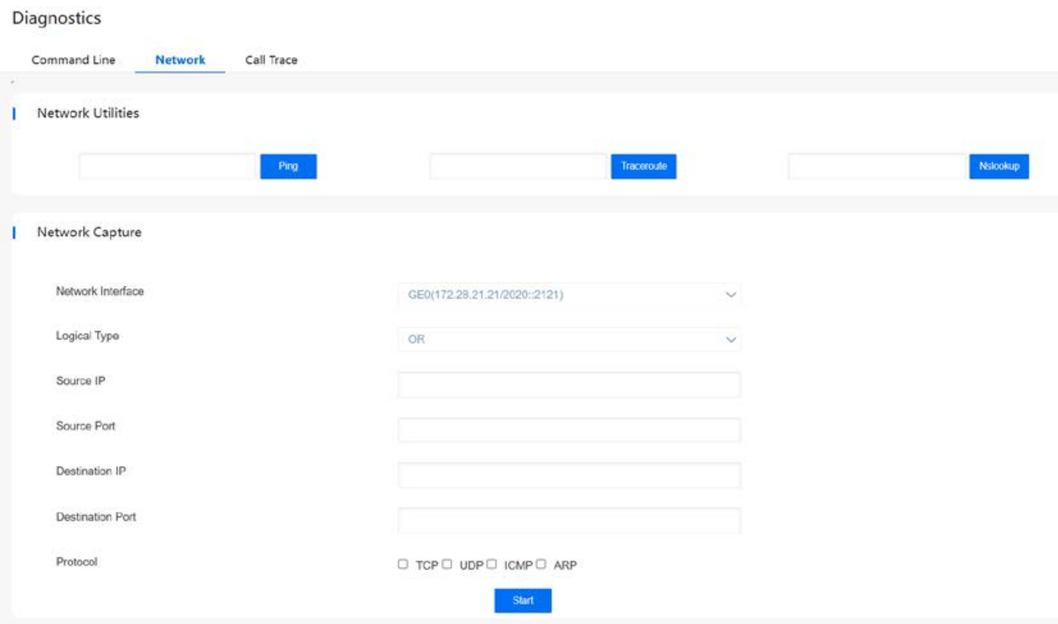
On the following interface, users can capture data packages of the available network ports. Users can also set source IP, source port, destination IP or destination port to capture the packages that users want.

There is a "and"/"or" logical type. The "and" relationship can only capture a one-way message, and "or" relationship to fetch the interaction message between a particular IP.

Note: If there are multiple source or destination IP addresses, please use '|' to separate them, for example, 172.16.115.12|172.16.115.15.

After package capturing is completed, save the captured packages on a computer and then use a tool to analyze them.

Figure-Parameters of Network



Call Trace

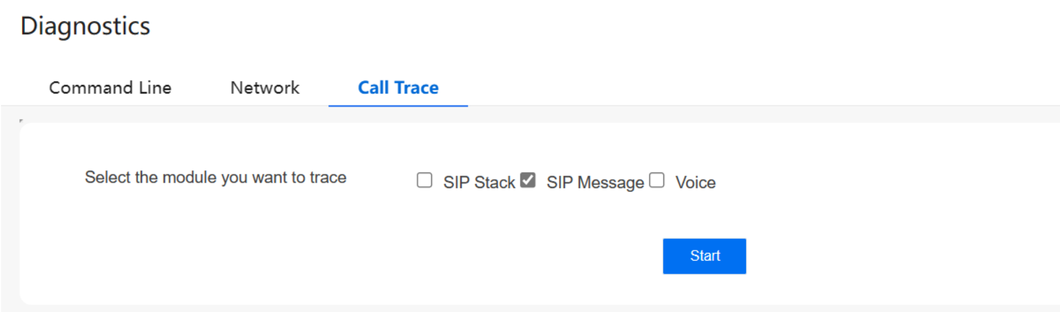
In case that call cannot be connected or voice has quality problem, users can enter the **Maintenance** > **Diagnostics** > **Call Trace** interface to collect fault-related information and then send it to technical support to identify the fault.

1. Select the module that need to be traced. For example, if a call from SIP to FXS has voice problem, users can select SIP message, SIP Stack and Voice, and then click the Start button.

2. Give a call, and come back to the **Maintenance** > **Diagnostics** > **Call Trace** interface after the call ends. Then click Stop and download the tracing file.

3. In order to locate faults more quickly, users sometimes need to enter into the **Maintenance > Log > Service Log** interface, click export, and then send this exported file and the tracing file to technical support.

Figure-Parameters of Call Trace



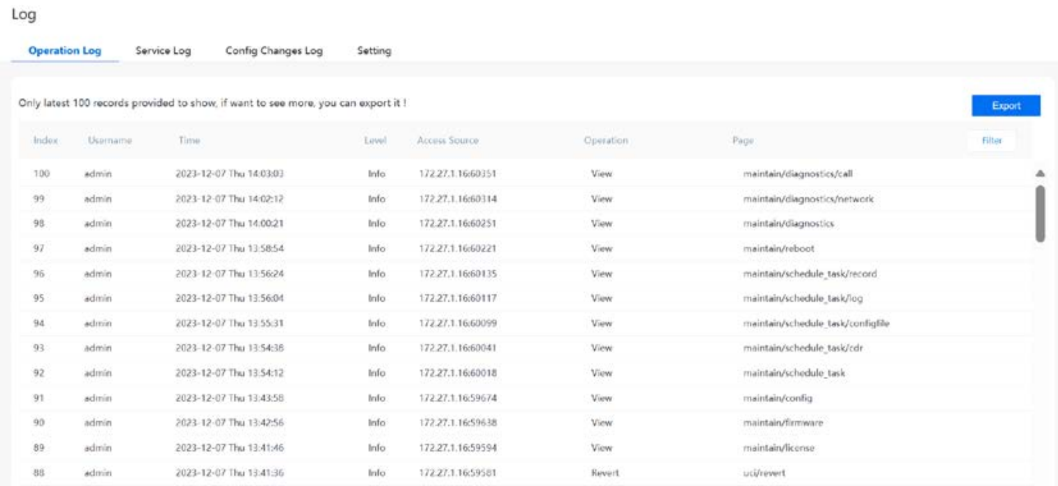
5.9.8 Log

Operation Log

The logs tracing the operations carried out on the Web can be queried on the **Maintenance > Log > Operation Log** interface. Users are allowed to set query criteria to query the logs that users want and to export the logs through clicking the Export button at the top-right corner.

Note: The operation log is mainly used by vendors to figure out problems.

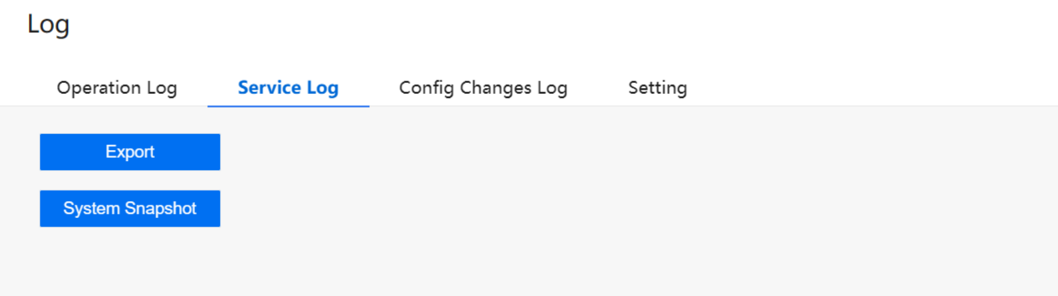
Figure-Parameters of Operation Log



Service Log

Service logs can be exported on the **Maintenance** > **Log** > **Service Log** interface. Those logs are generally used to identify system problems.

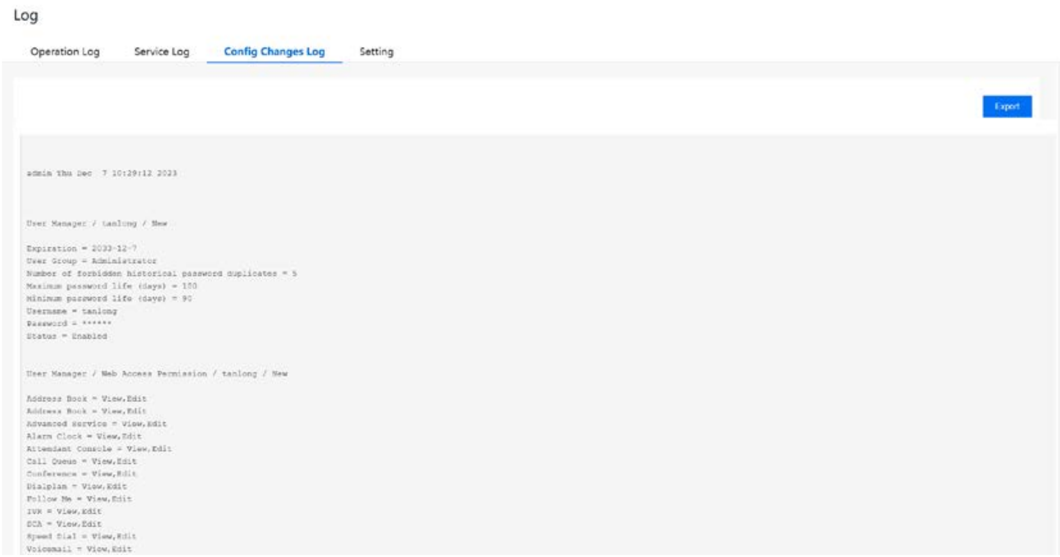
Figure-Parameters of Service Log



Config Changes Log

On the **Maintenance** > **Log** > **Config Changes Log** interface, the configurations changed by administrator on the Web of the gateway are recorded.

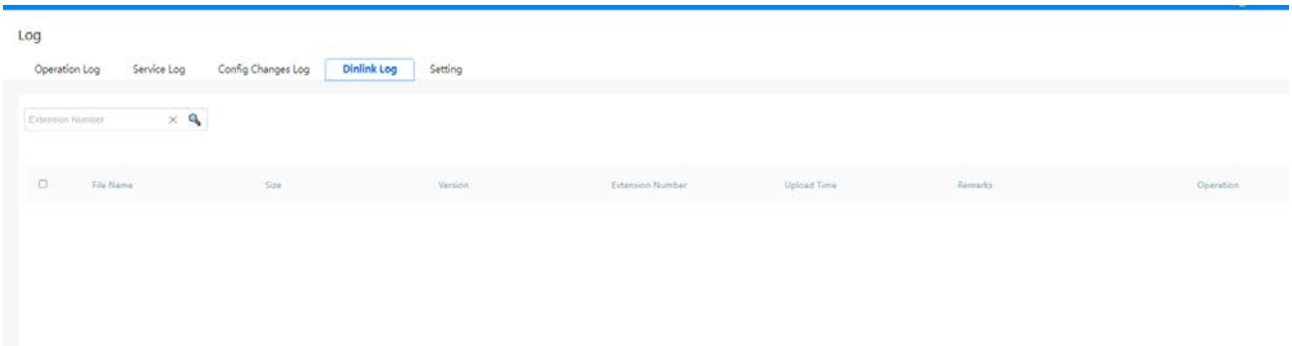
Figure-Parameters of Config Changes Log



Dinlink log

On the "**Maintenance ->Log->Dinlink Log**" page, users can use the feedback function on the Dinlink APP to feedback the problems encountered when using the APP, which will help R&D locate related problems.

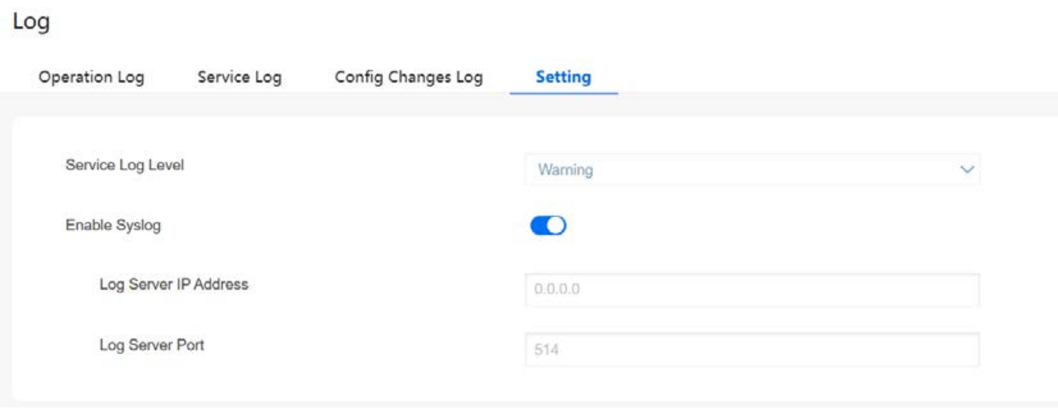
Figure-Parameters of Dinlink log



Setting

On the **Maintenance > Log > Setting** interface, User can configure the device remote logging function, specify the device logging level, set the log server IP address, receive real-time tracking device operation log, and understand the work of the device.

Figure-Parameters of Setting

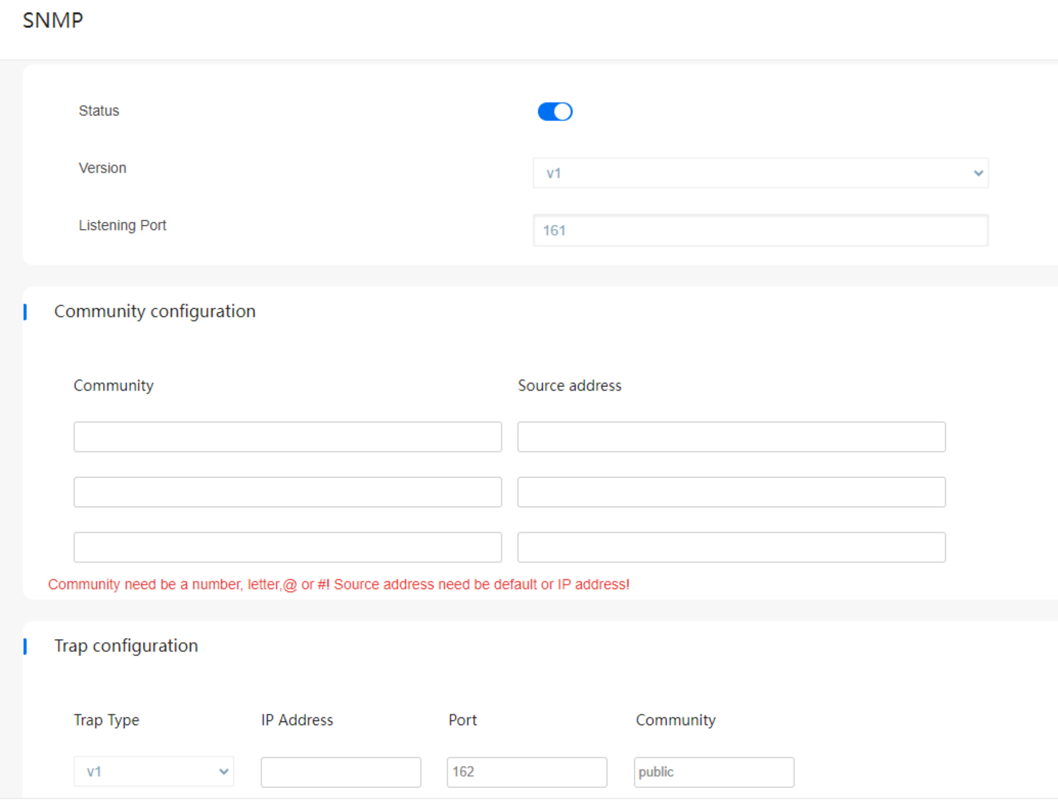


5.9.9 SNMP

SNMP stands for Simple Network Management Protocol, and originated from the Simple Gateway Monitoring Protocol (SGMP) , It's a powerful tool that facilitates the sharing of information among various devices on a network, regardless of their hardware or software. SNMP is designed to manage a wide range of hardware and software platforms from different manufacturers, conforming to the Internet standard network management framework. Currently, SNMP has progressed to its third version, SNMPv3, which offers significant improvements in security, functionality, and performance over earlier versions.

Note: Currently only UC350 supports SNMP, UC350 Pro doesn't support.

Figure-Parameters of SNMP



Parameter	Description
Status	Enable or disable SNMP.
Version	SNMP version, support v1, v2c and v3.
Listening Port	To configure SNMP listening port, (1~65535)
Community configuration	**Community:** To configure Community, equal to the password in authentication. **Source address:** Snmp sever address, need be default or IP address.

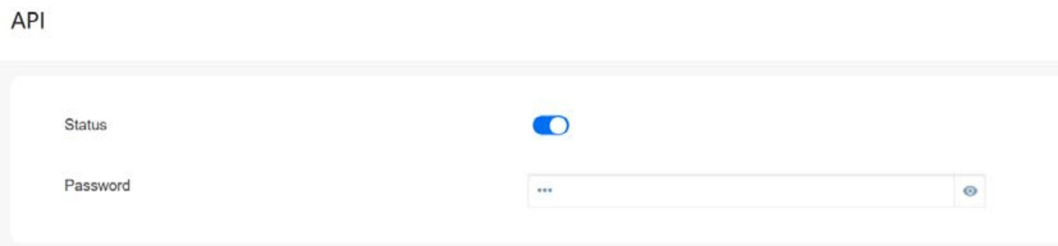
Trap configuration	**Trap Type:** Optional v1, v2c or v3. **IP Address:** Snmp sever address. **Port:** Snmp Server Port. **Community:** To configuration Community, default is public.

5.10 Service Integrations

5.10.1 API

The device opens the API interface. Users can enable the API status and set the password in the **Service Integrations > API** interface. when connecting three-party devices/platforms, the configured password will be used for verification to ensure the security of docking between the devices.

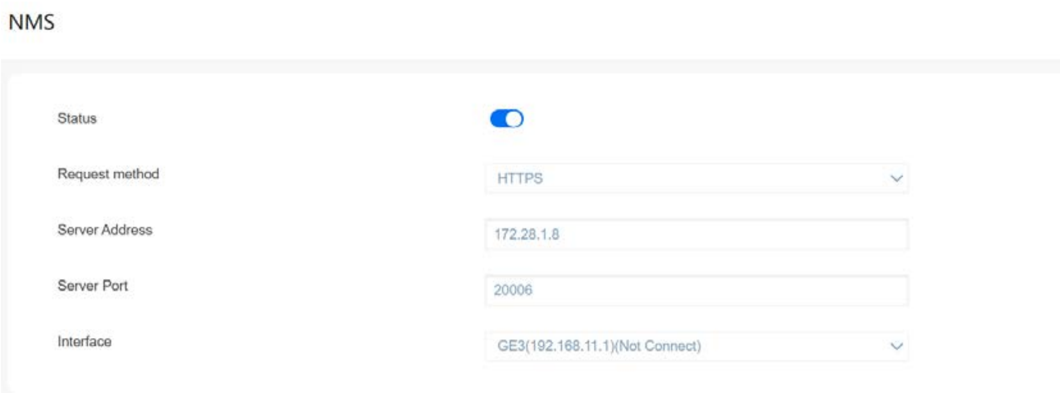
Figure-Parameters of API



5.10.2 NMS

UC350 Series IPPBX supports the network management system, which can help users to access devices, modify device configurations, upgrade devices and other operations.

Figure-Parameters of NMS



5.10.3 Event Report

UC350 Series IPPBX allows the following events to be reported through URL: call status, Register or deregister SIP extension, availability or unavailability of SIP trunks, CDR and Recording information.

For event report through URL, please see the following example:

1. On the **Service Integrations** > **Event Report** interface, select the events to be reported and the reporting method (URL).

2. Input the URL.

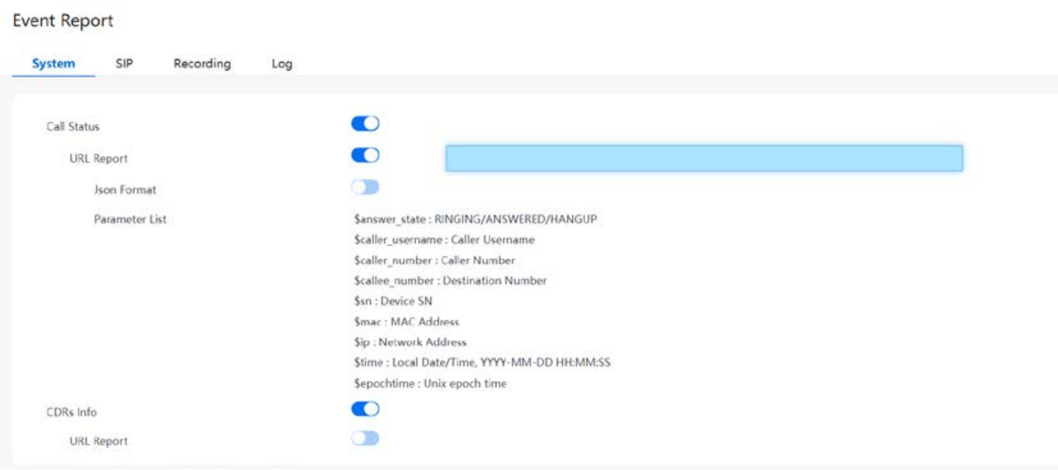
Format:

http://ip:port/event?key1=$value1&key2=$value2

Example: http://172.18.111.65:8080/sip?sn=$sn&mac=$username&key=$sip_status

Event refers to call status、sip、siptrunk、CDR and Recording, while value refers to the parameter that needs to be reported. Key can be defined by users, but it’s generally the same with value.

Figure-Parameters of Event Report



3. Use a softphone to register to an extension of UC350 Series IPPBX, and then the registration or deregistration of the softphone will be reported to UC350 Series IPPBX through the URL.

4. On the **Service Integrations** > **Event Report** > **Log** interface, users can view the report information.

Figure-Parameters of Event Report Log

6 Appendix

Abbreviation	Explanation
ARP	Address Resolution Protocol
CID	Caller Identity
DNS	Domain Name System
DDNS	Dynamic Domain Name Service
DHCP	Dynamic Host Configuration Protocol
DMZ	Demilitarized Zone
DND	Do NOT Disturb
DTMF	DTMF：Dual Tone Multi Frequency
FTP	File Transfer Protocol
HTTP	Hypertext Transfer Protocol
LAN	Local Area Network
L2TP	Layer 2 Tunneling Protocol
PPTP	Point-to-Point Tunneling Protocol
MAC Address	Media Access Control Address
NAT	Network Address Translation
Ping	Packet Internet Grope
SIP	Session Initiation Protocol

Abbreviation	Explanation
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
RTP	Real Time Protocol
PPPOE	Point-to-point Protocol over Ethernet
QoS	Quality of Service
UPnP	Universal Plug and Play
VLAN	Virtual Local Area Network
NTP	Network Time Protocol
STUN	Simple Traversal of UDP over NAT
PSTN	Public Switched Telephone Network
WLAN	Wireless Local Area Network