UC8000 IPPBX User Manual

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Preface

Welcome

Thanks for choosing the UC8000 IPPBX! We hope you will make full use of this rich-feature product. If you need any technical support you can contact us at +86 755 2645 6664.

About This Manual

This manual provides information about the introduction of the UC8000 IPPBX, and about how to install, configure or use the UC8000 IPPBX. Please read this document carefully before using the UC8000 IPPBX.

Intended Audience

This manual is aimed primarily at the following people:

1. Users
2. Engineers who install, configure, and maintain the device.

Document Information

Document Name	UC8000 IPPBX User Manual
Document Version	V1.0
Firmware Version	2.58.3.4

Conventions

All references to the system in this document refer to the UC8000 IPPBX. "Note" marked in the document is what users need to pay attention to.

Contents

1 Product Introduction

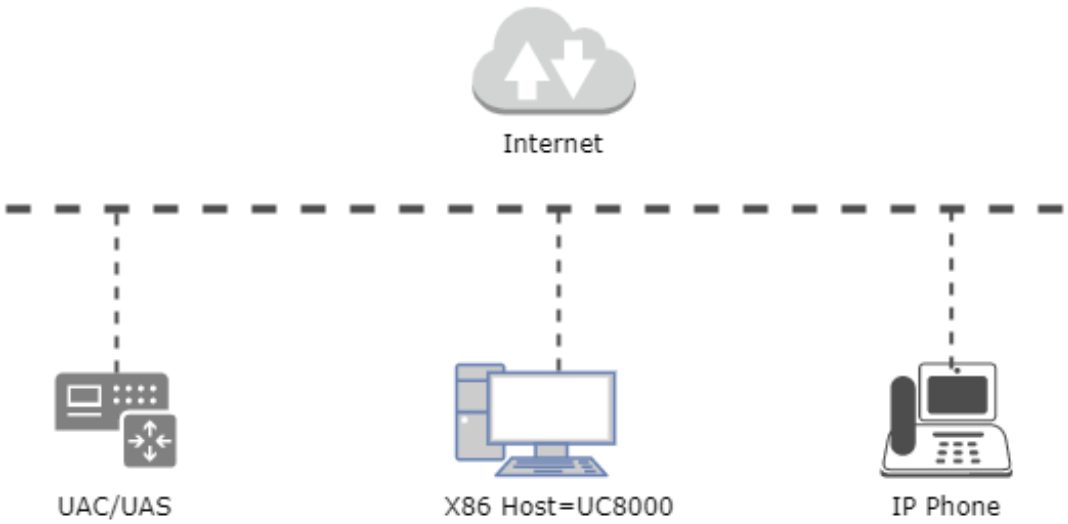
1.1 Overview

UC8000 is a new generation of large capacity unified communication system. It supports 20,000 extensions and 4,000 concurrent calls which are integrated voice, video, paging, conference, recording and other communication functions.

The UC8000 is deployed as a software based IPPBX when loaded on an X86 Architecture. It runs in a docker and can be loaded on Linux or Centos.

1.2 Application Scenario

The application scenario of UC8000 is shown as follow:



1.3 Product Form

Dinstar UC8000 is a software-based UC solution. It can be installed on any mainstream Linux based platform.

1.4 Features & Functions

1.4.1 Key Features

- Supports up to 20,000 extensions and 4,000 concurrent calls
- Support X86, ARM and Huawei KunPeng architecture, Docker, AWS, Google, Alibaba and Microsoft cloud
- Provide very flexible call routing based on time profile and number prefix
- Supports Multi-level IVR, helps to build personalized voice navigation for enterprise
- Flexible dialing plans with routing policies based on time, number, source, IP, etc.
- Supports PBX services such as call forward, three-way conference, voice conference, broadcast, intercom, etc.
- Support voicemail/voice recording
- User-friendly web interface, classification of web user's permission
- Support HA solution（active and standby), active and standby switch to ensure stable operation of UC8000

1.4.2 Physical Specifications

- CPU Processor: 2 or more cores
- RAM: 2G or more
- Hard disk: 100G or more
- Ethernet interface: a 100/1000 Base-T RJ45 or more

1.4.3 Voice Capabilities

- SIP over UDP/TCP/TLS, RTP/SRTP
- SDP, RTP/RTCP, SSL
- Voice Codec： G.711a/u, G.723, G.729, G.722, G.726, Opus
- Video Codec: VP8, H261, H263, H264, H263-1998, H263-2000
- Voice Activity Detection (VAD)
- Comfort Noise Generator (CNG)
- Adaptive Dynamic Buffer
- Adjustable Gain Control
- FAX: T.38 and Pass-through
- NAT: STUN/DDNS
- DTMF: RFC2833/Signal/Inband

1.4.4 PBX Call Service

- Call Forward (Always/No Answer/Busy)
- Call Waiting
- Call Holding
- Call Transfer
- Do-not-disturb
- Three-Way Conference
- Ring Group
- Call Queue
- Route Group

- Caller/Called Number Manipulation
- Routing Based on Time Period
- Routing Based on Caller/Called Prefixes
- Dialing Rule
- Failover Routing
- Multi-level IVR
- Auto-attendant Function
- CDRs
- Voicemail
- Voice Recording
- Up to 20,000 SIP Extensions
- Up to 4,000 Concurrent Calls
- Paging Group
- Event Report
- Email Client
- Voicemail to Email
- Broadcast/Broadcast Group
- Intercom/Intercom Group
- Emergency Number
- Blacklist/Whitelist
- Feature Code
- SCA
- Follow Me

- Alarm Clock
- Surpport Dinlink (APP)
- Attendant Console
- PMS (Property Management System)

1.4.5 Network Features

- Multi-service network port
- IPv4/IPv6
- VALN, QoS, NAT and Fail2ban
- Double-device Hot Standby

1.4.6 Maintenance

- Web GUI Configuration
- Command line management configuration
- Configuration Restore/Backup
- Multi-language support
- HTTP/TFTP firmware upgrade
- Web password change
- Ping, Traceroute and Nslookup Test
- Network Capture
- Network Quality Test
- Call List query and export
- API interface support

2 Installation

2.1 Attentions before installing

- Users install the X86 host on their computer, then install a mainstream Linux or Centos system on the X86 host, and install the programs required for the UC8000, etc.
- To reduce the interference to telephone calls, please separate power cables from telephone lines;
- To guarantee stable running of the UC8000 IPPBX, please make sure that there is enough network bandwidth;
- The software license of UC8000 will be changed after reinstalling the docker.

2.2 Power on the System

- Plugging the host into a monitor, keyboard, mouse, etc.
- Connect the host computer's Ethernet port to the network.
- Refer to UC8000 Installation Tutorial to install the operating system and UC8000 tar file.

2.3 Network Connection

The UC8000 uses the IP address of the server's physical machine network interface, so the UC8000 does not require network configuration; when users need to use the active and standby functions, they can configure a floating IP address for active and standby management.

2.4 Connecting devices to the network

2.4.1 Preparations for Login

Connect the network interface of the host with UC8000 to Ethernet or LAN, and configure the accessible IP address in advance, e.g. 192.168.11.1.

- Take the Windows 7 operating system as an example, set the local computer to the same network segment address as the default IP of the system.

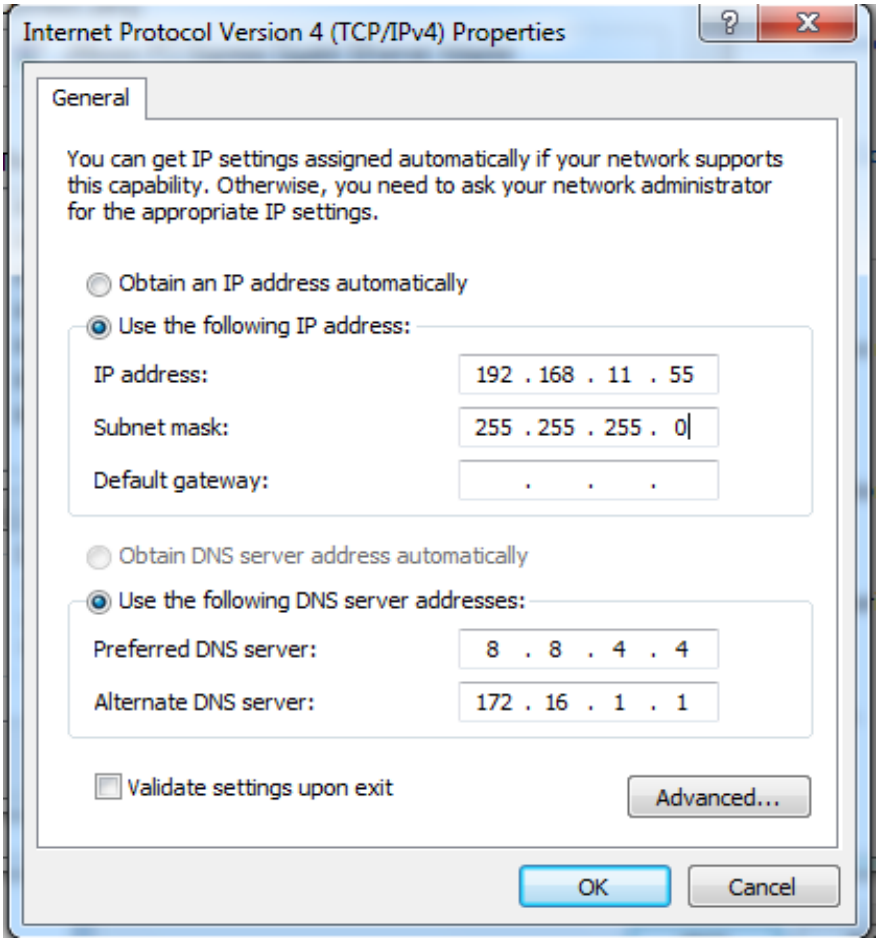


Figure 2.4.1 Modify the local computer address.

- Check the connectivity between the PC and the UC8000. Click Start -> Run of PC and enter cmd to execute ‘ping 192.168.11.1’ to check whether the IP address of the system is normal.

2.4.2 Log In Web

Enter the IP address of UC8000 in your browser, such as: "192.168.11.1", and press Login to enter the user login interface.

Figure 2.4.2-1 Login GUI of UC8000

Enter username and password (default is admin/admin@123#), and click Login to enter the web interface.

For the security purpose, login of the web will be limited:

- If three consecutive login failures, users need to slide to validate their user account;
- If ten consecutive login failures, the IP address of the UC8000 will be blacklisted, and users need to reset a new IP address for the system;
- Successful login or system restart will wipe out login failure records.

3 Basic Operation

There are a variety of basic call service operations in an IPPBX. The following examples describe only a few basic telephone operations. For a complete description of the IPPBX calling features, please refer to other sections of this document for instructions.

3.1 Phone call operation

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

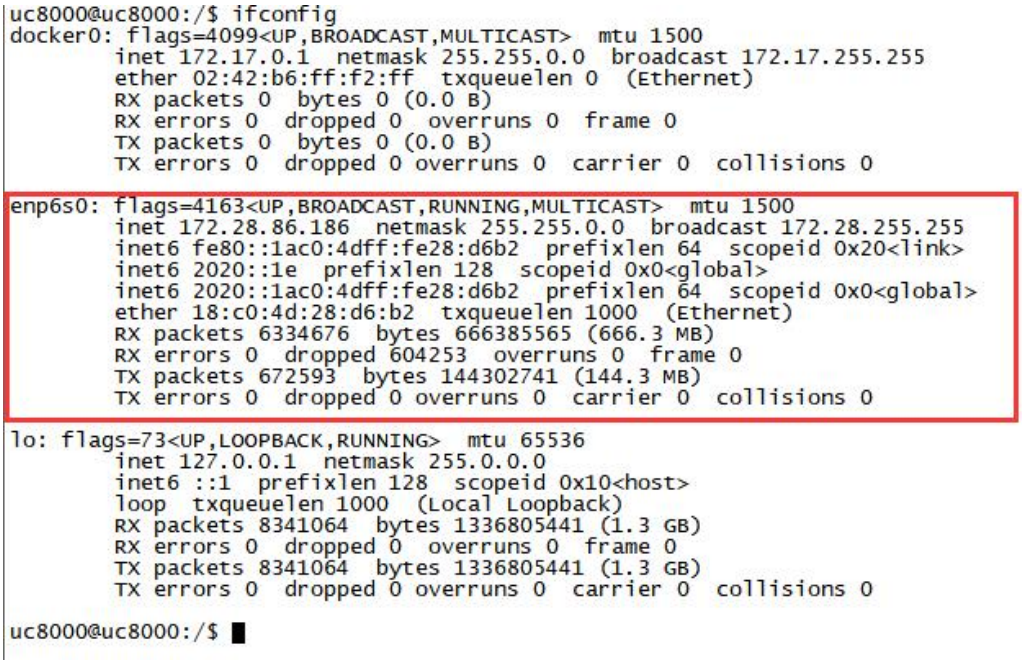
The current call can be held by pressing the "flash" key on the phone (if available), and then pressing the "flash" key again to resume the held call. If there is no "flash" key, you can use "hook flash" instead.

3.3 Call Waiting

When call waiting is enabled, if you hear the call waiting voice during a call, it indicates that a new call is incoming. You can switch between the incoming call and the current call through the Flash key or hook flash.

3.4 Query IP Address

Since UC8000 uses the IP address of the host system, and UC8000 cannot change this IP address, users can only check the IP address by logging into the host system, as shown in the following figure.

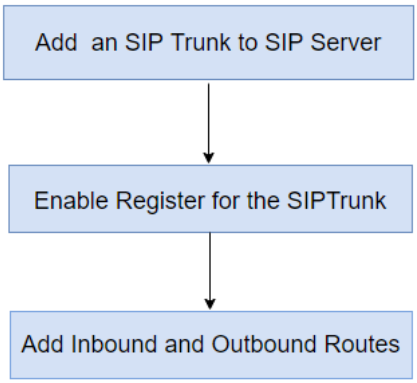


4 Configuration Wizard

The following are the common ways to configure the UC8000 IPPBX.

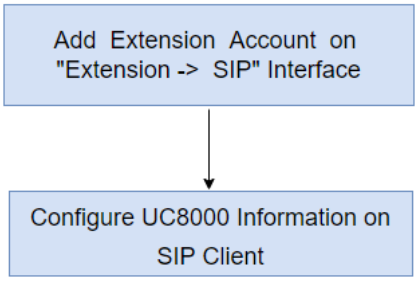
1. Register to the server as a terminal

The UC8000 is registered to the server as a terminal.

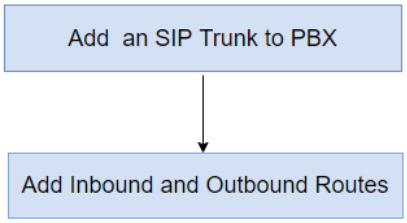


2. Other SIP Clients registered at UC8000

Under this mode, UC8000 is regarded as an SIP Server. Create an extension account first on the **Extension -> SIP** interface, and configure the listening port on the **Profile -> SIP** interface. Then configure the server and account on SIP client.



3. UC8000 Connected to PBX through SIP Trunk


Add an SIP Trunk to PBX
Add Inbound and Outbound Routes

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 Uc8000 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-Status

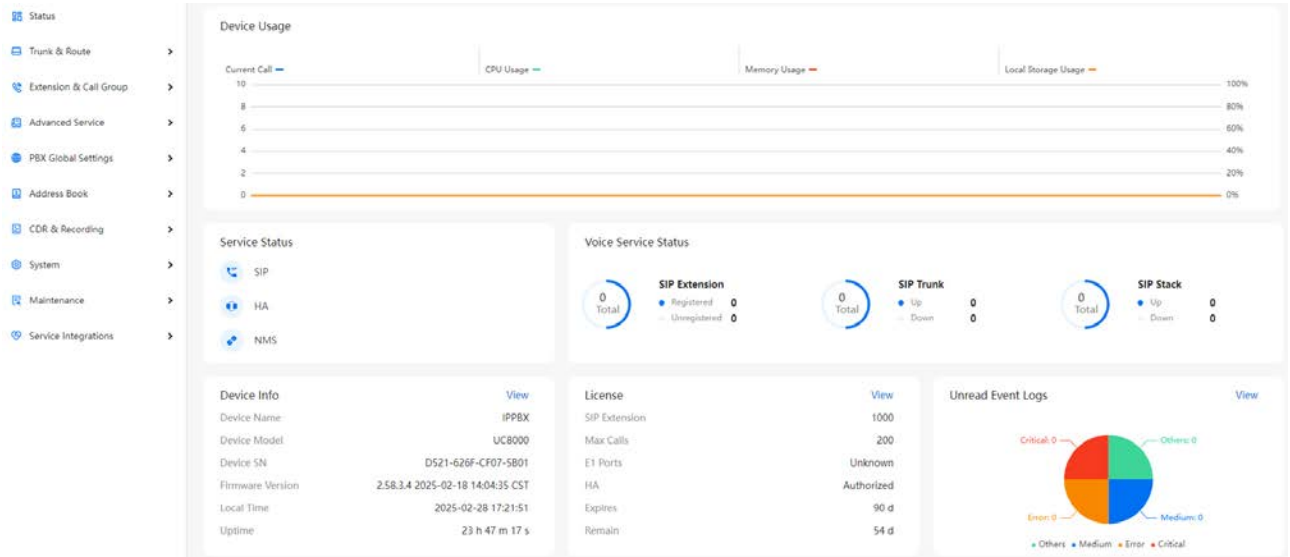


Table-Status

Index	Item	Description
1	Menu Bar	The menu bar of uc8000 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	Port Status	The status of 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, 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 uc8000 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

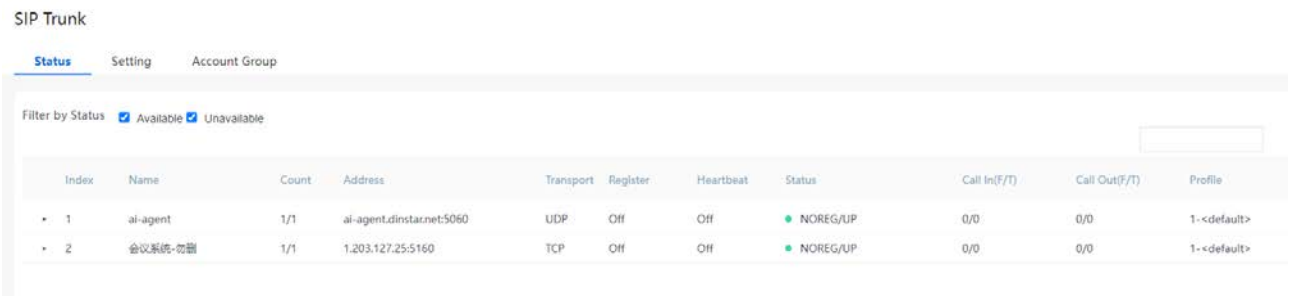


Figure-Parameters of SIP Trunk

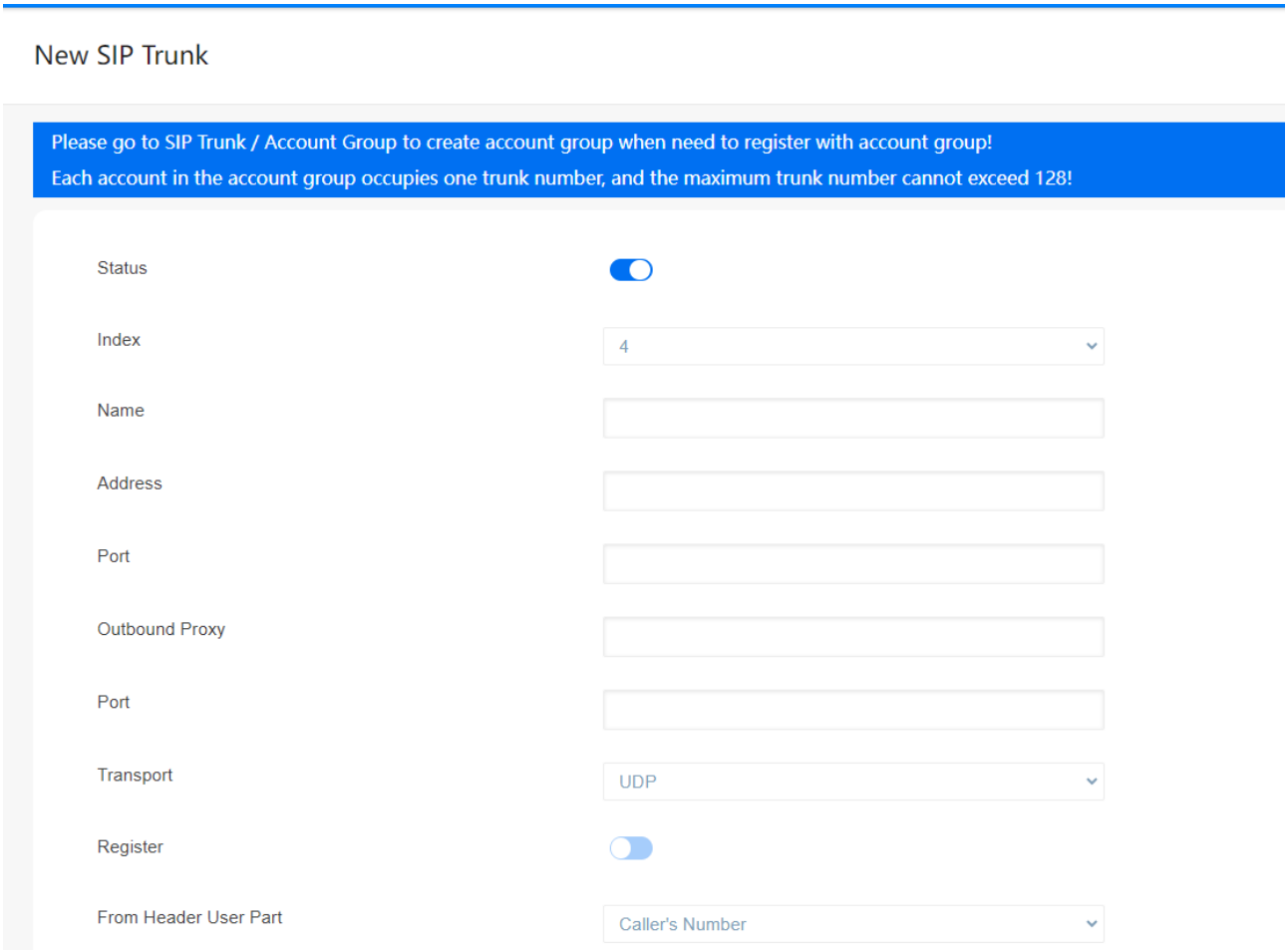


Table-Parameters of SIP Trunk

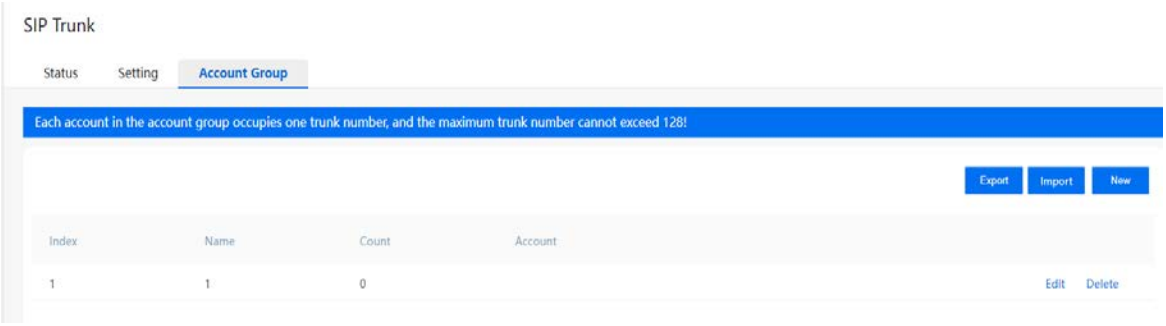
Parameter	Description
Status	Enable or disable SIP Trunk.
Index	Index of SIP Trunk. Range from 1 to 32.
Name	The name of the SIP trunk. The input value is text type and cannot be null. The value is up to 32 characters and cannot contain " ".
Address	The IP address or domain name of the peer SIP devices 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 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



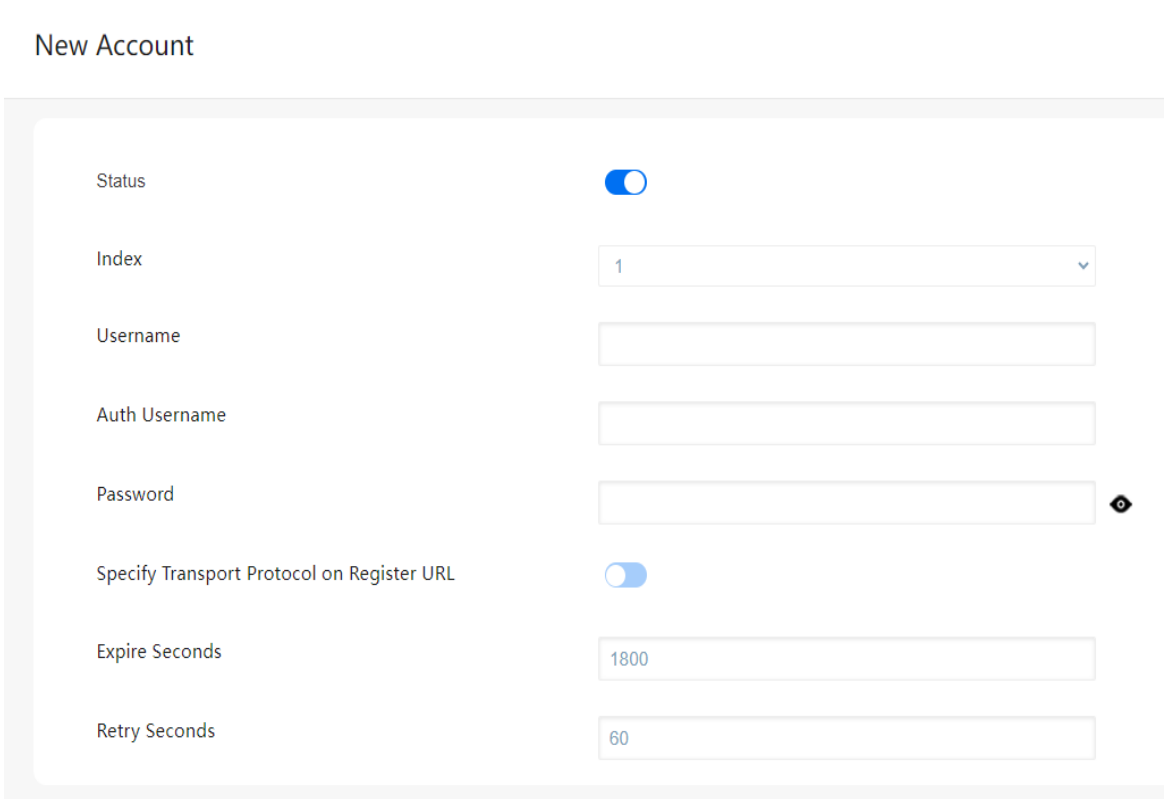


Table-Parameters of Account Group

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 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

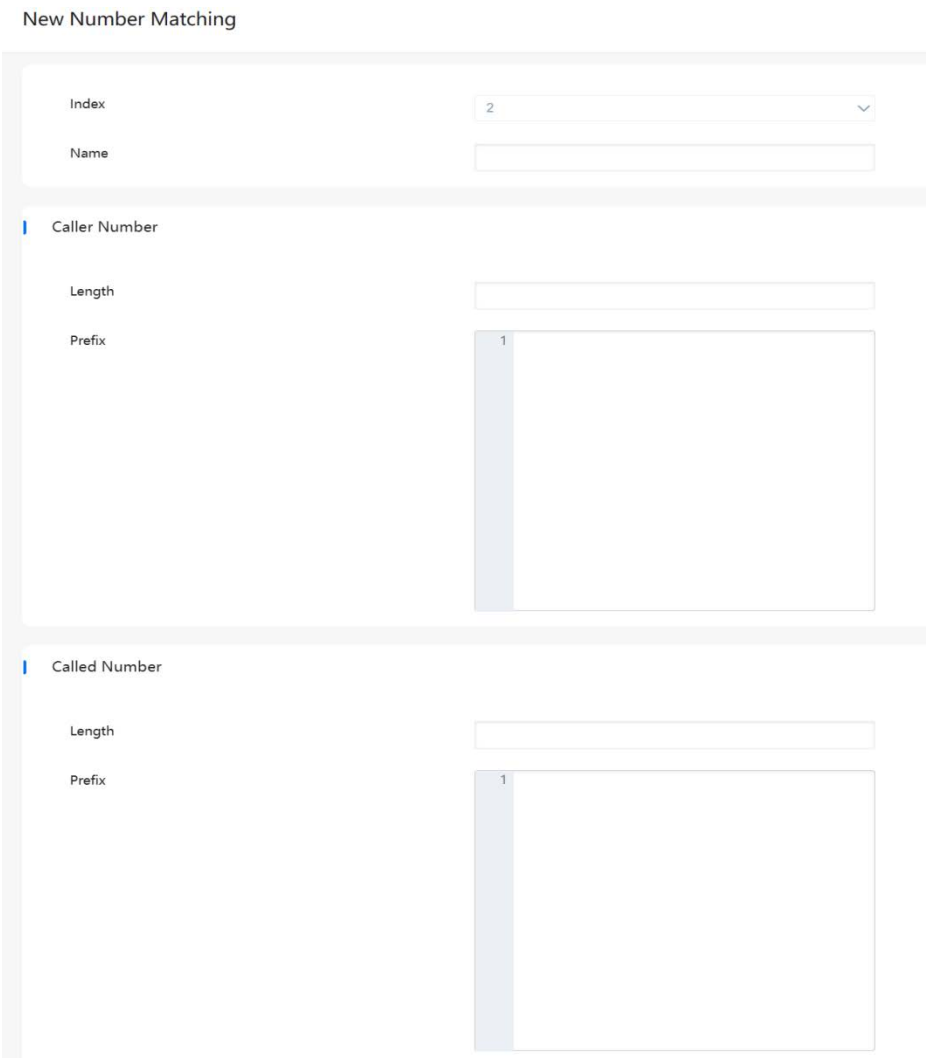


Table-Parameters of Number Matching

Parameter	Description
Index	The index of number matching rule. Range from 1 to 32.
Name	The name of the number profile.
Length	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.

Prefix of Caller Number	The prefix of the calling number. It supports multiple prefixes, multiple rules for "or" relationships. It supports regular expression.
Prefix of Called Number	The prefix of the called number. It supports regular expression. It Supports multiple prefixes, multiple rules for "or" relationships.

Regex(Regular Expression)Syntax

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

5.2.3 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

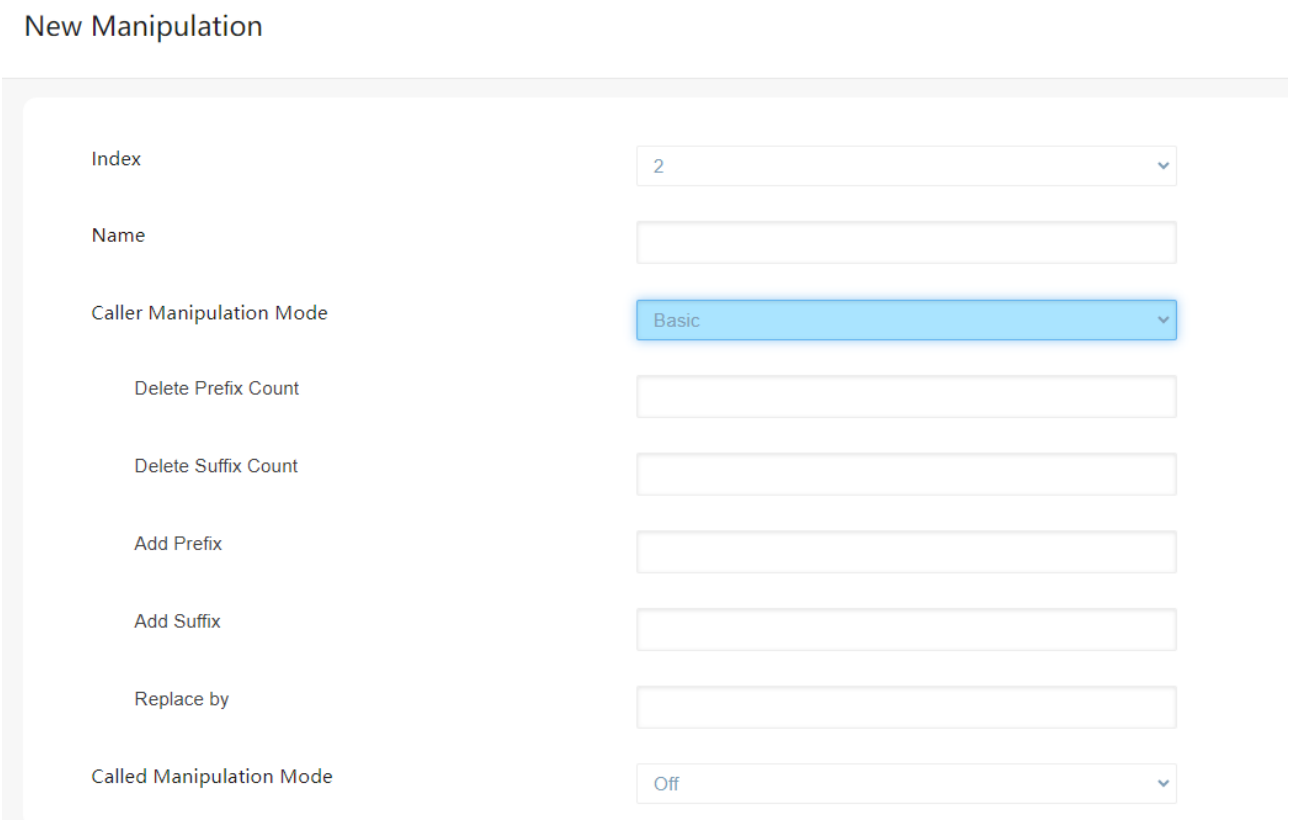


Table-Parameters of Manipulation

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

5.2.4 Route

This section is to configure routes or route groups for incoming and outgoing calls through uc8000 IPPBX.

Route

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

Figure-Parameters of Route

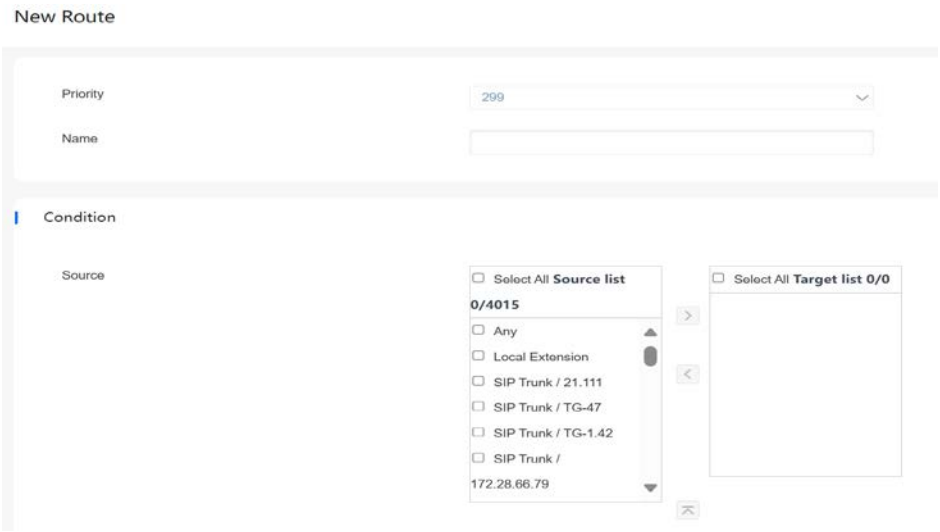


Table-Parameters of Route

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 SIP extension, a customized source or any.
Number Profile	The profile of the caller number and the called number. Please make reference to the **Trunk & Route> Number Matching** section. The default value is 'Off'. **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 extension 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

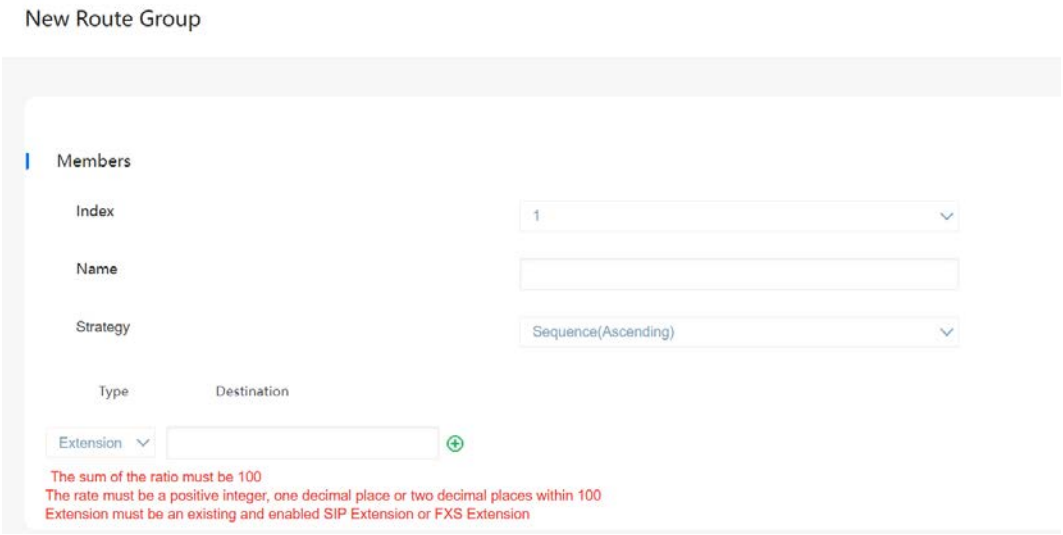


Table-Parameters of Route Group

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

Members	Select SIP extension, SIP 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.

5.2.5 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

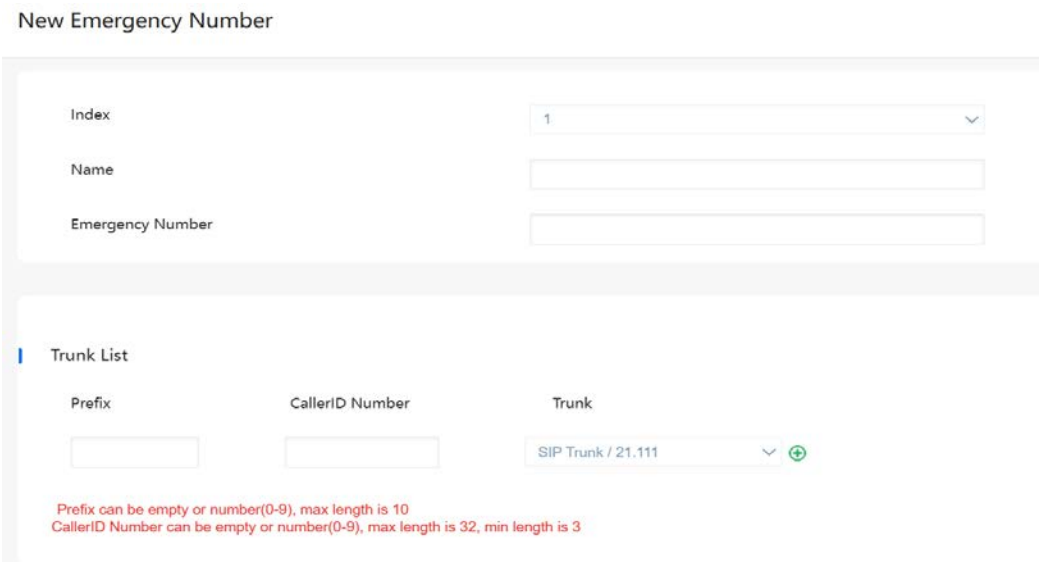


Table-Parameters of Emergency Number

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

5.2.6 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

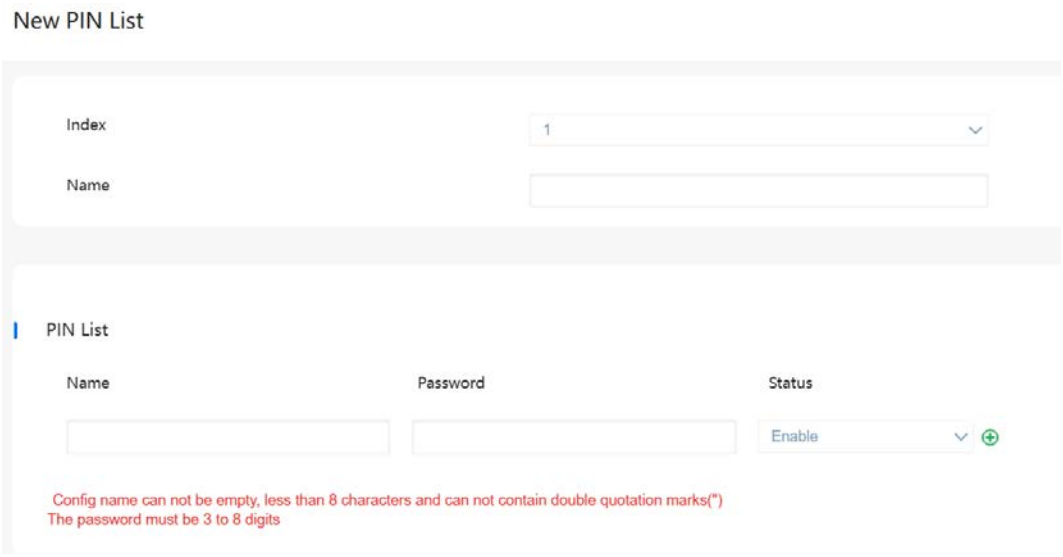


Table-Parameters of PIN List

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.

5.2.7 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

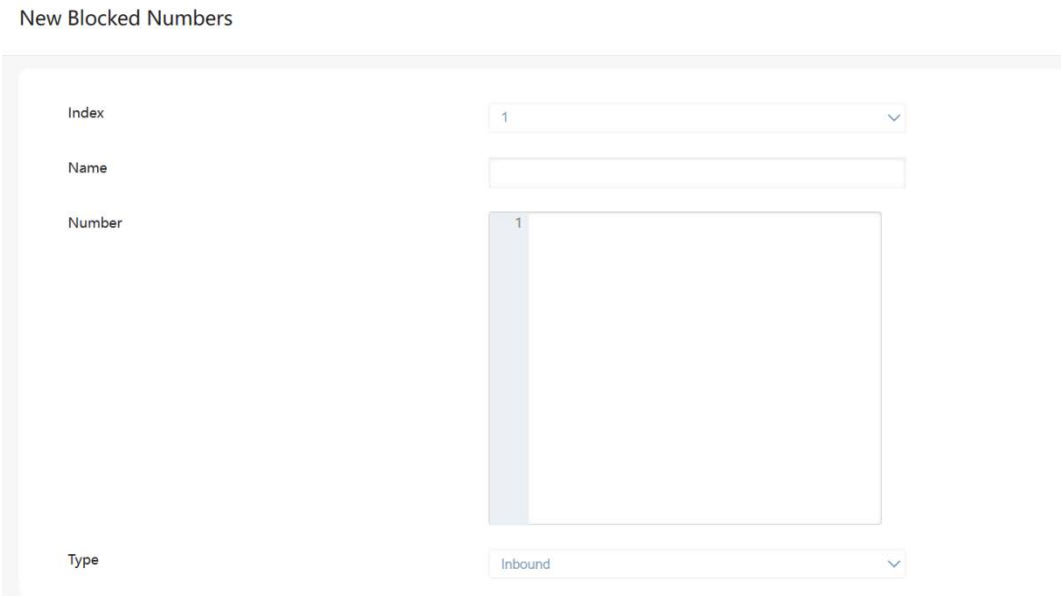


Table-Parameters of Blocked Numbers

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

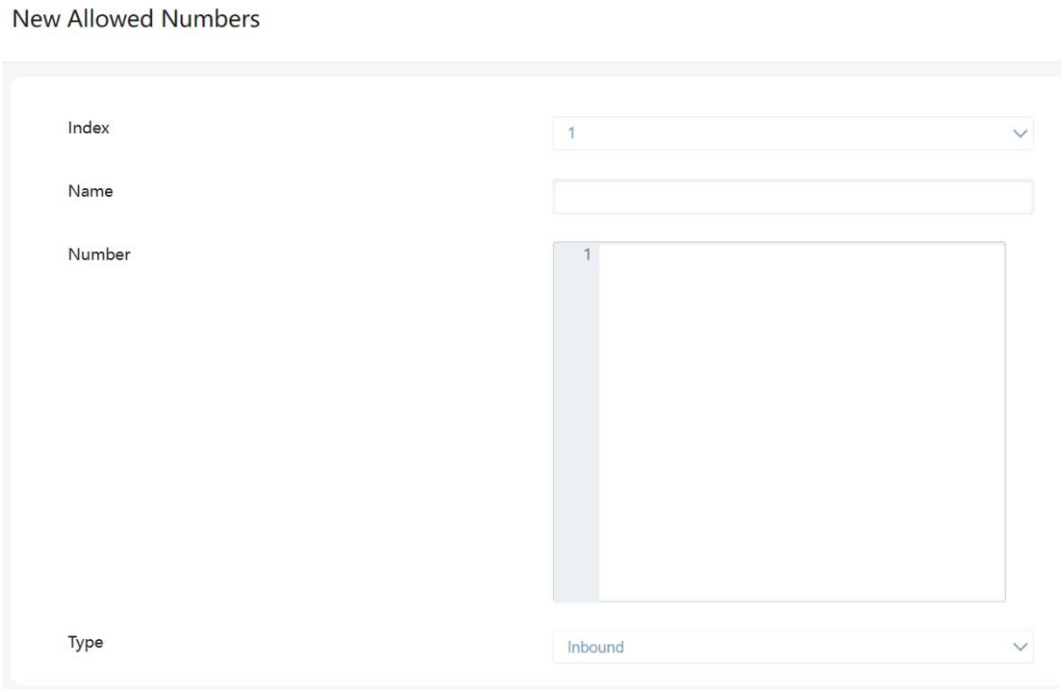


Table-Parameters of Allowed Numbers

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.

5.2.8 AutoCLIP

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

Figure-Parameters of AutoCLIP

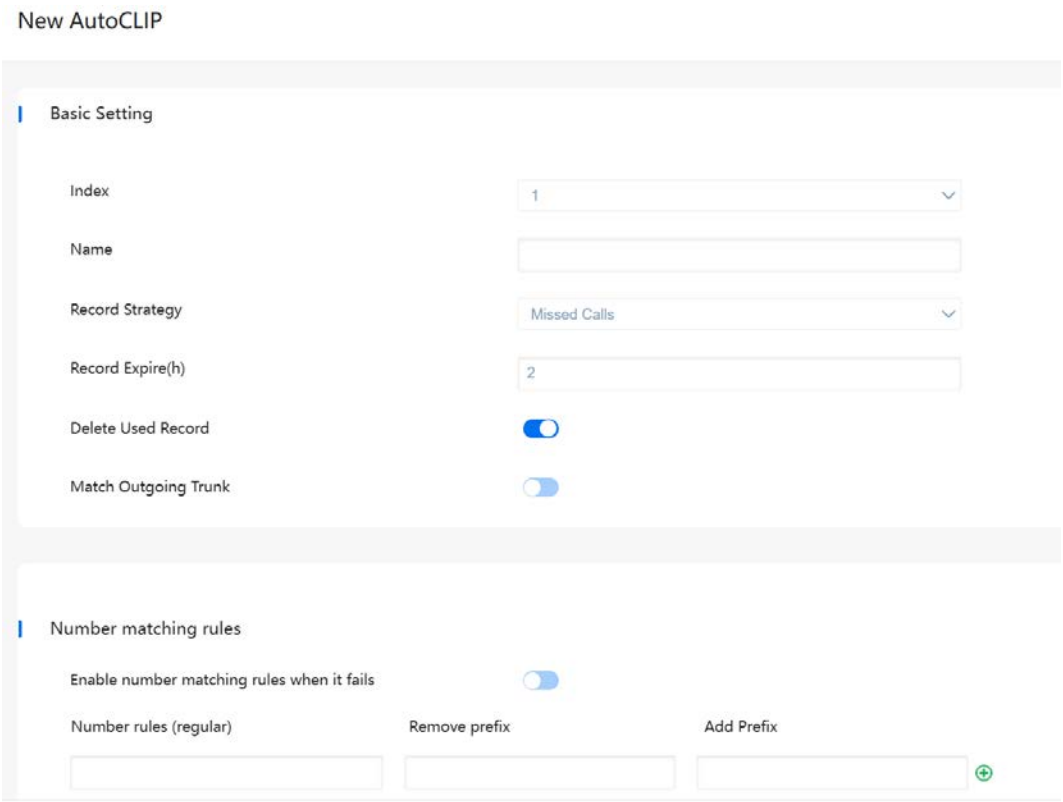


Table-Parameters of AutoCLIP

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

5.2.9 SMS Route

uc8000 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

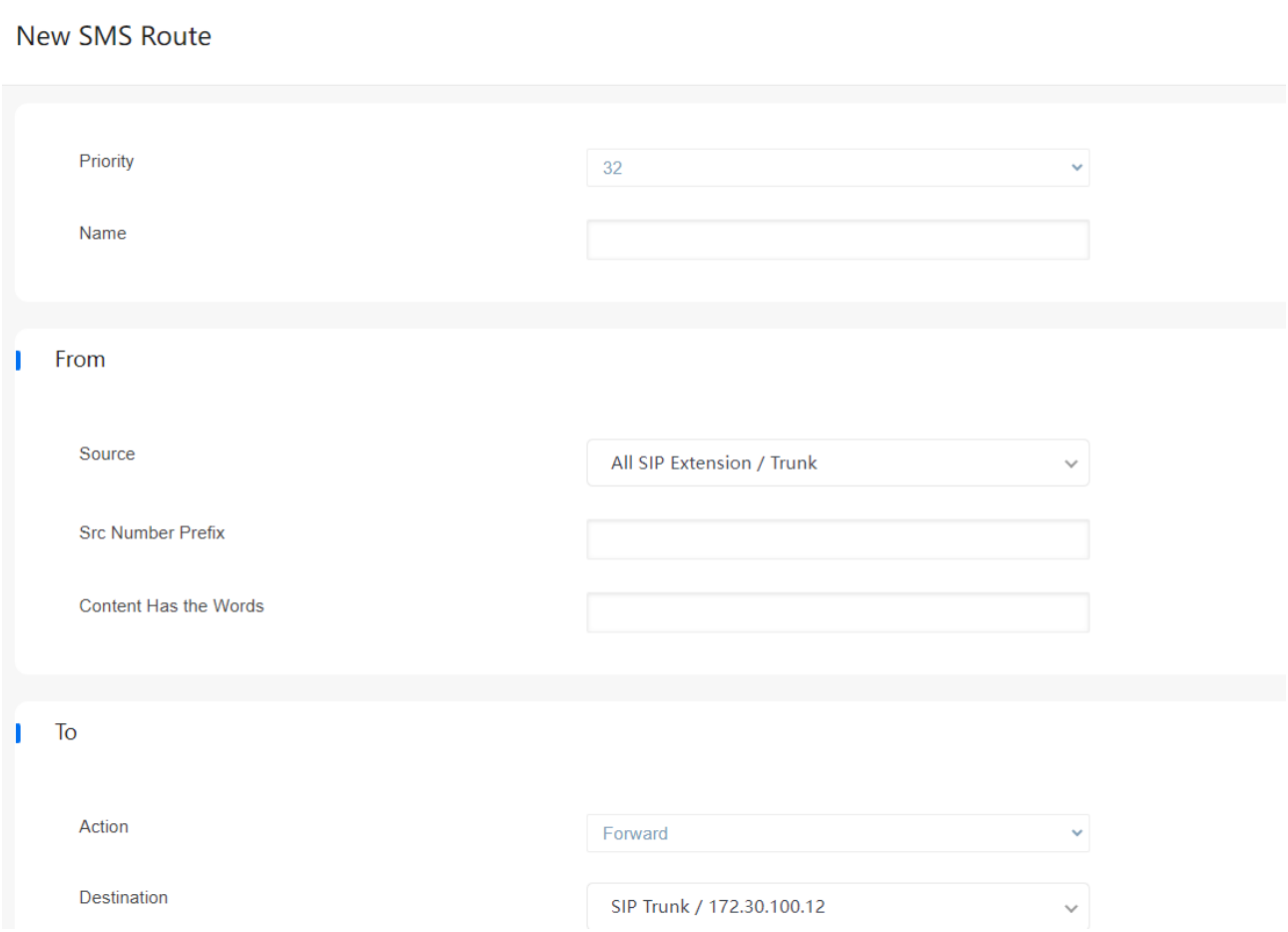


Table-Parameters of SMS Route

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 uc8000 IPPBX by SIP clients (hereby uc8000 IPPBX is regarded and act as a SIP server).

Figure-Status of SIP Extension

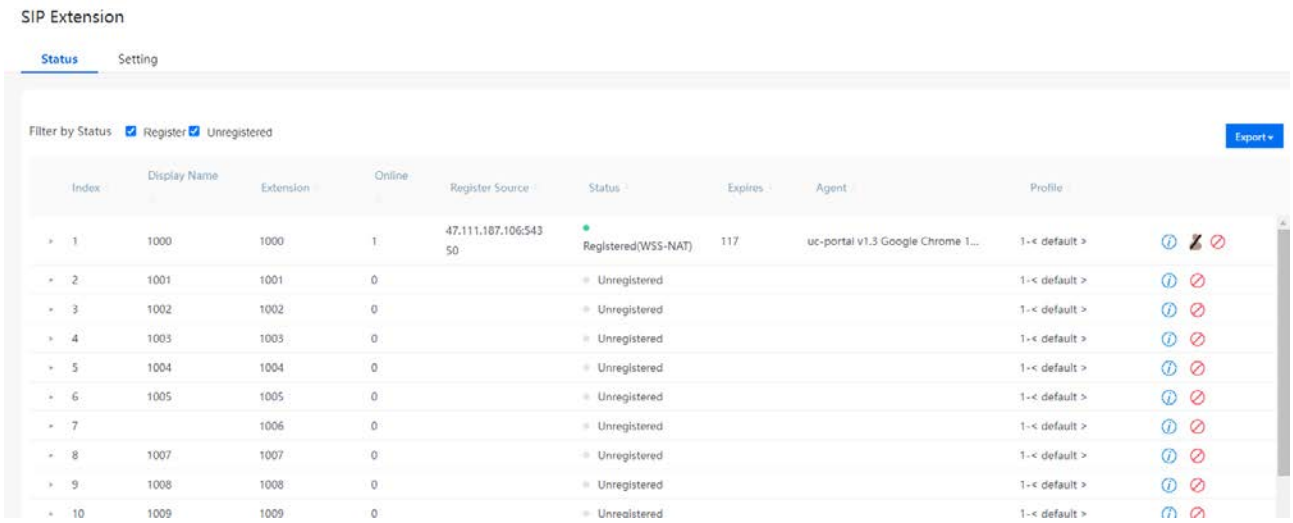


Figure-Parameters of SIP Extension configuration

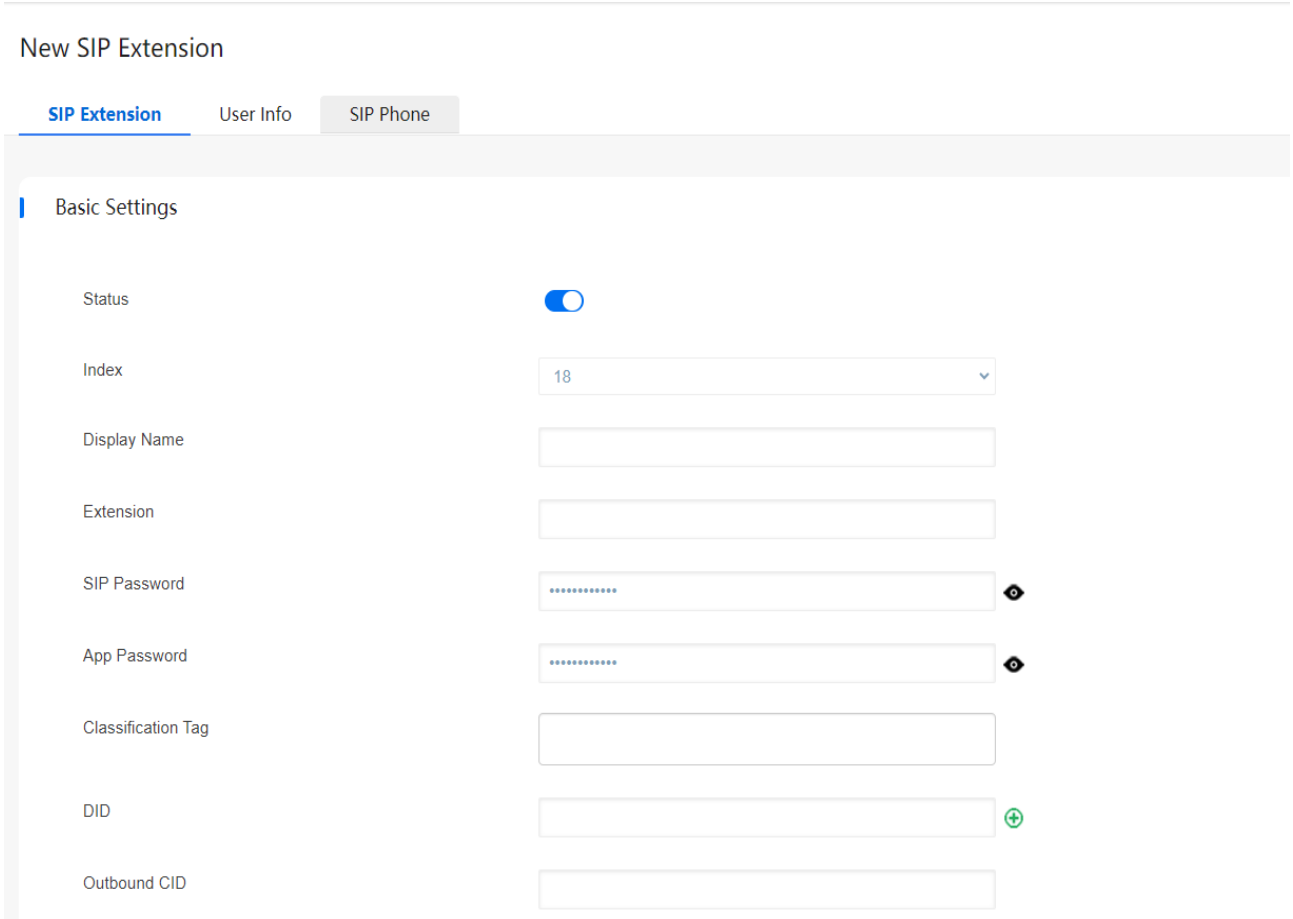


Table-Parameters of New SIP Extension

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 Uc8000 IPPBX by a SIP client.
Password	The password of the SIP account registered in Uc8000 IPPBX by 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.

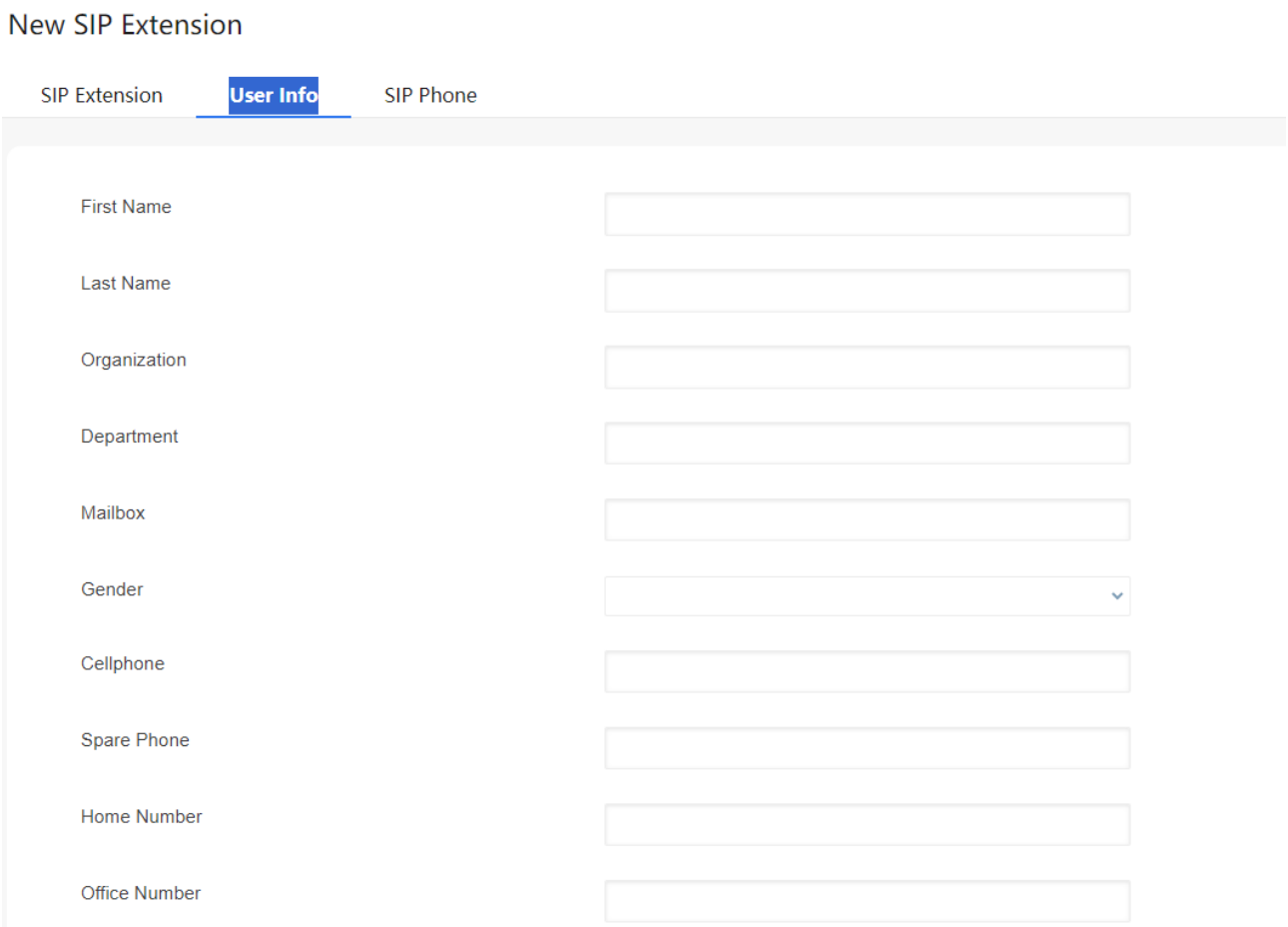
Call Forward No Reply	If 'Call Forward No Reply' feature is enabled, calls will be forwarded when nobody answer the calls during a specified period. **Designated Forward No Reply**: If value is empty or null, busy call forwarding will be activated for all incoming numbers, no reply call forwarding will 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.

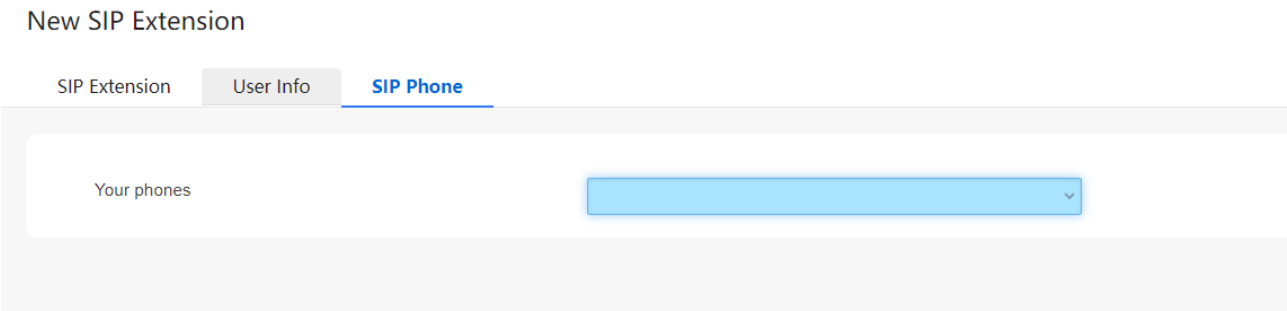
Max Call Duration(s)	Limit the duration of each phone call, unit: seconds
Original Called Number Location(Send INVITE)	When sending INVITE, configure the location of the original called number.
NAT	If NAT is enabled, the IP address of SIP extension in LAN will be bound into the outbound IP address of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



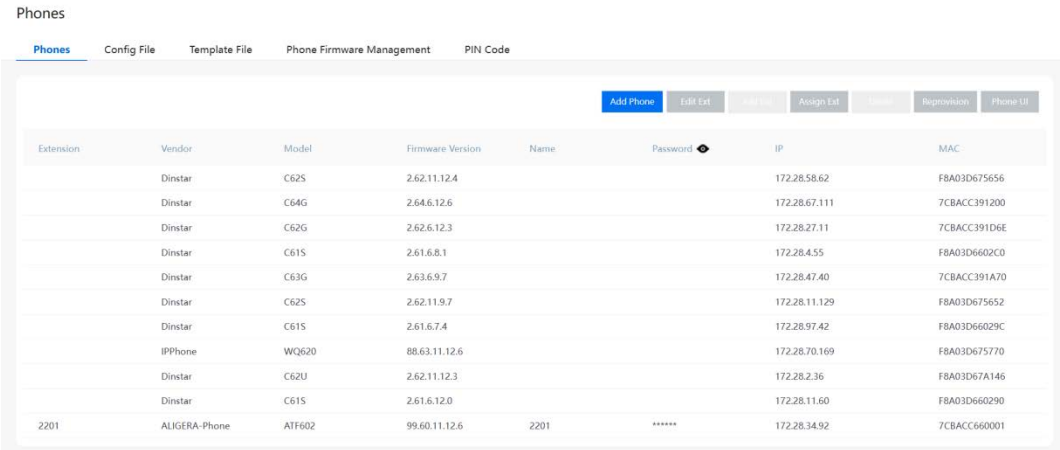
SIP phones are used to send configurations to IP phones

5.3.2 Phones

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

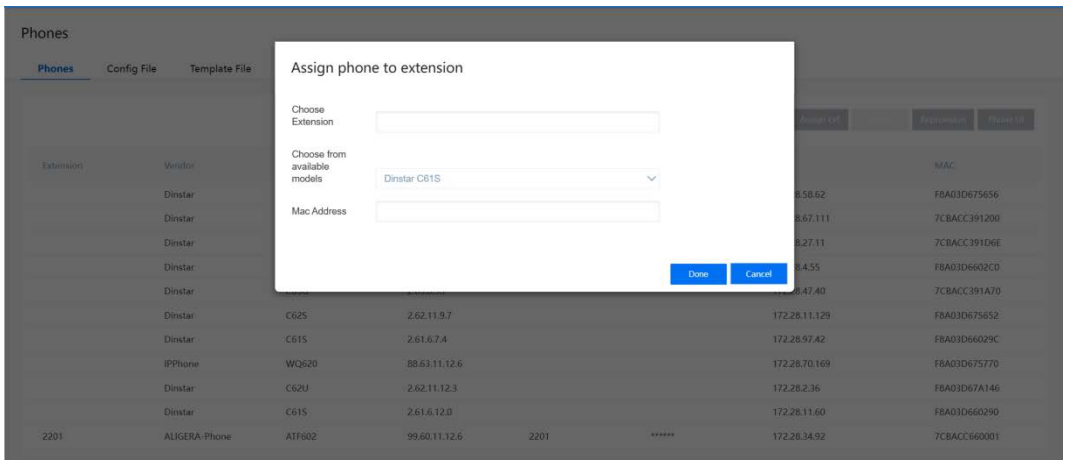
After enabling PNP, the phone will periodically send a Subscribe message to the multicast address. If the PBX receives the multicast message, it will list the phone models in the PBX configuration list.

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.

5.3.3 Ring Group

On the **Extension & Call Group > Ring Group** interface, users can group SIP extension(s) together and set strategy for choosing the 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

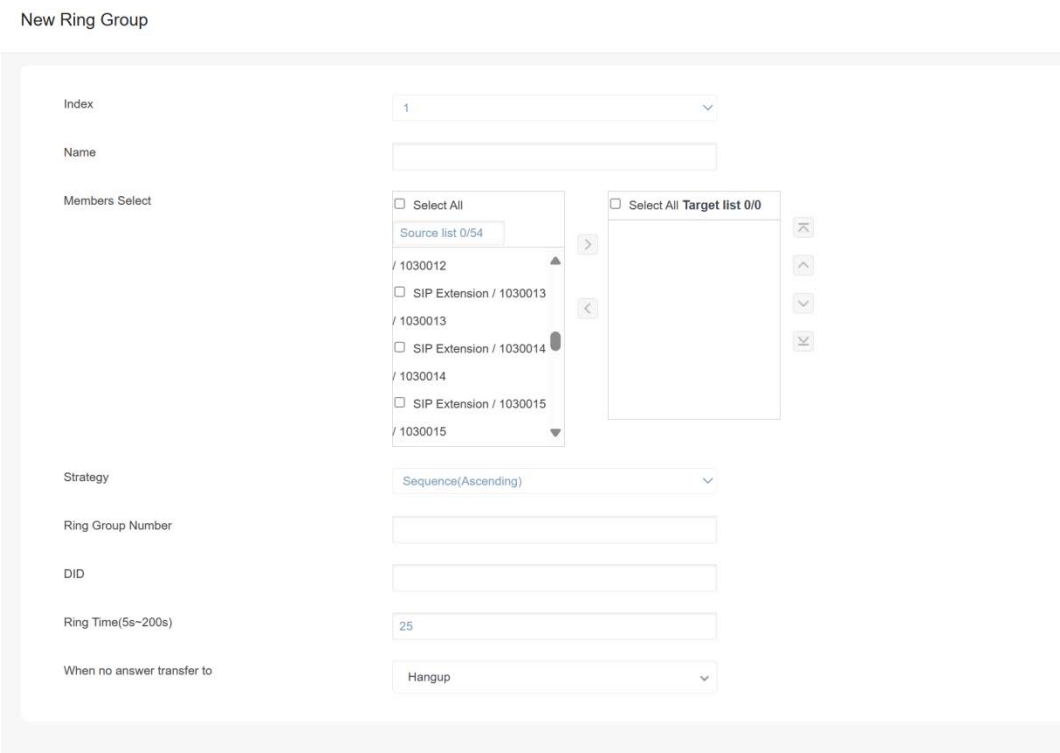


Table-Parameters of Ring Group

Parameter	Description
Index	The index of Ring Group.
Name	The name of this ring group.
Members Select	Select the 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.

5.3.4 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 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

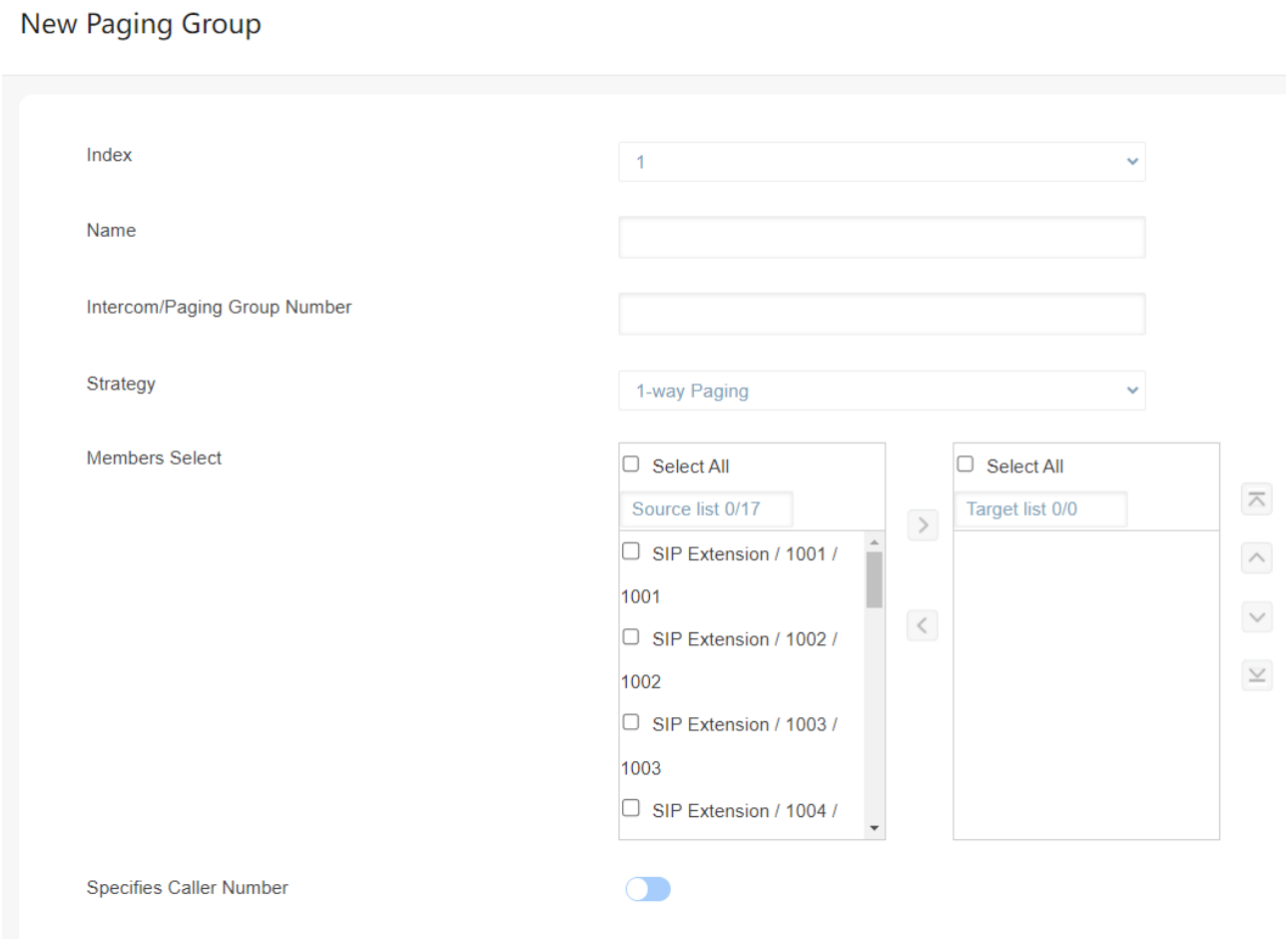


Table-Parameters of Paging Group

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 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

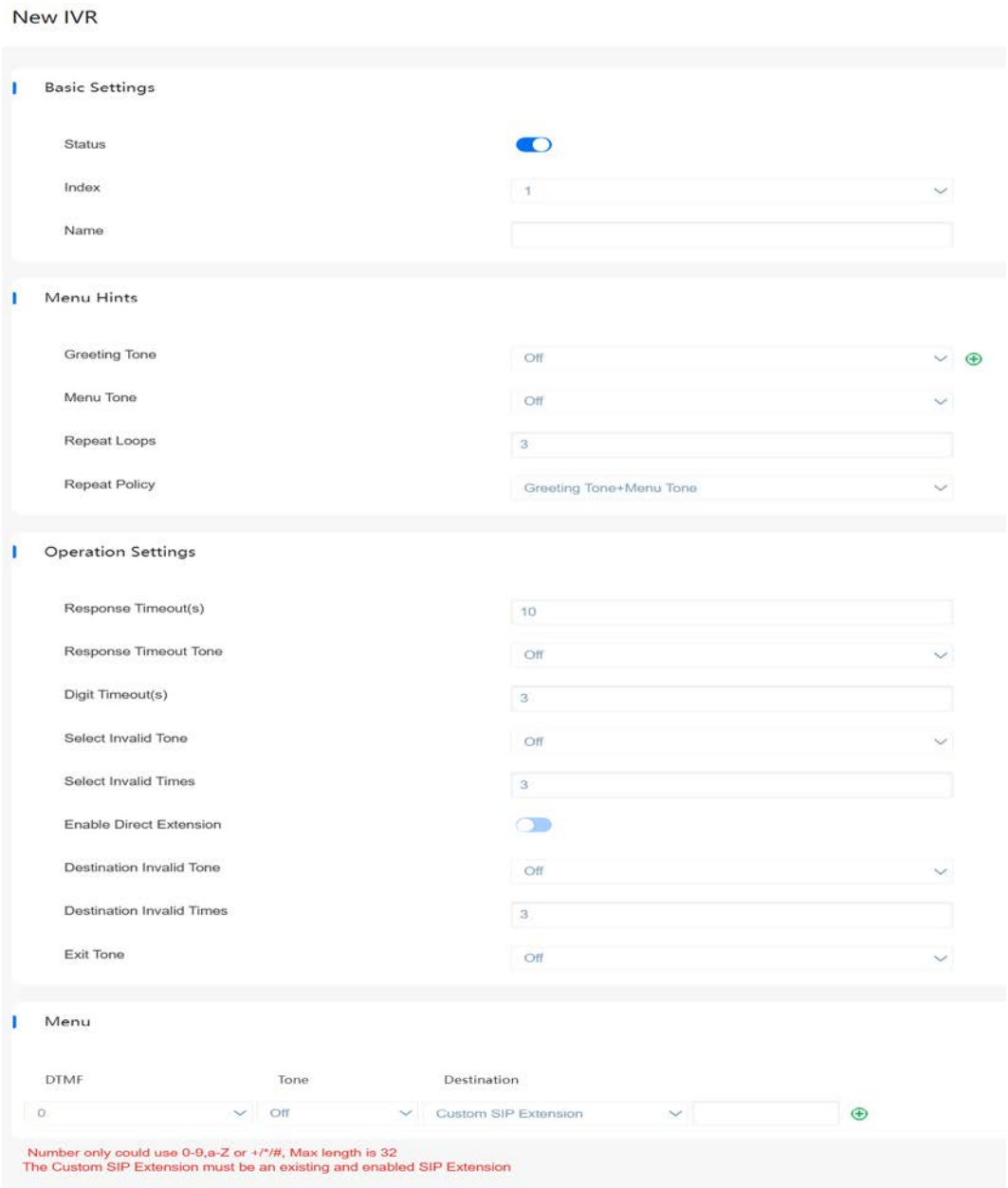


Table-Parameters of IVR

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 dials 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-Call Queue

Figure-Parameters of Call Queue

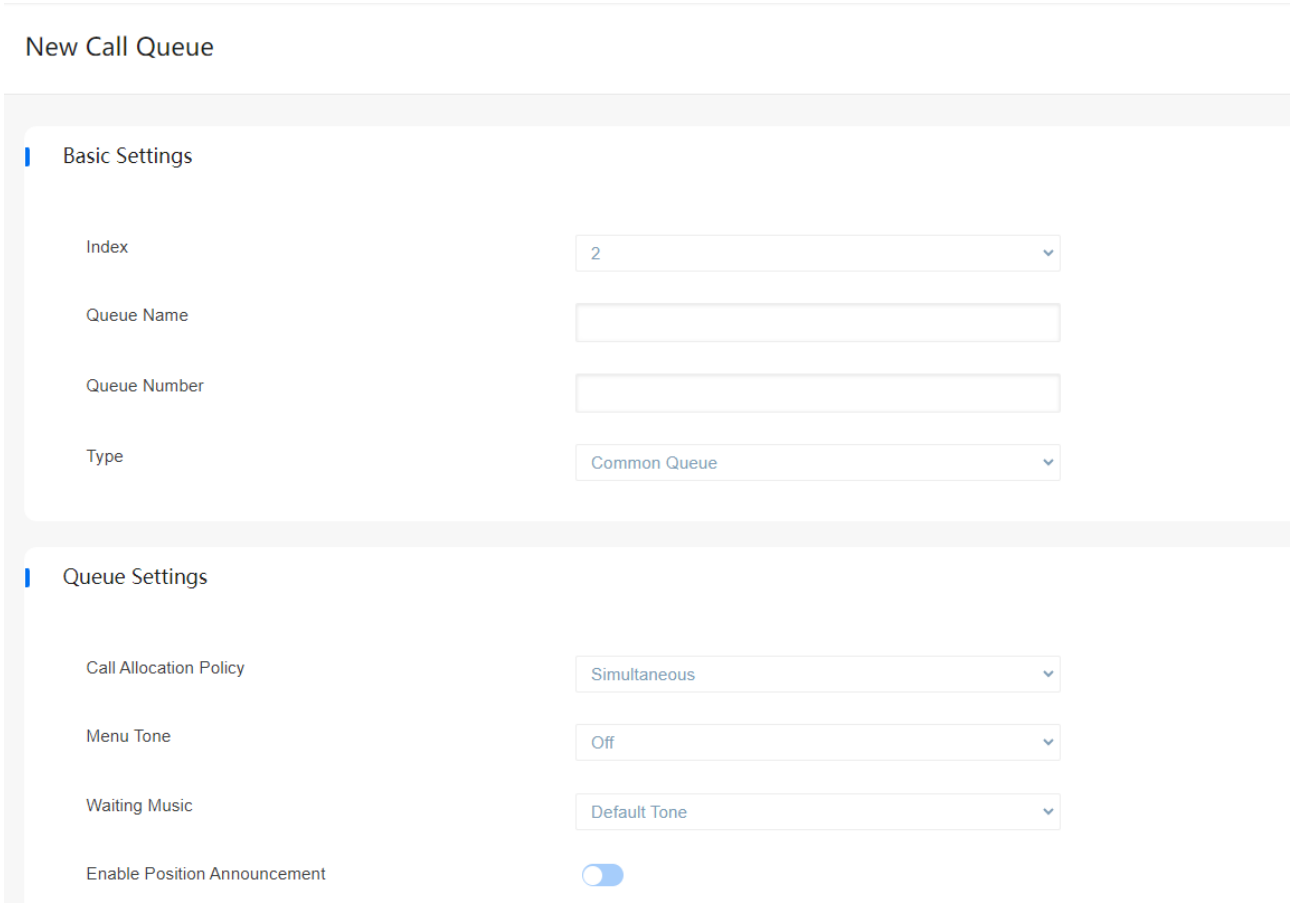


Table-Parameters of Call Queue

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 SIP extension, move left to add a member, move right to delete a member Attendant console queue

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, 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

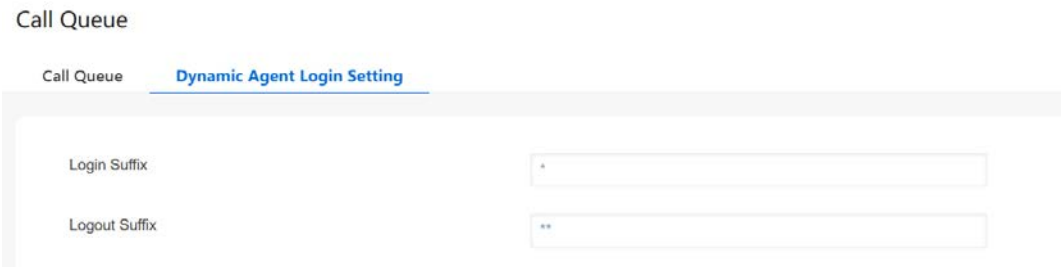


Table-Parameters of Dynamic Agent Login Setting

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

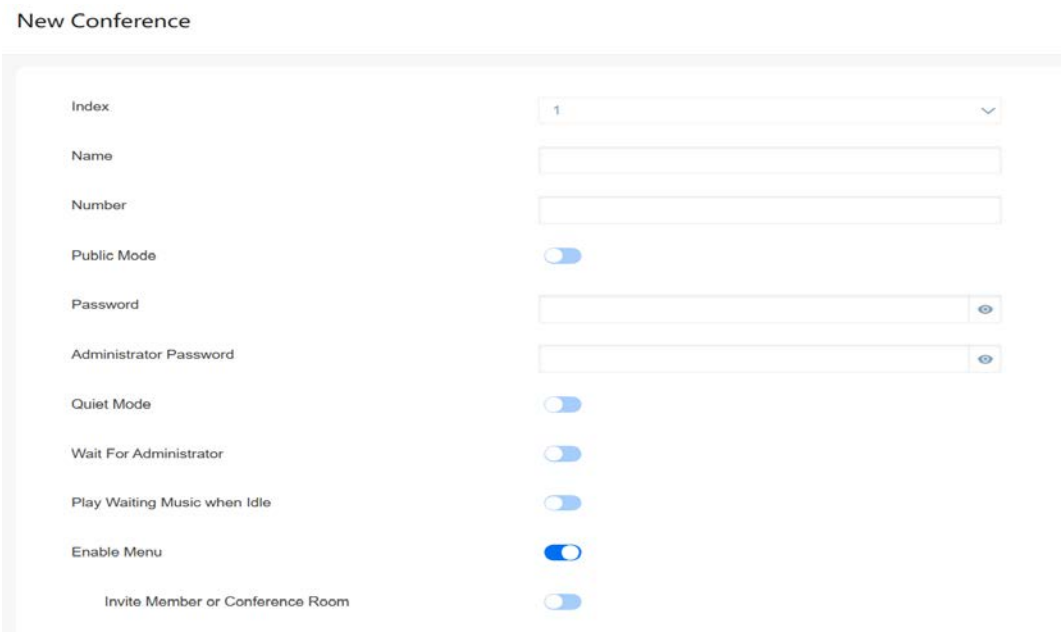


Table-Parameters of Conference

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

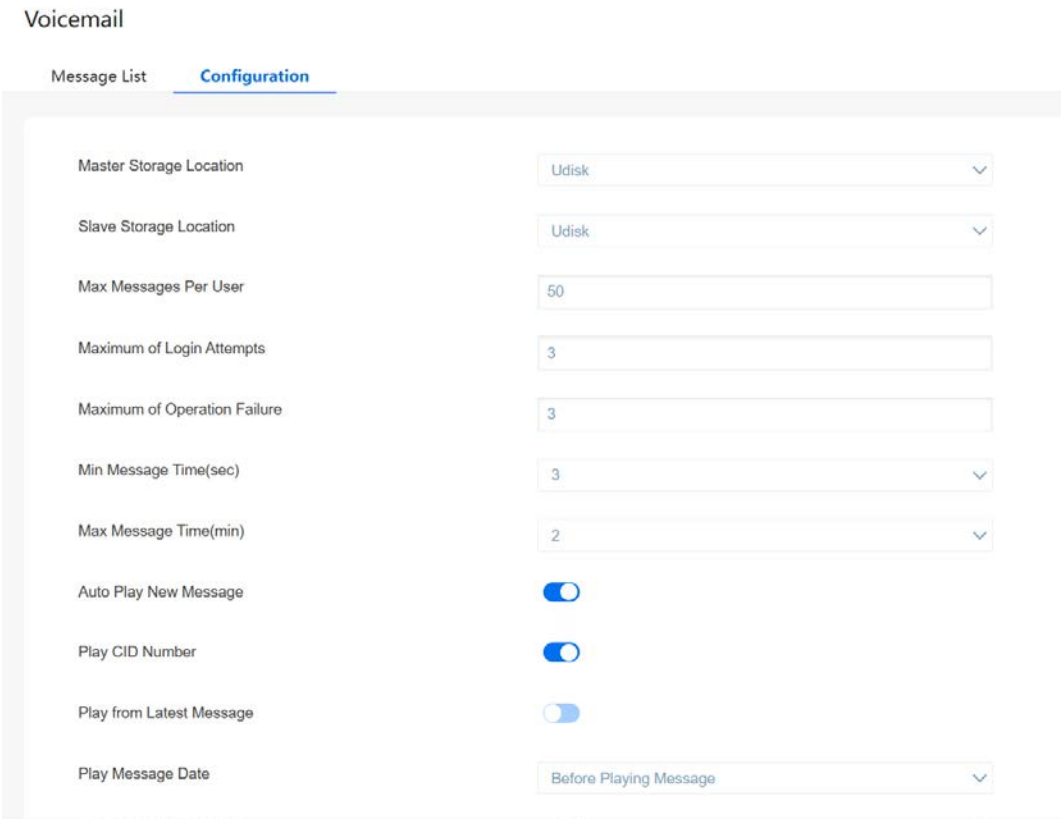


Table-Parameters of Voicemail Configuration

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

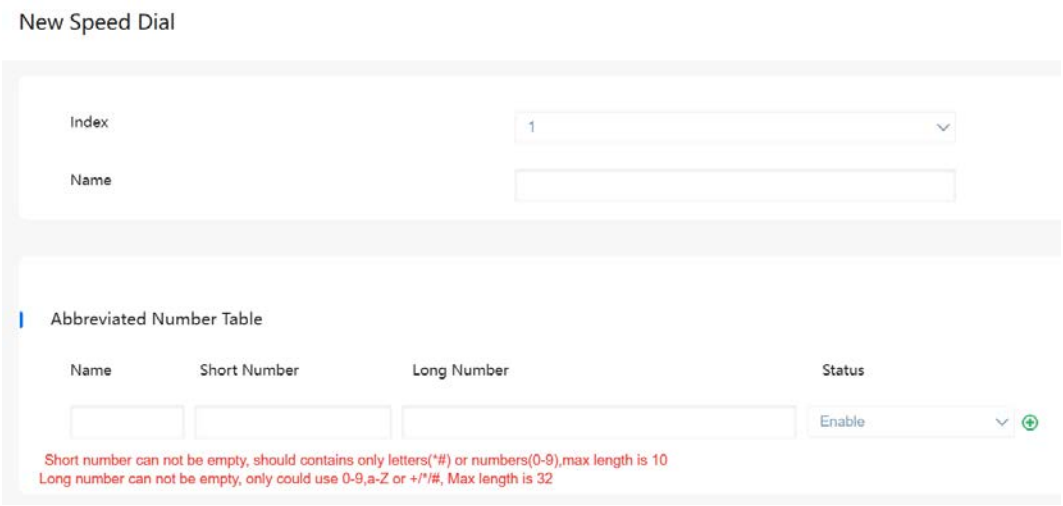


Table-Parameters of Speed Dial

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

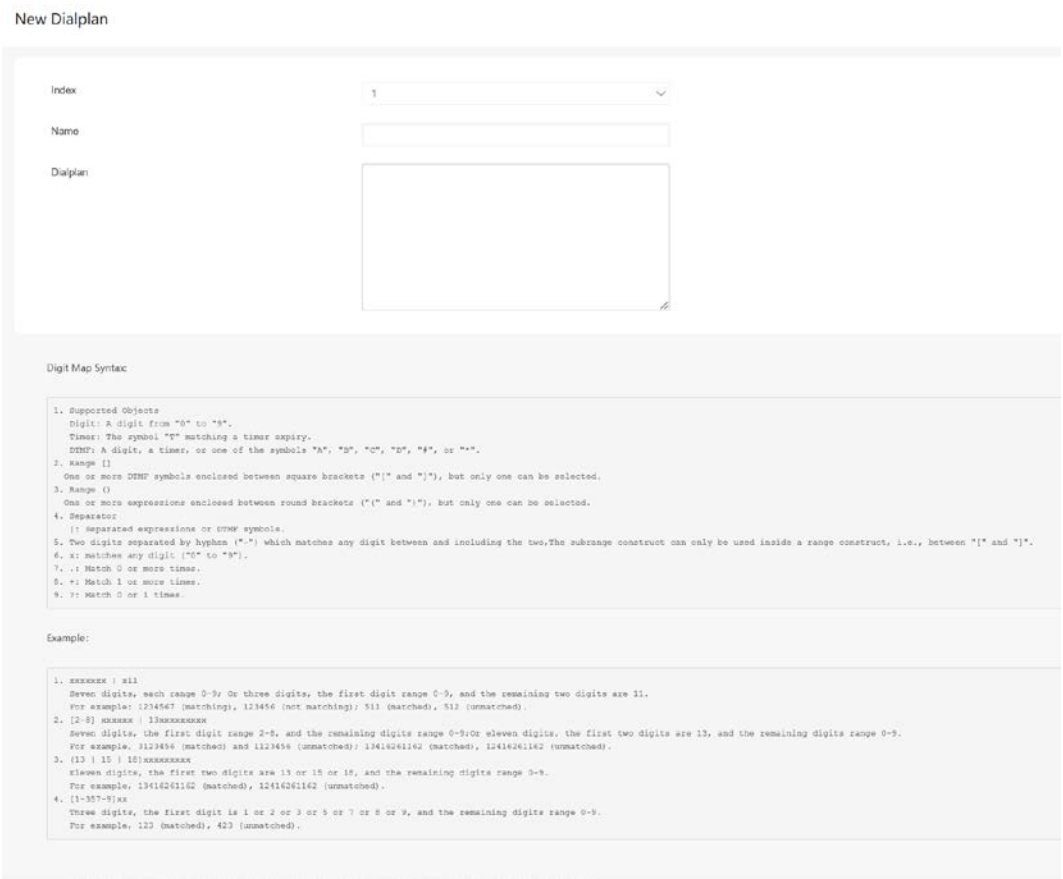


Table-Parameters of Dial plan

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

Table-Regex (Regular Expression) Syntax

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

Modifiers	?	Matches 0 or 1 times the preceding element

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

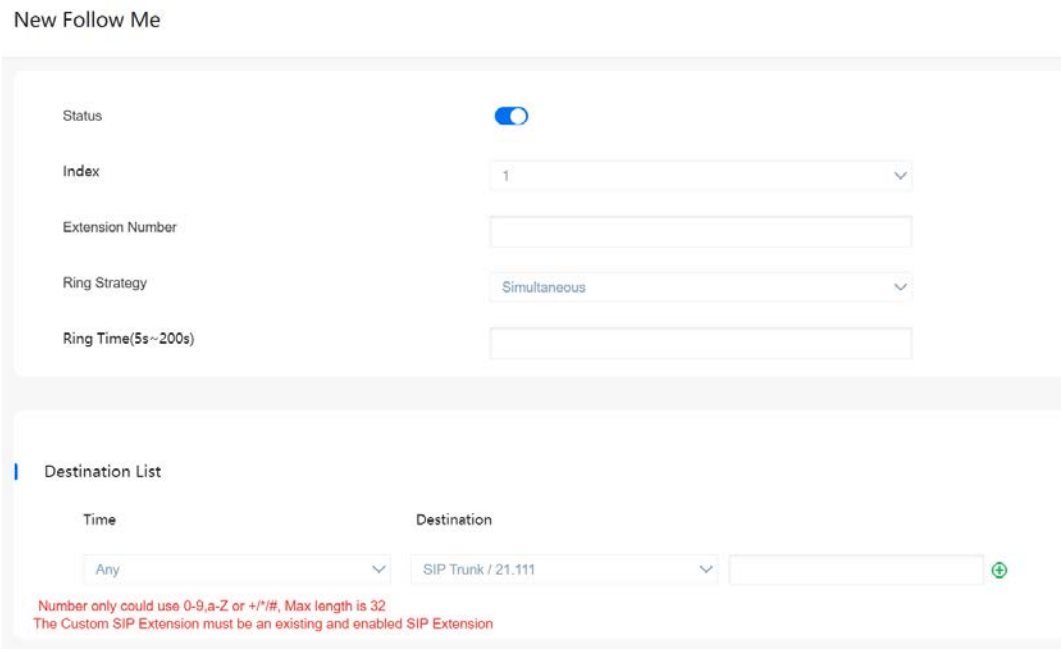


Table-Parameters of Follow Me

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

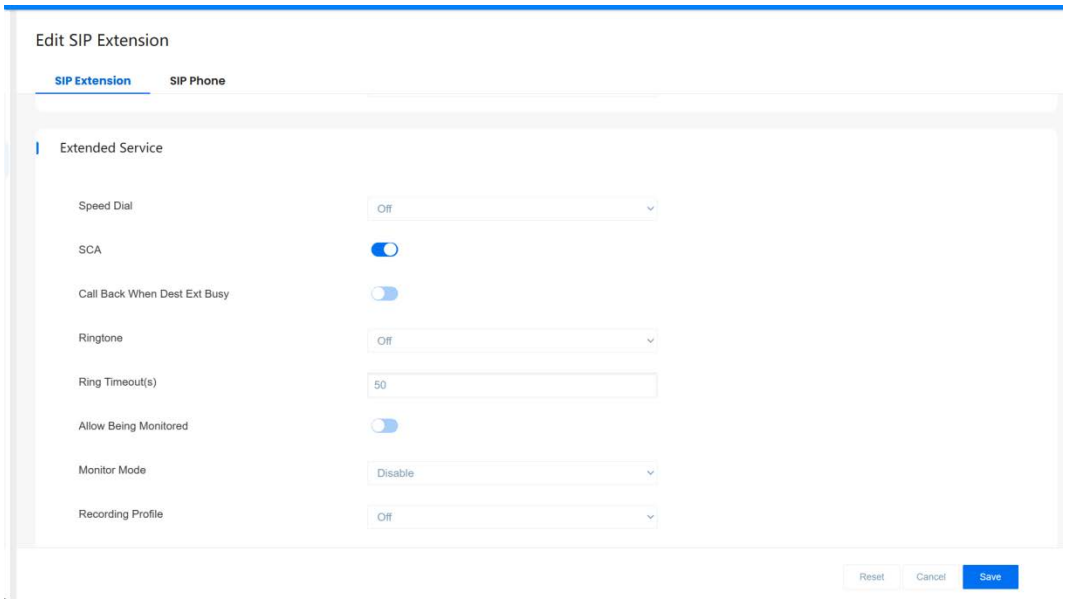
When someone calls a company manager, the secretary will receive the call first and determine whether to forward the person's call to the manager. Sometimes the manager wants to answer the call directly, so a switch is used to control whether the manager can receive the call directly. Managers and secretaries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

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

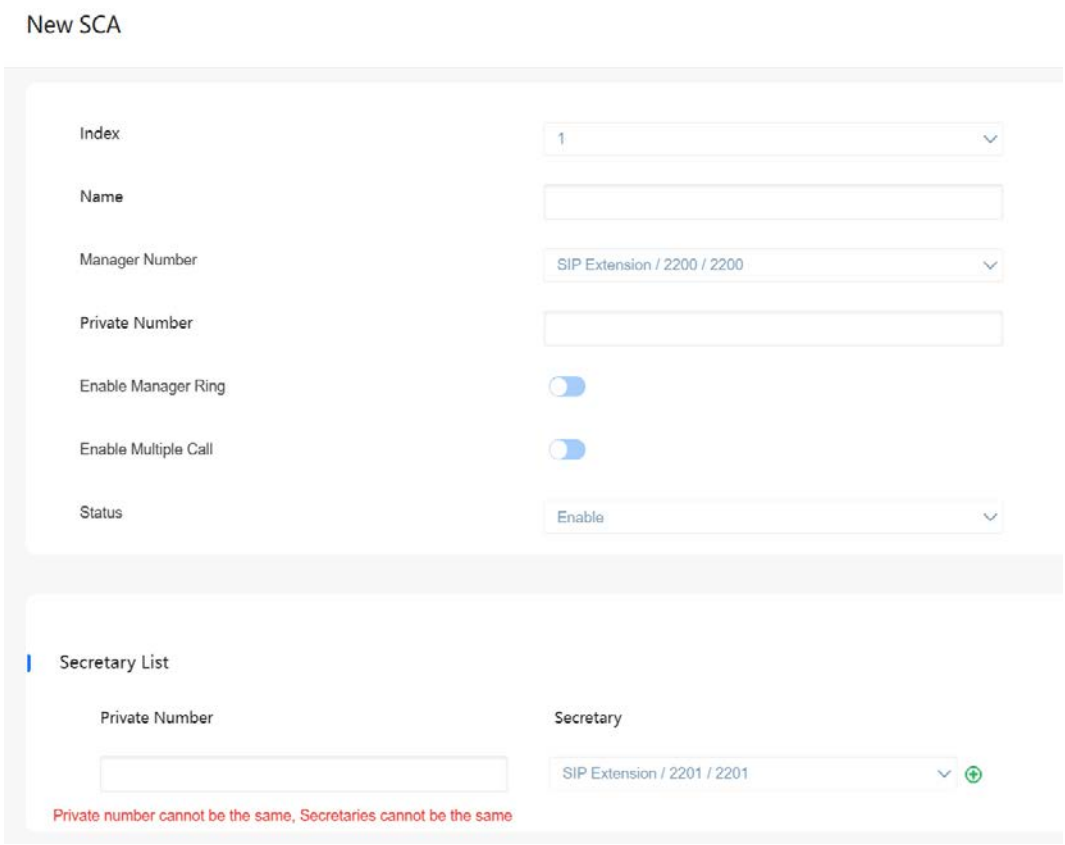


Table-Parameters of SCA

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

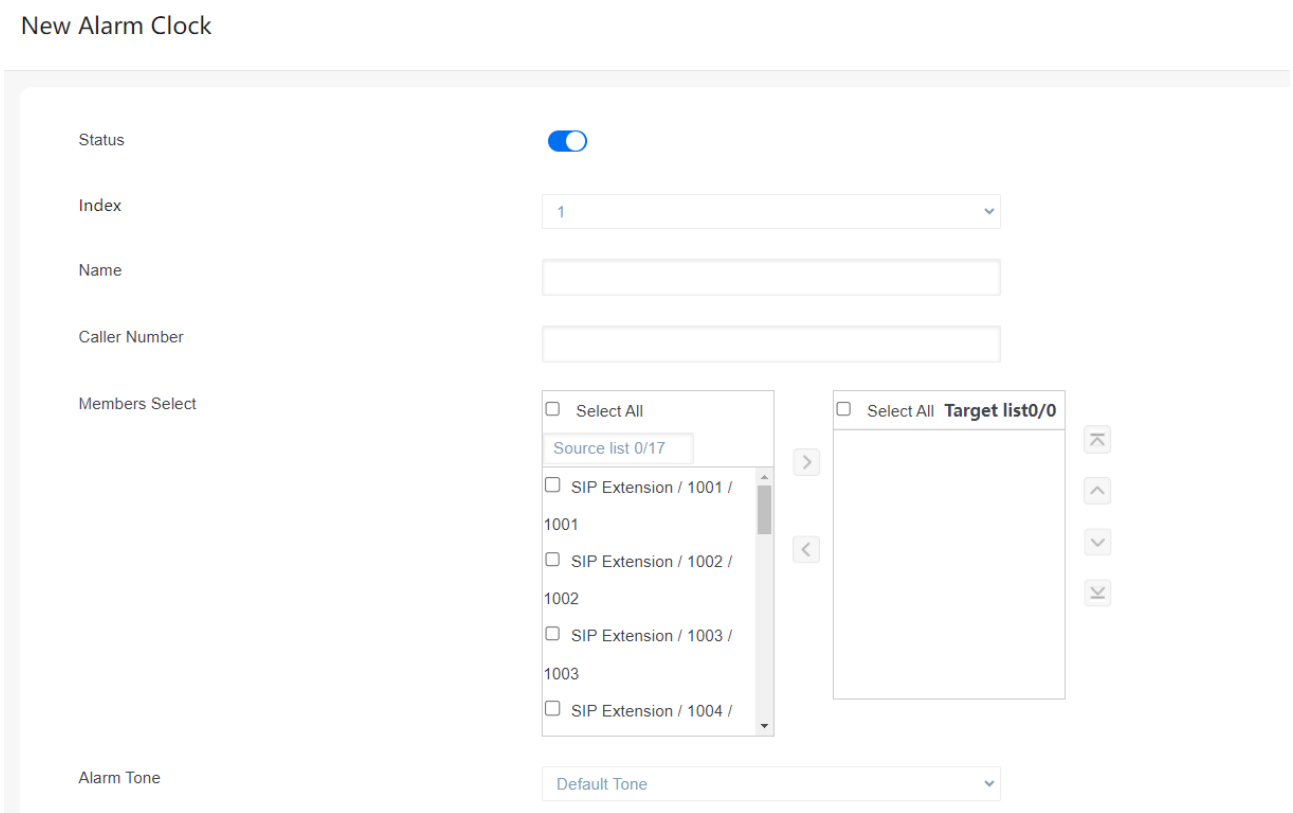


Table-Parameters of Alarm Clock

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 Uc8000 IPPBX device, so users can choose different SIP profiles according to different requirements.

Figure-Parameters of SIP Profile

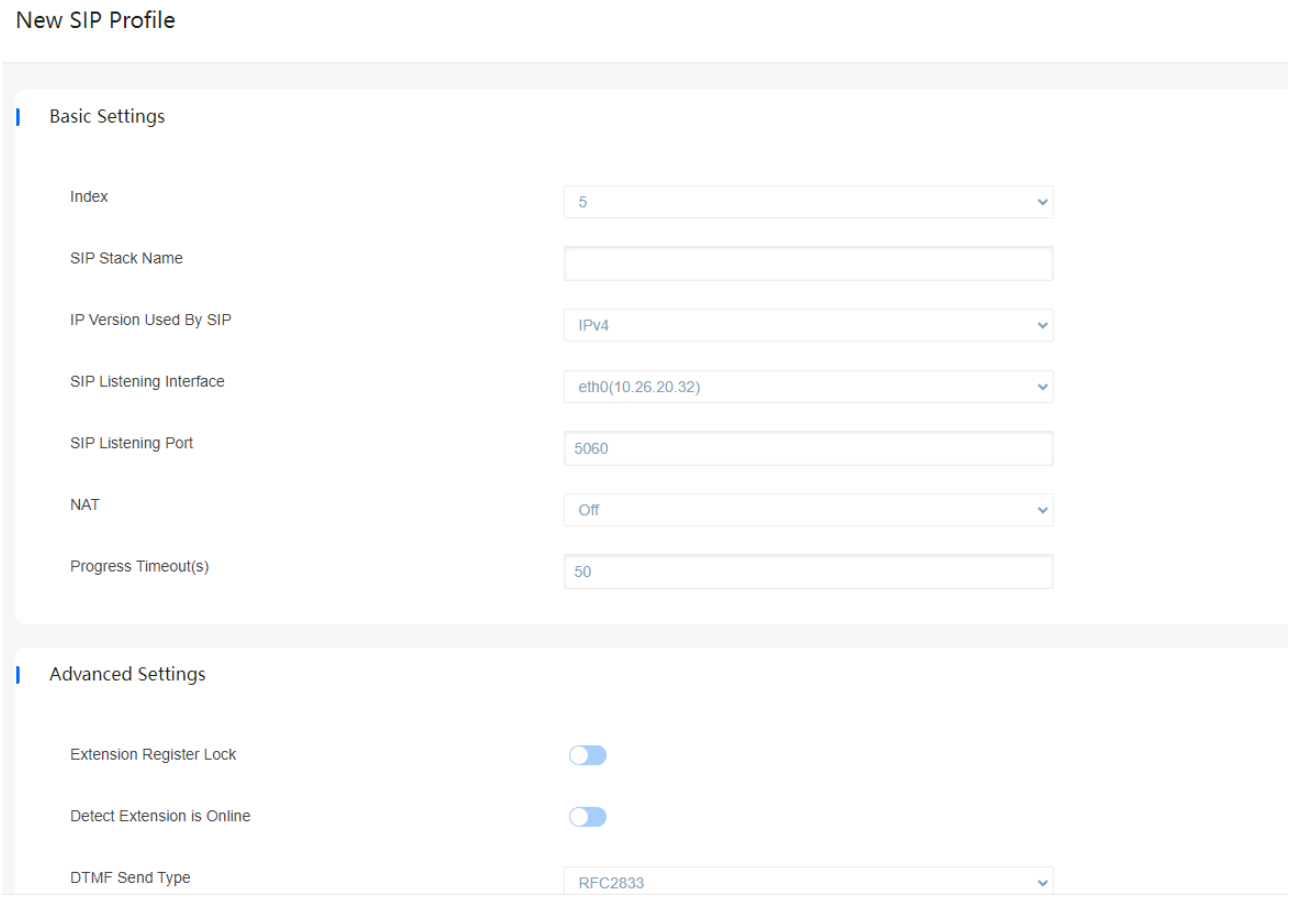


Table-Parameters of SIP Profile

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.

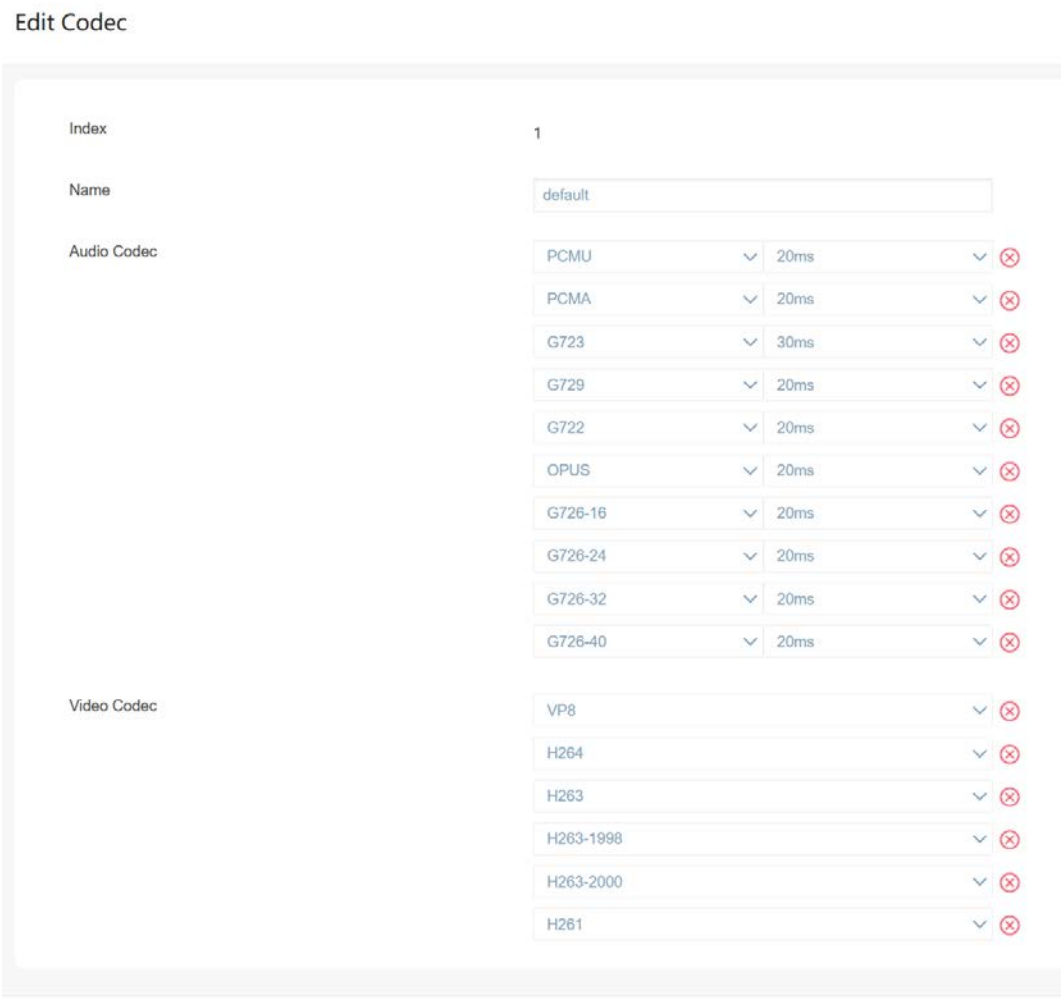
PickUp Caller Refresh Method	The default is disabled. Users can select re-INVITE or UPDATE.
QoS	Whether to enable QoS. QoS is a technology used to solve network delay or congestion.
User Agent	Then content of the 'user agent' field in SIP packets.
Timer T1(ms)	The value of timer T1 in SIP protocol. Default 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, Uc8000 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 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

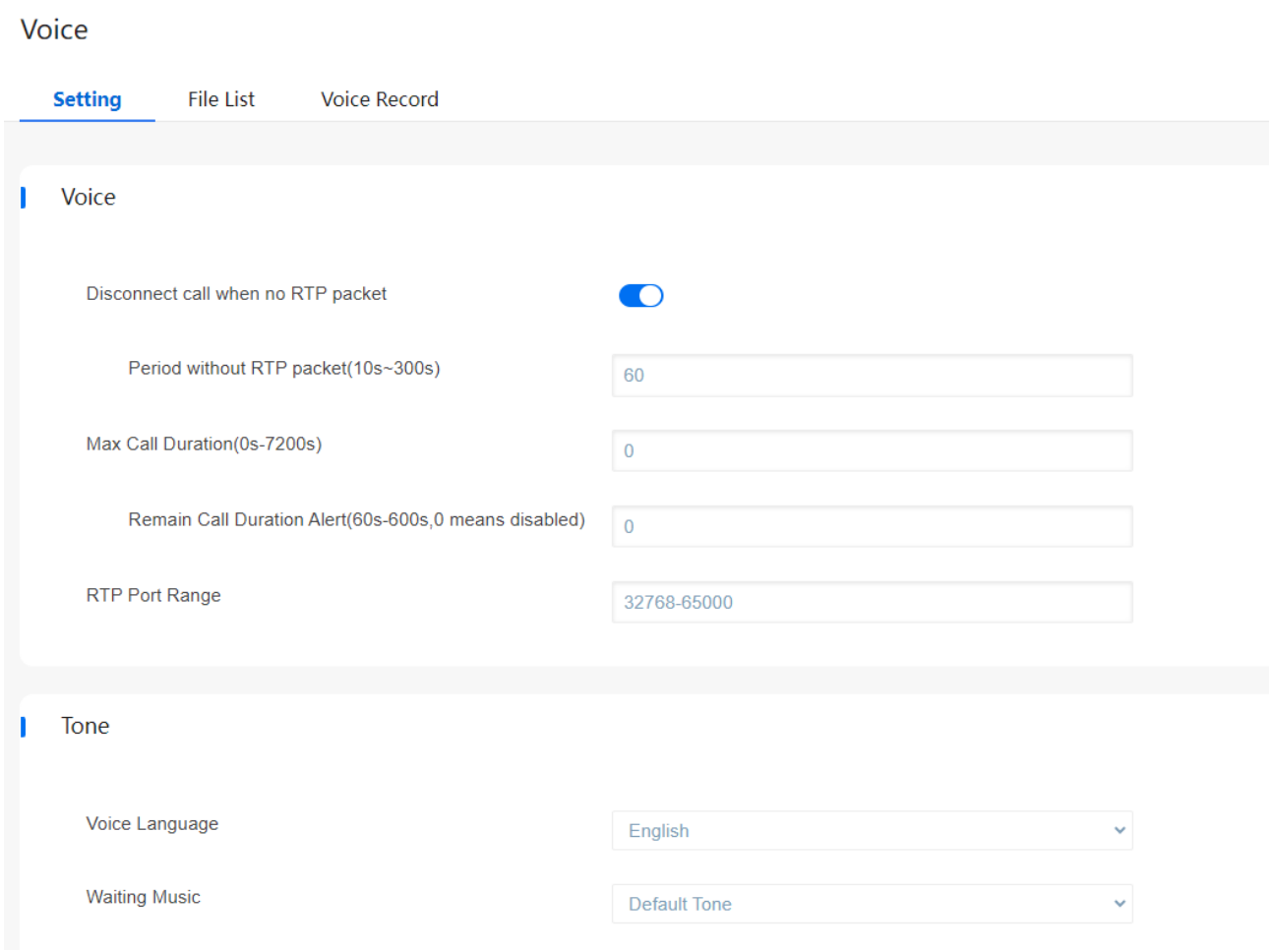


Figure-File List of Voice

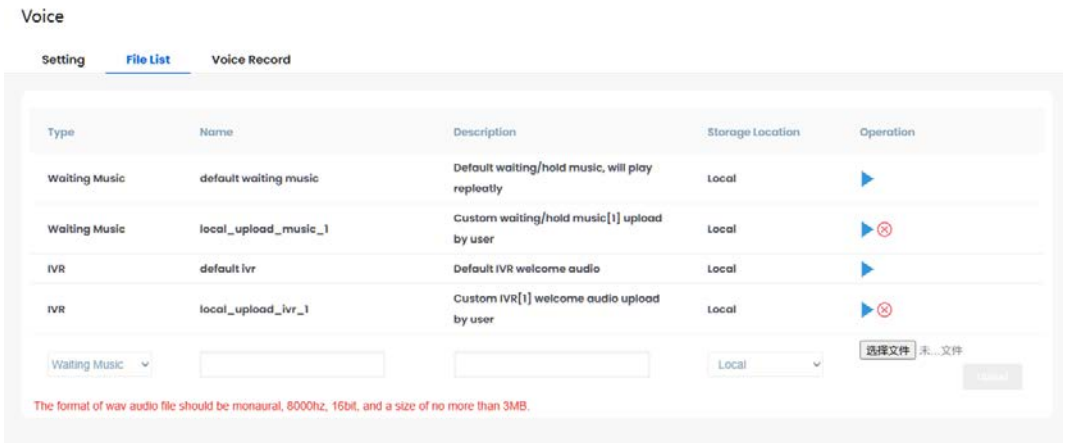


Figure-Voice Record of Voice

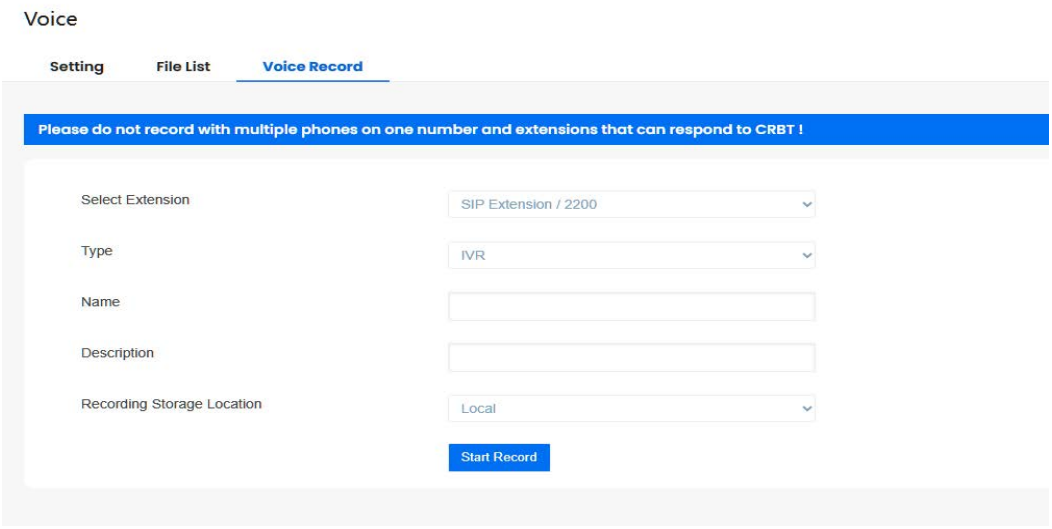


Table- Parameters of Voice Setting

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.

5.5.4 Feature Code

UC8000 provides convenient telephone functions.

The following is the corresponding function of each feature code:

Figure-Feature Code

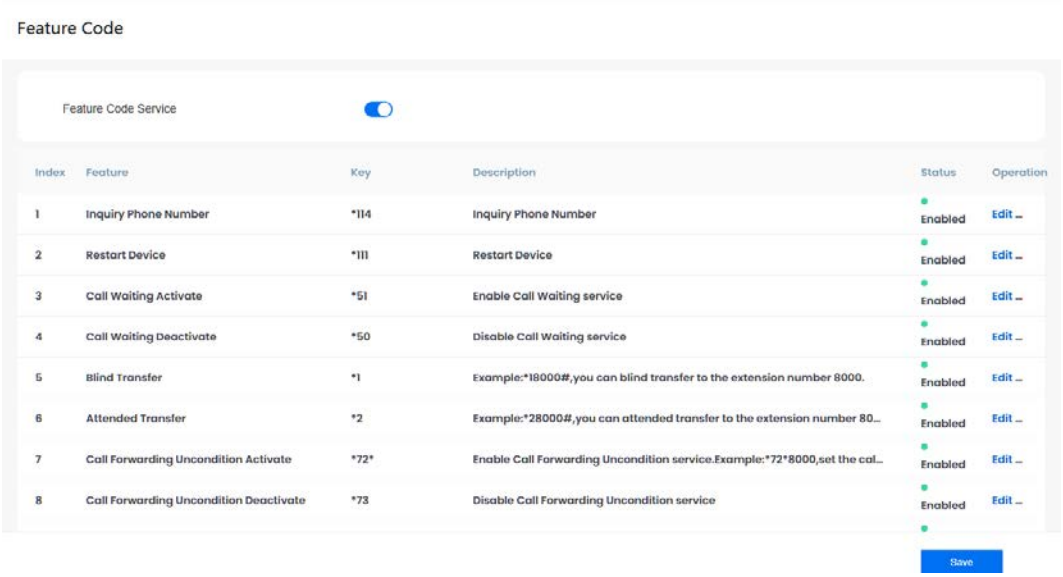


Table-Feature Code

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 attended 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 pickup 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

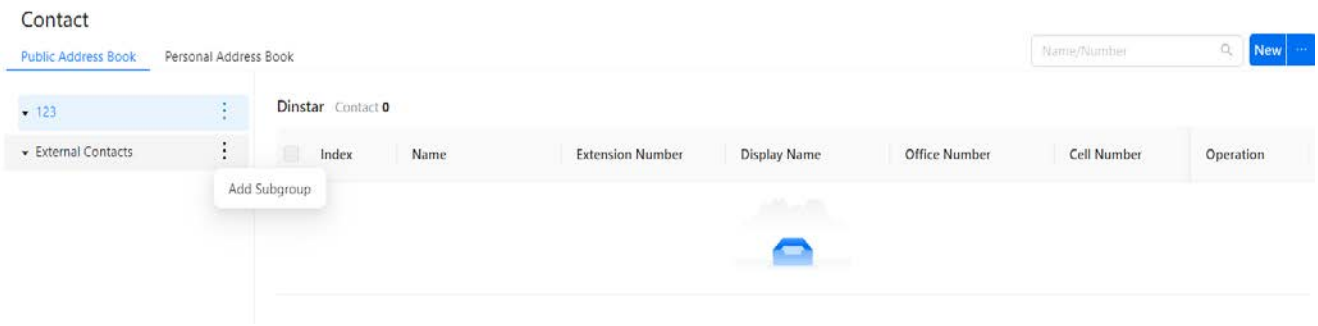


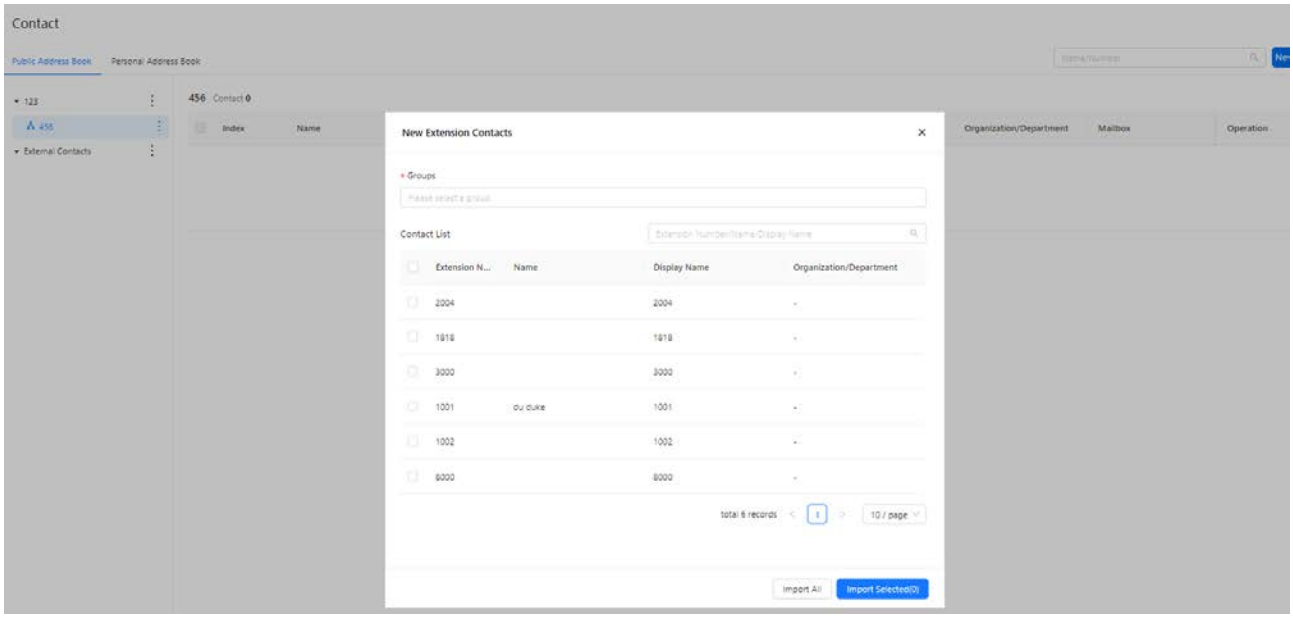
Table- Public Address Book

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



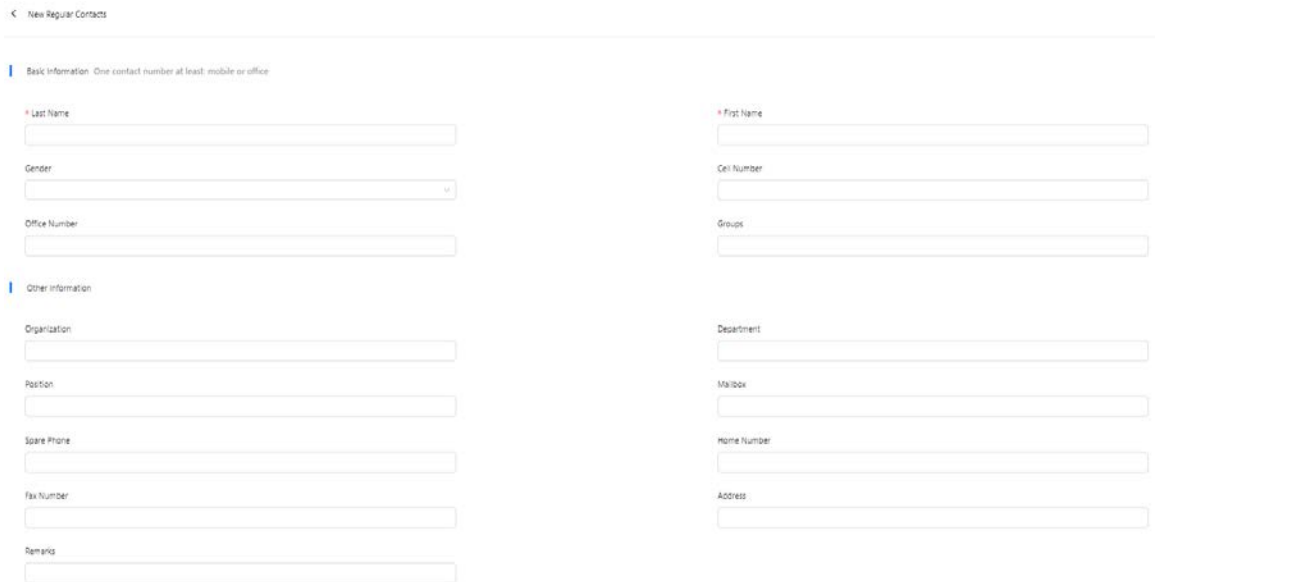


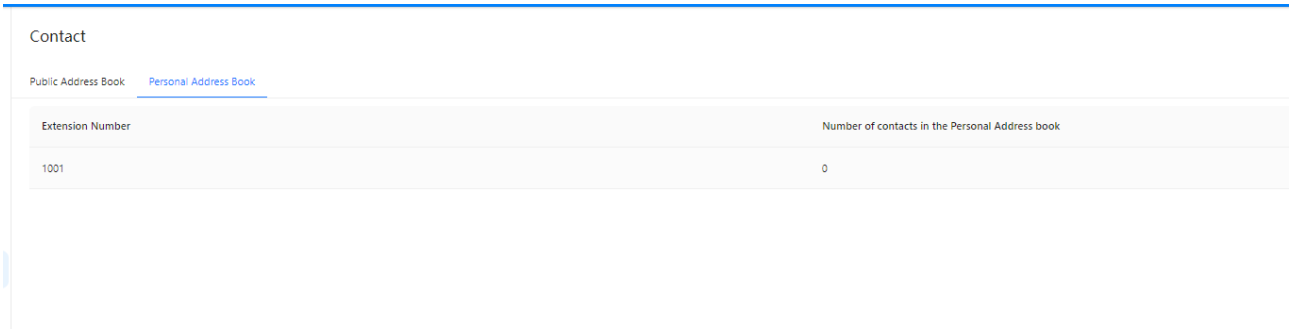
Table- New Regular Contacts

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

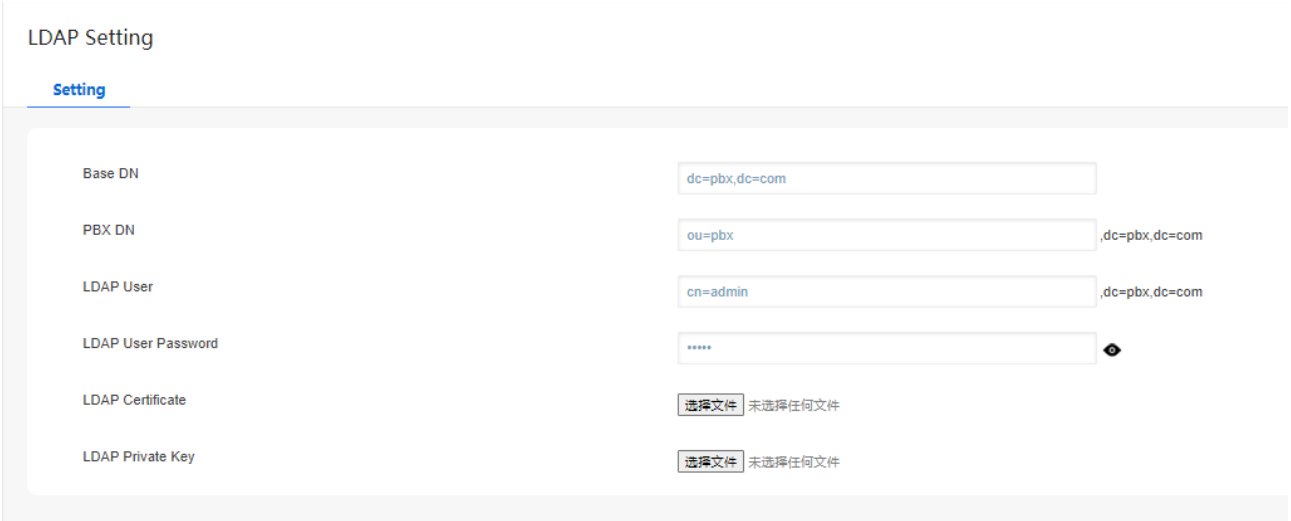


5.6.2 LDAP

Uc8000 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 contact 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-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);

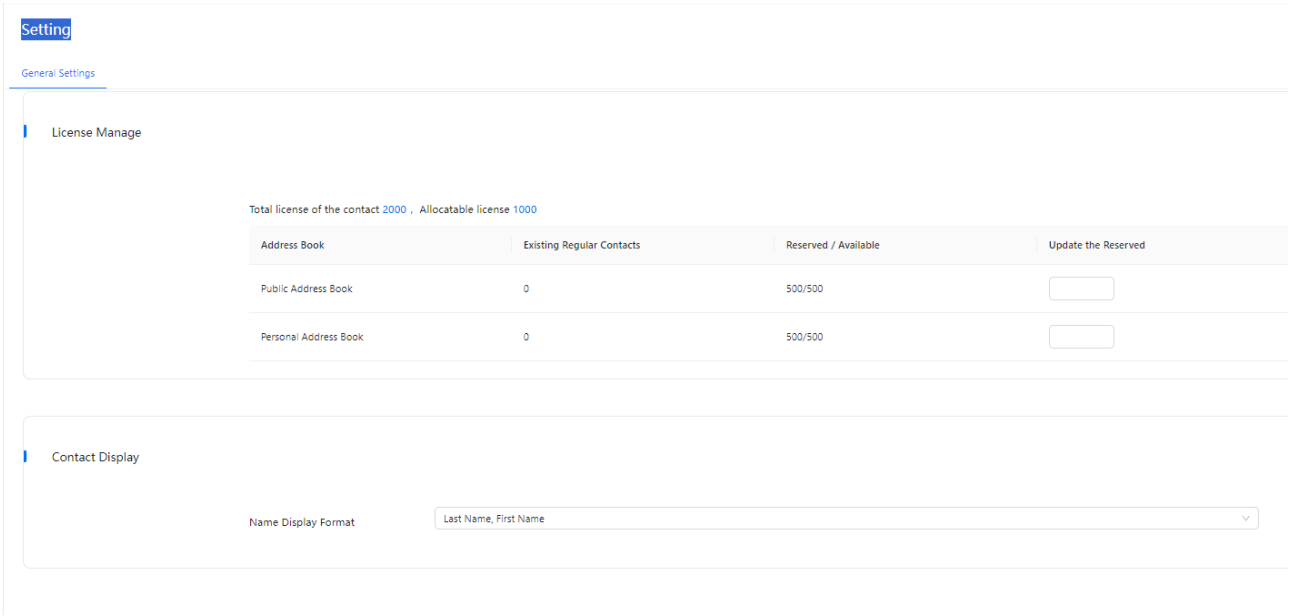
Ordinary contacts in the public address book and personal address book contacts are controlled using the same authorization; after importing the authorization, it is allocated as needed

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

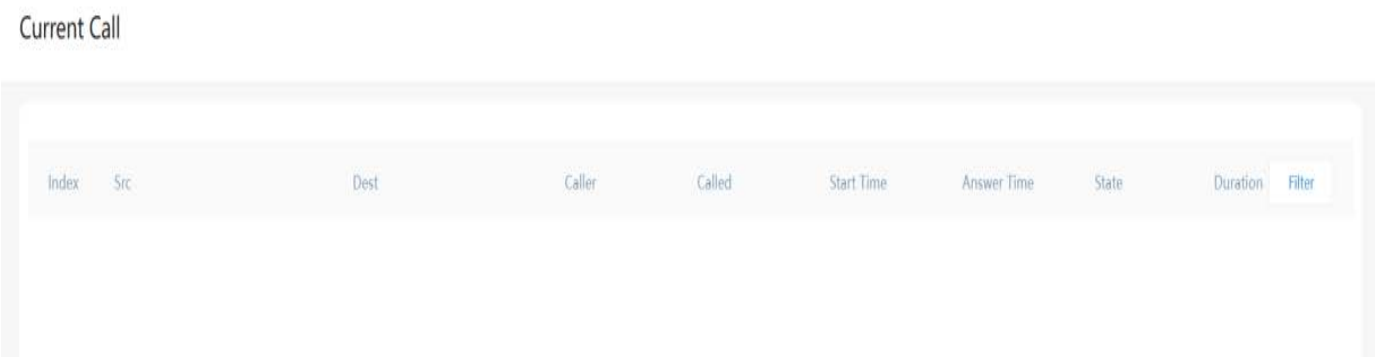


5.7 CDR & Recording

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.

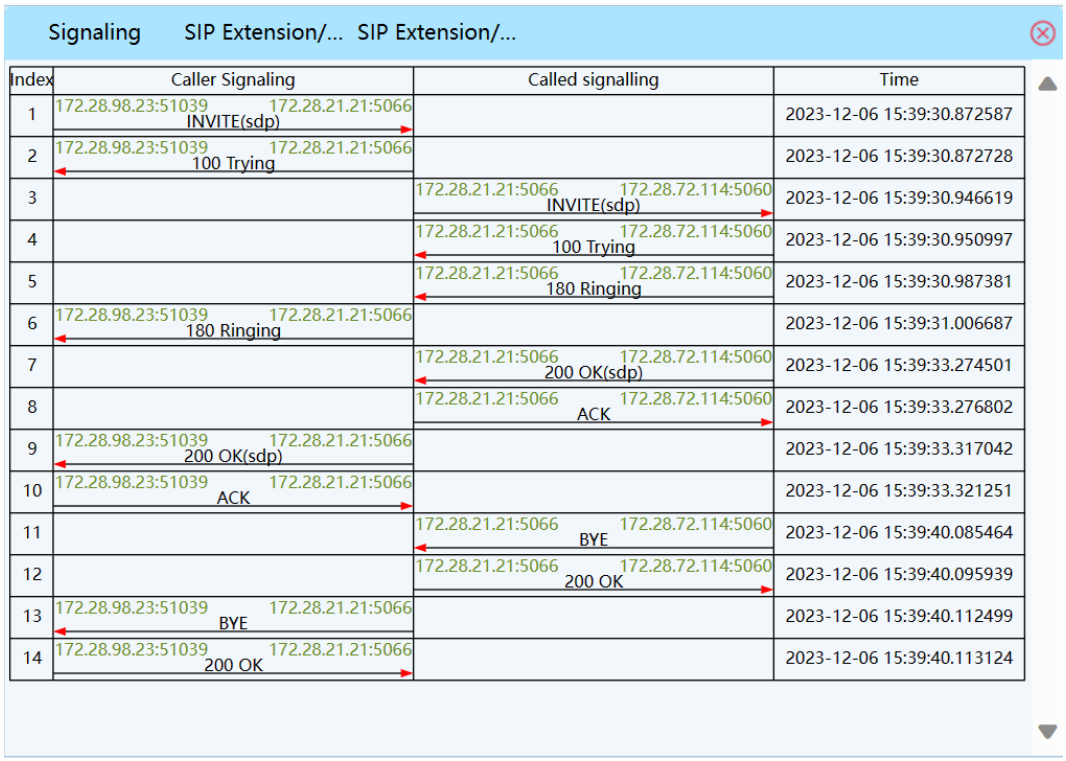
Figure-Parameters of Current Call



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



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

Figure-Parameters of Recording

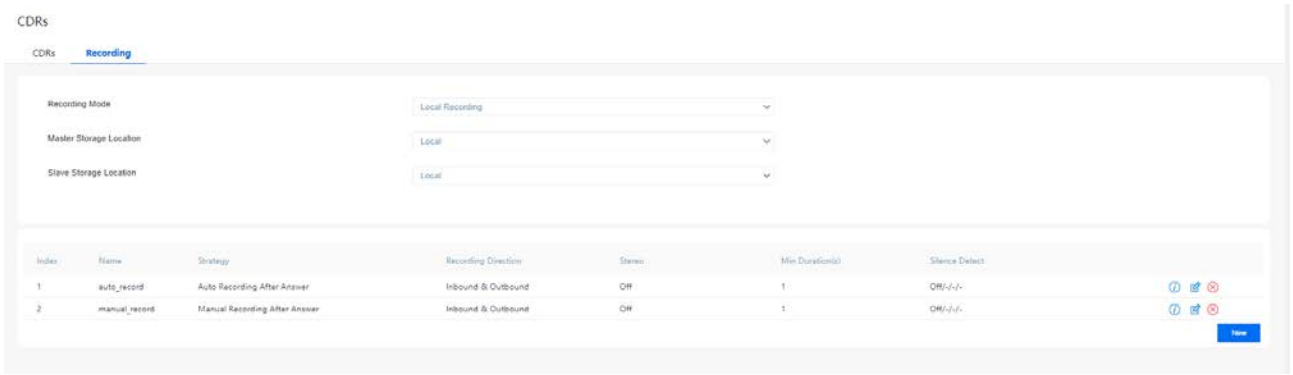


Table-Parameters of Operation

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

Figure-Parameters of Recording rules

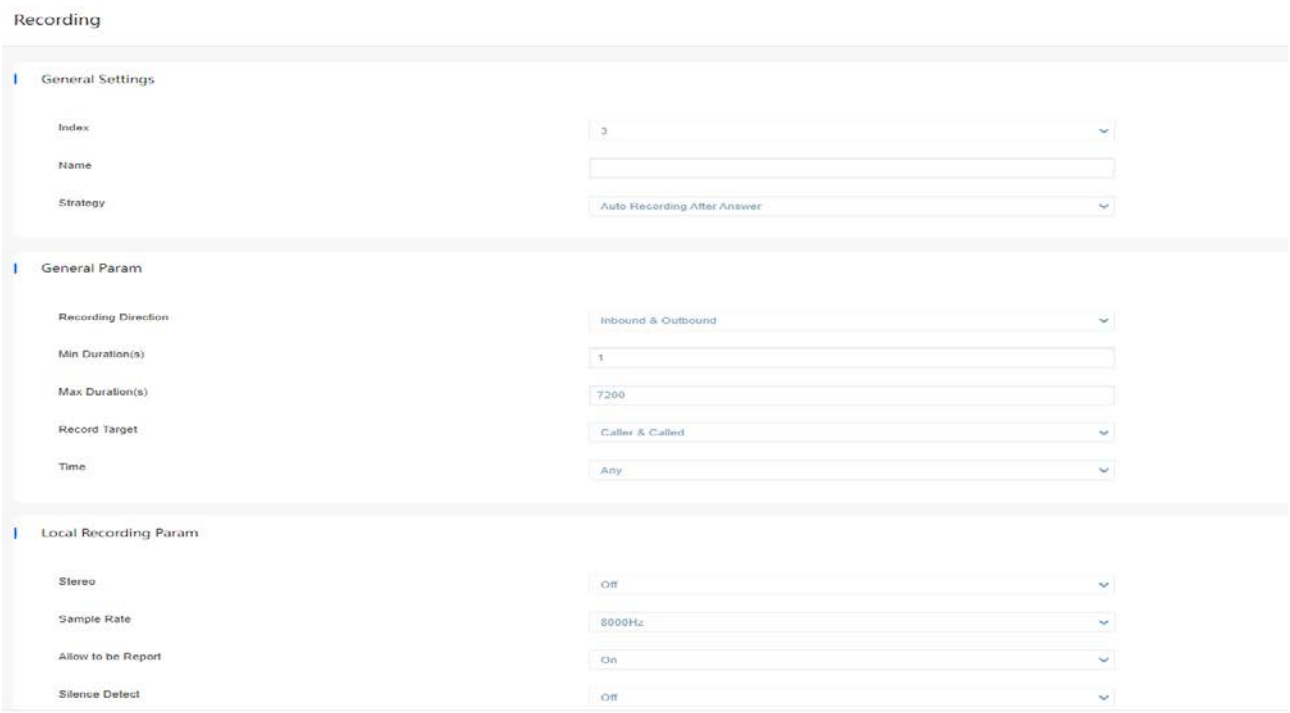


Table-Parameters of Recording rules

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 SIP extension, both inbound and outbound calls will be recorded. Inbound: If this recording profile is added to SIP extension, only inbound calls will be recorded. Outbound: If this recording profile is added to 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

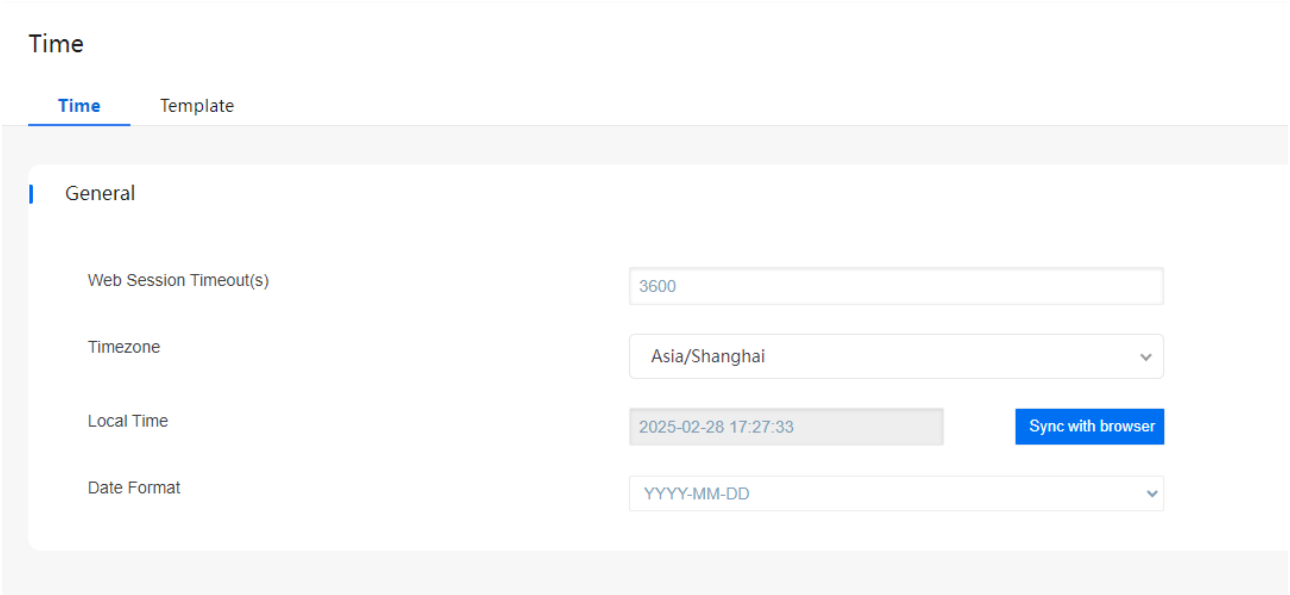


Figure-Parameters of Time Template

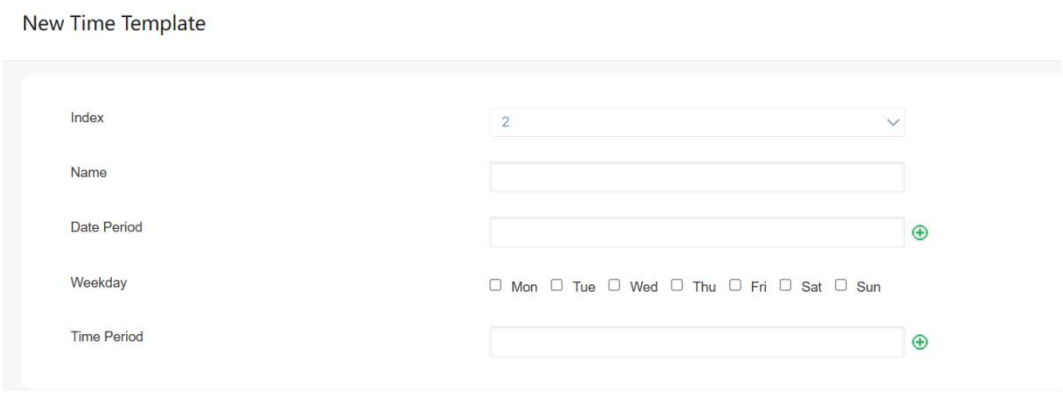


Table-Parameters of Time Template

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

Static Route

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

Figure-Parameters of Static Route

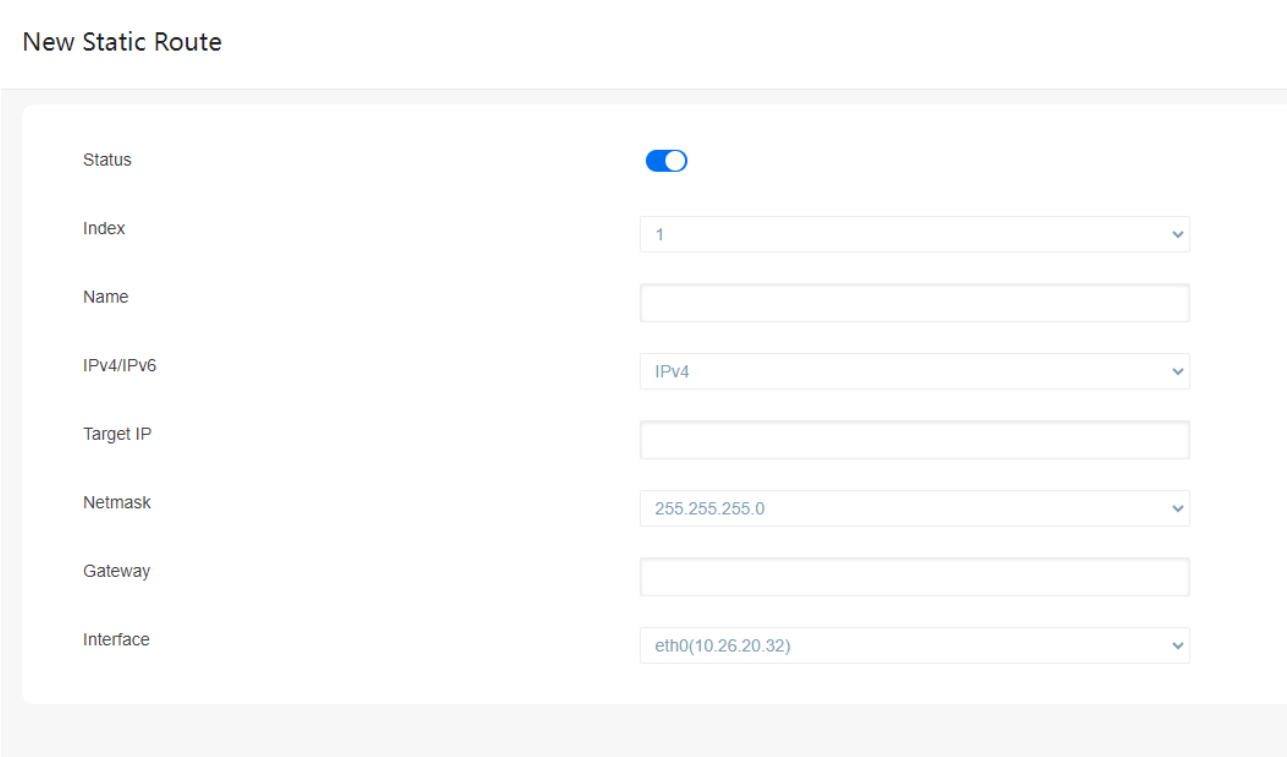


Table-Parameters of Static Route

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.

DDNS

On the **System > Network > DDNS** interface, users can use Uc8000 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

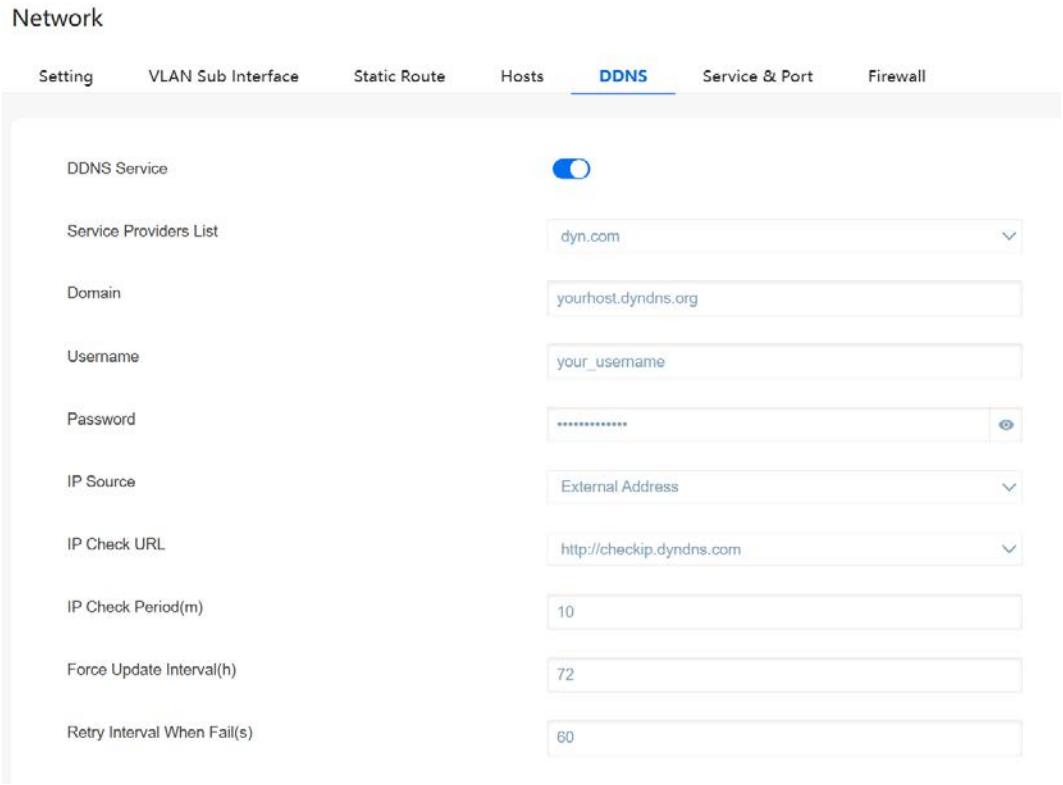


Table-Parameters of DDNS

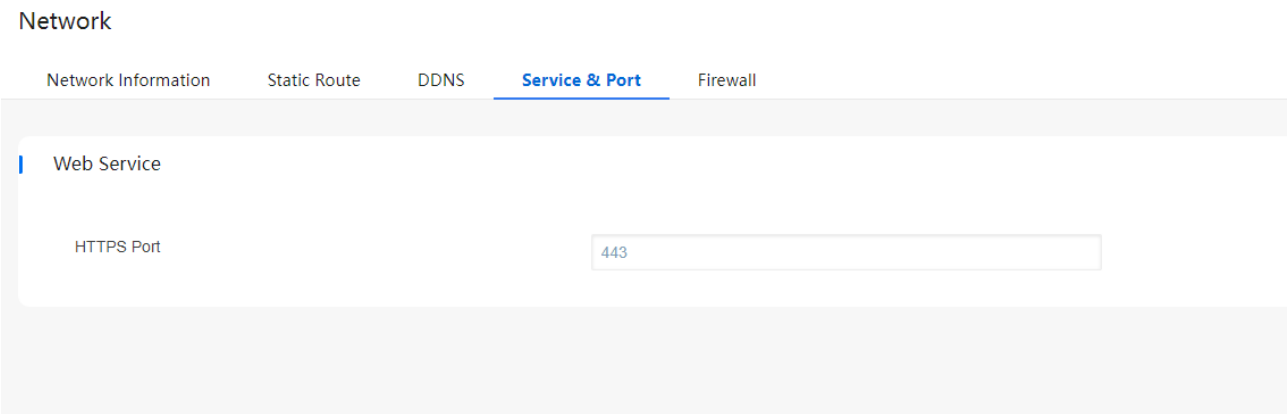
Parameter	Description
Service Providers List	Dynamic domain name service providers.
Domain	Domain name applied for on the service provider website.

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 Network interface 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

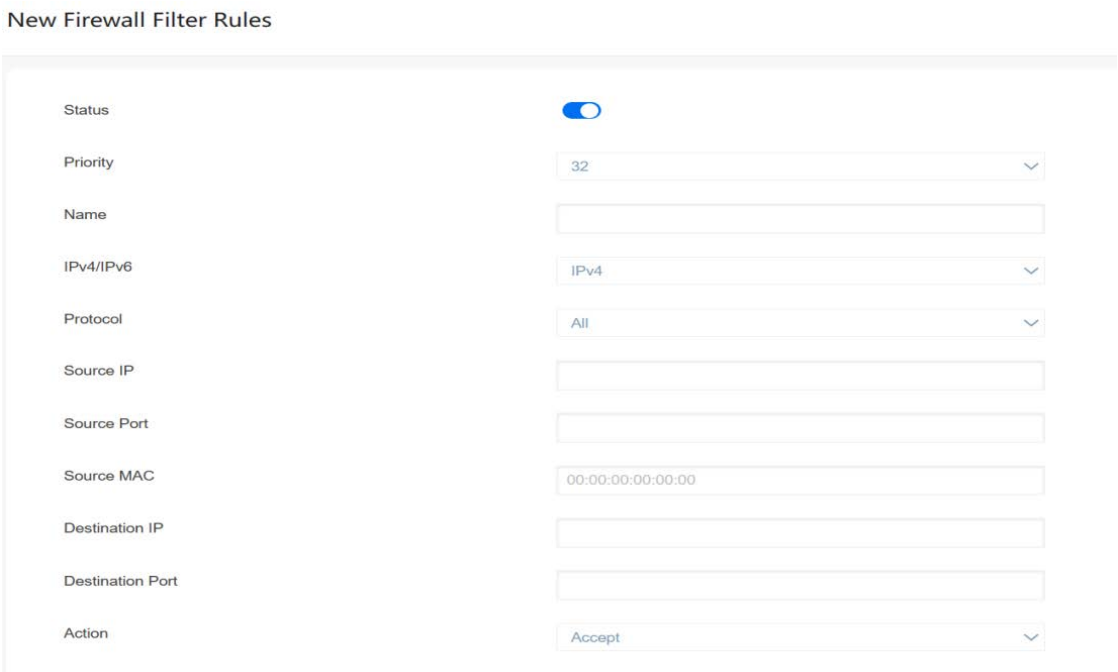


Table-Parameters of Firewall Filter Rules

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 Uc8000 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.

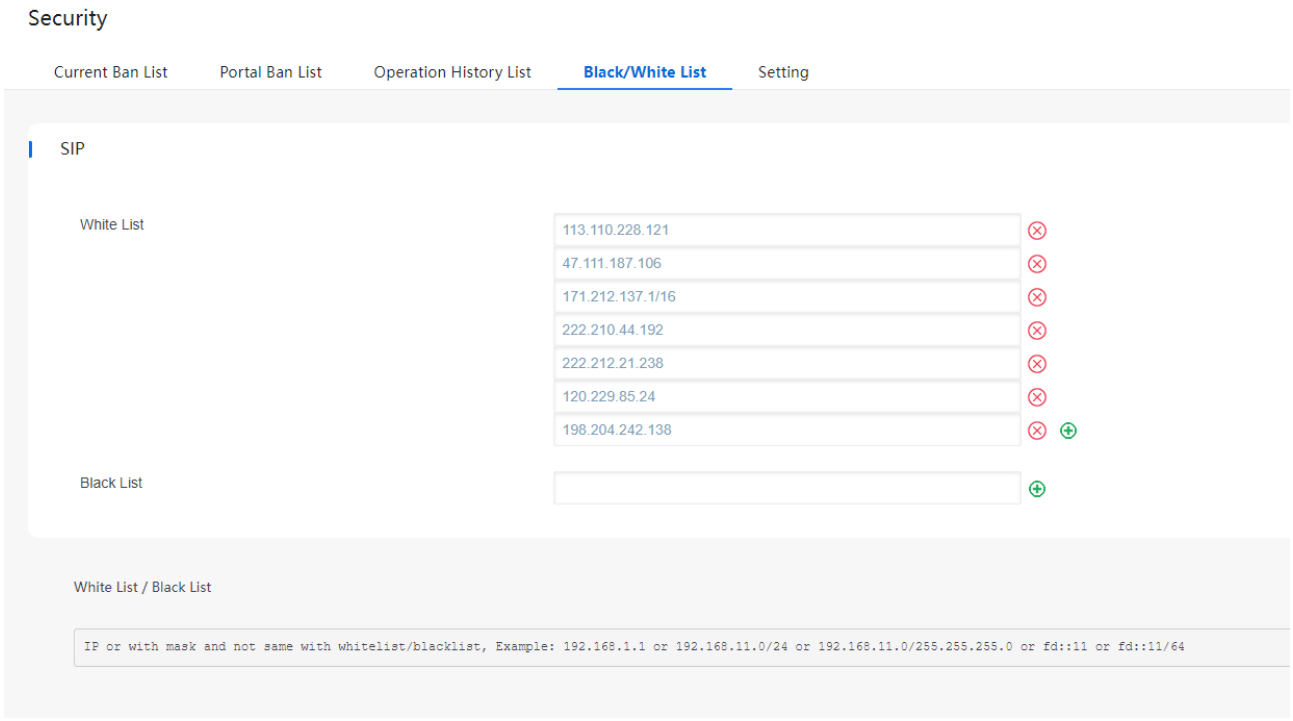
Source Port	The source port of host which the accepted or rejected IP address belongs to.
Source MAC	The source mac of the host which the accepted or rejected IP address belongs to.
Destination IP	The destination IP address that users want Uc8000 IPPBX to accept or reject. It is the destination IP address 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.

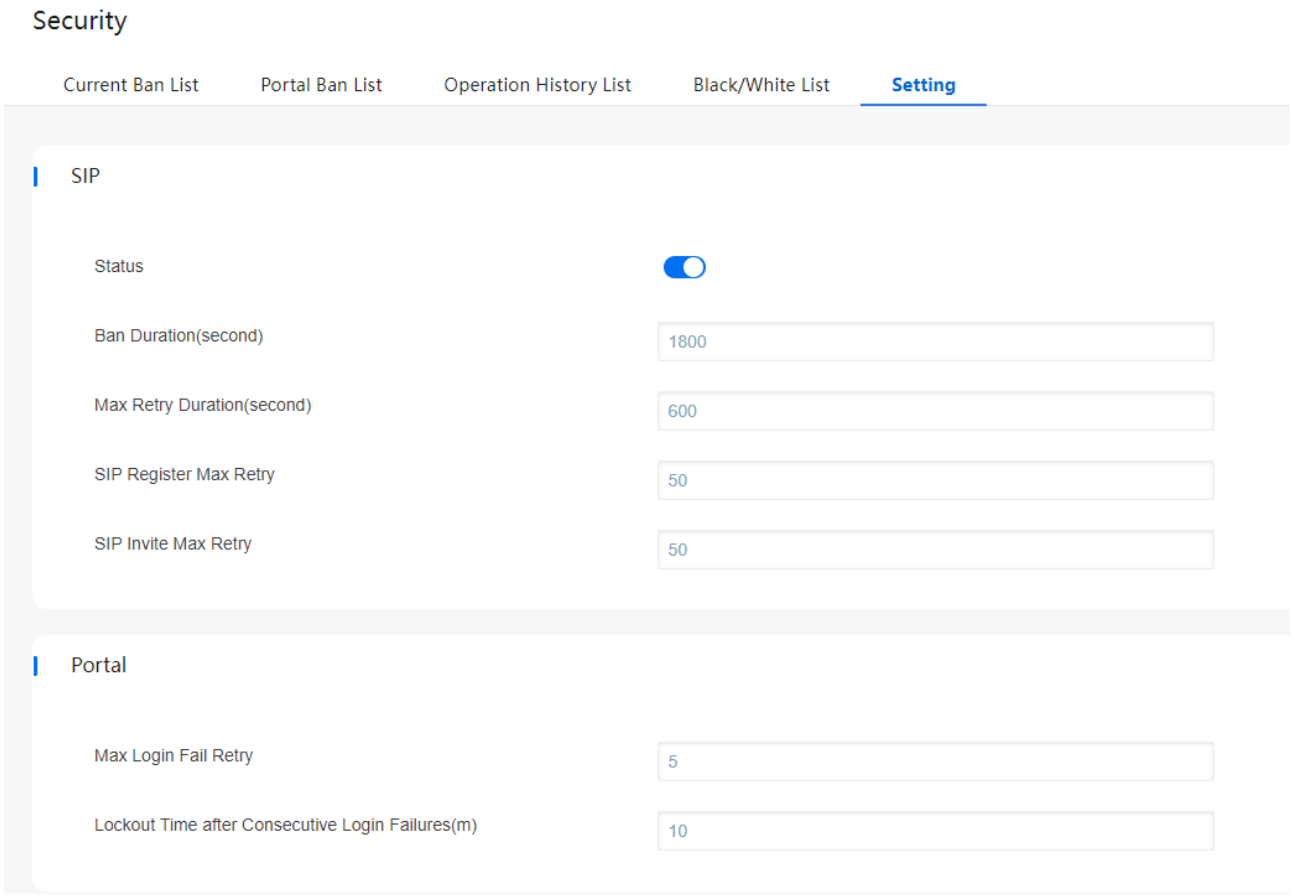
In the "System->Security->Black / White List" page, you can set the black and white list of SIP based on IP address, as shown in the following figure:

Figure- Black / White List Setting



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 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. Uc8000 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

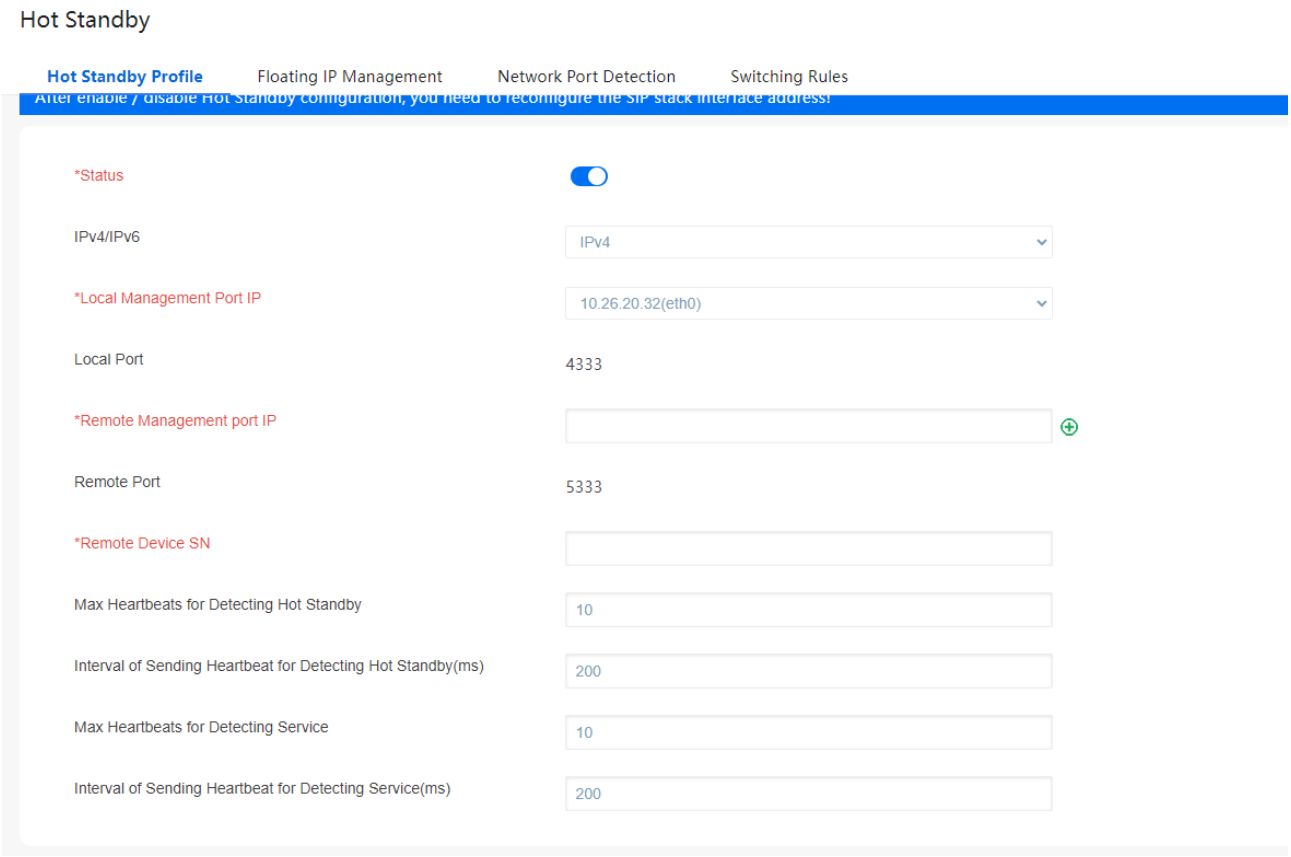
5.8.5 Hot Standby

In order to ensure that the Uc8000 IPPBX can work normally and stably, the uc8000 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



Note:

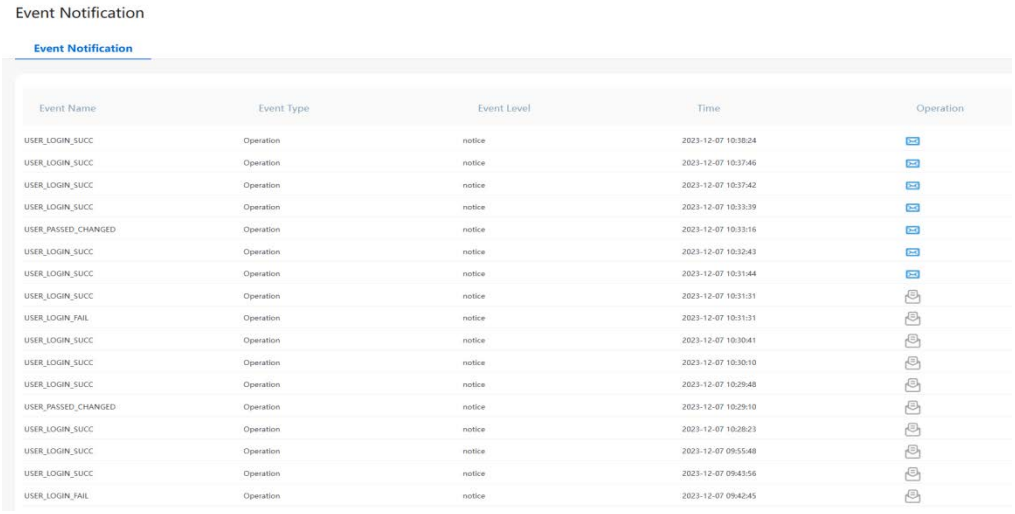
1. Modifications with * options may affect hot standby synchronization of other configurations except HA. Please modify and apply separately!

2. After enabling/disabling hot standby configuration, you need to reconfigure the interface address in the SIP protocol stack!

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

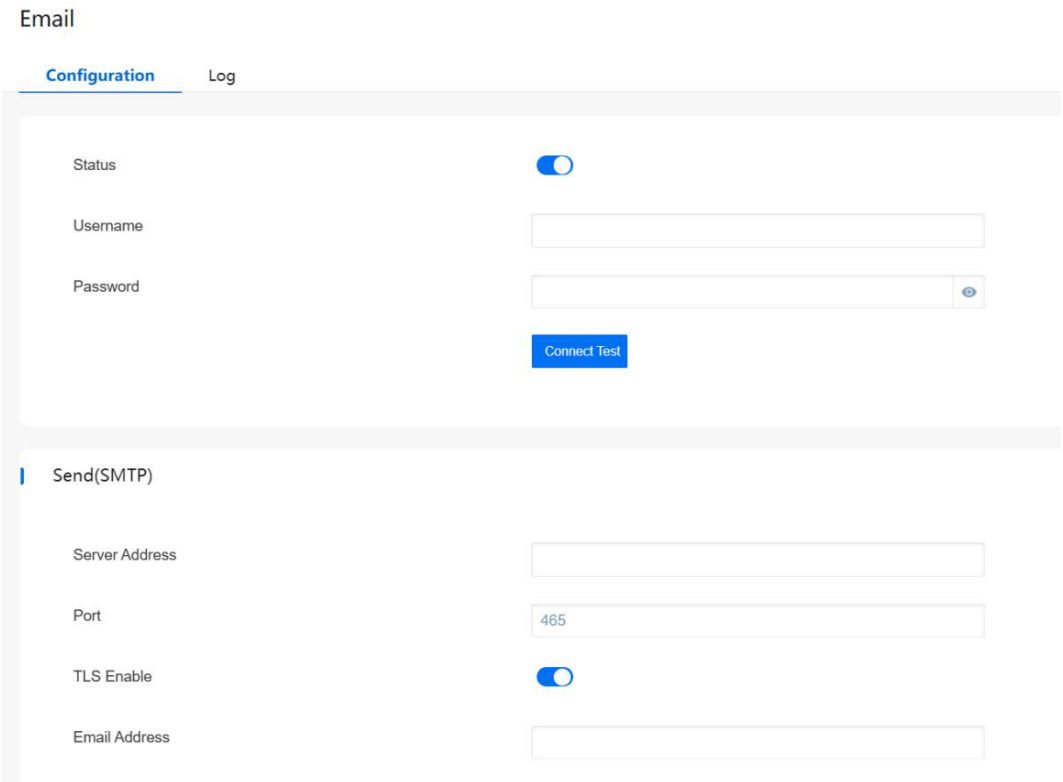


Table-Parameters of Email

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

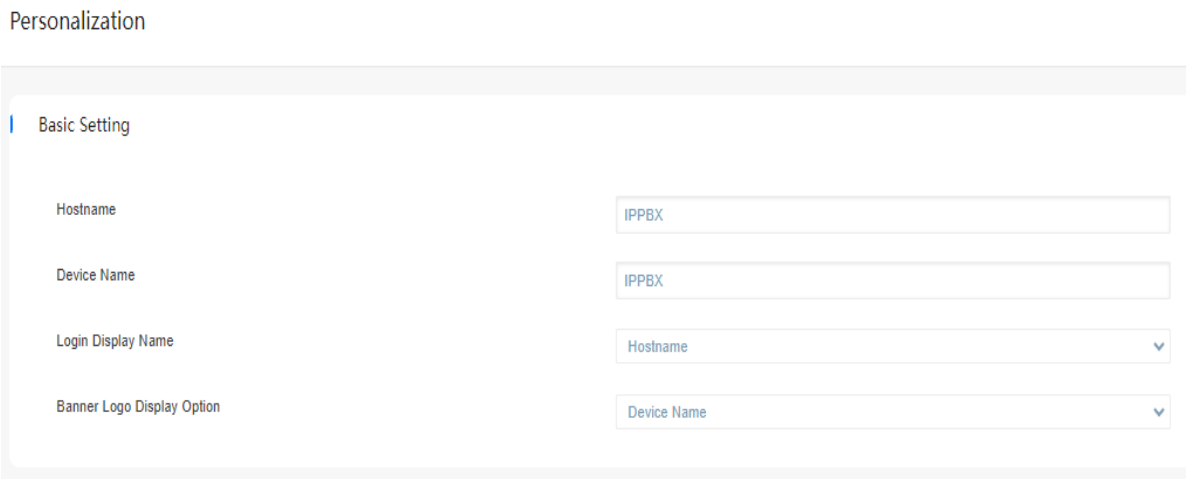


table-Parameters of Personalization

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

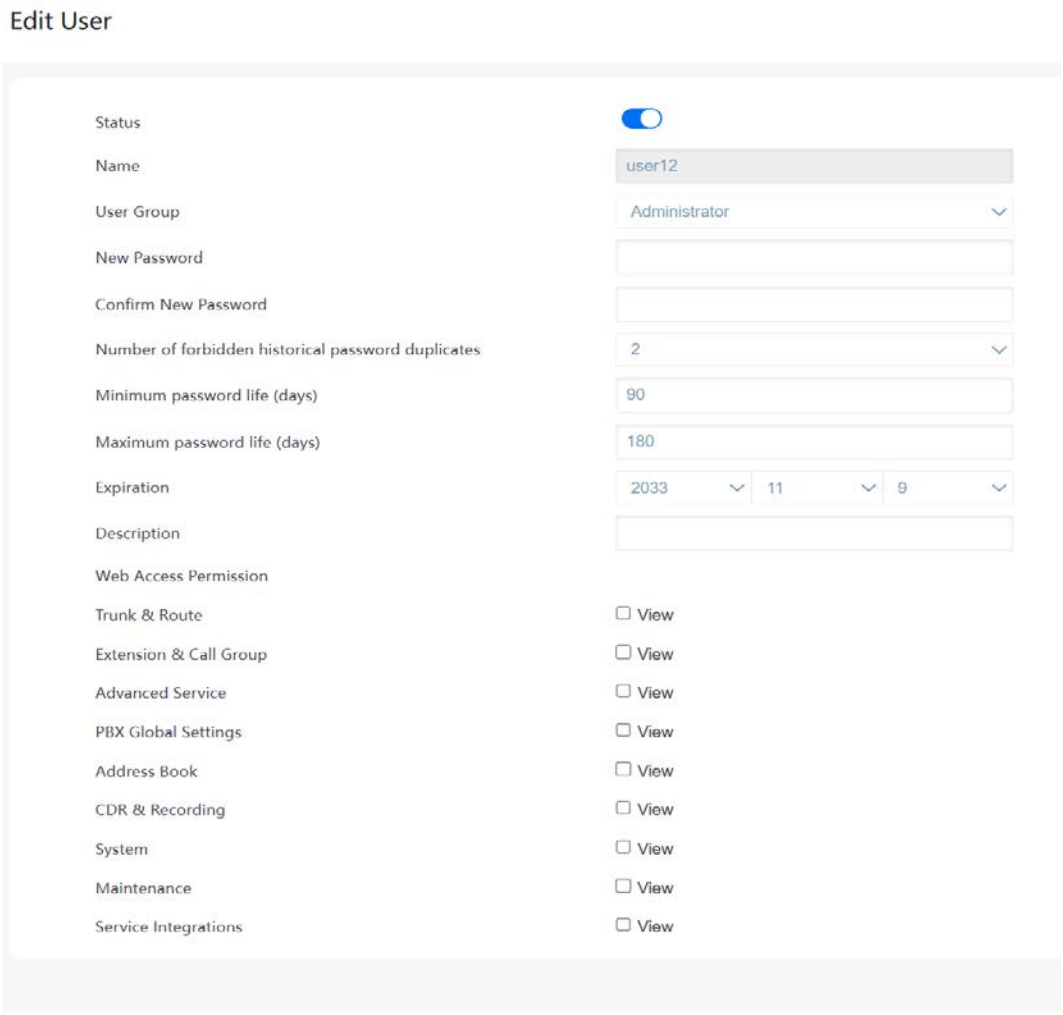


Table-Parameters of User

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

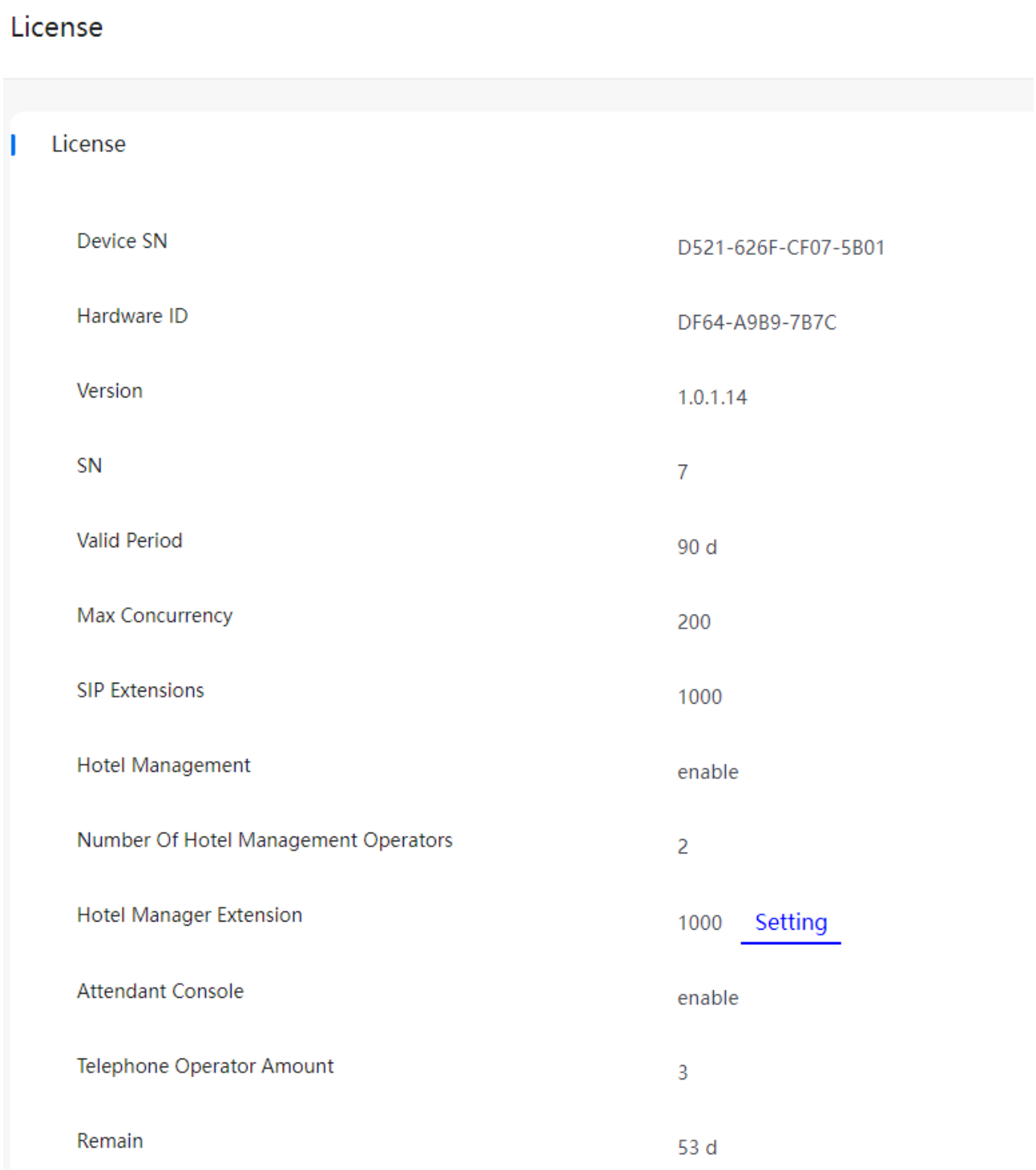


Table-Parameters of License

Parameter	Description

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)

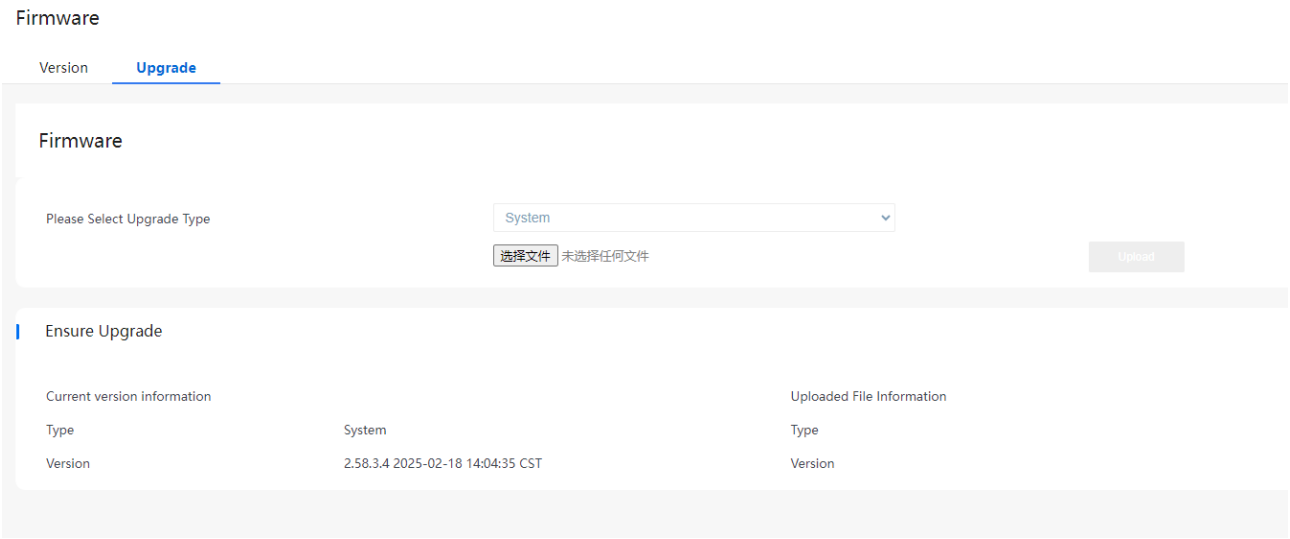
Telephone Operator Amount	When creating a new console type queue, the number of static seats in all console queues cannot exceed this number
Remain	Indicates how many days are left for the license to expire
Hot Standby	Indicates whether UC has authorized the dual-machine hot standby function

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, kernel image, 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.

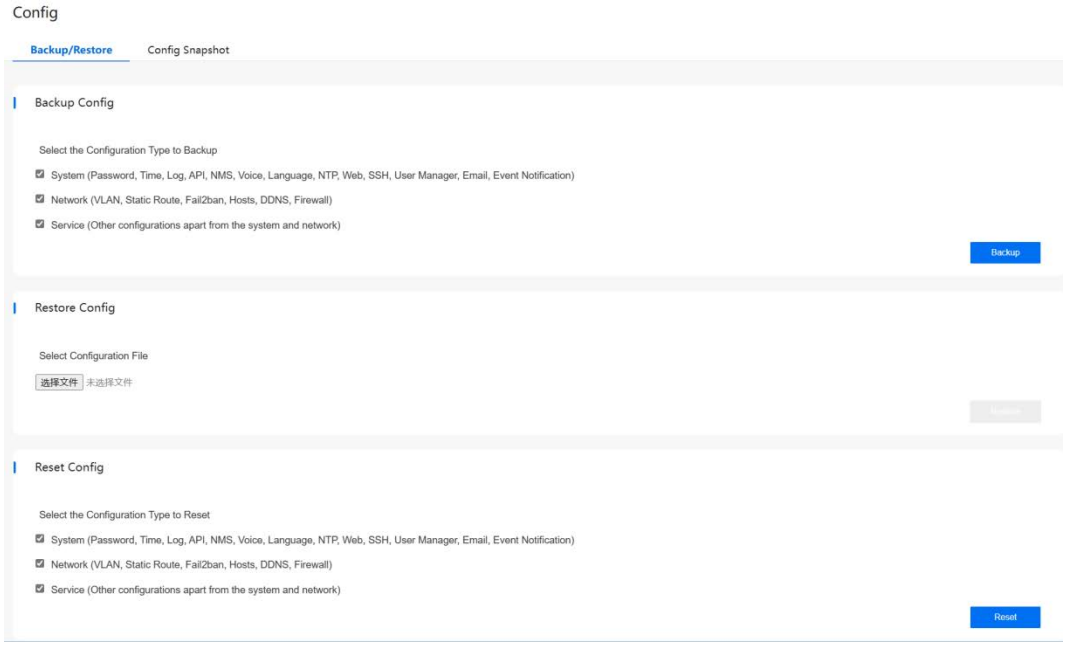
Figure-Parameters of Firmware



5.9.4 Config

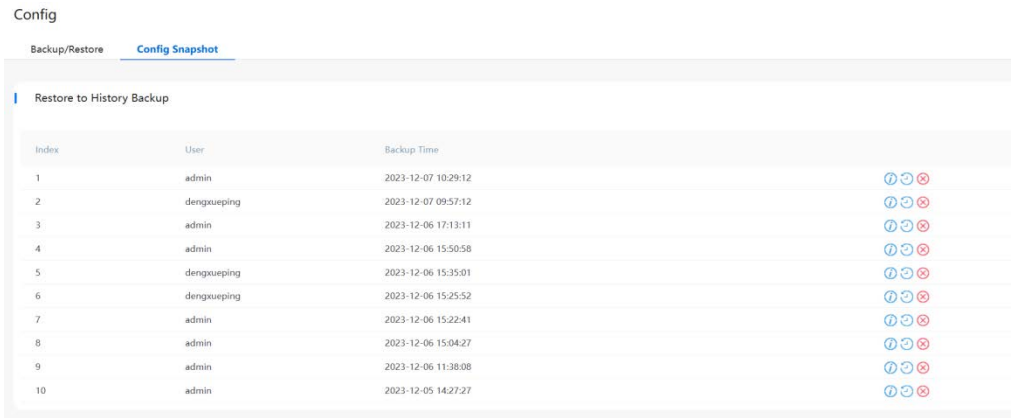
On the **Maintenance > Config** interface, users can back up or restore configuration files. But users need to restart the device for the change to take effect after executing restore.

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 Uc8000 IPPBX device, record backup, and back up CDRs, logs or configurations.

Figure-Parameters of Reboot

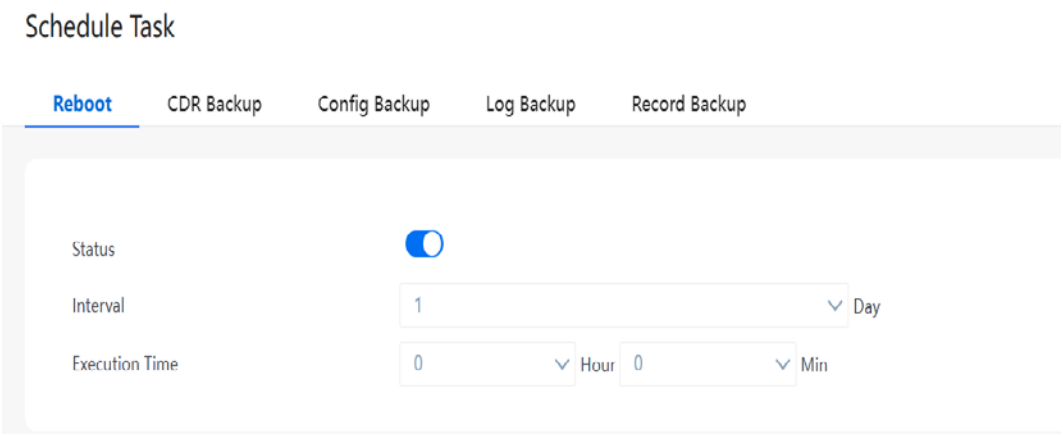


Figure-Parameters of CDR Backup

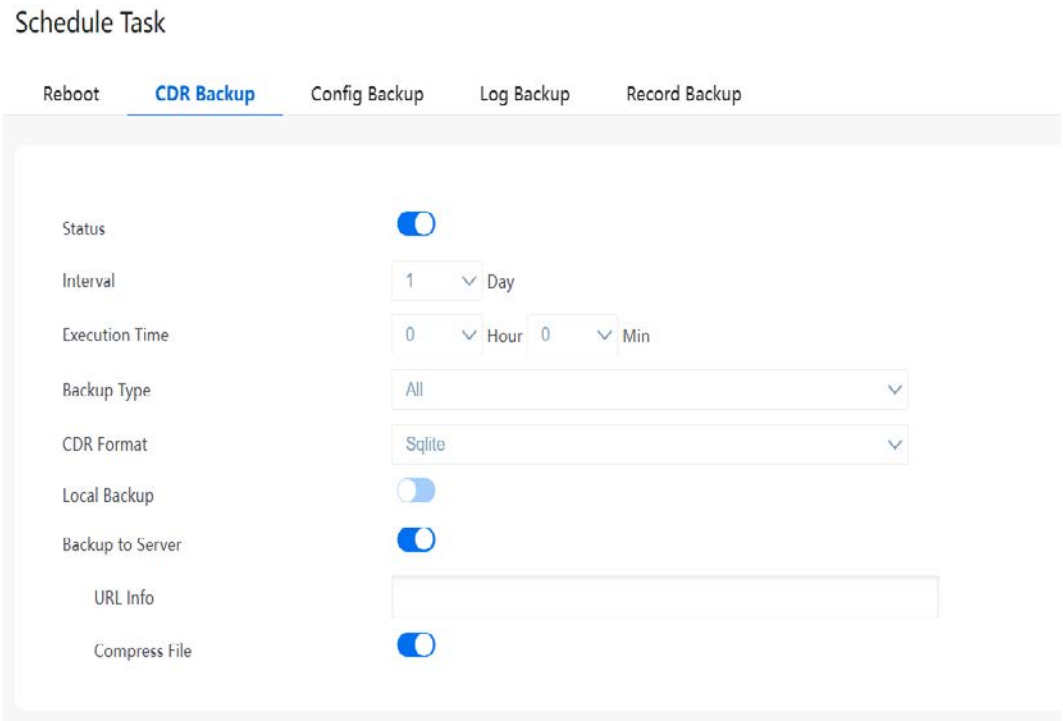


Figure-Parameters of Config Backup

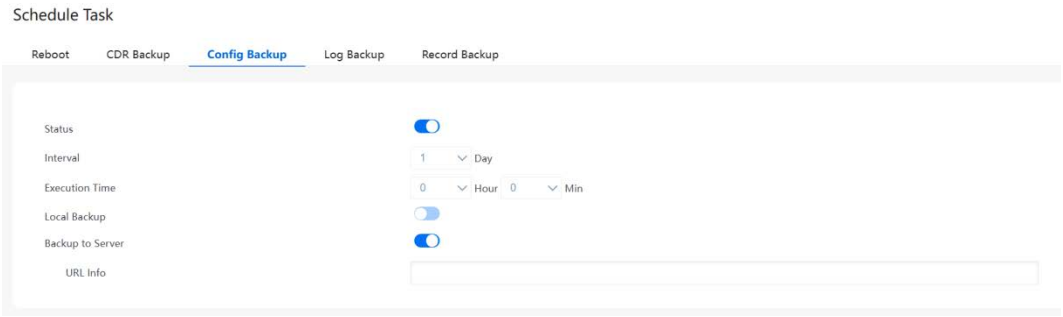


Figure-Parameters of Log Backup

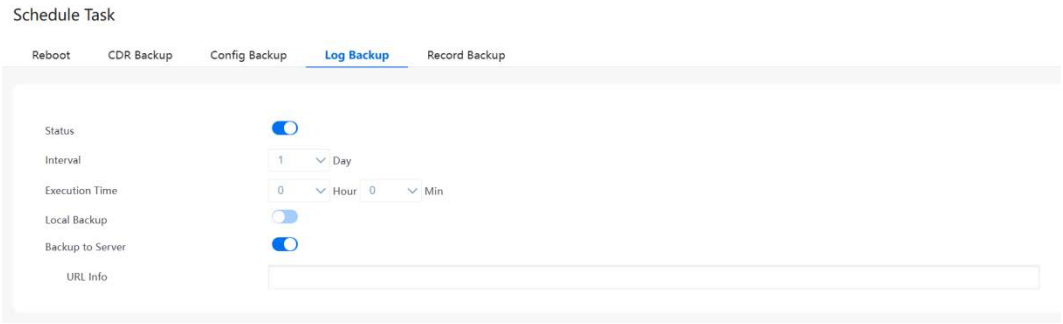
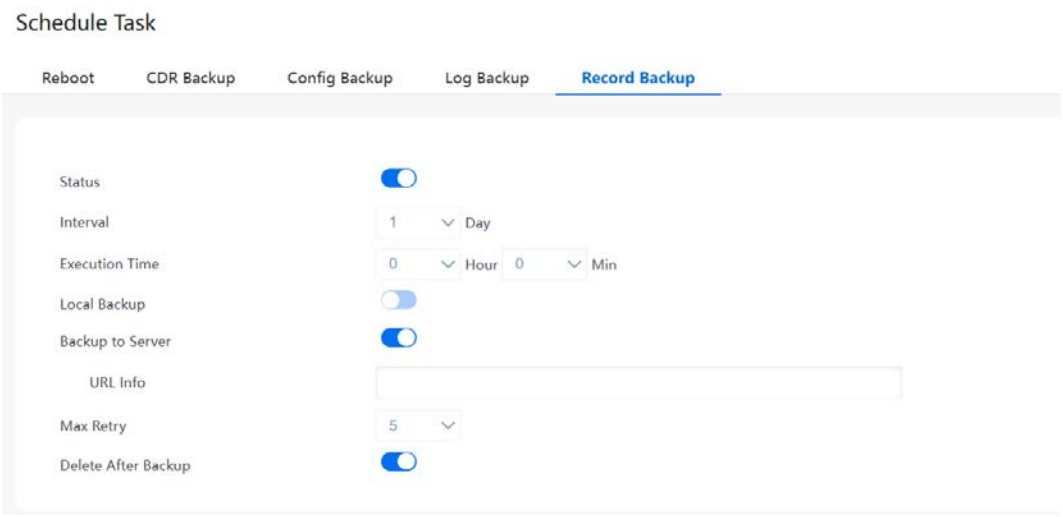


Figure-Parameters of Record Backup

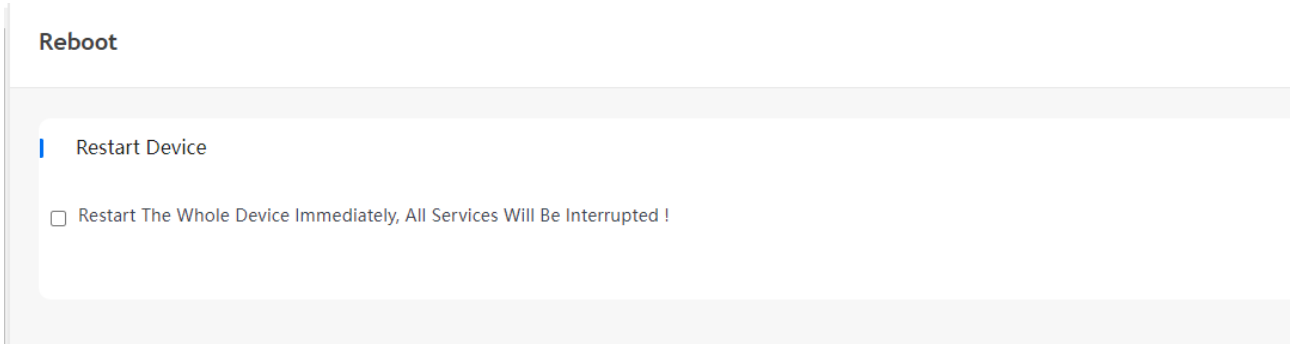


5.9.6 Reboot

On the **Maintenance > Reboot** interface, users can click Restart Device to reboot the Uc8000 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



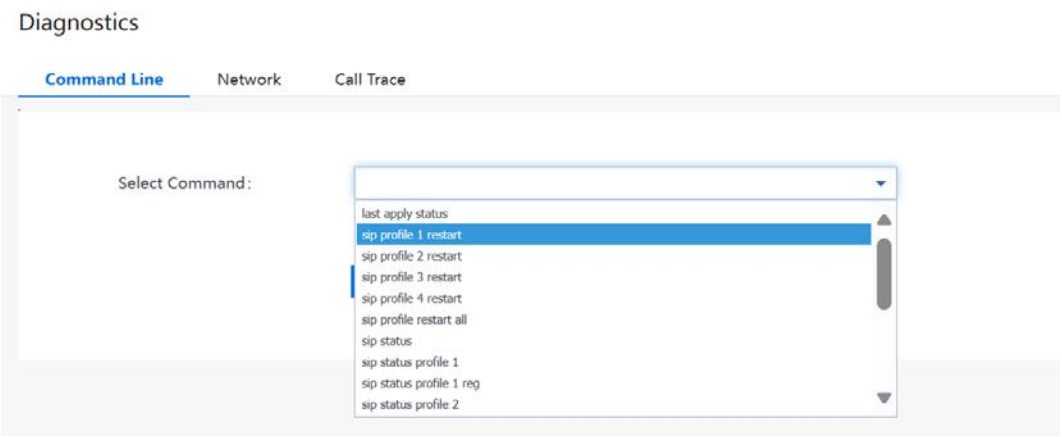
5.9.7 Diagnostics

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】

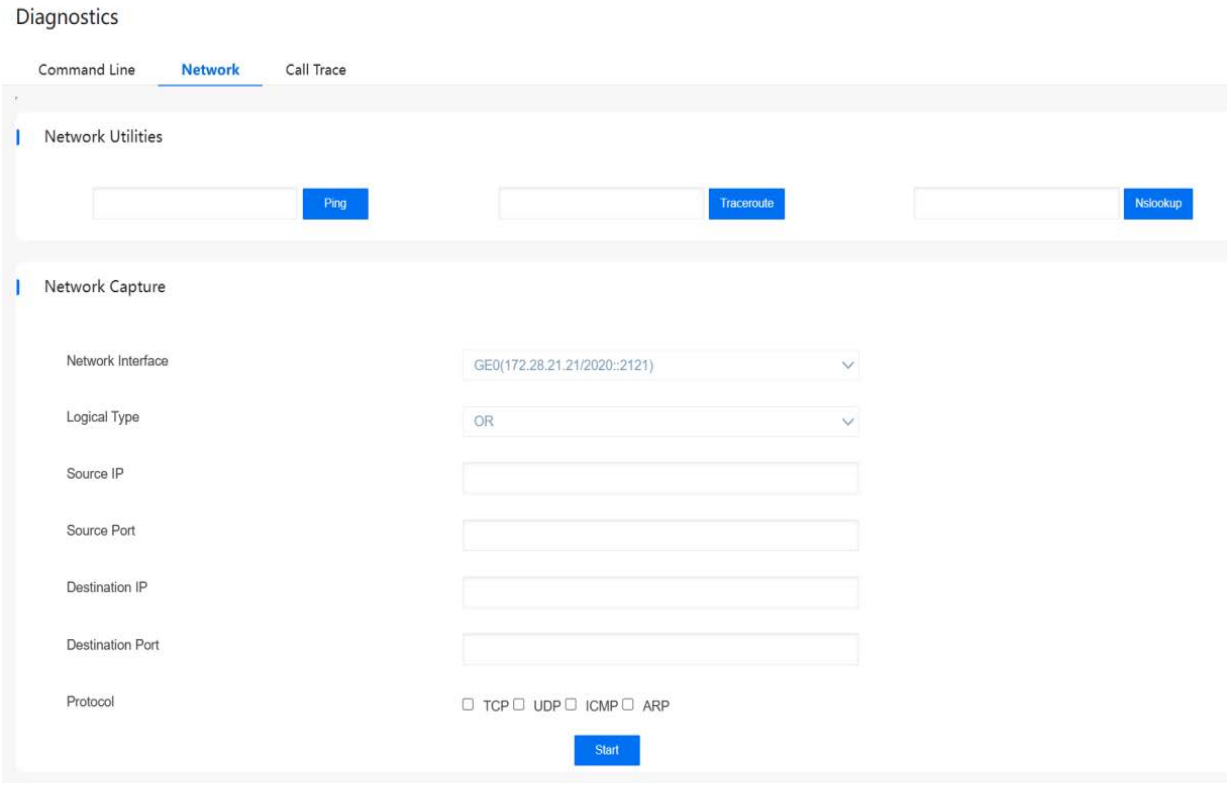
On the following interface, users can capture data packages of the available network ports. Users can also set source IP, source port, destination IP or destination port to capture the packages that users want.

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



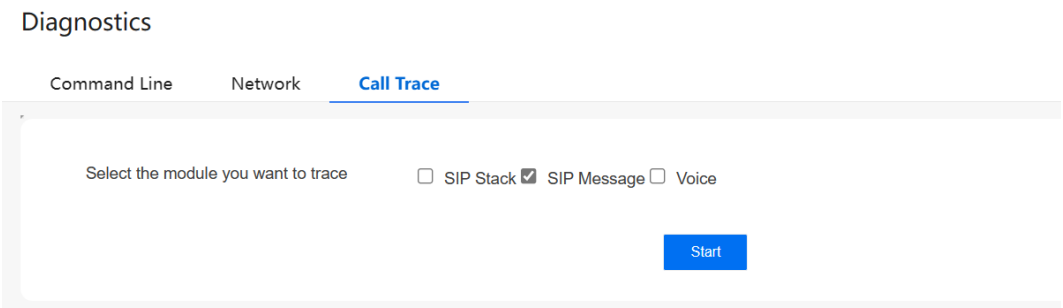
Call Trace

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.
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



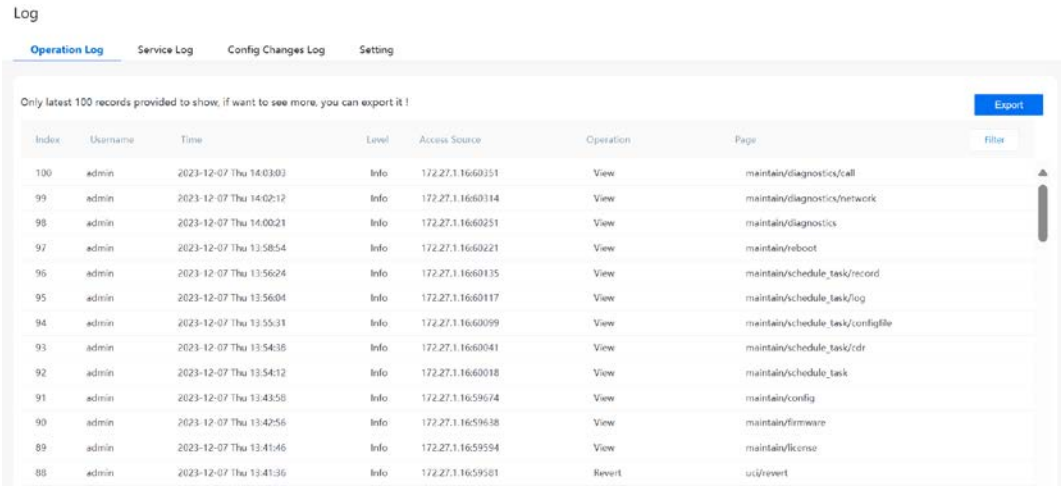
5.9.8 Log

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.

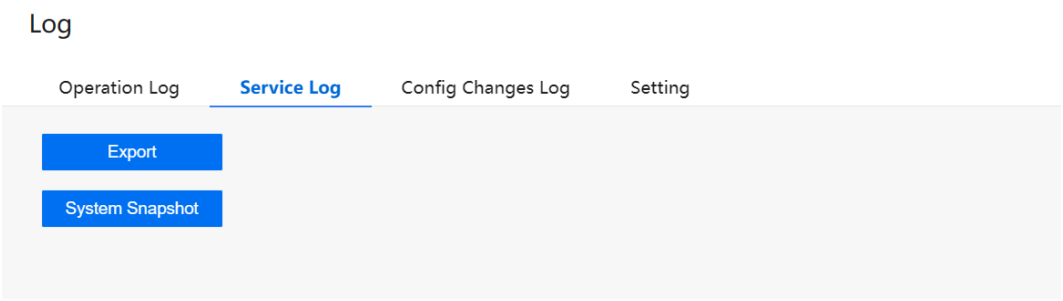
Figure-Parameters of Operation Log



Service Log

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

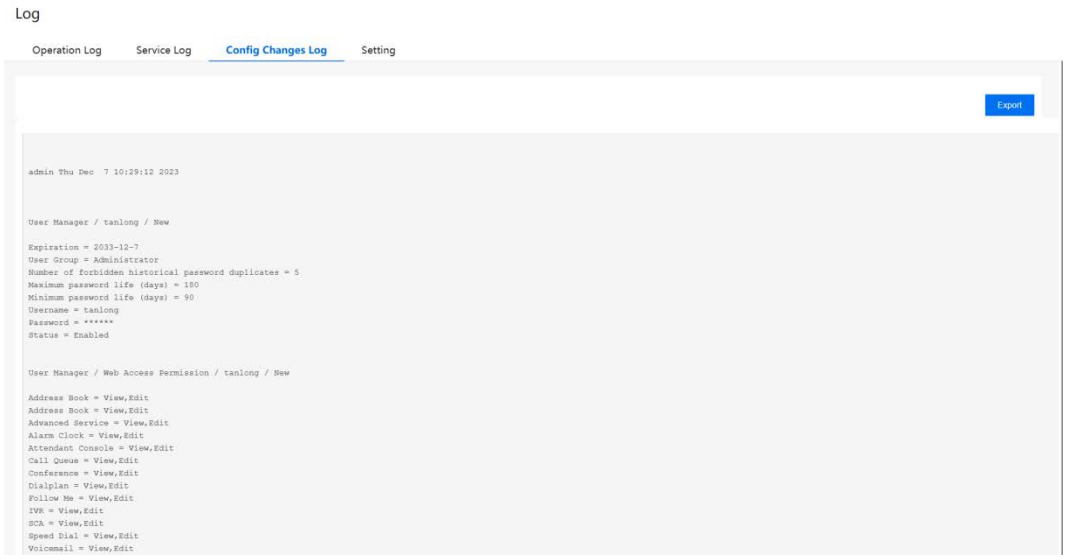
Figure-Parameters of Service Log



Config Changes Log

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

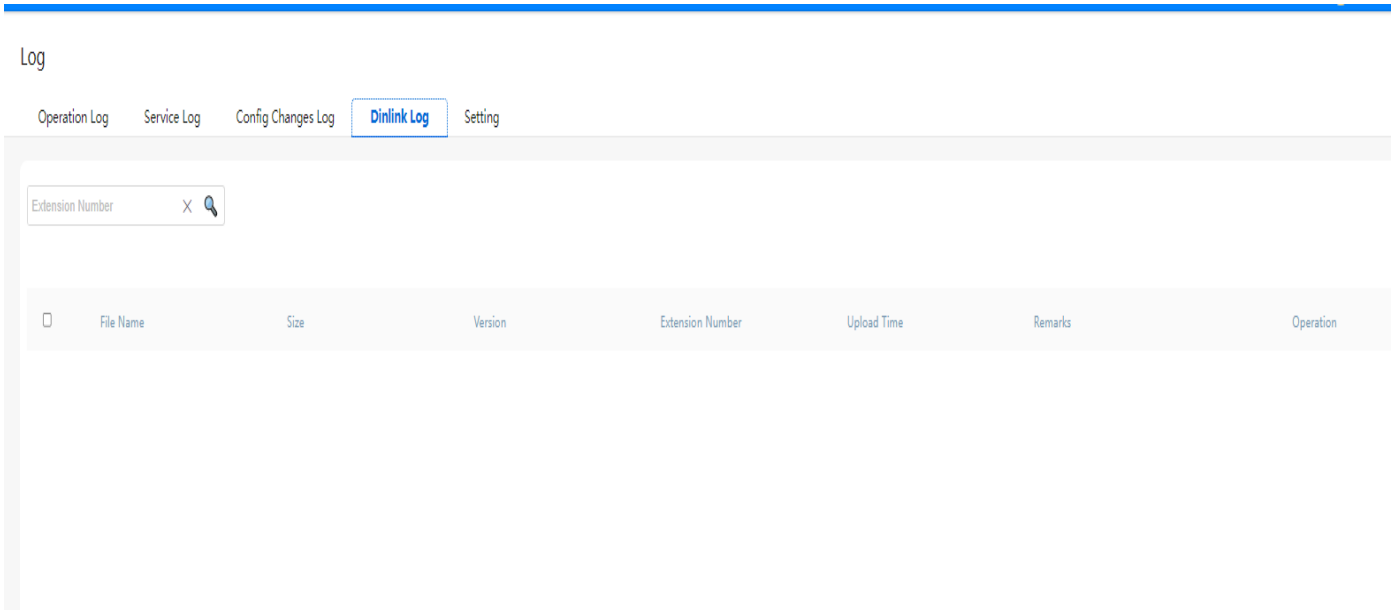
Figure-Parameters of Config Changes Log



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.

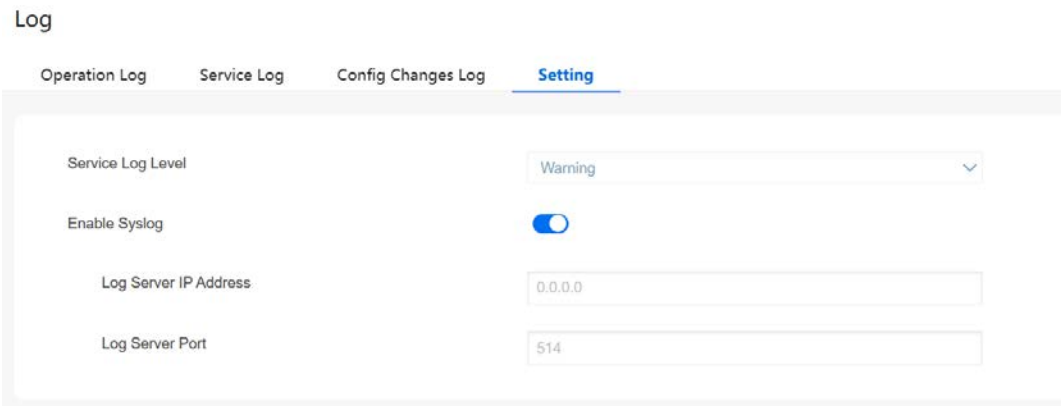
Figure-Parameters of Dinlink log



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.

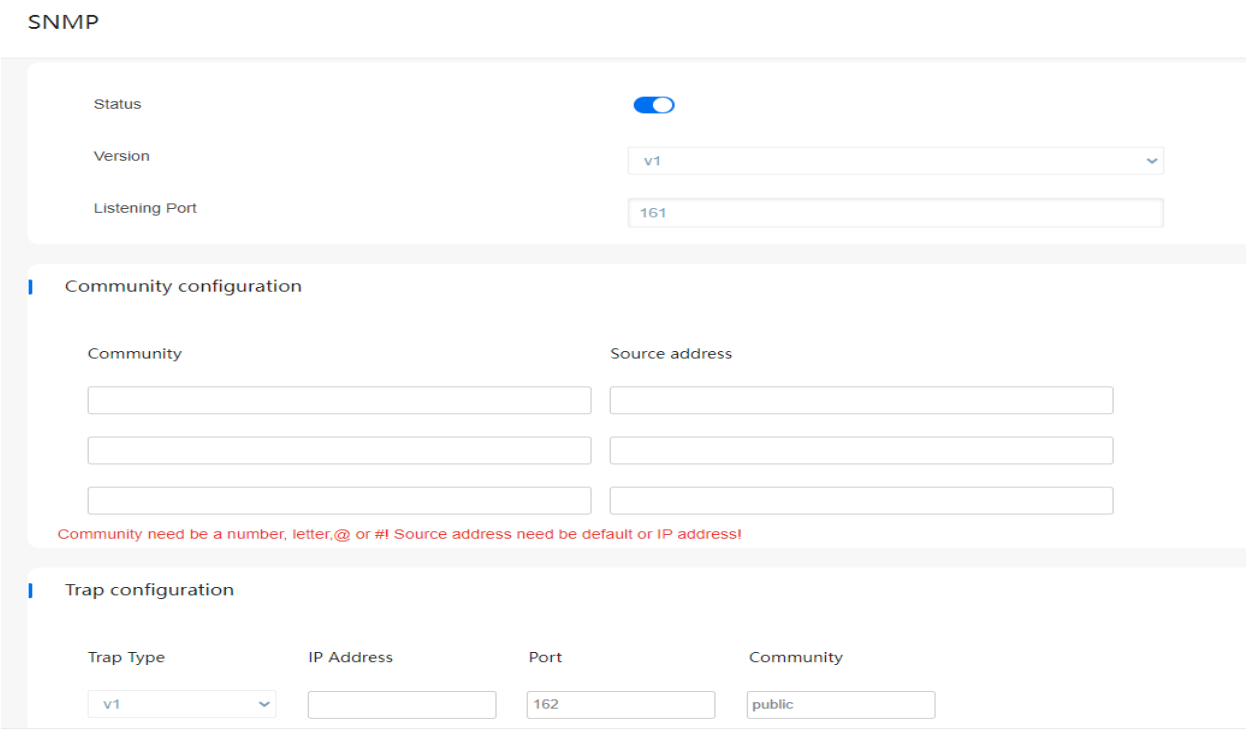
Figure-Parameters of Setting



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.

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.

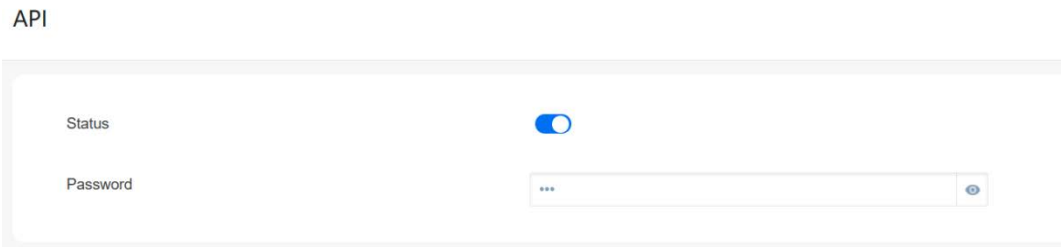
User configuration	**Username:** To configure auth username when SNMP version is v3. **Auth Type:** Supports MD5 and SHA. **Security Level:** Supports authPriv or authNoPriv. **Verify Password:** To configure the auth password. **Encryption Type:** Supports DES, AES and AES128. **Encryption Password:** To set encryption password.

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.

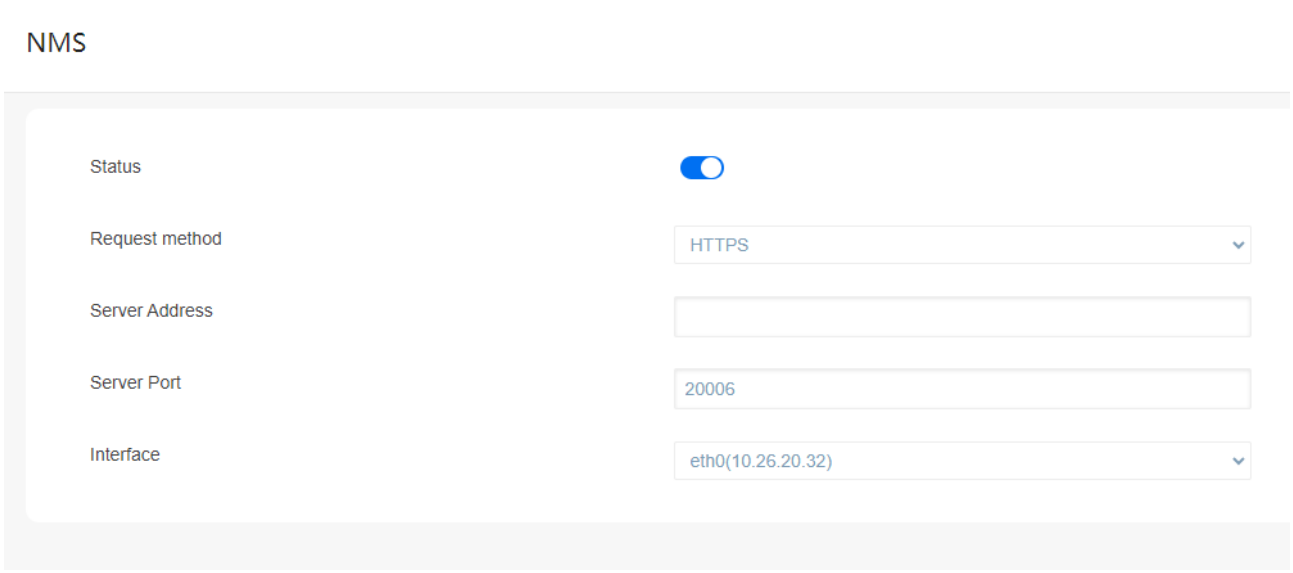
Figure-Parameters of API



5.10.2 NMS

Uc8000 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

Uc8000 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.

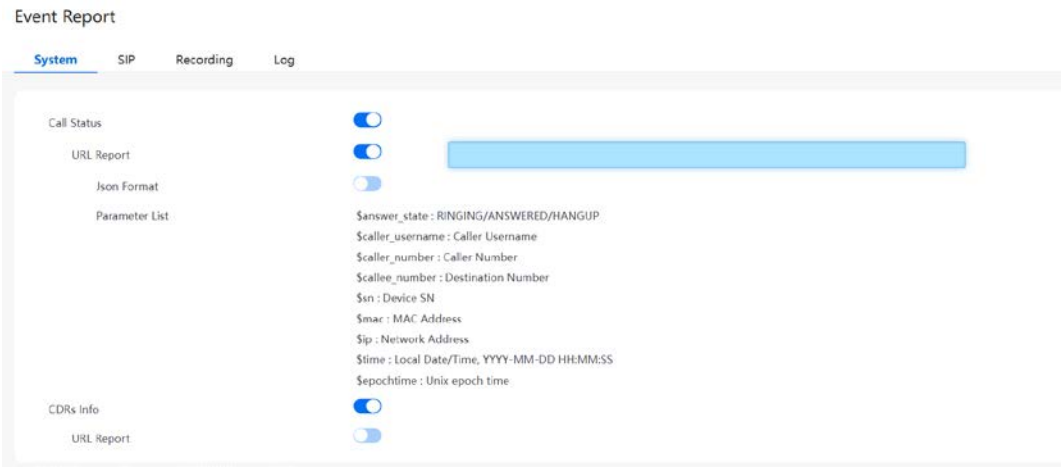
   Format:

   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、sip trunk、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 uc8000 IPPBX, and then the registration or deregistration of the softphone will be reported to uc8000 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
TCP	Transmission Control Protocol

Abbreviation	Explanation
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