UC350 Universal Gateway

User Manual V1.0

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Preface

Welcome

Thanks for choosing the UC350 Universal Gateway! We hope you will make full use of this rich-feature gateway. Contact us if you need any technical support: 0755-61919966

About This Manual

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

Intended Audience

This manual is aimed primarily at the following people:

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

Revision Record

Document Name	Document Version	Firmware Version
UC350 Universal Gateway User Manual V1.0	V1.0	

Conventions

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

Contents

1 Product Introduction

1.1 Overview

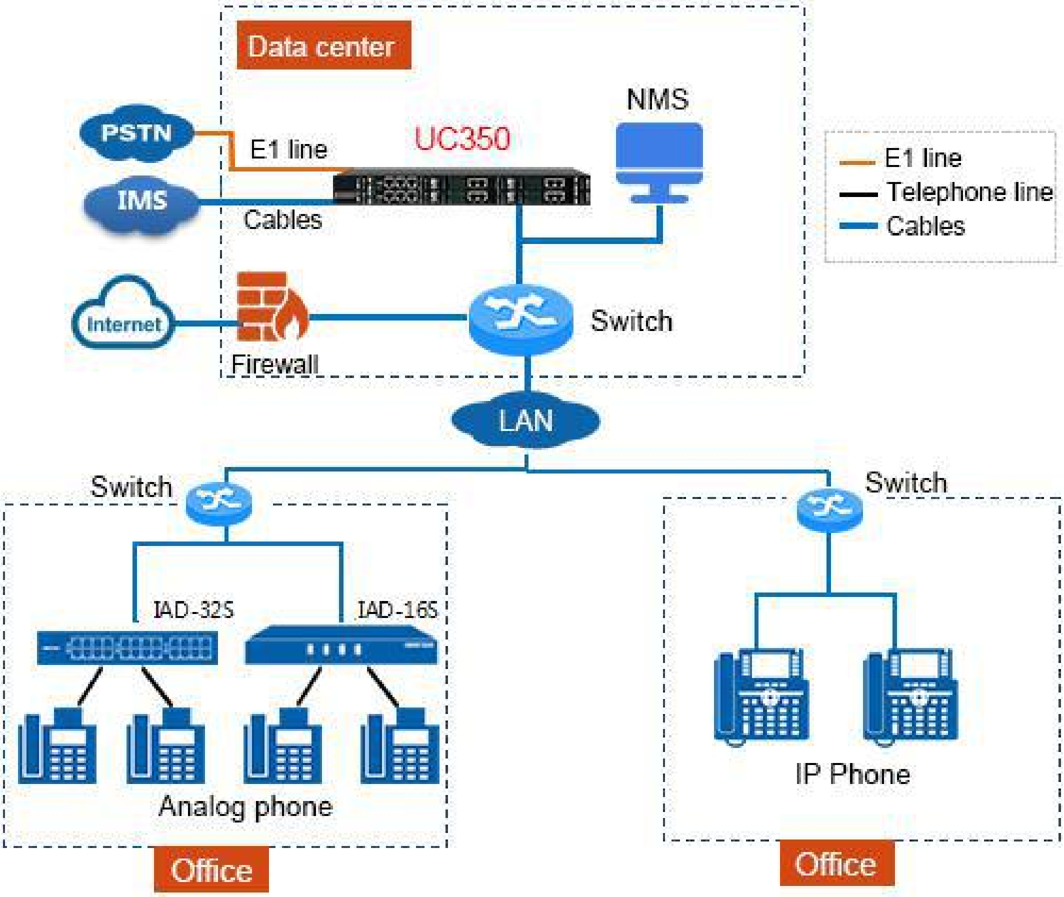
Dinstar UC350 is a new generation of large capacity unified communication. Based on a powerful hardware platform, It supports 1000 extensions and 200 concurrent calls which is integrated voice, video, paging, fax, conference, recording and other useful functions. It also provides four slots,which can be hot-plugged to install E1/T1 boards, FXS, and FXO boards, and can be flexibly configured and combined according to actual usage scenarios. It is not only suitable for helping to build the telephone system of large and medium-sized enterprises, but also to meet the needs of branch offices of large group enterprises and government agencies, and help enterprises and industry customers to establish convenient and efficient IP telephone system.

In addition, UC350 provides an intuitive and easy-to-use graphical interface for user management and maintenance.With the open API, uses can easily integrate with the third-party CRM systems and dispatching platform to provide enterprises with convenient and intelligent one-stop telephone solutions.

1.2 Application Scenario

The application scenario of UC350 universal gateway is shown as follows:

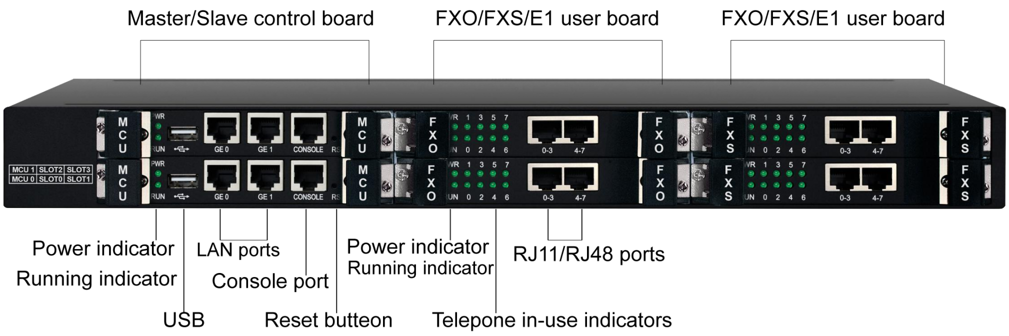
Figure 1- 2 Application Scenario of UC350



1.3 Product Appearance

Back View:

Front View:



Note: The installation of three types of user boards (FXO/FXS/E1) is supported. If the installation type is E1 user board, the RJ48 ports is connected to the E1 line; if the installation type is FXS user board, the RJ11 ports is connected to the telephone wire; if the installation type is FXO, the RJ11 ports is connected to the PSTN line.

No.	Definition	Description
1	Power switch	Power switch button
2	Power jack	Power interface: 100~240VAC, 50~60HZ
3	USB	Provide an outgoing USB interface, through which USB devices can be connected
4	Console port	Console port for debugging and configuring the device
5	CE0/CE1	Connect to IP network via DSL modem or router or LAN switch

1.4 Description of User Boards

Type	Indicator	Definition	Status	Description
Master/Slave control board	PWR	Power Indicator	Off	There is no power supply or power supply is abnormal.
			On	The device is powered on.
	RUN	Running Indicator	Flashing	The device is initialized successfully and running normally
			On	The system is initializing.
			Off	The device is not running normally.
FXS/FXO user board	PWR	Power Indicator	On	The power supply is normal
			Off	The power supply is not normal
	RUN	Running Indicator	Off	The system is starting up
			Fast Flashing	Part of the port registered successfully
			Slow Flashing	All ports are registered
	FXS/FXO	FXS/FXO Indicator	On	The FXS port is in off-hook (in-use) status.
			Off	The FXS port is in on-hook status.
E1 user board	PWR	Power Indicator	Off	There is no power supply or power supply is abnormal.
			On	The device is powered on.

	RUN	Running Indicator	Slow Flashing	The device is initialized successfully and running normally
			On	The system is initializing.
			Off	The device is not running normally.
	E1	E1 Indicator	On	E1 line is connected
			Off	E1 line E1 is disconnected

1.5 Features & Functions

1.5.1 Key Features

- Supports up to 1000 SIP users and 200 concurrent calls
- Supports E1/T1, FXO and FXS ports with the flexible and alternative capability
- Distributed multi-core CPU, greatly improves the call processing capacity
- Flexible dial rules based on time, number or source IP etc.
- Supports Multi-level IVR, helps to build personalized voice navigation for enterprise
- Built-in VPN server/client
- Support voicemail/ Voice recording
- User-friendly web interface, classification of web user's rights

1.5.2 FXS

- Connector: RJ45
- Caller ID: Bellcore Type 1&2, ETSI,BT,NTT and DTMF
- Answer and Disconnect Signaling: Answer, Disconnect, Busy Tone
- Polarity Reversal
- Hook Flash

1.5.3 FXO

- Connector: RJ45
- Caller ID: FSK, DTMF
- Polarity Reversal
- Answer Delay
- Detection of Busy Tone
- Detection of No Current
- Auto Match of FXO Impedance

1.5.4 PSTN

- 4* E1/T1 Ports at max
- Interface: RJ48 (120Ohm)
- ISDN PRI:
- 23B+D(T1),30B+D(E1),NT or TE
- ITU-T Q.921, ITU-T Q.931, Q.Sig
- Signal 7/SS7:
- ITU-T, ANSI， ITU-CHINA
- MTP1/MTP2/MTP3, TUP/ISUP
- R2 MFC

1.5.5 Voice Capabilities

- VoIP Protocols: SIP over UDP/TCP/TLS,RTP/SRTP
- Codecs: G.711,G.723.1, G.729A/B, G.722,G.726, OPUS
- Silence Suppression
- Comfort Noise Generator(CNG)
- Voice Activity Detection(VAD)
- Echo Cancellation: G.168 with up to 128ms
- Dynamic Jitter Buffer
- Adjustable Gain Control
- Automatic Gain Control （AGC）
- Call Progress Tones: Dial Tone, Ring Back Tone, Busy Tone
- FAX: T.38 and Pass-through
- NAT: STUN/UPnP
- DTMF: RFC2833/Signal/Inband
- VPN Server/Client

1.5.6 PBX Services

- Call Forward (Unconditional/No Answer/Busy)
- Call Waiting
- Call Holding
- Call Transfer
- Hotline
- Do-not-disturb
- 3-Way Conference

- Ring Group
- Call Queue
- Routing Groups
- Caller/Called Number Manipulation
- Routing Based on Time Period
- Routing Based on Caller/Called Prefixes
- Routing Based on Source Trunks
- Dial Rules
- Failover Routing
- Multi-level IVR
- Auto-attendant Function
- CDRs
- Voicemail
- Voice Recording
- Up to 1000 SIP Extensions
- Up to 200 Concurrent Calls
- Paging
- Event Report
- Email Client
- Voicemail to Email

1.5.7 Environmental

- 1+1 Power Supply, 100-240VAC, 50-60 Hz
- Power Consumption: 50W
- Operating Temperature: 0 ℃ ～45 ℃
- Storage Temperature: -20 ℃ ~80 ℃
- Humidity: 10%-90% Non-Condensing
- Dimensions (W/D/H): 437×345×49mm
- Unit Weight: 5.7 kg

1.5.8 Maintenance

- Web GUI Configuration
- Telnet Management
- Configuration Restore/Backup
- Multiple Languages Supported

- HTTP/TFTP/FTP Firmware Upgrade
- Auto Provision
- Syslog
- Ping and Tracert
- Traffic Statistics: TCP,UDP,RTP
- Network Capture
- NTP
- Classification of Web Users' Rights
- HTTP&HTTPS/NATS API

2 Installation

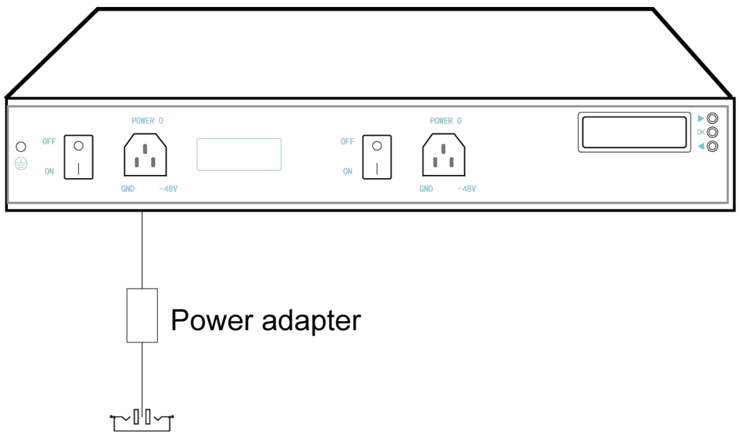
2.1 Installation Attentions

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

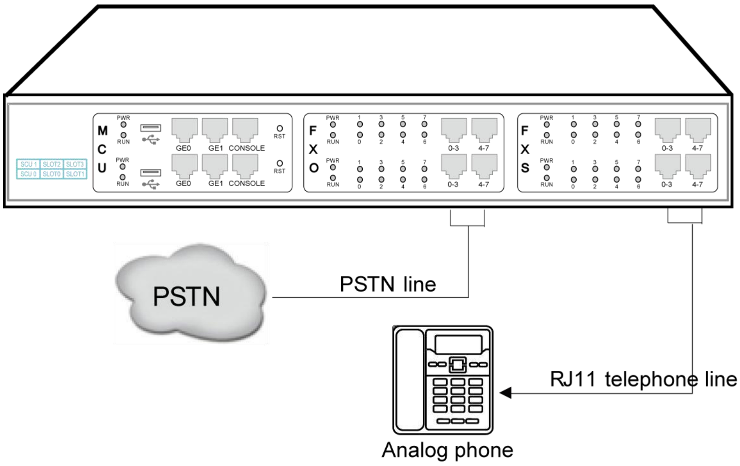
- The adapter of the gateway accepts DC220V 10A dual power input. Please ensure stable and safe power supply;
- To reduce the interference to telephone calls, please separate power cables from telephone lines;
- To guarantee stable running of the gateway, please make sure that there is enough network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

2.2 Installation Steps

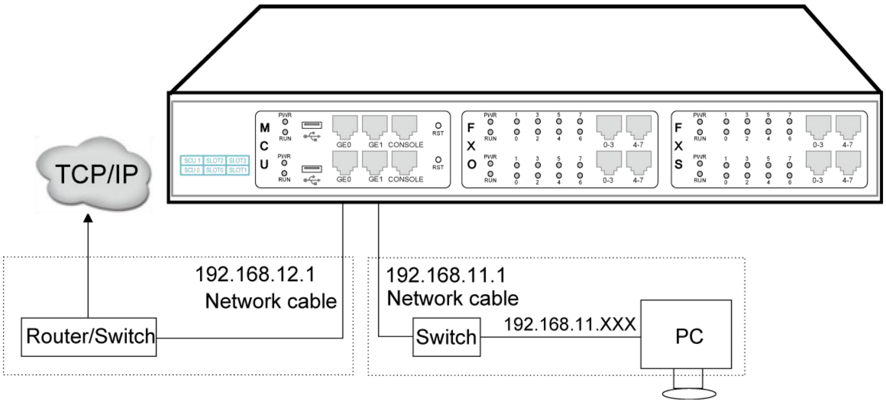
1) Connect the power adapter to the power jack;



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

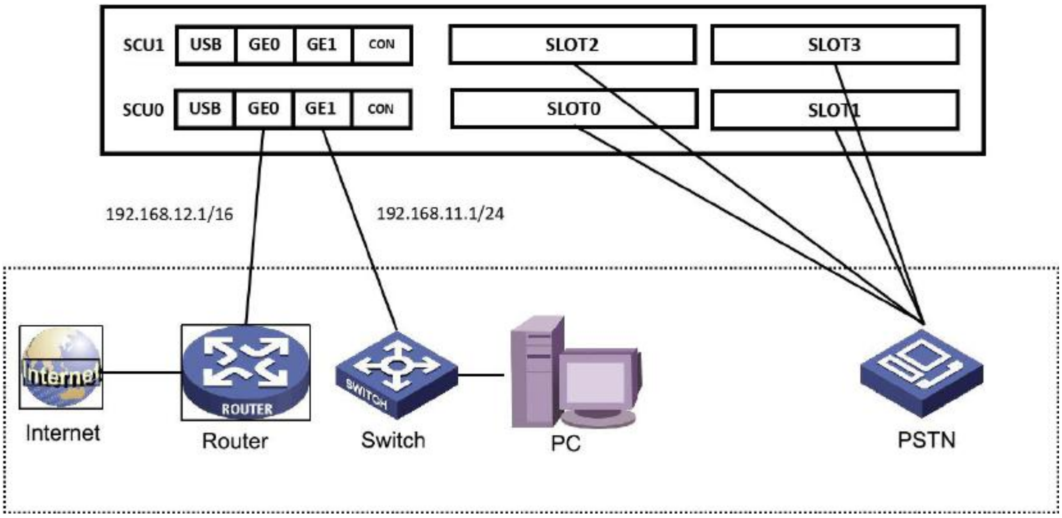


3) Connect network cable to the GE0 port, and the GE1 port is connected to the PC.



2.3 Network Connection

UC350 provides 3 RJ45 ports, of which GE1 port is the management port, GE0 port is service port, which is mainly used to access the network. GE0 port is a static IP address and needs to be configured with an IPv4 address in the same network segment as the uplink, as shown in Figure: 2.3 Network Connection.

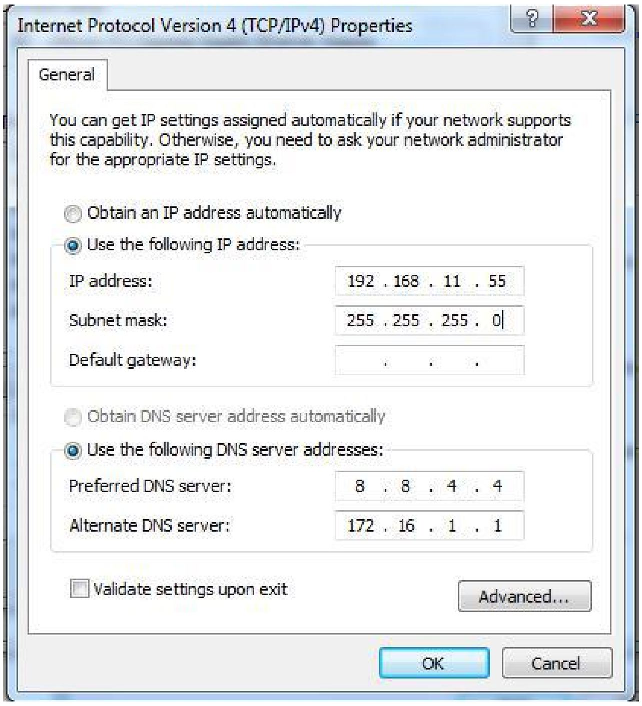


2.4 Connect Gateway to Network

2.4.1 Preparations for Login

Modify the IP address of the PC to make it at the same network segment with the UC350 gateway, since the default IP address of GE1 port is 192.168.11.1.

Take the Windows 7 operating system as an example, make the IP address of the local computer and the gateway to be in the same network segment. Figure 2.4.1 Modify the local computer address.



Check the connectivity between the PC and the UC350. Click **Start → Run** of PC and enter **cmd** to execute 'ping 192.168.11.1' to check whether the IP address of GE1 port runs normally.

2.4.2 Log In Web

Open a browser and enter the IP address of GE1 port (the default IP is 192.168.11.1). Then the login GUI will be displayed.

You also can enter the IP address of GE0 port, but it's required to modify the IP address of PC to make it at the same network segment with that of GE0 port.

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

Figure 2.4-2 Login GUI of UC350

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

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

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

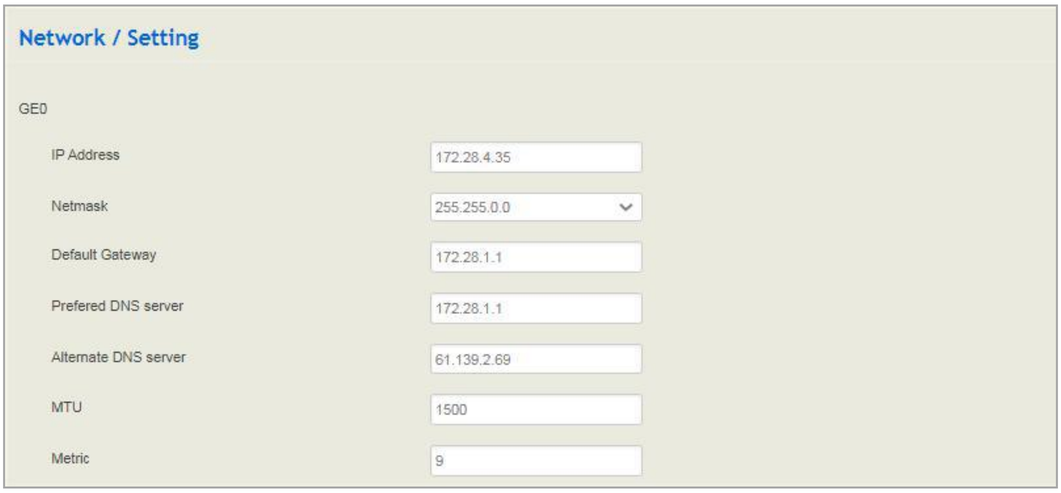
2.4.3 Modify the IP address of GE0 port

Connect the UC350 to the network(refer to figure 2.3 network connection), but the GE0 port of the UC350 defaults to a static IP address: 192.168.12.1. At this time, you need to modify the IP address of the GE0 port so that the GE0 port and the upstream network are in the same network segment. There are two ways to modify the IP address of the GE0 port:

Method 1:

As shown in methods 2.4.1 and 2.4.2, log in to the Web of the gateway and navigate to "**Network -> Settings"** to modify the IP address.

Figure 2.4-3 Modify IP address via Web



Method 2:

Step 1: Connect the phone to the FXS port of the user board with a telephone line.

Step 2: Modify the IP address of GE0 port by dialing *152, for example: dial *152*192*168*1*10# to set the IPv4 address to 192.168.1.10.

Step 3: Modify the gateway IP address of GE0 port by dialing *156, for example: dial *156*192*168*1*1# to set the IPv4 gateway to 192.168.1.1.

Step 4: Modify the mask of the GE0 port by dialing *153, for example: dial *153*255*255*0*0# to set the IPv4 mask to 255.255.0.0.

Step 5: Restart the device by dialing *111, the IP address of the GE0 port is the modified IP address.

3 Basic Operation

3.1 The Methods of Dialing

There are two methods to dial telephone or extension number:

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

3.2 Call Holding

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

3.3 Call Waiting

When call waiting is enabled, if you hear the call waiting voice (three beeps of the FXS extension) 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 Instruction of Hook Flash

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

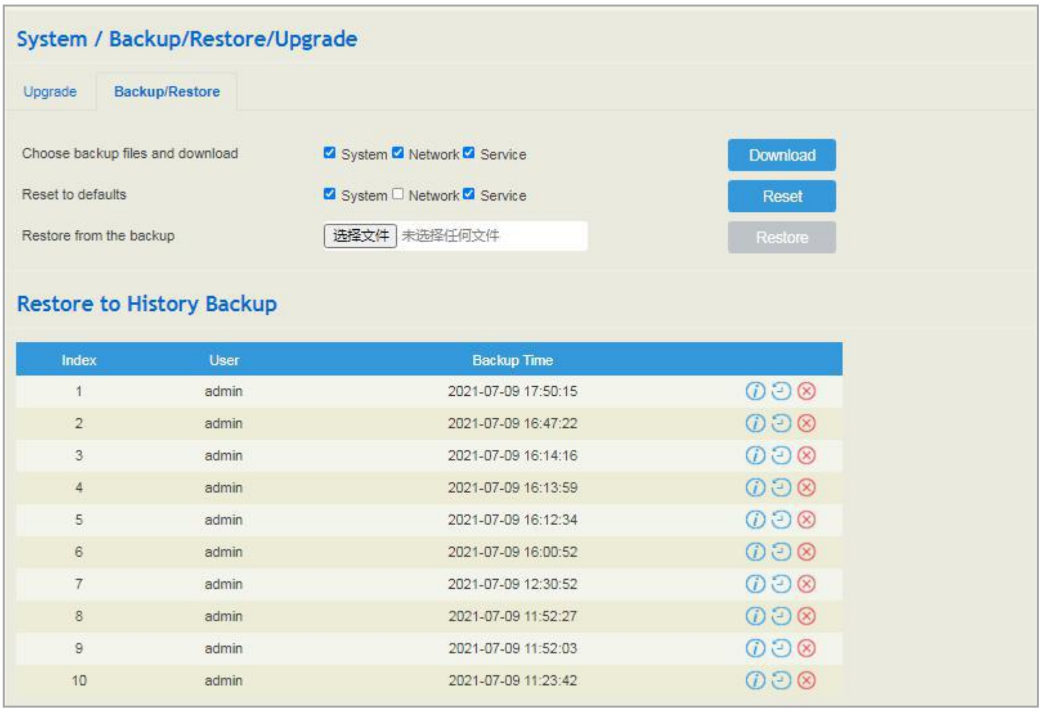
3.5 Query IP Address and Restore Default Settings

After connecting the phone to the FXS port of the gateway with a telephone line, dial feature code *158 to query the IP address of the GE1 port, and dial feature code *159 to query the IP address of the GE0 port.

If you want to restore UC350 to default settings, you can press the **RST** button for 6 to 12 seconds or you can configure it on the Web interface.

After logging in to the Web, navigate to **System → Backup/Restore/Upgrade**, select the parts (system, network or service) that need to be restored in the "Restore to default settings" option,click **Reset** to restart the device, and the selected parts will be restored to default settings.

.Figure 3.5 Restore default settings



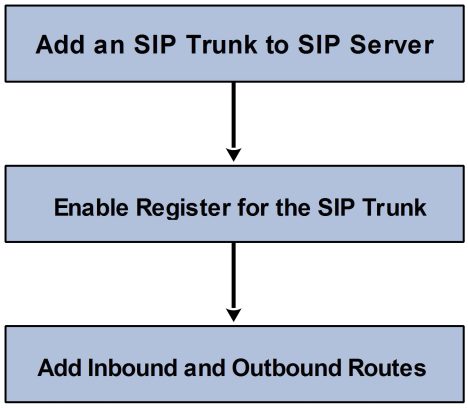
4 Configuration Wizard

4.1 Configuration Wizard

The following are the common ways to configure the UC350 gateway.

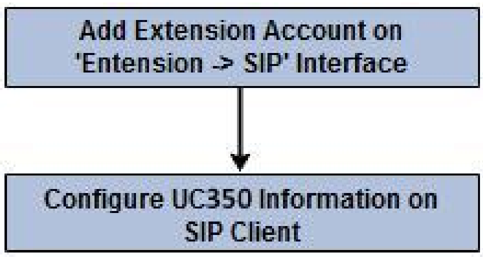
4.1.1 UC350 Regarded as Terminal and Registered to SIP Server

1. UC350 Registered to SIP Server

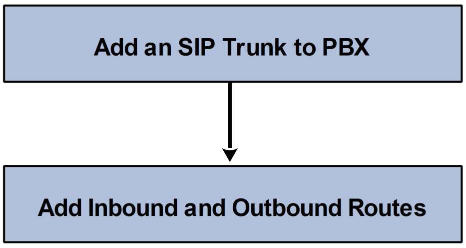


4.1.2 Other SIP Clients registered to UC350

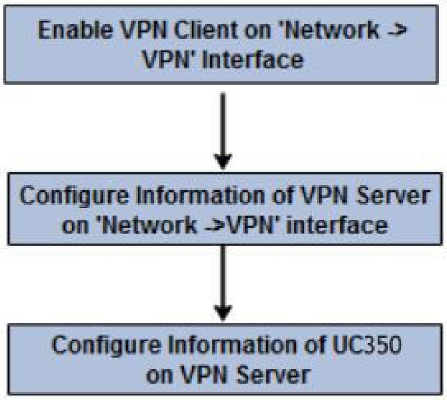
Under this mode, UC350 is regarded as an SIP Server. Create an extension account first on the **Extension →SIP** interface, and configure listening port on the **Profile → SIP** interface. Then, configure the server and account on SIP client.



4.1.3 UC350 Connected to PBX through Trunking



4.1.4 UC350 Serving as VPN Client



5 Web Platform

5.1 Introduction to Web Interface

Modify the IP address of PC to make it at the same network segment with that of GE1 port of the UC350 gateway (the default IP of GE1 port is 192.168.11.1).

Open a web browser on the PC and then enter the IP address of GE1 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 5- 1 Introduction to login GUI

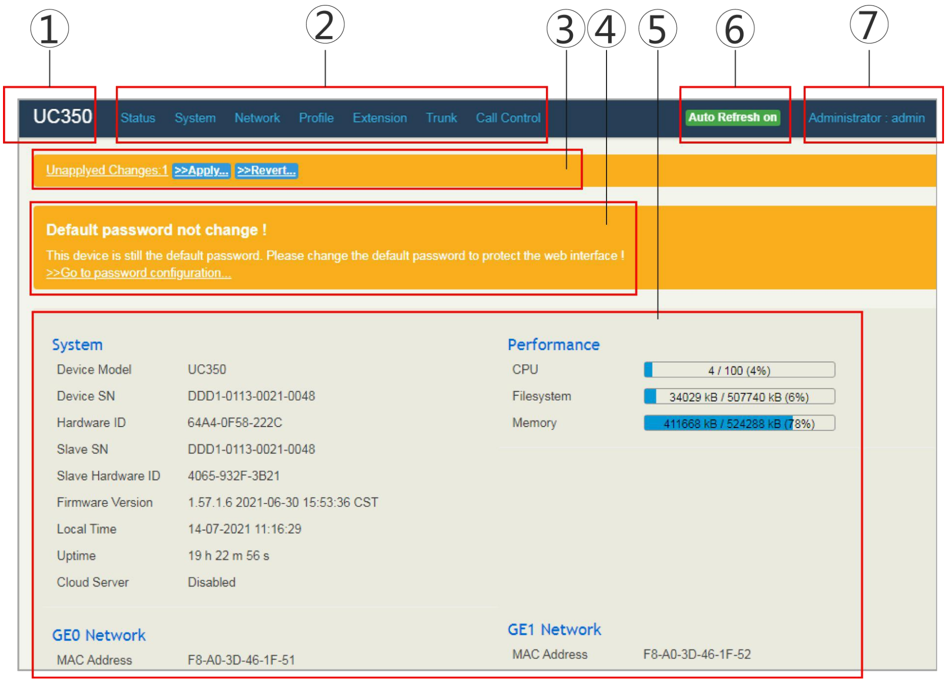


Table 5- 1 Introduction of Web Interface

Index	Item	Description
1	UC350	The name of the gateway; it can be edited on the **System → Setting** interface.
2	Menu Bar	The menu bar of UC350
3	Unsaved Changes	All changes to the configuration of the gateway need to be saved. Click **Apply** to enter into the interface to save the changes; click **Revert** to return to original configuration.

4	Password modification	It will be prompted when the default password of the admin account is not changed, and it will not be prompted after the default password is changed.
5	Detailed Interface	The detailed configuration interface or display interface
6	Auto Refresh Button	The button can be enabled or disabled. If it is enabled, the information on the **Status → Overview/SIP/PSTN/Current Call** interfaces will be refreshed automatically
7	User Role	The role of the current user logging into the Web. And the "exit" sign will pop up when the mouse moves over there. You can log out of the web from there

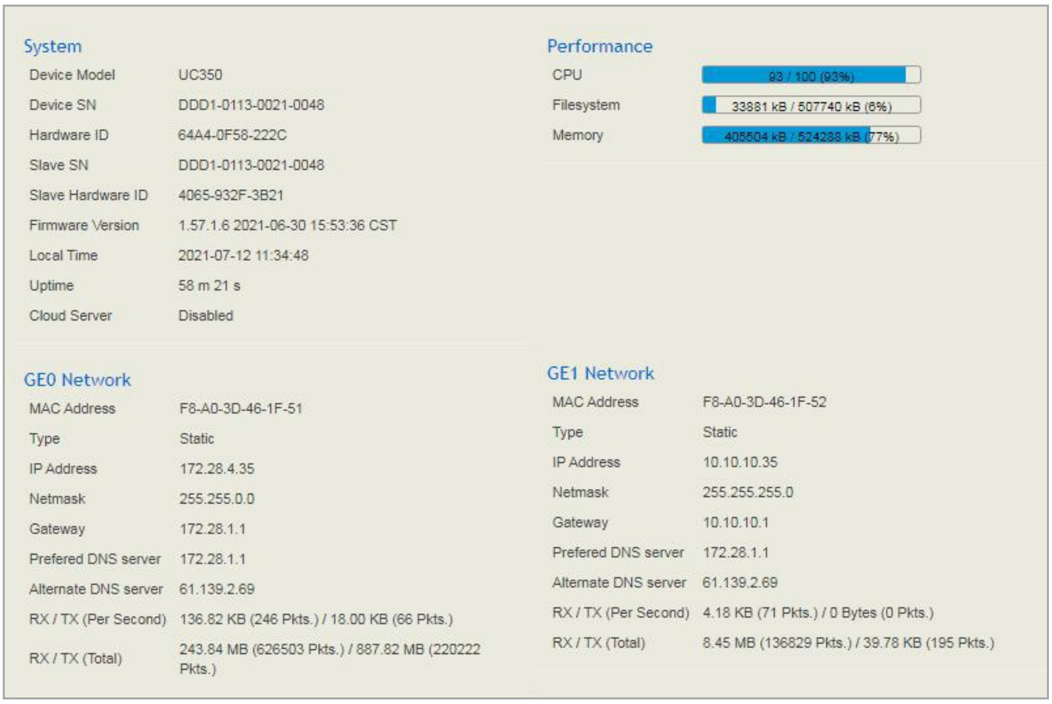
5.2 Status

The 'Status' menu mainly displays all kinds of status information. It includes the following sub-menus: Overview, SIP, PSTN, Fail2ban, Current Call, CDRs, Service, performance and About etc.

5.2.1 Overview

Log in the Web interface of UC350, click **Status → Overview**, and the following interface will be displayed. On the interface, device model, firmware version, device running time, Mac address, IP address, performance parameters, etc.

Figure 5.2-1 Overview



5.2.2 SIP

Click **Status → SIP**, and the following interface will be displayed. On the interface, information of SIP profile, SIP Trunk and SIP extension is shown.

Figure 5.2-2 Status of SIP Profile, SIP Trunk and SIP Extension

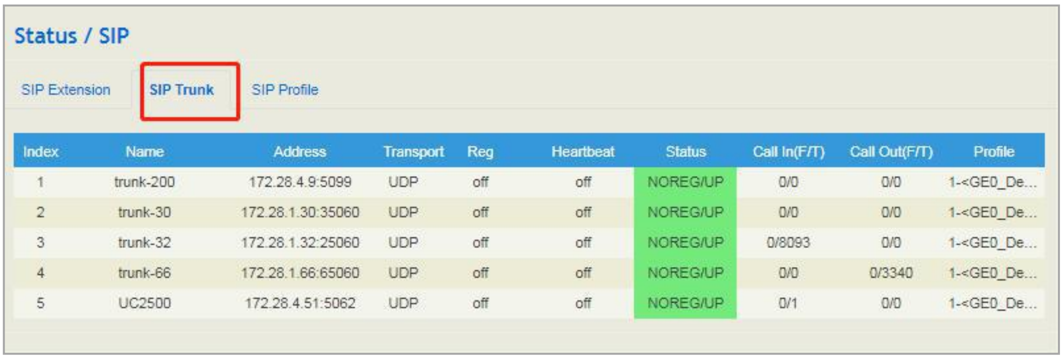


Table 5.2.2 Explanation of SIP Parameters

Belong To	Parameter	Explanation
Profile	Name	The name of the SIP profile
	Listening Address	The current listening address and port of SIP
	State	Green color means normal running, while red color means listening address and port of SIP is unavailable. There are two states :Running and Down
SIP Trunk	Heartbeat	If heartbeat is enabled, option message will be sent to peer device (the peer device is reachable)
	Status	Green color means available, while red color means abnormal, unavailable or prohibited. There are five statuses: Running, Reged/Up, Noreg/Up, Trying-Down, Fail-Wait
	Profile	The profile that is used by the SIP trunk
SIP Extension	Profile	The profile that is used by the SIP extension
	Status	SIP extension is registered or not. There are two statuses: Registered and Unregistered

5.2.3 PSTN

On the **Status → PSTN** interface, the configuration status of the FXS port, FXO port and E1 port, the line state and hook status, and the channel status of the line and E1 port.

Figure 5.2-3 FXS port\FXO port status and E1 channel status

FXS

Slot	Port	Exension Number	Hook State	Line State
0	0	300	ONHOOK	IDLE
0	2	301	ONHOOK	IDLE
0	4	302	ONHOOK	IDLE
0	6	303	ONHOOK	IDLE
1	0	310	ONHOOK	IDLE
1	2	311	ONHOOK	IDLE
1	4	312	ONHOOK	IDLE
1	6	313	ONHOOK	IDLE

FXO

Slot	Port	Number	Hook State	Line State
0	1		OFFLINE	IDLE
0	3		OFFLINE	IDLE
0	5		OFFLINE	IDLE
0	7		ONHOOK	IDLE
1	1		OFFLINE	IDLE
1	3		OFFLINE	IDLE
1	5		OFFLINE	IDLE
1	7		OFFLINE	IDLE
2	0		ONHOOK	IDLE
2	1		ONHOOK	IDLE

E1 Trunk

Slot	Port	Port Status	Channel Status
			0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31

NOTES	Activated	Disable	Not Authorized	LOS Alarmd
	RAI Alarm	AIS Alarm	ISDN/SS7 Signal Alarm	Auto Closed
	Frame-Sync	IDLE	Signal	Progress
	Ring	Talk	Release	Fault
	Disable	L-blocked	R-blocked	B-blocked

5.2.4 Fail2ban

On the **Status →Fail2ban** interface, you can see currently-banned IP addresses and historic banned IP addresses. You can also unban those IP addressed that have been blocked before.

Fail2ban is a log-parsing application that monitors system logs for symptoms of an automated attack on your device. When an attempted compromise is located, using the defined parameters, Fail2ban will add a new rule to block the IP address of the attacker, either for a set amount of time or permanently. Fail2ban can also alert you through email that an attack is occurring.

Figure 5.2-4 Banned IP Addresses


Status / Fail2ban

Current Ban List

Index	IP	Ban time	Release time	Type	Action

Operation History List

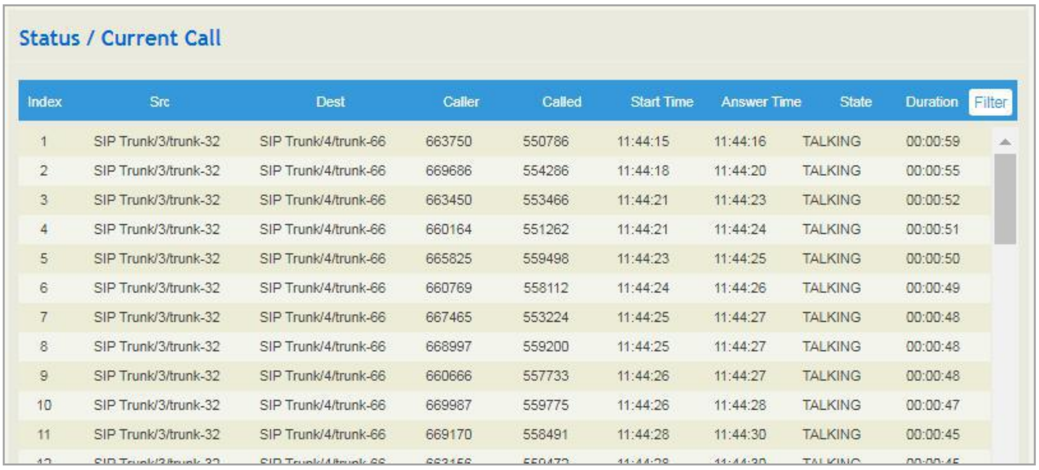
Index	IP	Common Ban Duration	Type	Action	Operation time	Filter

For the explanation of parameters related to fail2ban, please refer to the "Network ->Fail2ban" section.

5.2.5 Current Call

On **Status →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 shown

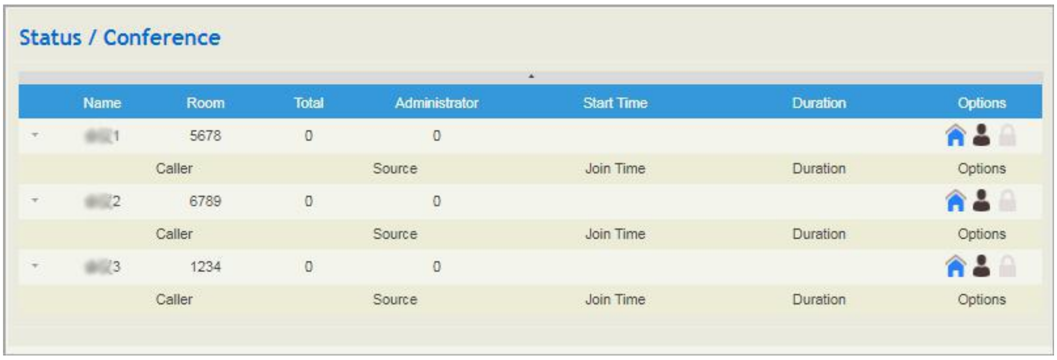
Figure 5.2-5 Current Call Information


Status / Current Call

Index	Src	Dest	Caller	Called	Start Time	Answer Time	State	Duration
1	SIP Trunk/3/trunk-32	SIP Trunk/4/trunk-66	663750	550786	11:44:15	11:44:16	TALKING	00:00:59
2	SIP Trunk/3/trunk-32	SIP Trunk/4/trunk-66	669686	554286	11:44:18	11:44:20	TALKING	00:00:55
3	SIP Trunk/3/trunk-32	SIP Trunk/4/trunk-66	663450	553466	11:44:21	11:44:23	TALKING	00:00:52
4	SIP Trunk/3/trunk-32	SIP Trunk/4/trunk-66	660164	551262	11:44:21	11:44:24	TALKING	00:00:51
5	SIP Trunk/3/trunk-32	SIP Trunk/4/trunk-66	665825	559498	11:44:23	11:44:25	TALKING	00:00:50
6	SIP Trunk/3/trunk-32	SIP Trunk/4/trunk-66	660769	558112	11:44:24	11:44:26	TALKING	00:00:49
7	SIP Trunk/3/trunk-32	SIP Trunk/4/trunk-66	667465	553224	11:44:25	11:44:27	TALKING	00:00:48
8	SIP Trunk/3/trunk-32	SIP Trunk/4/trunk-66	668997	559200	11:44:25	11:44:27	TALKING	00:00:48
9	SIP Trunk/3/trunk-32	SIP Trunk/4/trunk-66	660666	557733	11:44:26	11:44:27	TALKING	00:00:48
10	SIP Trunk/3/trunk-32	SIP Trunk/4/trunk-66	669987	559775	11:44:26	11:44:28	TALKING	00:00:47
11	SIP Trunk/3/trunk-32	SIP Trunk/4/trunk-66	669170	558491	11:44:28	11:44:30	TALKING	00:00:45

5.2.6 Conference

The "**Status -> Conference**" interface displays the current conference status, conference call information, operation records, etc.



5.2.7 Call Queue

The user can view the related information of the queue status on the "**Status -> Call Queue**" interface, the queue configuration change will restart the agent, and the dynamic agent will disappear.

click the ▽on the far left to view more detailed agent status.

5.2.8 Parking Lot

You can use the parking feature to park a call, and then retrieve the call either from your phone or another phone. After you park a call, the call is placed on hold, you can continue the conversation after retrieving it.

On the **Status -> Parking Lot** interface, the numbers that are parked and the parking duration are shown.

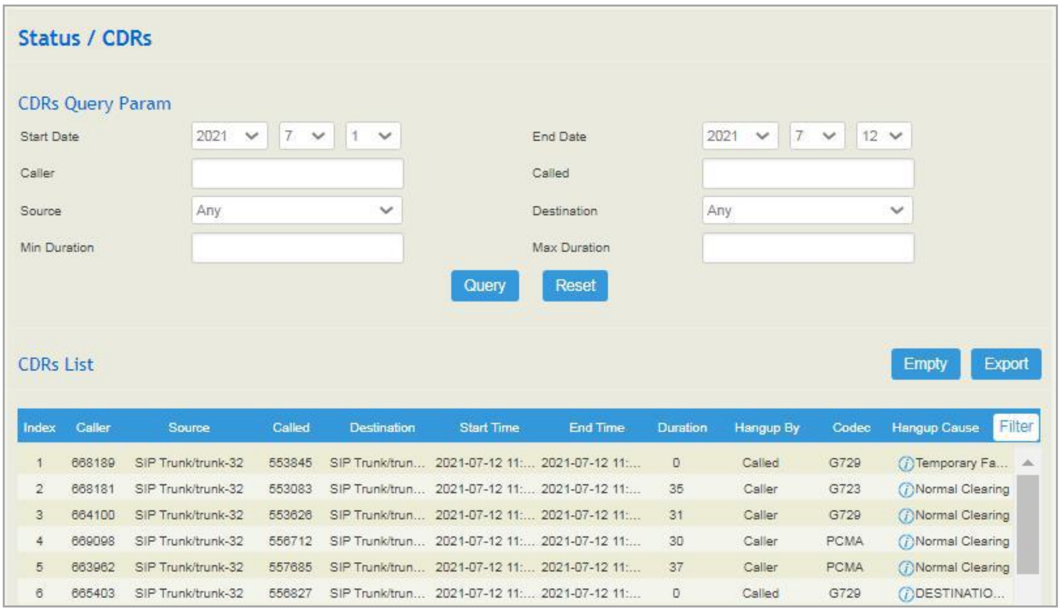
Figure 5.2-8 Call Parking Status

5.2.9 CDRs

Click **Status → CDRs,** and you can set query criteria to query the CDRs (Call Detailed Records) that you want on the displayed interface. Meanwhile, you are allowed to clear CDRs or export CDRs through clicking the **Empty** or **Export** button. The maximum number of CDRs that can be saved is 5000.

CDRs cannot be saved on the **Status → CDRs** interface unless the CDRs function has been enabled on the **System → Setting** interface.

Figure 5.2-9 CDRs



5.2.10 Service

Click **Status → Service**, and the service status of UC350 is displayed. This function is enabled by default. The Web, SSH and Telnet service can be disabled and their ports can be modified on the **Network → Access Control** interface. The remote proxy and NATS server are enabled by default, and can be disabled and modified on the "System -> Cloud Service" interface; other services cannot be disabled. If no running status is shown, it means exception has occurred on UC350

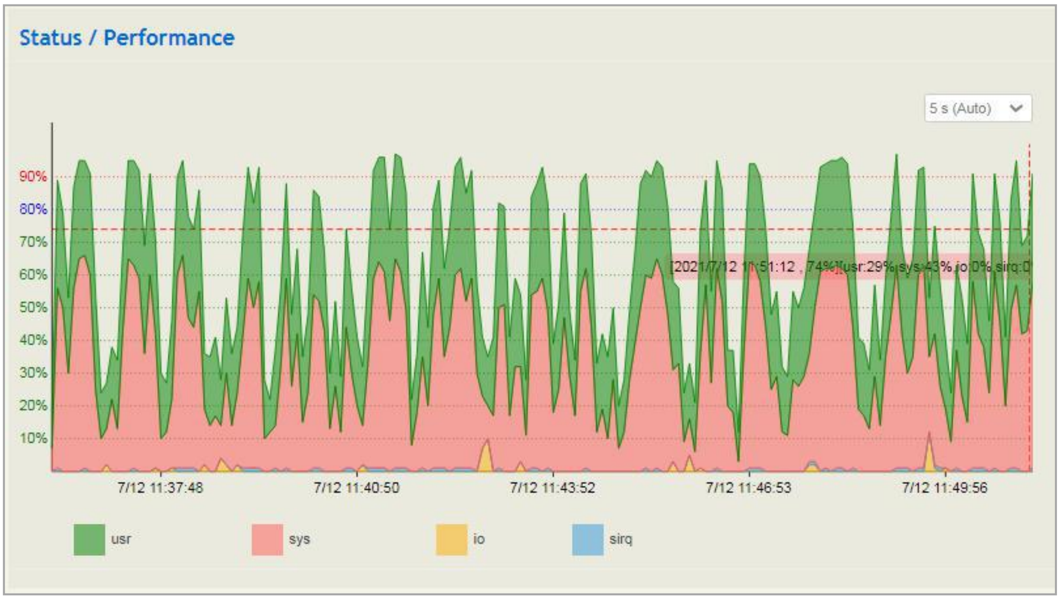
Besides, if syslog is disabled on the **System → Setting** interface, the logs cannot be uploaded to the server, but log service is still running.

Figure 5.2-10 Service Status

5.2.11 Performance

On the **Status ->Performance** Interface displays the performance statistics of the system. The parameters of the device such as usr, sys, io, and sirq are displayed at the 5th second, 1 minute, 5 minutes, and 1 hour.

Figure 5.2-11 System Performance



5.2.12 About

On the **Status → About** interface, the device model, device SN, hardware ID, MAC address, boot image, root image, firmware Version of the UC350 device are displayed.

Figure 5.2-12 Device and Company Information


Status / About

System

Device Model	UC350
Device SN	DDD1-0113-0021-0048
Hardware ID	64A4-0F58-222C
Slave SN	DDD1-0113-0021-0048
Slave Hardware ID	4065-932F-3B21
MAC Address	F8-A0-3D-46-1F-51
Boot Image	21
Root Image	49
Firmware Version	1.57.1.6 2021-06-30 15:53:36 CST
Userboard 0 Version	IAD-4S4O 1.81.80.02 PCB 3 LOGIC 0 BIOS 1, 2021-06-24 12:25:47 (13 / 14 / 12)
Userboard 0 DSP	ARM_32_9 Dec 29 2018 17:01:36
Userboard 1 Version	IAD-4S4O 1.81.80.02 PCB 3 LOGIC 0 BIOS 1, 2021-06-24 12:25:47 (13 / 14 / 12)
Userboard 1 DSP	ARM_32_9 Dec 29 2018 17:01:36
Userboard 2 Version	IAD-8O 1.81.80.02 PCB 2 LOGIC 0 BIOS 1, 2021-06-24 12:25:47 (13 / 14 / 12)
Userboard 2 DSP	ARM_32_9 Dec 29 2018 17:01:36
Userboard 3 Version	02.06.25.03 2021-06-07 17:44:26(25/39)
Userboard 3 DSP	[dsp0:succ-128][dsp1:succ-128]

License

Version	1.0.0.3
SN	2
Valid Period	90 d
Max Concurrency	150
SIP Extensions	1000
Remain	35 d

5.3 System

Configurations for timezone, login username & password, other user name, provision, operation log, service log, upgrade/backup/restore, Command Line, cloud server and device reboot can be carried out in the System section.

5.3.1 Setting

On the **System → Setting** interface, you can modify the device name, set a new timezone, synchronize local time and enable CDRs, Syslog as well as built-in NTP server.

Figure 5.3-1 Basic Setting

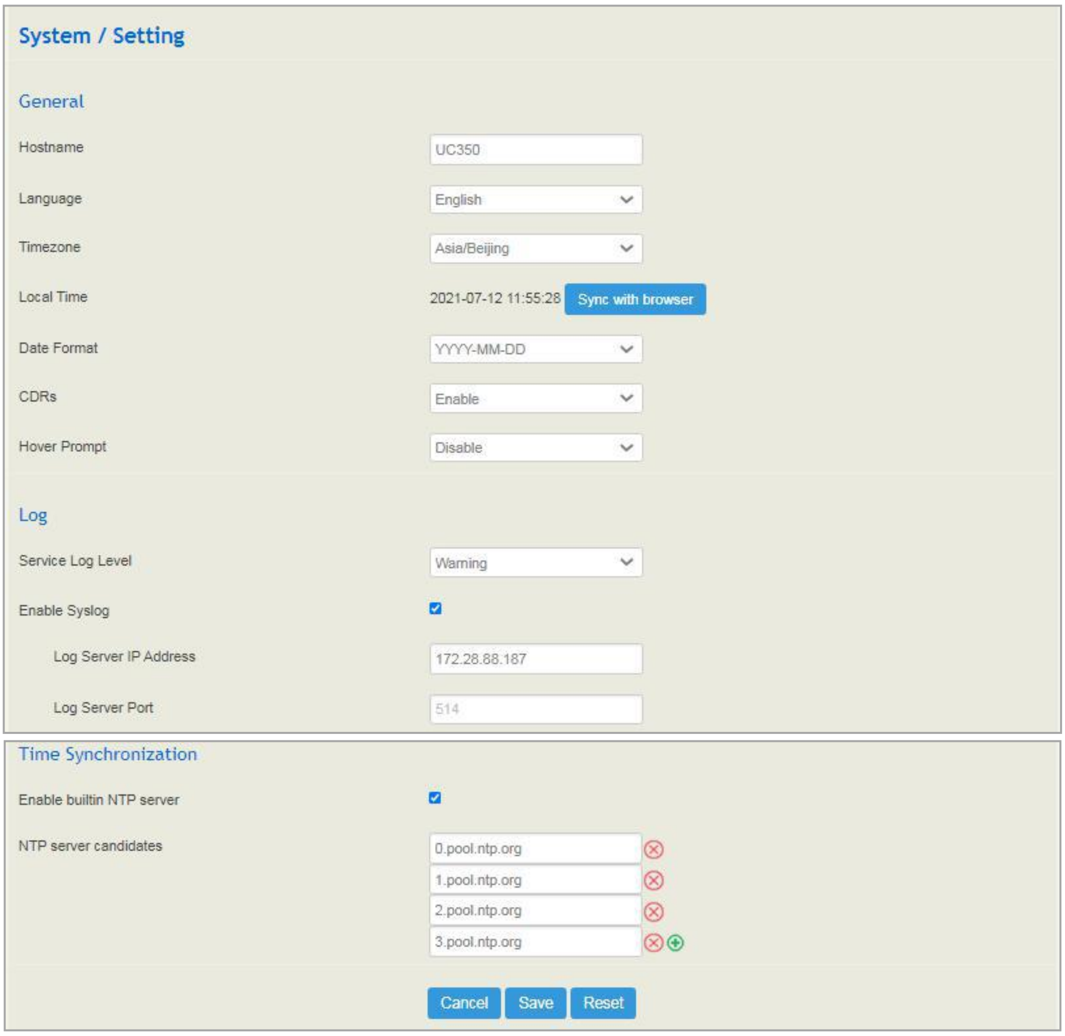


Table 5.3.1 Explanation of Basic Setting Parameters

Parameter	Explanation
Hostname	The name of the gateway. After it is configured, the name will be displayed on the left of the menu bar.
Timezone	You can choose a time zone you want. The default value is UTC (Universal Time Coordinated)
Local Time	The current time based on current time zone. It is synchronized with NTP.
CDRs	If it is enabled, CDRs will be saved automatically. 5000 CDRs call be saved at most and they can be queried on the **Status → CDRs**

	interface. If it is disabled, CDRs will not be saved
Service Log Level	There are eight levels, including Debug, Info, Notify, Warning, Error, Critical, Alert and Emergency
Enable Syslog	Whether to enable syslog
Time Synchronization	If NTP server is enabled, the UC350 can be synchronized with the world standard time. Meanwhile, you're able to add or reduce NTP servers. Please consult local telecom operators or surf the internet for the address of NTP servers.
	Delete a NTP Server
	Add a NTP Server

5.3.2 User Manager

Click **System → User Manager**, and you can modify the username name and password for logging in the UC350 gateway. Factory defaults for username name and password are **admin and admin@123#** respectively, so it is advised to modify them for security consideration.

The above mentioned username and password are also used to log in Web Interface, Telnet and SSH.

The super administrator of the device can add different users to the device and assign different roles for them, like observer, operator and administrator. Different roles can be allocated with different permissions to the functions.

Figure 5.3.2-1 Modify Username ,Password and Manager Users

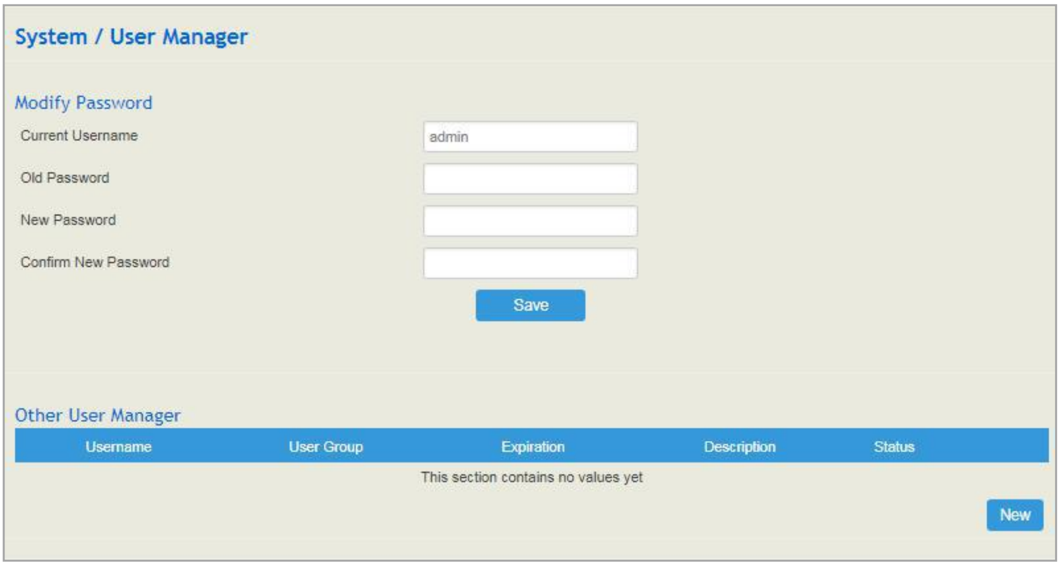


Figure 5.3.2-2 Add New User

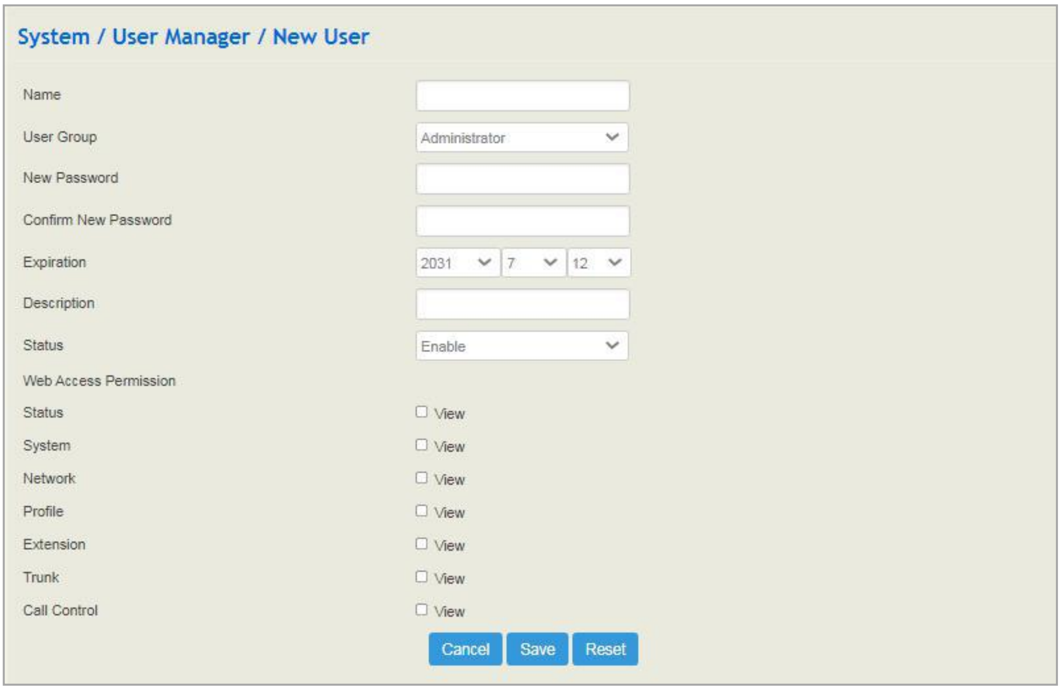


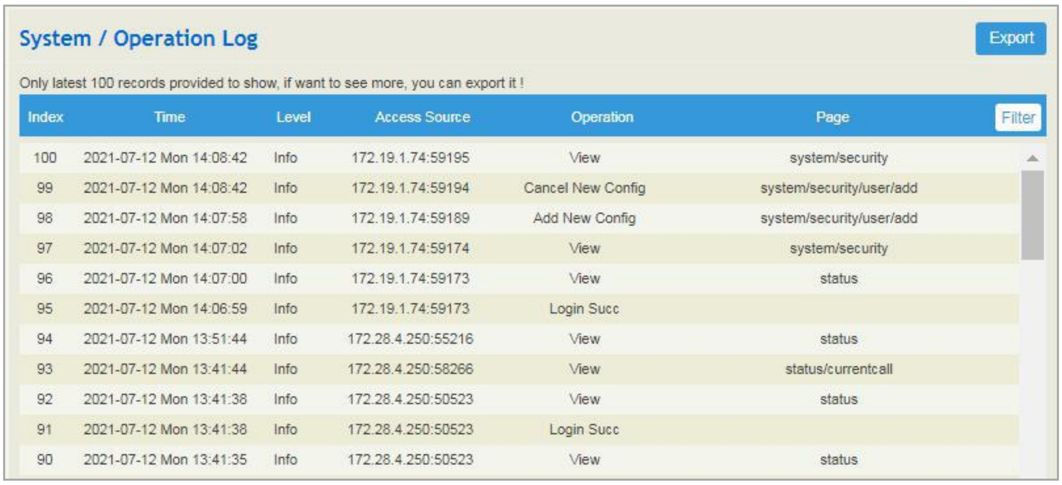
Table 5.3.2 Explanation of Provision Parameters

Parameter	Explanation
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 device.
User Group	You 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.
Expiration	The expiry date when the user cannot log in the device any more.
Status	Choose enable or disable.
Web Access Permission	The permissions to view status, system, network, profile, extension, trunk and call control.

5.3.3 Operation Log

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

Figure 5.3-3 Operation Logs


System / Operation Log

Export

Only latest 100 records provided to show, if want to see more, you can export it !

Index	Time	Level	Access Source	Operation	Page
100	2021-07-12 Mon 14:08:42	Info	172.19.1.74:59195	View	system/security
99	2021-07-12 Mon 14:08:42	Info	172.19.1.74:59194	Cancel New Config	system/security/user/add
98	2021-07-12 Mon 14:07:58	Info	172.19.1.74:59189	Add New Config	system/security/user/add
97	2021-07-12 Mon 14:07:02	Info	172.19.1.74:59174	View	system/security
96	2021-07-12 Mon 14:07:00	Info	172.19.1.74:59173	View	status
95	2021-07-12 Mon 14:06:59	Info	172.19.1.74:59173	Login Succ	
94	2021-07-12 Mon 13:51:44	Info	172.28.4.250:55216	View	status
93	2021-07-12 Mon 13:41:44	Info	172.28.4.250:58266	View	status/currentcall
92	2021-07-12 Mon 13:41:38	Info	172.28.4.250:50523	View	status
91	2021-07-12 Mon 13:41:38	Info	172.28.4.250:50523	Login Succ	
90	2021-07-12 Mon 13:41:35	Info	172.28.4.250:50523	View	status

Note: Operation logs are generally used to locate faults by device manufacturer.

5.3.4 Service Log

Service logs (the running logs of UC350) can be exported on the **System → Service Log** interface. Those logs are used for analyzing where a fault has occurred on the gateway.

Figure 5.3-4 Service Log

5.3.5 Config Changes Log

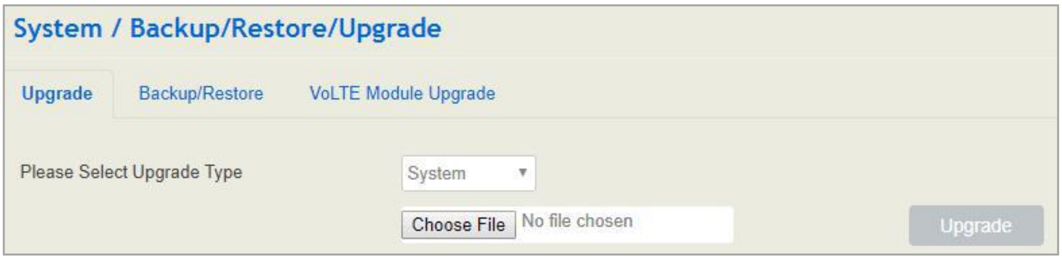
On the **System→Config Changes Log** interface, the configurations changed by administrator on the Web of the gateway are recorded.

Figure 5.3-5 Config Changes Log

5.3.6 Backup/Restore/Upgrade

On the **System → Backup/Restore/Upgrade** interface, you can back up or restore configuration files, and can upgrade UC350 to a new version. But you need to restart the device for the change to take effect after executing restore or upgrade.

Figure 5.3.6-1 Upgrade the Device



Note: The file you choose to be upgraded on the above interface is a local file.

Figure 5.3.6-2 Back up files

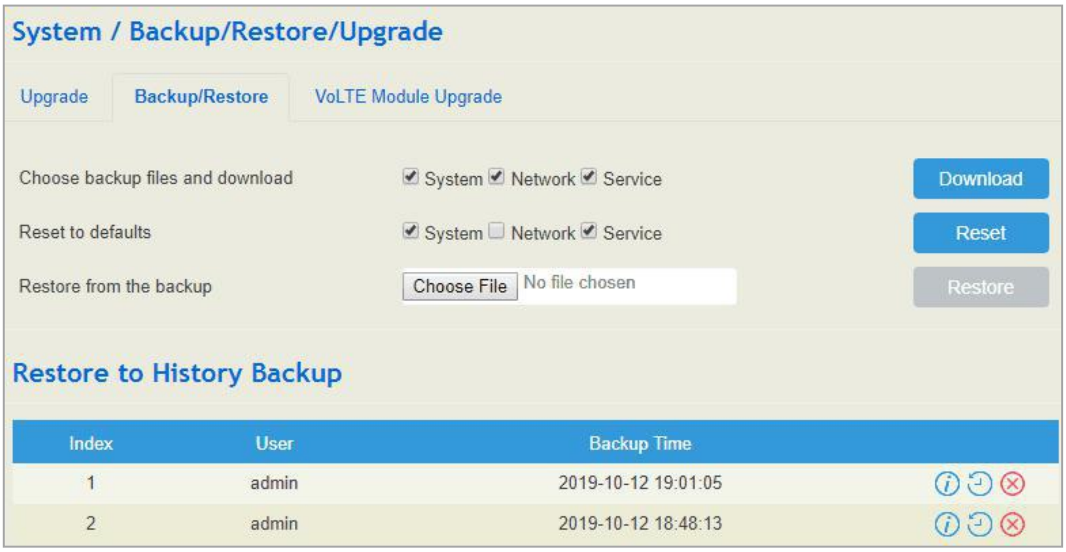


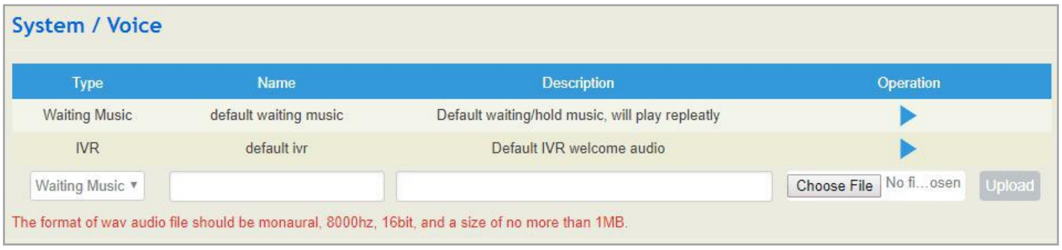
Table 5.3.6 Explanation of Backup/Restore/Upgrade

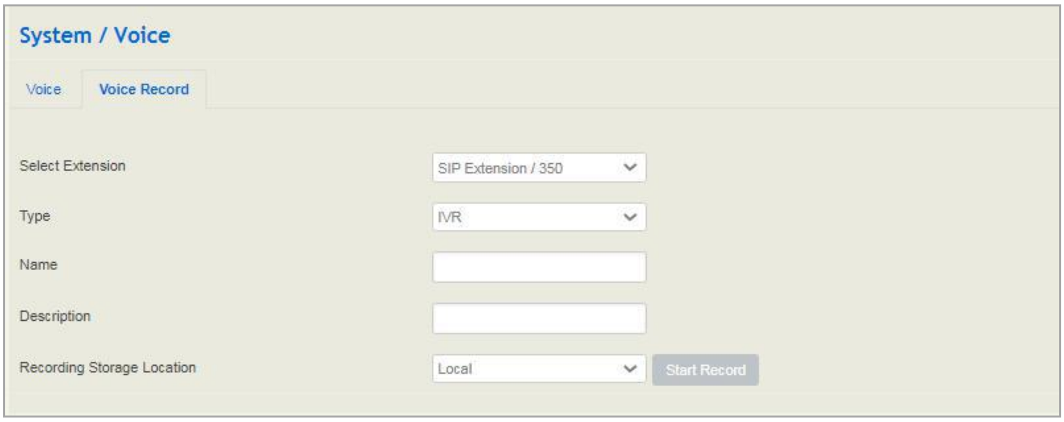
Download	You can download the configuration data to be backed up. Select any of the checkboxes on the left of System, Network and Service, and then click **Download**
Reset	Select any of the checkboxes on the left of System, Network and Service, and then click **Reset**, and configurations related to the selected part will be restored to factory defaults.
Restore	Choose a backup file, and then click **Restore**.
Upgrade	Choose a file to be upgraded (which is provided by Shenzhen Dinstar Co., Ltd.), and then click **Upgrade**.

5.3.7 Voice

On the **System → Voice** interface, you can upload an IVR file according to your needs. At present, only wav audio file is allowed. The format of the uploaded wav audio file must be: monaural, 8000hz, 16bit, and size of no more than 1M.

Figure 5.3.7 Upload IVR File



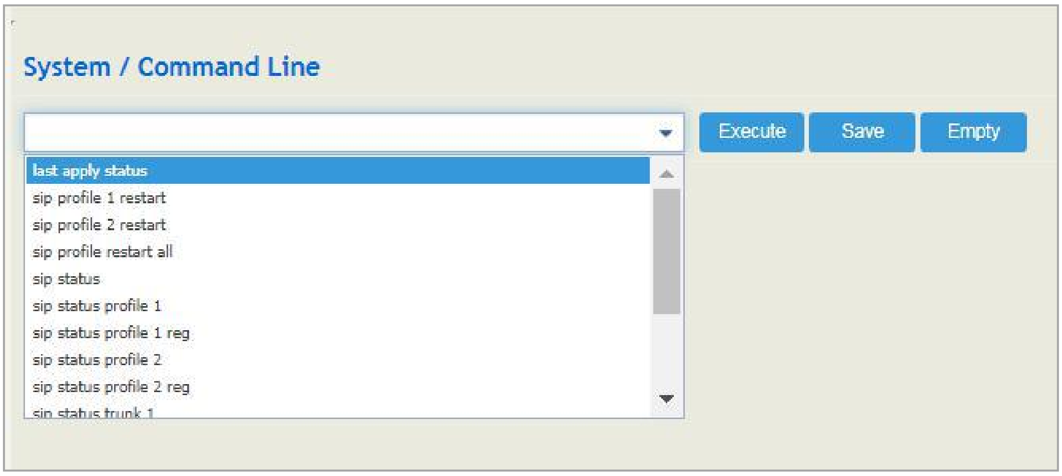


5.3.8 Command Line

On the **System → 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 fxo config, fxo status, fxs config, fxs status, sip status, sip profile and so on.

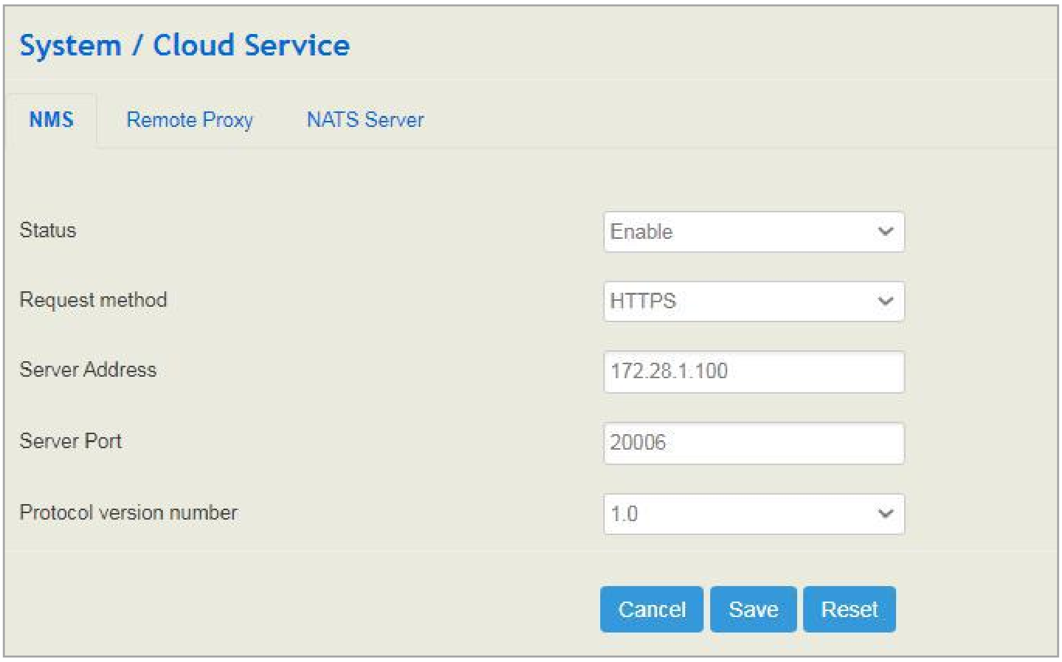
Figure 5.3-8 Command Line

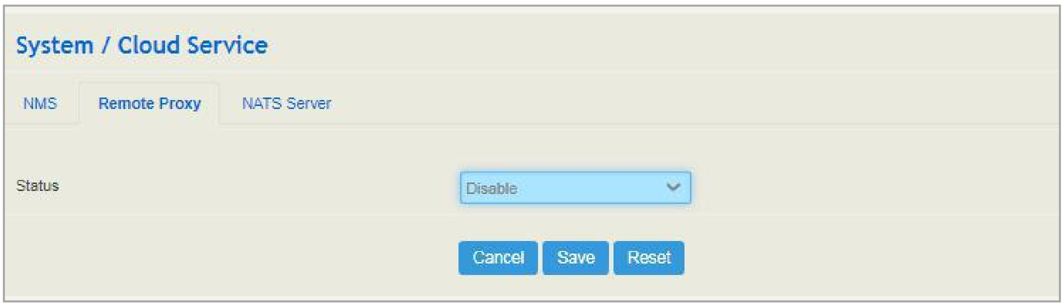


5.3.9 Cloud Service

Cloud service is mainly used to centrally manage all kinds of devices. Through cloud service, you can query the system status of a device, upgrade at batch, log in and configure remotely. UC350 provides two cloud services:remote proxy and NATS server, enter the IP address, service port and password of the Cloud server, and then the gateway will connect to the cloud server.

Figure 5.3.9-1 Cloud Server



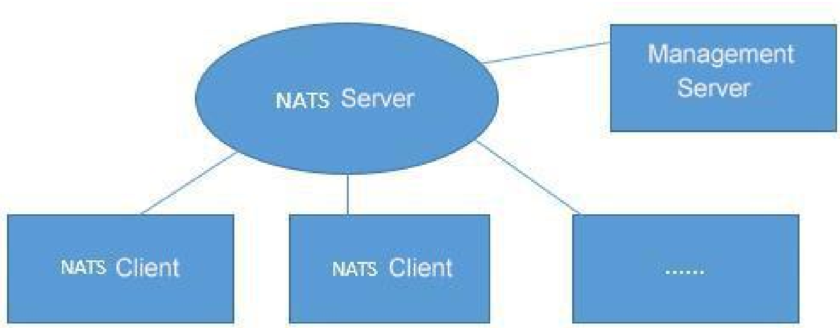


NATS Server:

UC350 can work as a NATS client to send messages to a NATS server, and then the NATS server will open related ports to facilitate the connection with those clients or servers of users.

How NATS works:

1) NATS network topology



2) Operating mode

Both NATS client and management server need to connect to the NATS server;

The management server subscribes to from the NATS server, and the NATS client is responsible for publishing messages. After receiving the subject published by the client, the NATS server forwards the message to the management server. The flows of subscribing to subject message: management server -> NATS server. The flows of reporting a message: NATS client -> NATS server -> management server;

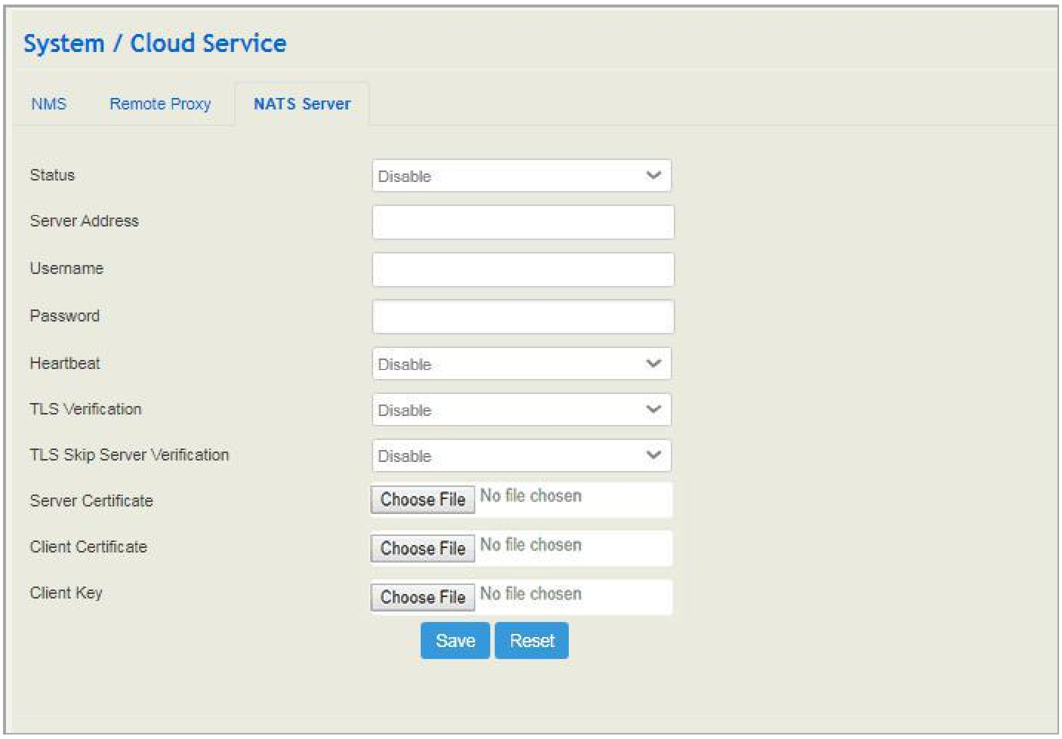
NATS clients can also subscribe to subjects and receive messages published by the management server, but this way of working is unavailable in the event reporting function.

Configuration steps:

1) Configure NATS server on the web interface (UC350 acts as a client):

On the "System -> Cloud Server" interface, enable the NATS server and configure other parameters, such as IP address, port (when TLS is disabled, the port defaults to 4222, you can choose not to fill in), user name and password (can be empty), enable heartbeat etc.

Figure 5.3.9-2 NATS server configuration



(3) Configure the same NATS server information on the management server;

(4) Enter the command line on the management server and enter the command to subscribe to the subject; command format: nats_client subscribe xxxxxx （example: nats_client subscribe *.server.register）

The following are the three subjects that the management server must subscribe to:

Register subject(used to receive the registration information of the NATS client): nats-client subscribe *.server register

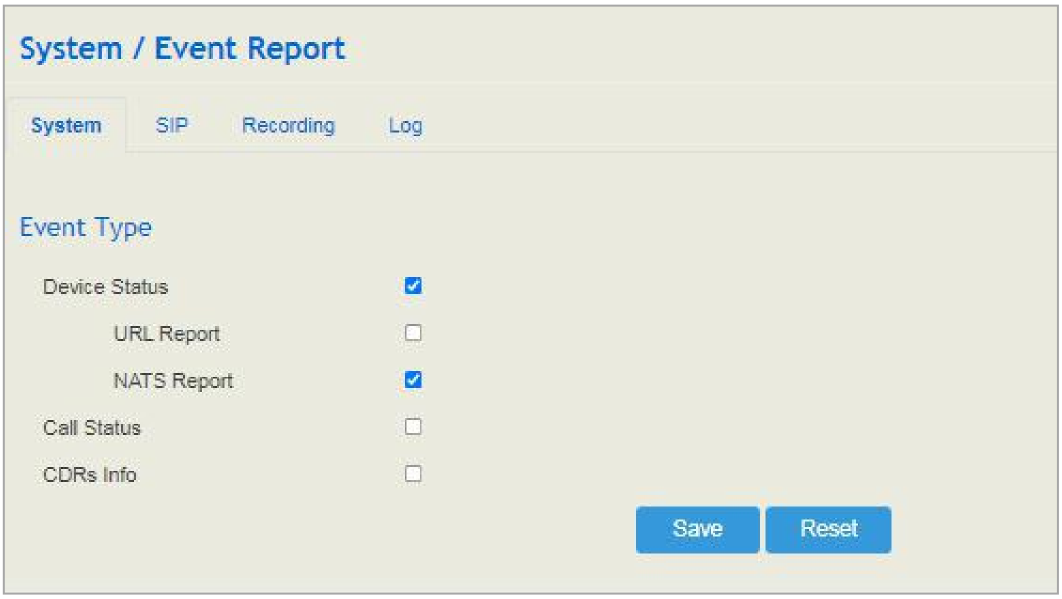
Event subject (used to receive event messages from NATS clients): nats_client subscribe *.server.event

Heartbeat subject (used to receive heartbeat messages from NATS clients): nats_client subscribe *.server.heartbeat

Note:	After subscribing , you can use the *nats-client status* command to view the registration status and the subjects that have been subscribed. The command format for unsubscribing subjects is: *nats-client unsubscribe *.server.register.*

(5) Enable NATS report on the "**System -> Event Reporting**" interface of UC350.

Figure 5.3.9-3 Enable NATS report



To make NATS report work normally, the following three points need to be met:

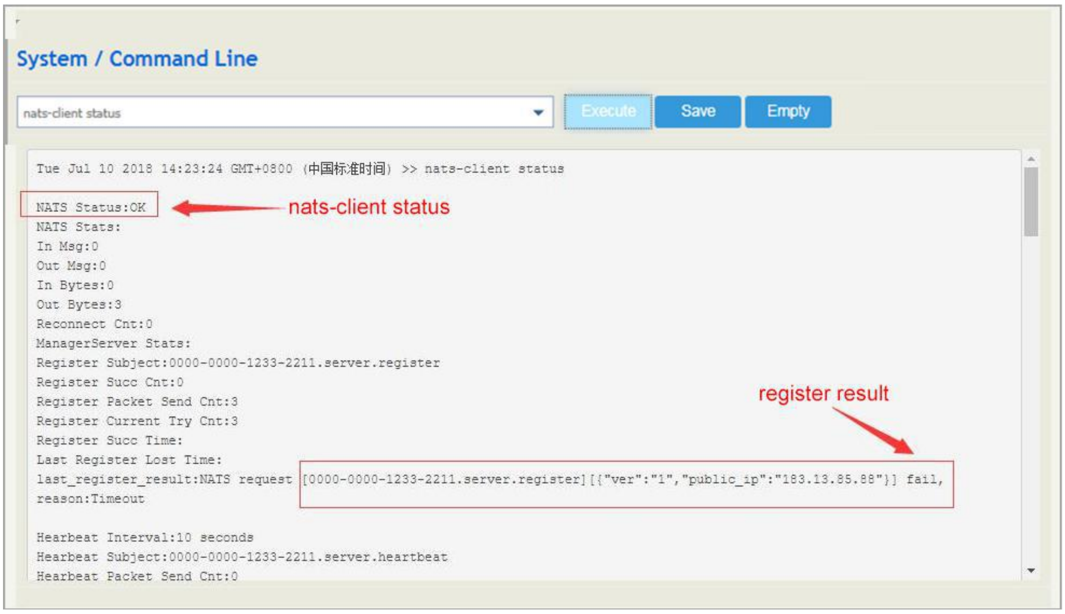
1) The NATS client (UC350) has enabled the NATS report;

2) The connection between the NATS client and the NATS server has been established;;

3) The NATS client has been registered to the management server.

Note:	You can view the status of the NATS client by entering "*nats-client status*" on the "System -> Command Line" interface. If the status displays "NATS Status: OK", it means that the NATS client and NATS server are successfully connected.

Figure 5.3.9-4Enter the command to view the status of the NATS client



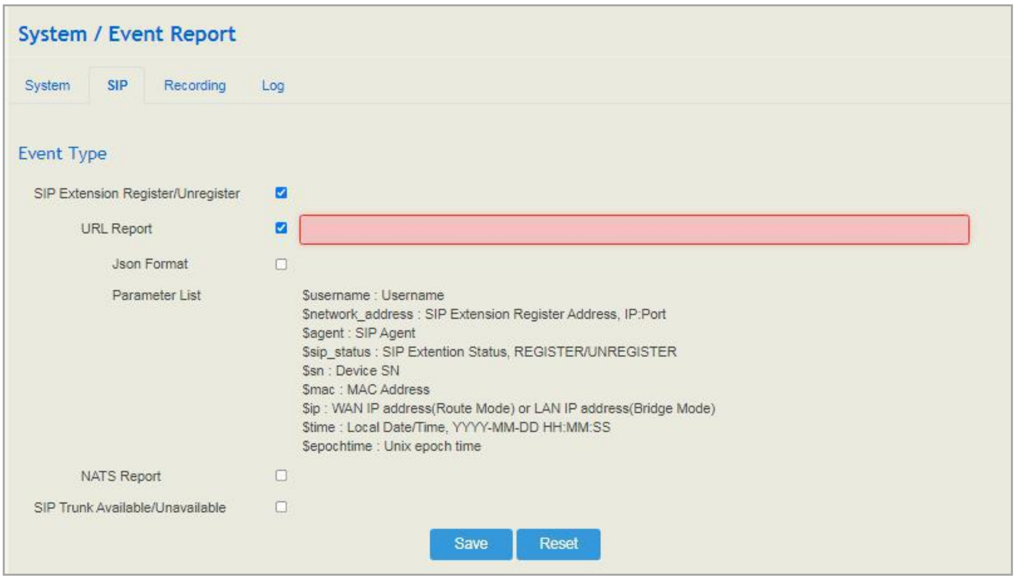
5.3.10 Event Report

UC350 allows the following events to be reported through NATS or URL: device startup, call status,Register or deregister SIP extension, availability or unavailability of SIP trunks, off-hook or on-hook of FXS phone, FXO status and update of CDR information.

For event report through NATS, please refer to the configuration steps of NATS in the Could Server section.

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

1) On the "System -> Event Report" interface, select the events to be reported and the reporting method (URL);



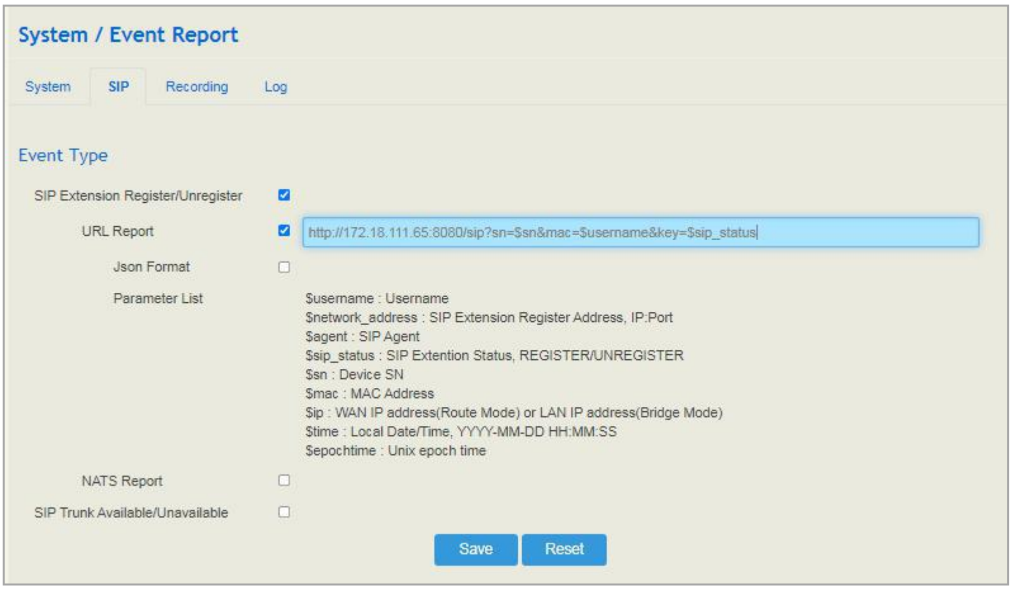
2) Input the URL.

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 startup, callstatus, sip, siptrunk, fxs, fxo, gsm, volte, vpn and cdr, while value refers to the parameter that needs to be reported. Key can be defined by yourself, but it’s generally the same with value.

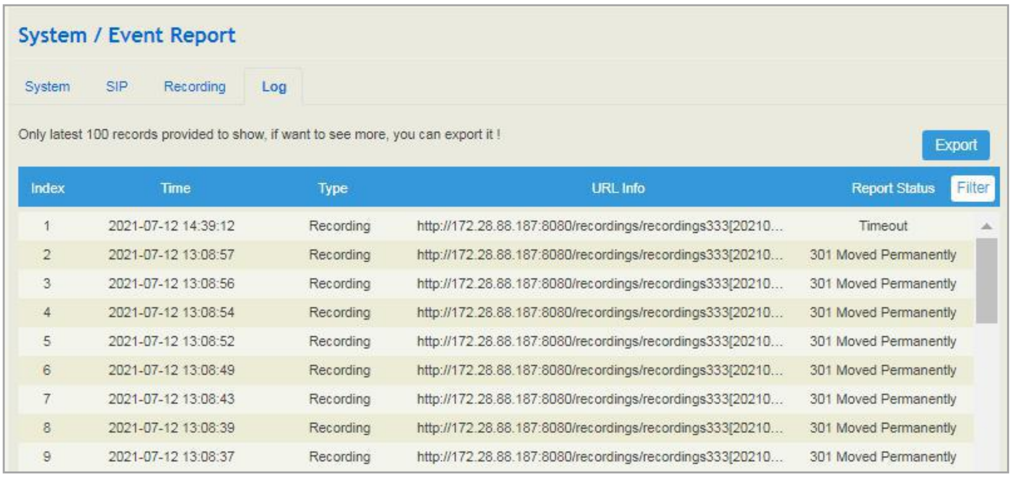
Figure 5.3.10-1 Input URL



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

4) On the System→Event Report→Log interface, you can view the report information.

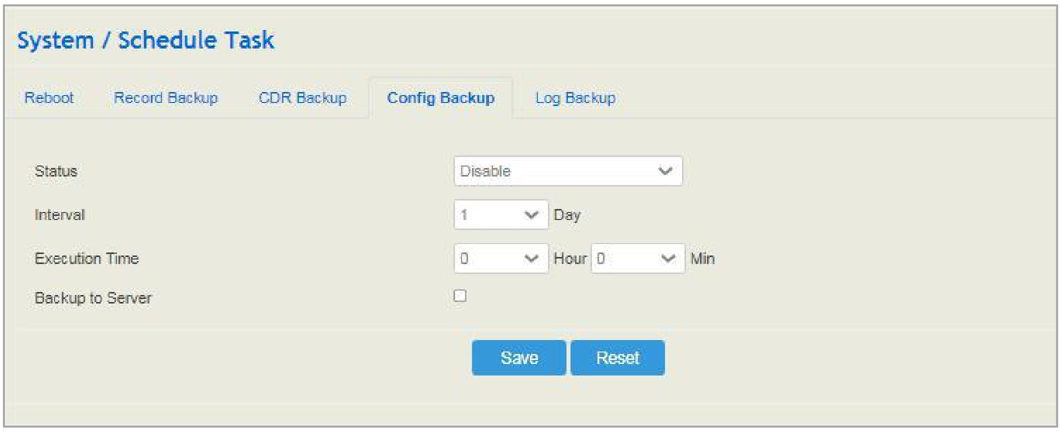
Figure 5.3.10-2View Report Log



5.3.11 Schedul Task

On the **System → Schedule Task** interface, you can set a scheduled to reboot the UC350 device, record backup, and back up CDRs, logs or configurations.

Figure 5.3-11 Scheduled Task



5.3.12 Email

On the **System → Email** interface, you can configure a email client on UC350, which can be used to connection test for sending and receiving mail. But on top of that, SMTP, IMAP and POP 3 services need to be enabled for the email client.

When the email client is used with SMS routing, email and SMS are bound, which brings great convenience, for example, you can receive an email , although someone is sending you an SMS message. Meanwhile, logs will be generated can be viewed on the **System → Email → Log** interface.

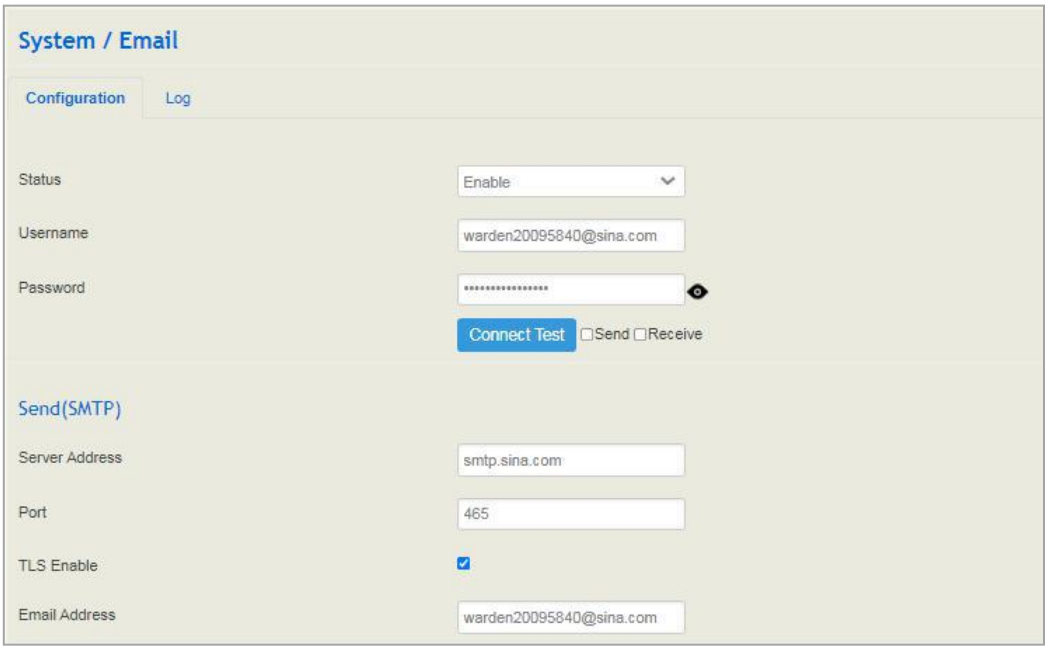


Figure 5.3-12 Email Client

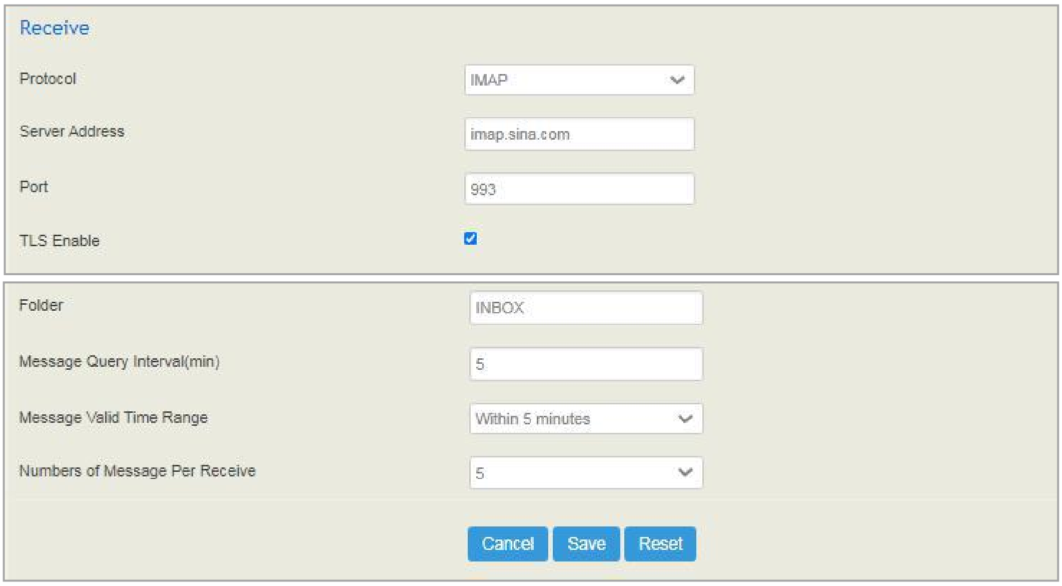


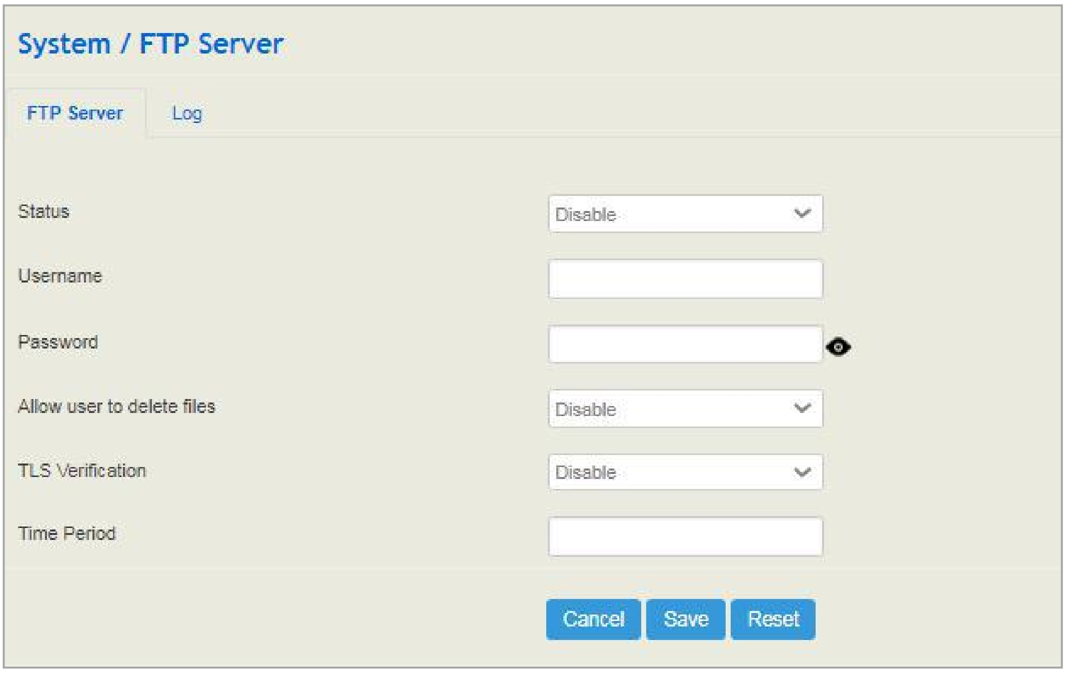
Table 5.3.12 Explanation of Email Client Parameters

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
Protocol	Choose IMAP or POP3. When POPS is selected, TLS port is 995 by default.
Message Query Interval (min)	The time interval to check whether there is a new email.
Message Valid Time Range	Only those emails received during this time range are addressed.
Number of Message Per Receive	The maximum number of emails that are received at one time. If the number exceeds, they will be received in batches.

5.3.13 FTP Server

On the **System → FTP Server** interface, you can enable the FTP server function of UC350 and configure related parameters such as username, password and access permissions. You can connect FTP clients to this FTP server and then access those files (like recording files and system logs) that are open on the UC350 through the 21 port.

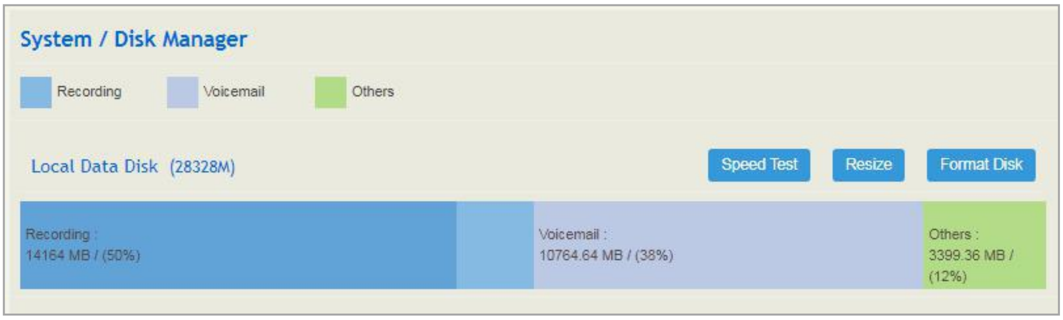
Figure 5.3-13 FTP Server



5.3.14 Disk Manager

On the **System →Disk Manager** interface, you can view the memory usage of device. The memory are divided into three categories, including voicemail(38%), recording (50%) and Others(12%). You can also re-division the proportion of each category or execute formatting on this interface.

Figure 5.3-14 Disk Manager



Note: UC350 currently does not support U disk/SD card.

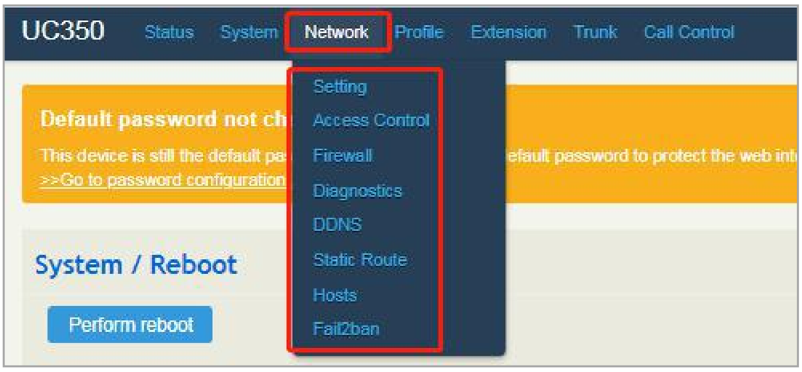
5.3.15 Reboot

On the **System → Reboot** interface, you can click **Perform Reboot** to reboot the UC350 gateway. After the device is rebooted, those configurations that have been saved will remain unchanged.

Figure 5.3-15 Reboot Device

5.4 Network

UC350 provides three RJ45 ports, namely GE0, GE1 and Console ports. GE0 is a network interface for access to Ethernet; GE0 is a management port for PC access to management equipment; a console port is used for access commands line.



5.4.1 Setting

On the **Network → Setting** interface, you can set the IP address of GE0 port.

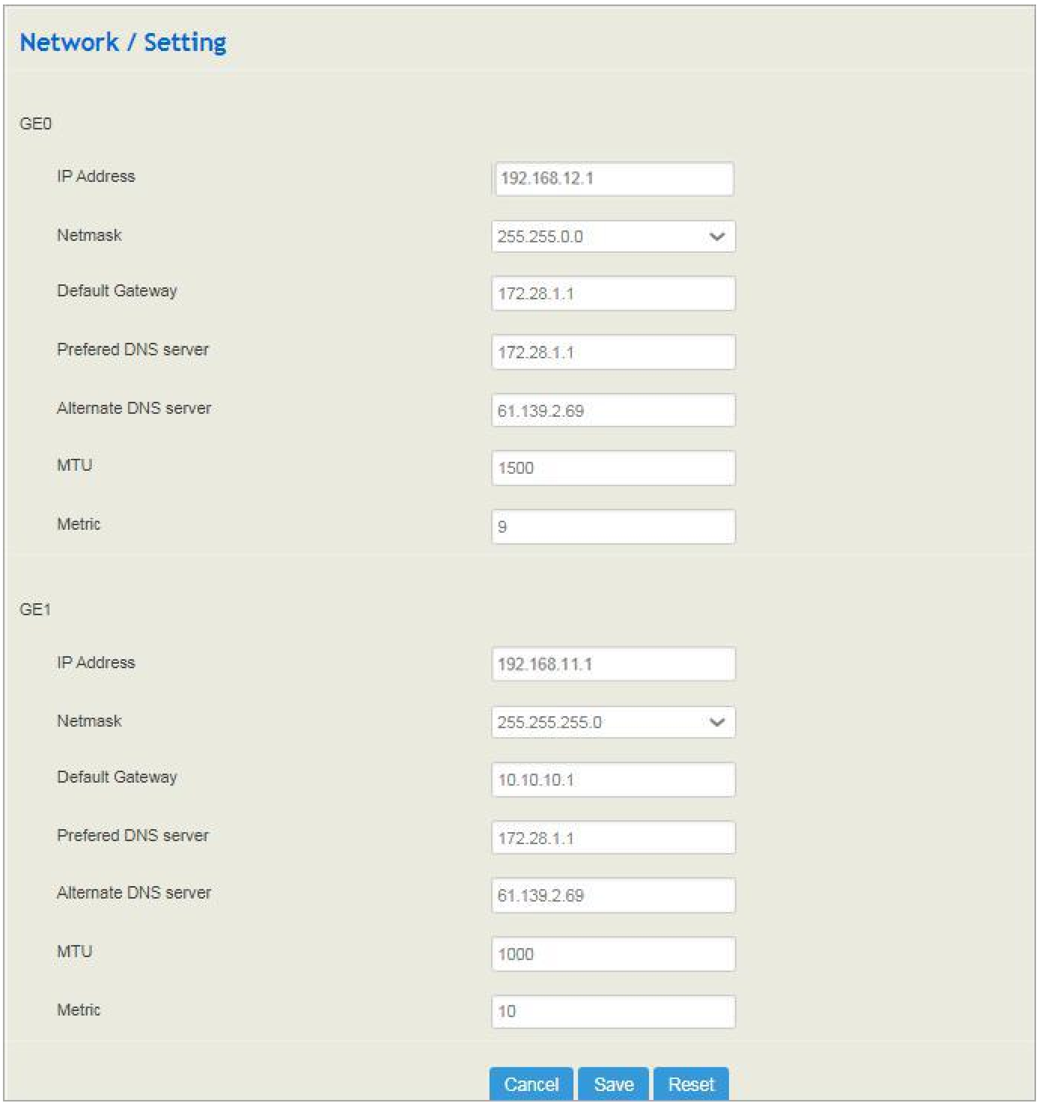
The GE0 port of UC350 can only be a static IP address (an IP address outside the 192.168.12.1/24 network segment), and the IP address of the GE1 port is 192.168.11.1.

Static IP Address:

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

- IP Address: the IP address of the GE0 port of the UC350;
- Netmask: the net mask of the router connected the UC350;
- Default Gateway: the gateway IP address of the router connected the UC350;
- Use custom DNS server: the IP address of the DNS server

Figure 5.4-1 Static IP address of GE0 port

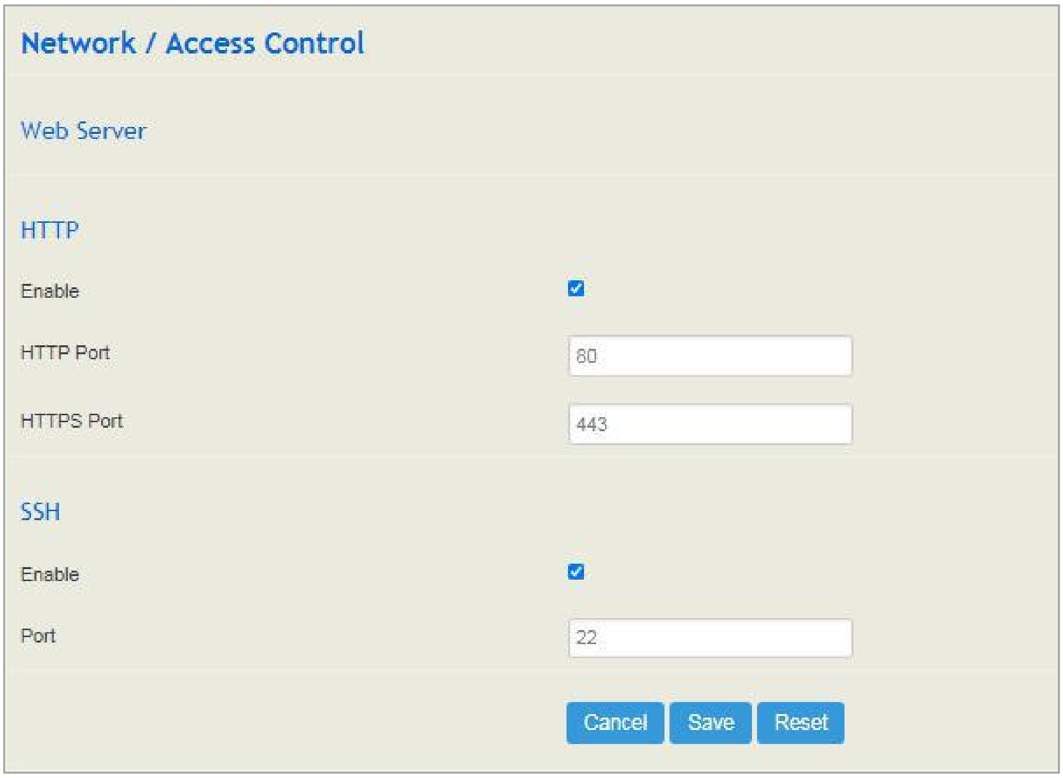


The GE0 port of UC350 is a static IP address: 192.168.12.1/24. The PC can be connected to the GE0 port with a network cable and enter 192.168.12.1 in the browser to access the UC350 Web interface (the PC needs to be configured with a static IP address, Such as: 192.168.12. 100).

5.4.2 Access Control

The access ports of Web, Telnet and SSH, as well as relevant on-off controls, can be configured on the Network → Access Control interface.

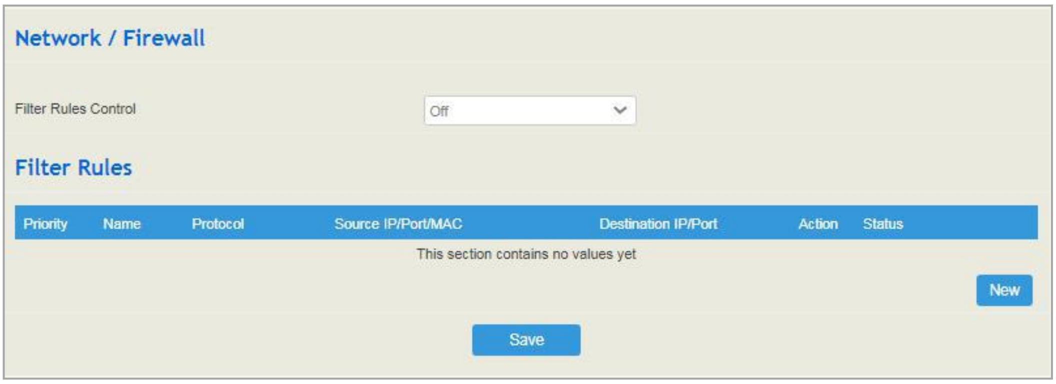
Figure 5.4-2Access Control



5.4.3 Firewall

You 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 5.4.3-1 Firewall



Configuration Procedures:

1) Select **On** in the drop-down box on the right of **Filter Rules Control**
2) Select filter action, accept or reject;
3) Click the **New** button;

4) Fill in information of filter rule (Filtering rule information: IP, port, mac address can all be empty, no judgment will be made when it is empty, otherwise all three must be meet after configuration);

5) Click the **Save** button to save the configuration.

Figure 5.4.3-2 Create Filter Rule

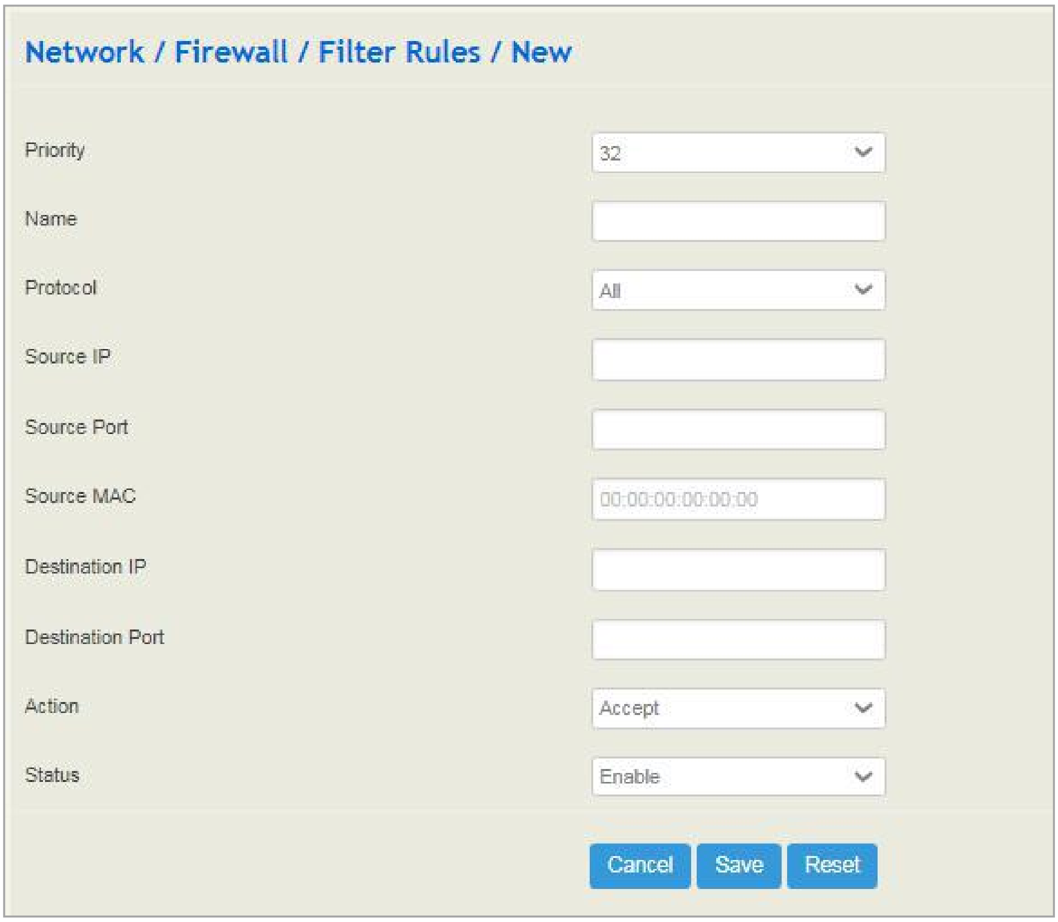


Table 5.4.3 Explanation of Filter Rule

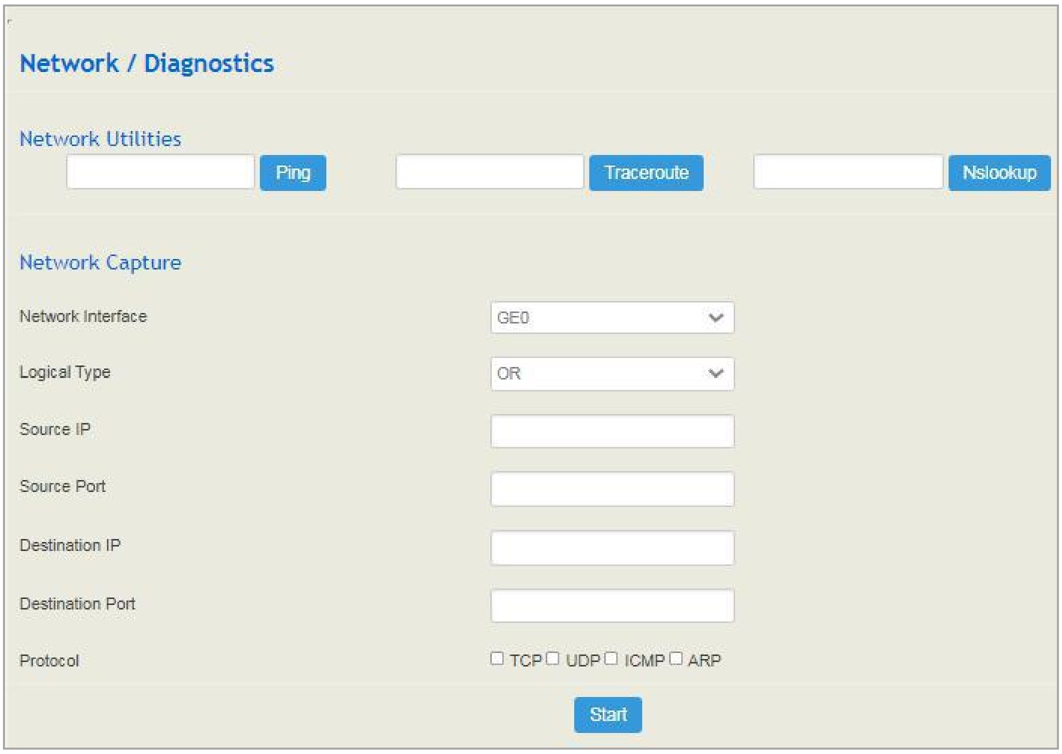
Source IP	The source IP address that you want UC350 to accept or reject. It is the source IP address of the message; it can also be a string of IP addresses, for example, 172.16.11.1/15.
Source Port	The source port of host which the accepted or rejected IP address belongs to
Source MAC	The source mac of the host which the accepted or rejected IP address belongs to
Destination IP	The destination IP address that you want UC350 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

5.4.4 Diagnostics

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

Figure 5.4-4 Network Diagnostics



- **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, you can capture data packages of the available network ports. You can also set source IP, source port, destination IP or destination port to capture the packages that you 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.

5.4.5 DDNS

On the "Network -> DDNS" interface, you can use UC350 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 5.4-5 Dynamic Domain Name

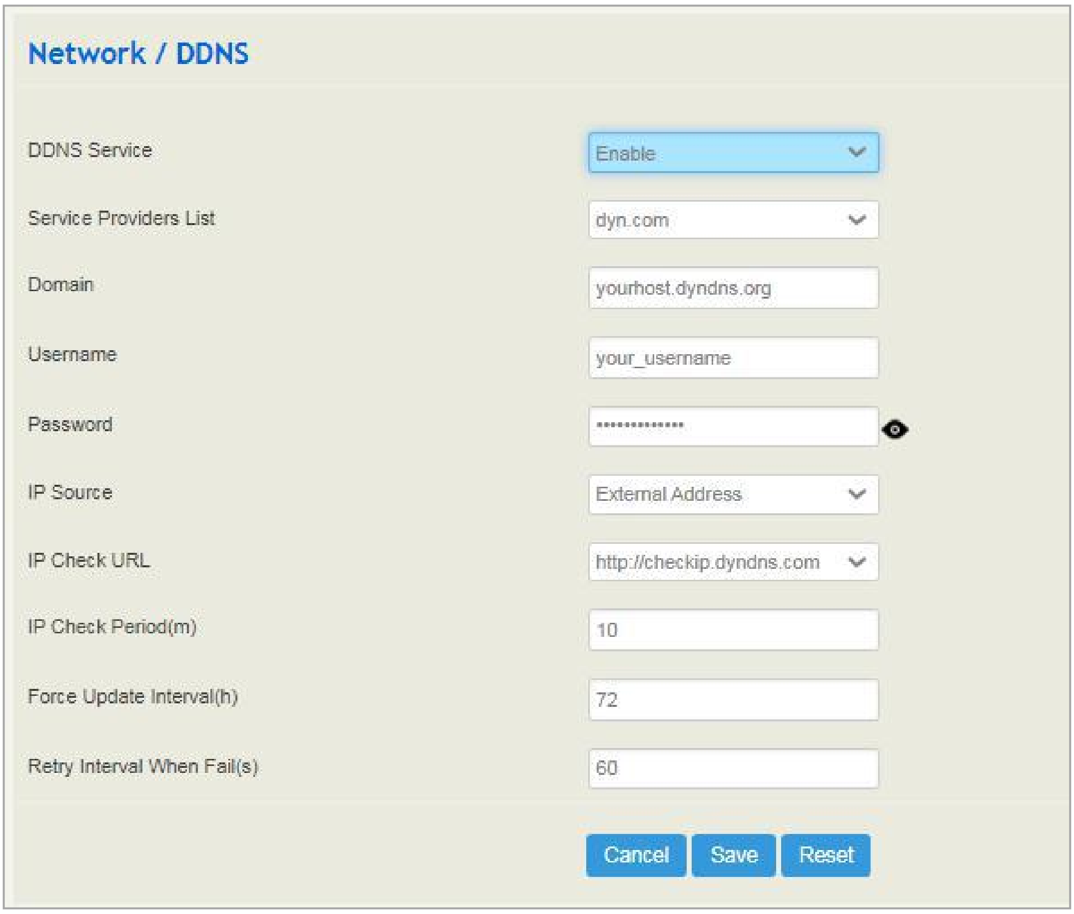


Table 5.4.5 Explanation of DDNS Parameters

Service Providers List	Dynamic domain name service providers
Domain	Domain name applied for on the service provider website
Username	The user name when applying for a domain name on the service provider website
Password	The password when applying for a domain name on the service provider website
IP Source	The external address and the device address can be selected, the external address is the current network export public network IP address, and the device address is the GE0 port IP address
IP Check URL	Server address that detects whether the IP address is updated
IP Check Period(m)	Check whether the IP address has changed detection period
Force Update Interval(h)	Force update within the configured time interval and report the IP address to the DDNS server
Retry Interval When Fail(s)	Set the retry interval when updating the IP address fails

5.4.6 Static Route

On the **Network → Static Route** interface, you can configure static routes for the network.

Figure 5.4-6 Create Static Route

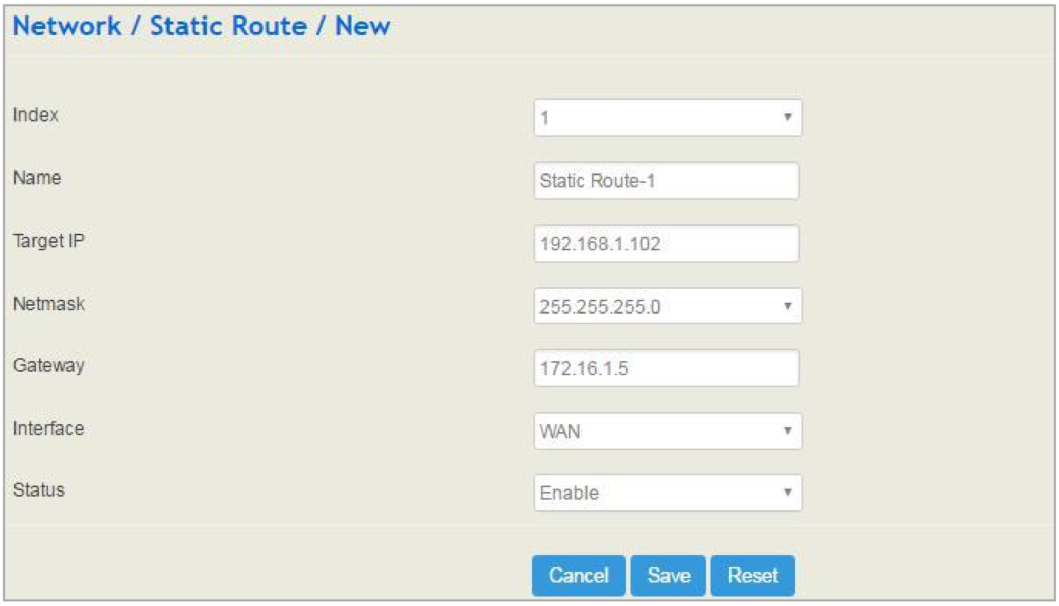


Table 5.4.6 Explanation of Static Route Parameters

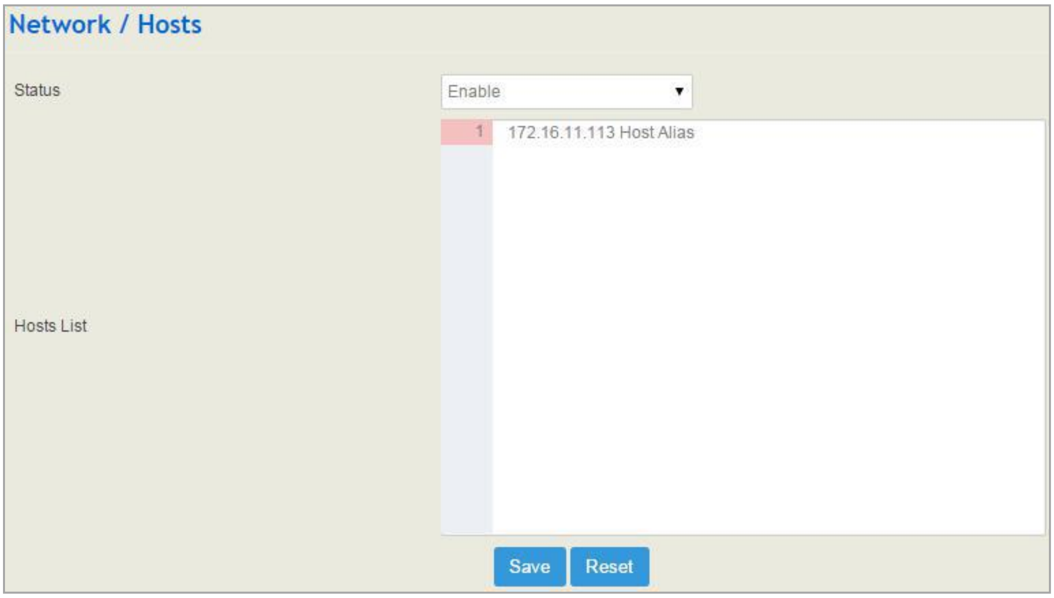
Name	The name of the static route
Target IP	The destination IP address of the static route
Netmask	The netmask of the static route, default: 255.255.255.0
Gateway	The IP address of the outbound gateway of the static route
Interface	The outbound interface of the static route, namely GE0 port or GE1 port
Status	The static route is enabled or disabled

5.4.7 Hosts

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

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

Figure 5.4-7 Enable Hosts File



5.4.8 Fail2ban

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 Network → Fail2ban interface, you can configure rules for Fail2ban. Fail2 ban is generally targeted SSH and SIP. The Fail2ban of UC350 only works on the GE0 port, and does not disable the IP of the 192.168.11.1/24 network segment of the GE1 port.

Figure 5.4-8 Fail2ban Rules

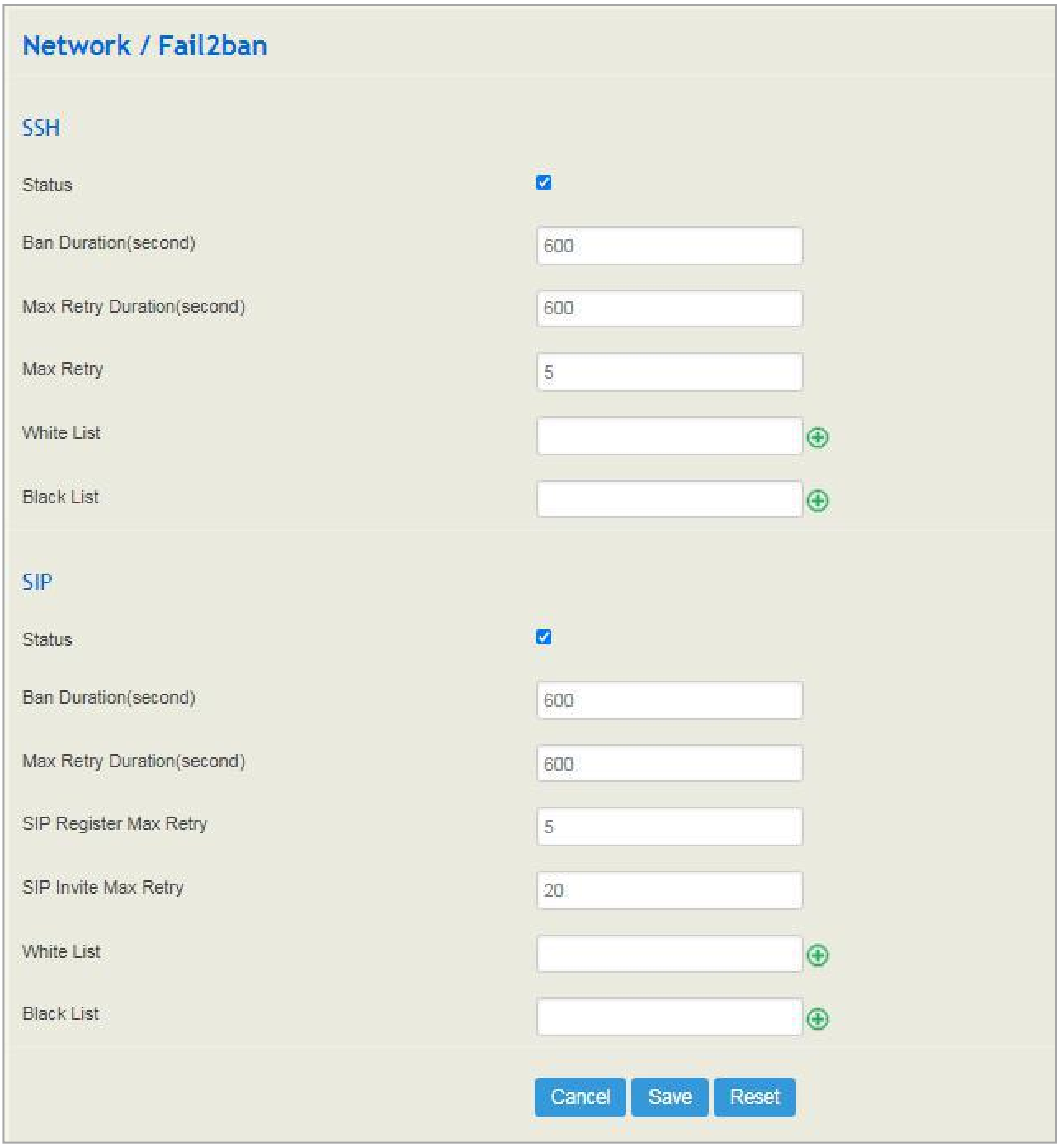


Table 5.4.8 Explanation of Fail2ban Parameters

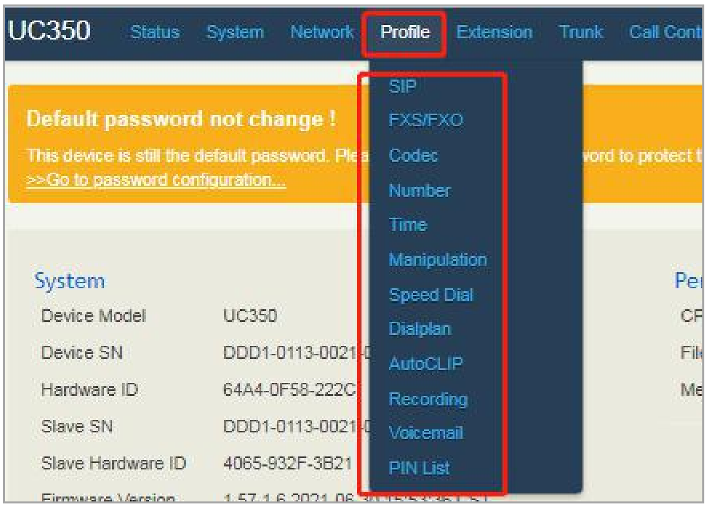
Ban Duration (Second)	The time period during which the IP addresses that conform to the banning rule or are in the backlist are prohibited. Range: 60-315360000 seconds
Max Retry Duration(second)	The time period during which the maximum retries have been executed and then the corresponding IP address will be banned. For example, if this parameter is set as 60 seconds and the maximum number of retries is set as 10, an IP address will be banned in case that it has tried 10 times during 60 seconds. Range: 5-3600
Max Retry	The maximum number of retries during a specific time. For example, if this parameter is set as 10 and the max retry duration is set as 60 seconds, an IP address will be banned in case that it

	has tried 10 times during 60 seconds. Range: 5-3600
White List	Those IP addresses that are in the white list will not be banned by Fail2ban.
Black List	Those IP addresses that are in the black list will not be banned by Fail2ban.

Note: If the SSH/SIP sent by an IP has an exception such as "The network is reachable but not responding", you can go to the **Status → Fail2ban** interface to check whether the IP address is banned or not.

5.5 Profile

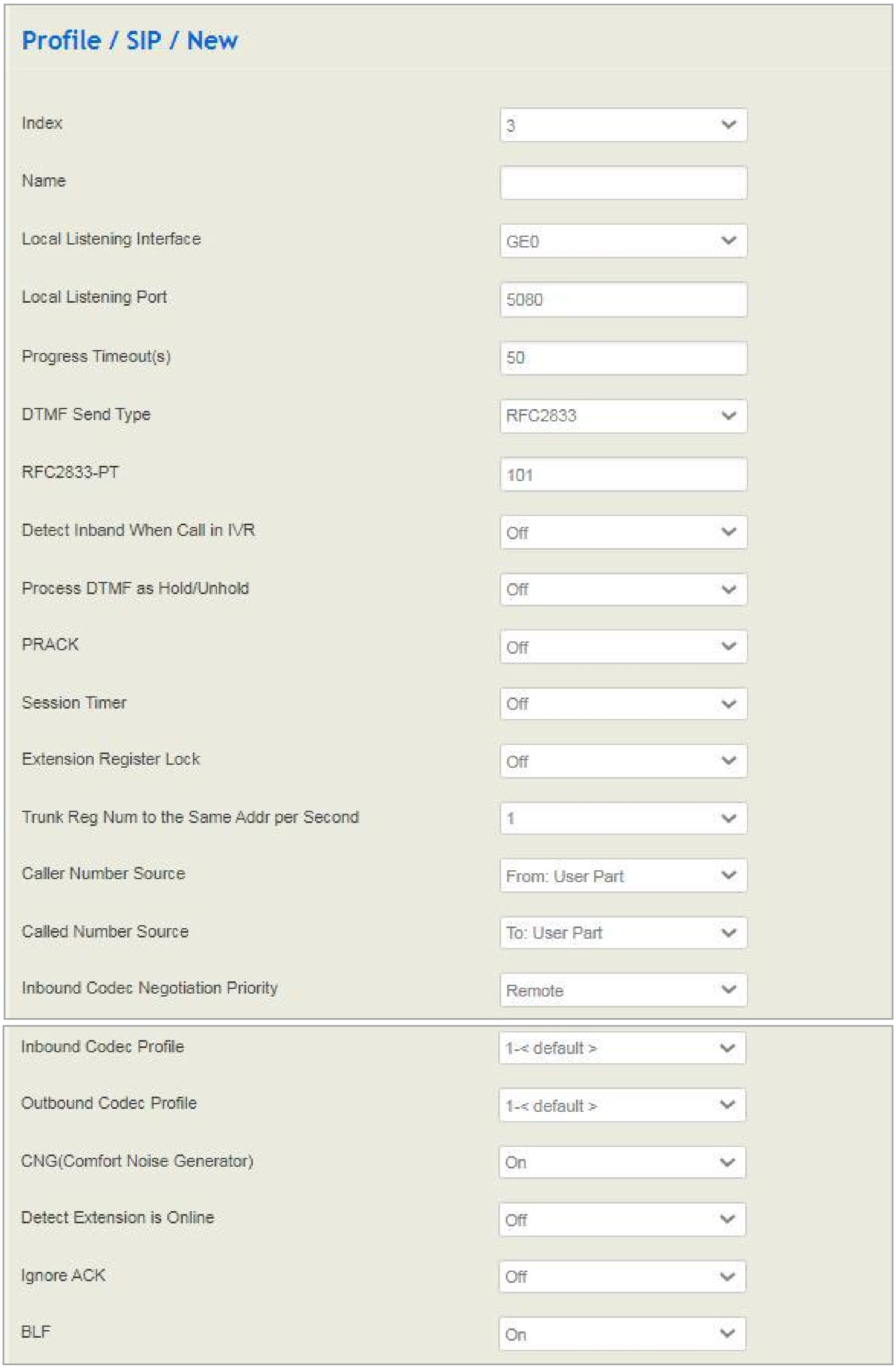
The Profile menu includes the following sub-menus: SIP, FXS/FXO, Codec, Number, Time, Manipulation, Dialplan, Recording, Voicemail etc.



5.5.1 SIP

On the **Profile → SIP** interface, you can set SIP information such as listening port, which will be used in extension and trunk. Multiple SIP profiles can be configured for one UC350 device, so you can choose different SIP profiles according to different needs.

Figure 5.5-1 SIP Profile



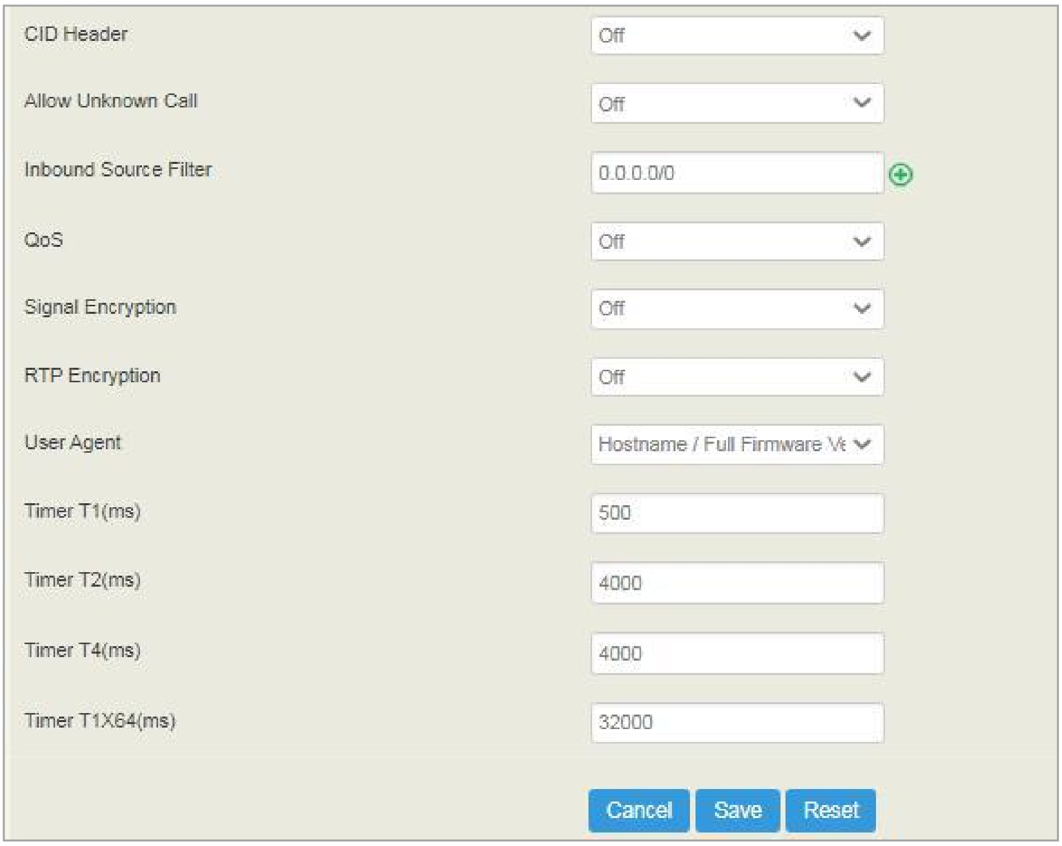


Table 5.5.1 Explanation of SIP Profile

Name	The name of the SIP profile
Local Listening Interface	The local listening interface of this SIP profile. It can be GE0 port and GE1 port. If the SIP profile is used by a SIP trunk, the interface filled in here is the listening port for the SIP trunk.
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.
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.
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
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
Extension Register Lock	When enabled, only the first successfully registered client is allowed to register
Trunk Reg Num to the Same Addr per Second	When multiple trunk are registered to 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.
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
Detect Extension is Online	Whether to detect the SIP extension using this SIP profile is online or not
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, you 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
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
QoS	Whether to enable QoS. QoS is a technology used to solve network delay or congestion
Signal Encryption	After enabling, the gateway will transmit signaling via TLS
RTP Encryption	Select encrypted SRTP for RTP stream transmission. SRTP is a secure real-time transmission protocol to ensure the security of voice communication, SRTP is a secure real-time transmission protocol to ensure the security of voice communication
User Agent	Then content of the 'user agent' field in SIP packets
Timer T1	The value of timer T1 in SIP protocol. Default value is 500ms
Timer T2	The value of timer T2 in SIP protocol. Default value is 4000ms
TimerT4	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

Description for SIP server

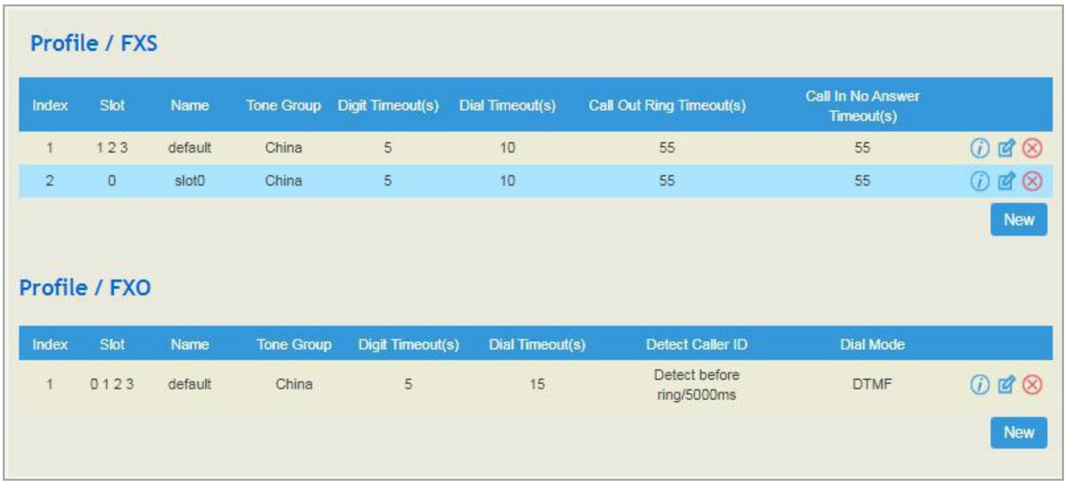
SIP server is the main part of the VoIP network, responsible for establishing all SIP calls. SIP server is also called SIP proxy server or registration server. Both IPPBX and softswitch can act as the role of SIP server. and UC350 is set in the "Extension-SIP" interface to act as this role.

Note: For more details on how to configure SIP server, please refer to 5.7.1

5.5.2 FXS/FXO

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

Figure 5.5.2-1 FXS/FXO Profile



Profile / FXS

Index	Slot	Name	Tone Group	Digit Timeout(s)	Dial Timeout(s)	Call Out Ring Timeout(s)	Call In No Answer Timeout(s)	
1	1 2 3	default	China	5	10	55	55	
2	0	slot0	China	5	10	55	55	

New

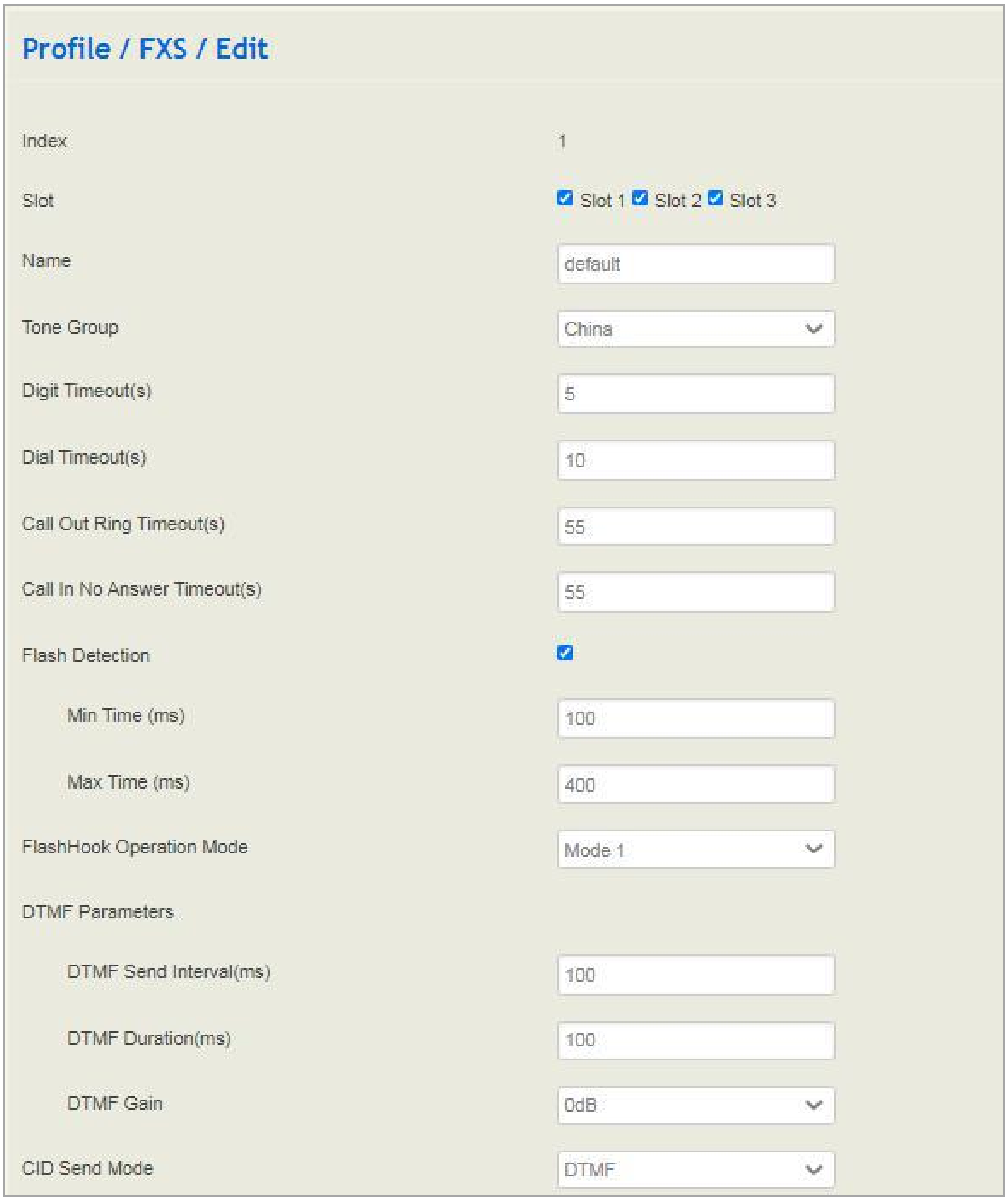
Profile / FXO

Index	Slot	Name	Tone Group	Digit Timeout(s)	Dial Timeout(s)	Detect Caller ID	Dial Mode	
1	0 1 2 3	default	China	5	15	Detect before ring/5000ms	DTMF	

New

Click , and configuration interface will pop up.

Figure 5.5.2-2 FXS Profile



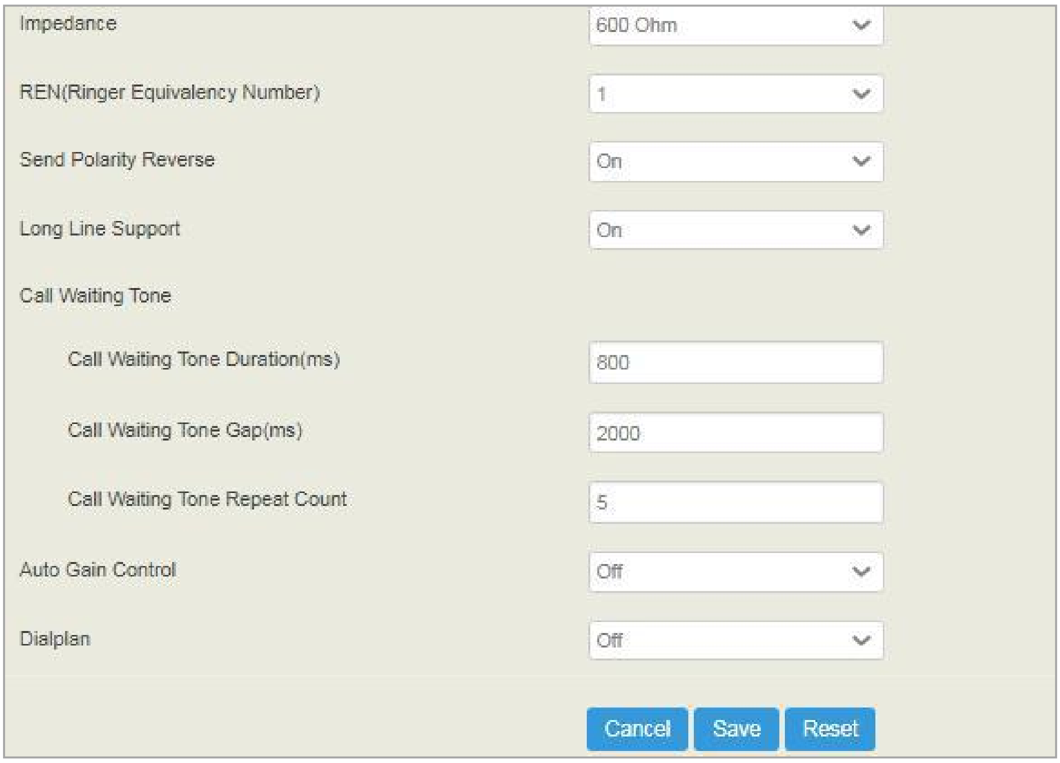
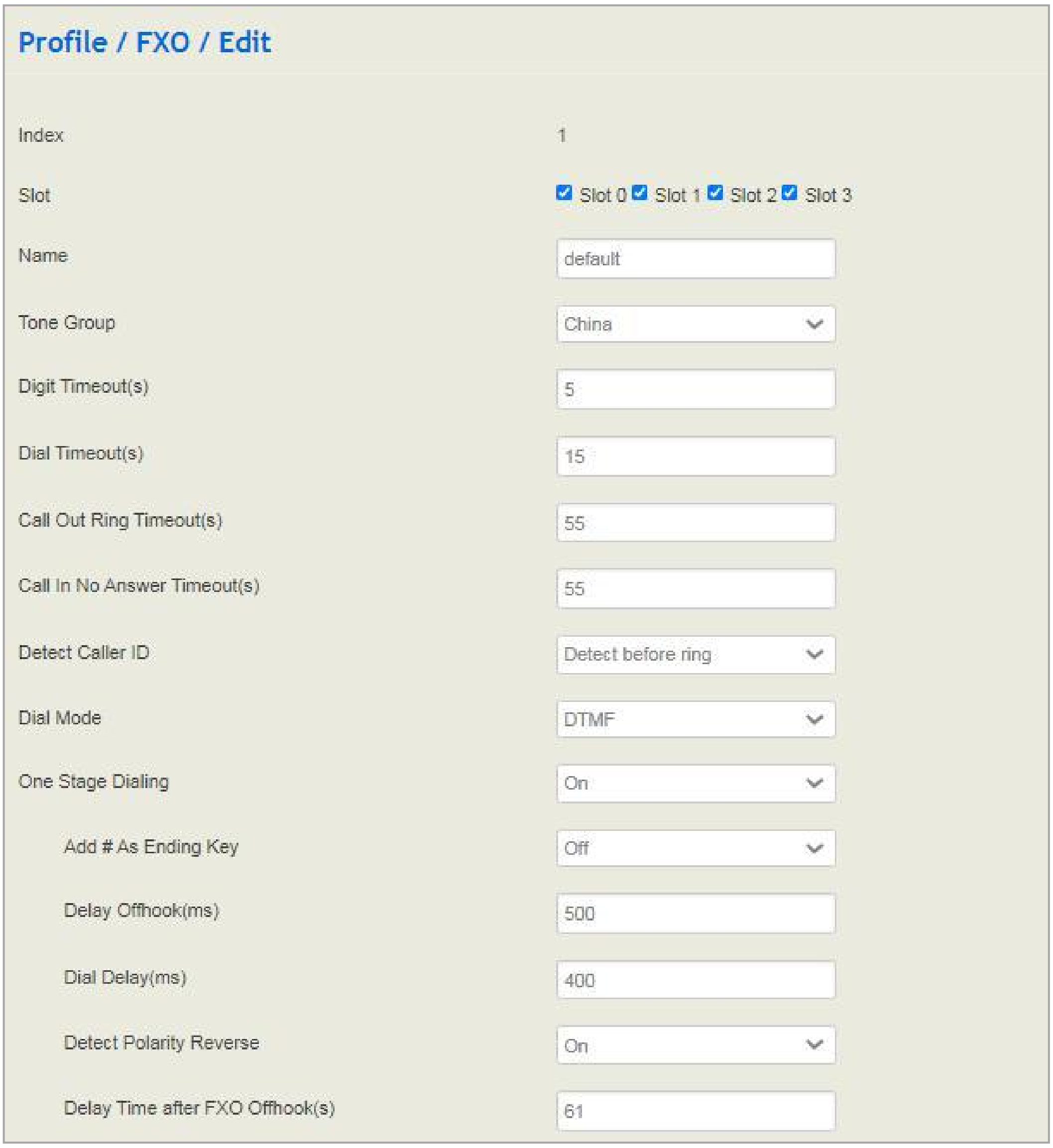


Table 5.5.2.2 Explanation of FXS Parameters

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

	phone will be hanged up.
DTMF Send Interval(ms)	The minimum interval between the sending of two DTMF tone DTMF: Dual Tone Multi Frequency
DTMF Gain	Signal gain of DTMF
DTMF Duration (ms)	The minimum duration of a DTMF tone
Impedance	The impedance （SLIC） matched with analog phones
REN(Ringer Equivalency Number)	The equivalent number of ringing phones. It is used to determine how many devices can be connected by telephone lines. Range:1to 4
Send Polarity Reverse	If polarity reverse is on, call tolls will be calculated based on the changes in voltage. If polarity reverse is off, you need to set the time for offhook detect and call tolls will be calculated starting from the set time.
Long Line Support	The UC350 supports up to 8km wiring length. When the length of the telephone line is less than 1km, the long line mode cannot be enabled
Call Waiting Tone	Configure the duration, gap and repeat count for the call waiting tone
Auto Gain Control	Automatically adjust the gain after enabling
Dialplan	The rules for dialing. The UC350 device supports regular expression. Please make reference to **Profile → Dianplan** section.

Figure 5.5.2-3 FXO Profile



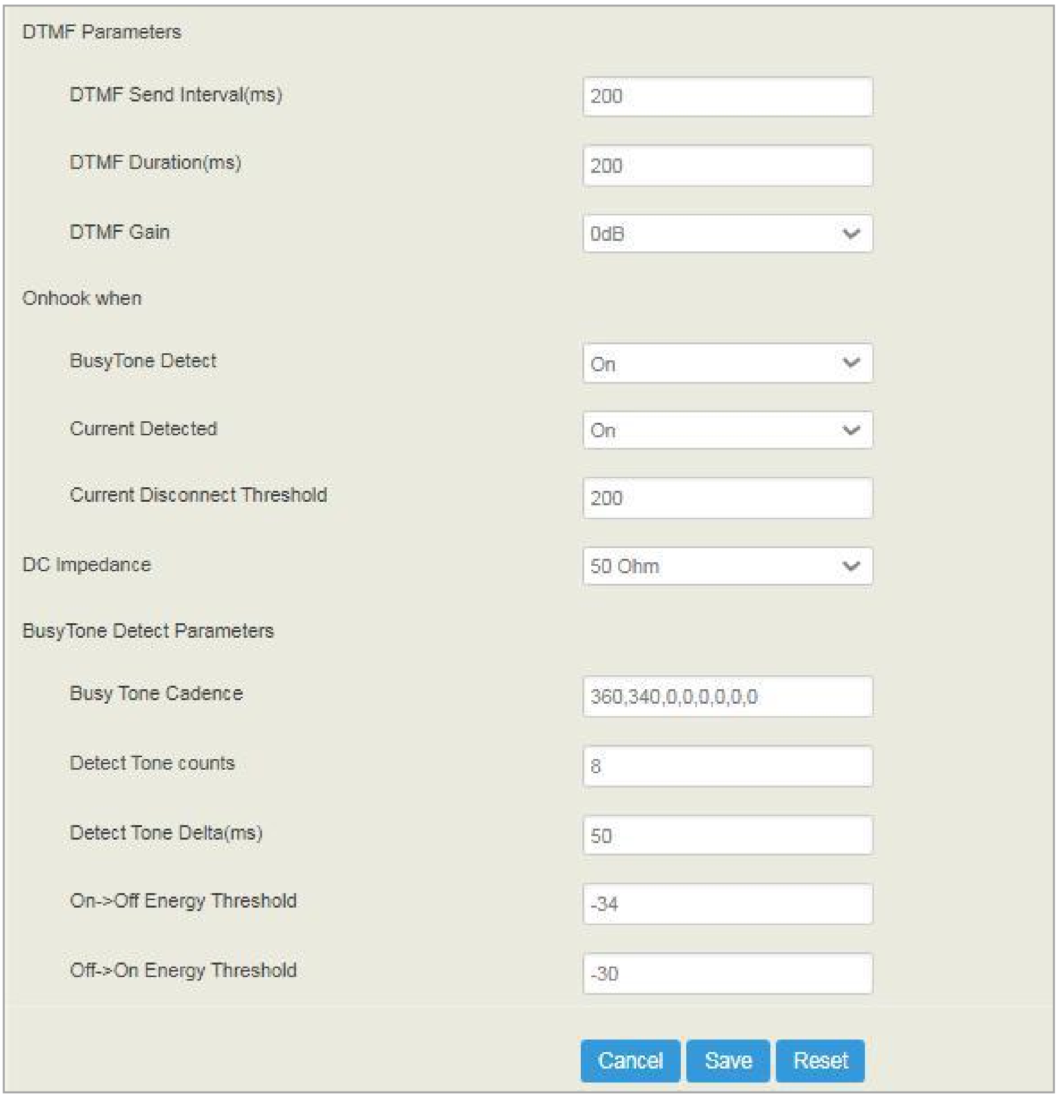


Table 5.5.2.3 Explanation of FXO Parameters

Name	The name of this FXO profile
Tone Group	The national standard of dialing tone, busy tone and ring tone; default value is China
Digit Timeout (s)	The timeout value for dialing a digit of a telephone number; When the time of dialing a digit exceeds this value, the system will think the dialing has completed; Default value is 4 seconds
Dial Timeout (s)	The timeout value for dialing the first telephone number after off-hook; Default value is 10 seconds
Call Out Ring Timeout (s)	The timeout value for the ringing of the analog phones of the FXS port when calling out
Call In No Answer Timeout (s)	The timeout value for ending a call which goes out through the FXS port, when nobody answers the call.

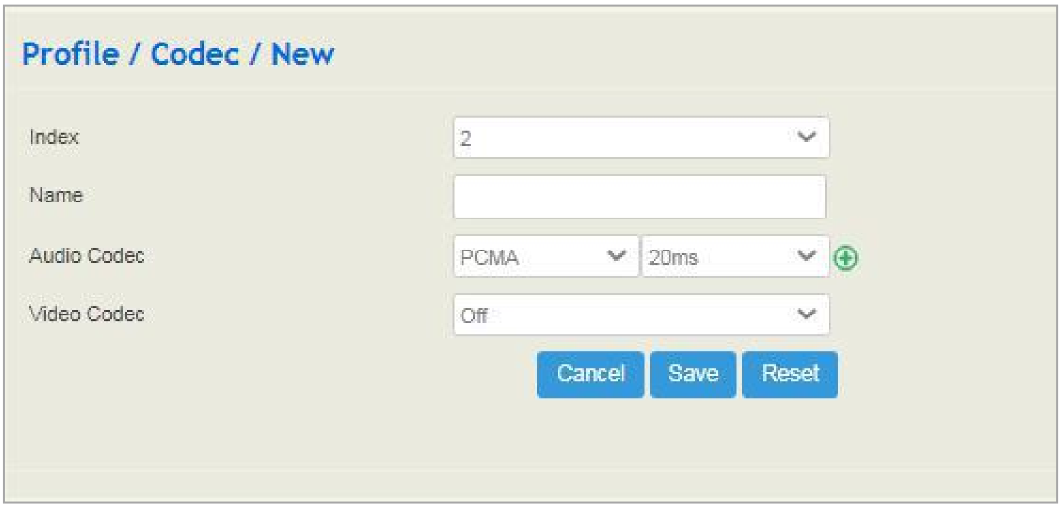
Detect Caller ID	Detect before ring: the CID will be shown before ringing; otherwise, CID will be displayed after ringing; Detect after ring: the CID will be shown after ringing; otherwise, CID will be displayed before ringing Off: the CID will not be shown
Dial Mode	The dialing mode when FXO port calls the PSTN side (supports 3 dialing modes)
One Stage Dialing	The FXO call mode means that when the FXO port makes an outgoing call, the called number in the SIP message is sent to the PSTN side digit by digit at a time.
Add #As Ending Key	After it is turned on, the FXO port makes an outgoing call, it will automatically add # after the original number as a ending key
Dial Delay(ms)	The delay time of dialing. Default value is 400ms
Detect Polarity Reverse	Whether to enable 'detect polarity reverse'. If 'detect polarity reverse' is on, call tolls will be calculated based on the changes in voltage. If 'detect polarity reverse' is off, you need to set the time for offhook delay and call tolls will be calculated starting from the set time.
Delay Time after FXO Offhook(s)	When the FXO port calls the PSTN side, the delay time from the port is on-hook to the port is off-hook (default 1000ms).
DTMF Send Interval(ms)	The minimum interval between the sending of two DTMF tone DTMF: Dual Tone Multi Frequency
DTMF Gain	Signal gain of DTMF
DTMF Duration (ms)	The minimum duration of a DTMF tone
On-hook When (Busy Tone Detect or Current Detected)	When FXO calls the PSTN side, the conditions for the FXO port to on-hook: detect busy tone and detect current. Busy Tone Detect: After enabling, if FXO detects a busy tone, FXO hangs up. Current Detected:After enabling, if FXO detects that there is no current, FXO will hang up
Current Disconnect Threshold	This current threshold is used to determine whether a phone is onhook.
DC Impedance	Matching impedance value when FXO and PBX or PSTN are interconnected.
Detect Tone counts	Set the number of busy notes to check
Detect Tone Delta	Set the error size to check the busy tone
On→ Off Energy	Busy tone signal On→Off energy threshold

Threshold	
Off→ On Energy Threshold	Busy tone signal Off→On energy threshold
Dialplan	The rules for dialing. The UC350 device supports regular expression. Please make reference to **Profile → Dianplan** section.

5.5.3 Codec

UC350 supports voice codec and video codec, including G729, G723, PCMU PCMA, eat. You can adjust the priority of these codec according to you needs.

Figure 5.5-3 Add or Delete Codec Profile



: Edit codec profile.

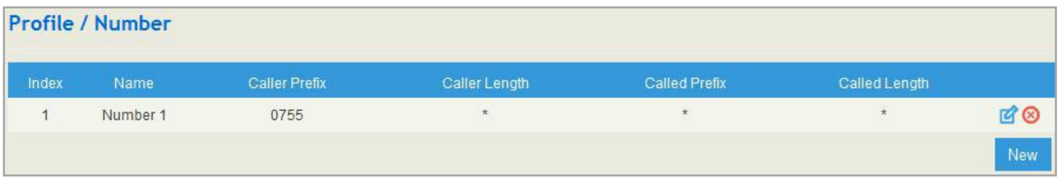
: Delete the codec profile or a codec mode.

: Create a new codec profile.

5.5.4 Number

On the **Profile → Number** interface, you 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 5.5.4-1 Number Profile



: Edit number profile.

: Delete the number profile

Click 

, and you will see the following interface:

Figure 5.5.4-2 Create Number Profile

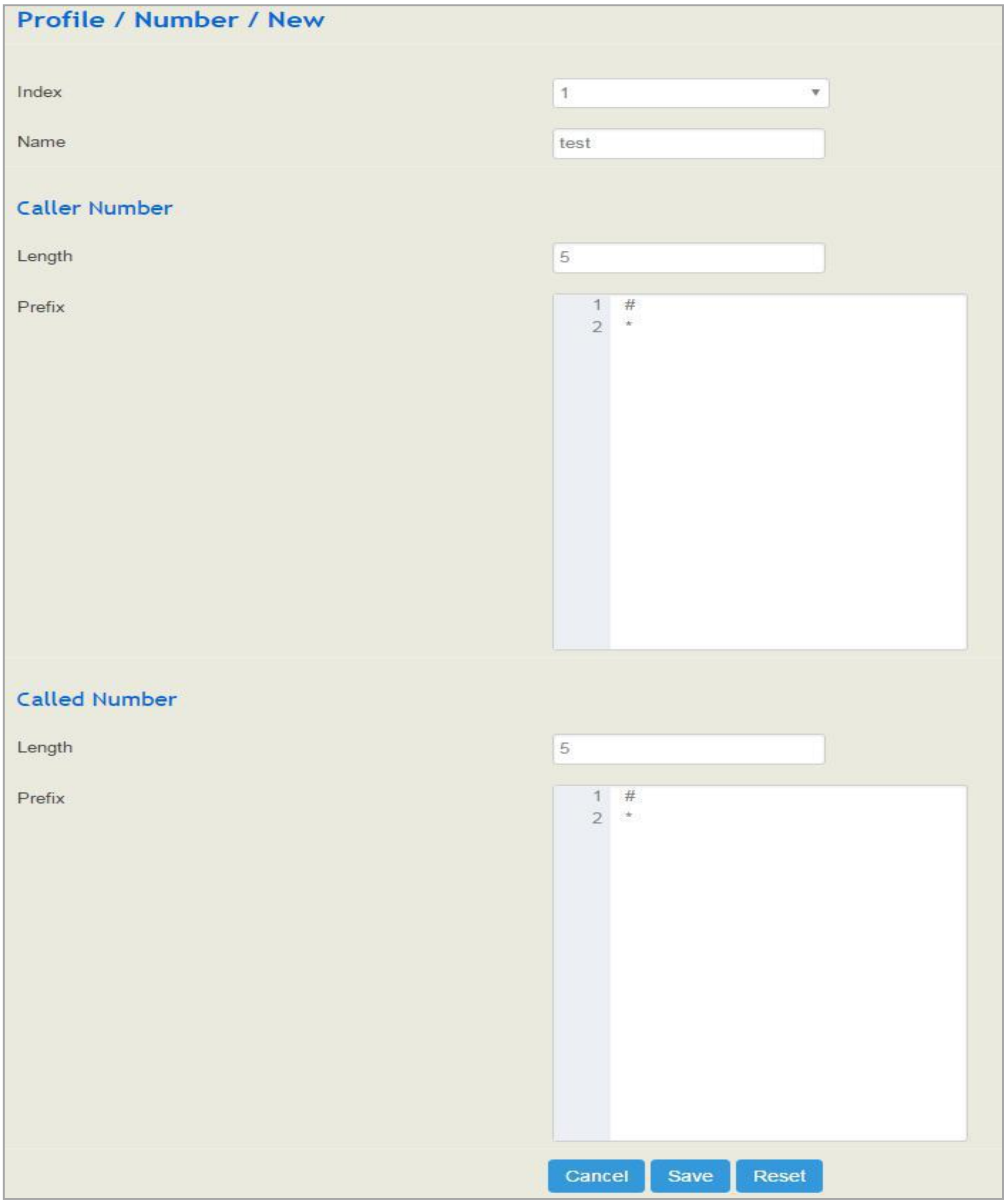


Table 5.5.4-1 Explanation of Number Parameters

Name	The name of the number profile
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.
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

Regex (Regular Expression) Syntax

Table 5.5.4-2 Explanation of frequently-used metacharacters in Regex

^	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 2,3or 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 element. 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	Mark any digit, equal to [0-9]
\D	Mark any character that is not a digit, equal to [^0-9]
\s	Mark any blank character such as a space or a tab.
\S	Mark any character that is not a blank character

Examples:

^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.

5.5.5 Time

On the **Profile → Time** interface, you 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.

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

Figure 5.5-5 Time Profile

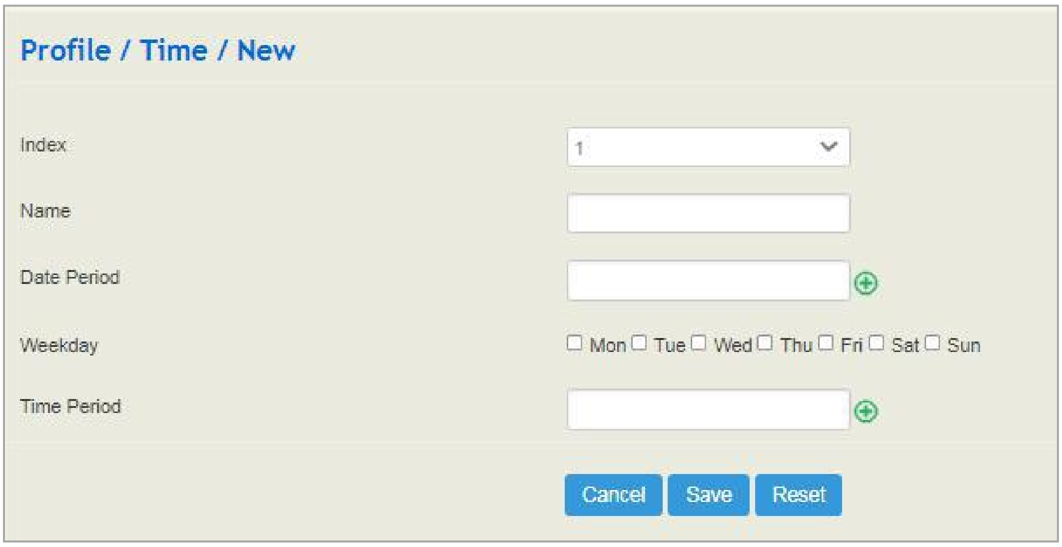


Table 5.5-5 Explanation of Time Parameters

Name	The name of the number profile
Date Period	Configure the starting date and ending date of a period : Add a date period : Delete a date period
Weekday	Choose a weekday
Time Period	Choose the starting time and ending time of a day

5.5.6 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 you will see the following interface:

Figure 5.5-6 Create Manipulation Profile

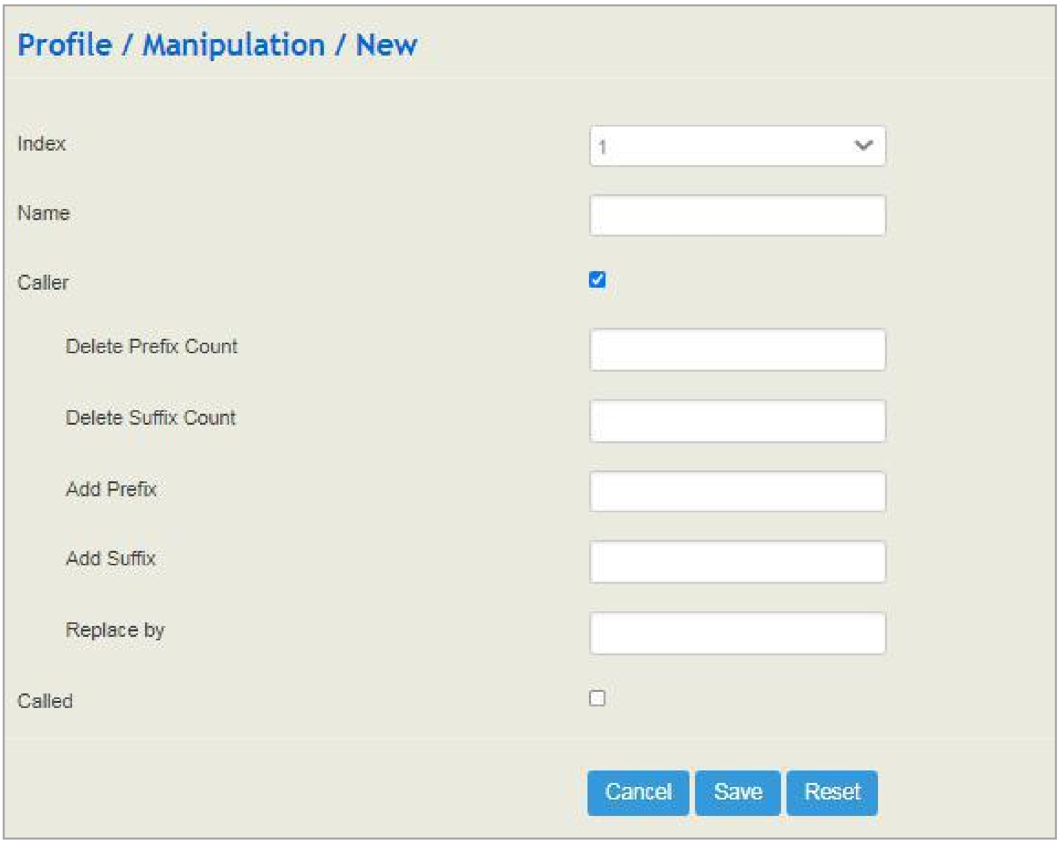


Table 5.5.6 Explanation of Manipulation Parameters

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 replace the caller number or the calling number
☑	If the checkbox on the right of Caller is selected, it means the caller number will be manipulated; if the checkbox on the right of Called is selected, it means the called number will be

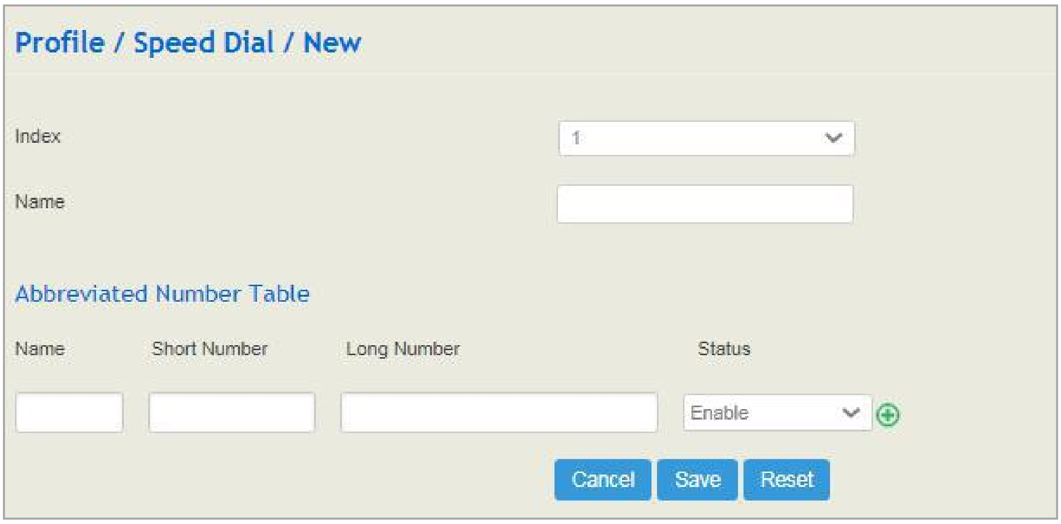
	manipulated.

Note: During number manipulation, deletion rules are carried out first, followed by adding rules. If ‘Replace by’ has been set, deletion rules and adding rules are invalid.

5.5.7 Speed Dial

On the **Profile → Speed Dial** interface, you can set one-digit or two-digit speed dial numbers for FXS/SIP calls. 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 FXS/SIP extension, the FXS/SIP extension only needs to dial 1 and the call will be directed to the extension number of 8000.

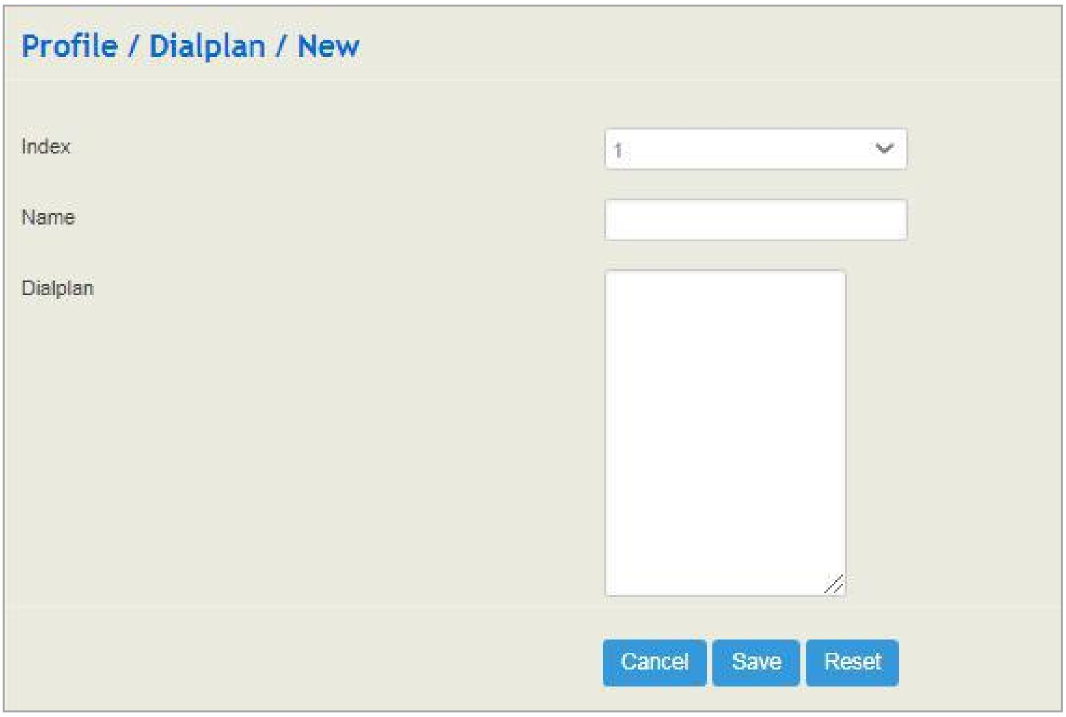
Figure 5.5-7 Add Speed Dial Profile



5.5.8 Dialplan

Dialplan is used for number dialing of calls through FXS ports. It supports Regular Expression (Regex) and DigitMap.

Figure 5.5-8 Add Dialplan



Regex (Regular Expression) Syntax

<table>
<tr><td>^</td><td>Matches the starting position in a number string. For example, ^134 matches the numbers starting with 134</td></tr>
<tr><td>$</td><td>Matches the ending position of a string. For example, 2$ matches the numbers ending with 2.</td></tr>
<tr><td>|</td><td>Separates alternate possibilities. For example, 2|3|4 means 2,3or 4.</td></tr>
<tr><td>\</td><td>Marks the next character as a special character, a literal, a backreference, or an octal escape</td></tr>
<tr><td>[ ]</td><td>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".</td></tr>
<tr><td>[^]</td><td>Matches any one character except those enclosed in [ ]. For example, [^9] matches any character except 9.</td></tr>
<tr><td>.</td><td>Matches any single character except the newline character. For example, 3.4 matches 314, 324, 334, 344.</td></tr>
<tr><td>?</td><td>Indicates there is zero or one of the preceding element. For example, colou?r matches both color and colour</td></tr>
<tr><td>*</td><td>Indicates there is zero or more of the preceding element. For example, ab*c matches ac, abc, abbc, abbbc, and so on.</td></tr>
<tr><td>+</td><td>Indicates there is one or more of the preceding element. For example, ab+c matches abc, abbc, abbbc, and so on, but not ac</td></tr>
<tr><td>/d</td><td>Mark any digit, equal to [0-9]</td></tr>
<tr><td>/D</td><td>Mark any character that is not a digit, equal to [^0-9]</td></tr>
<tr><td>/s</td><td>Mark any blank character such as a space or a tab.</td></tr>
</table>

/S	Mark 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.

DigitMap 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 of the preceding element

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

5.5.9 AutoCLIP

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

Figure 5.5-9 AutoCLIP Rule

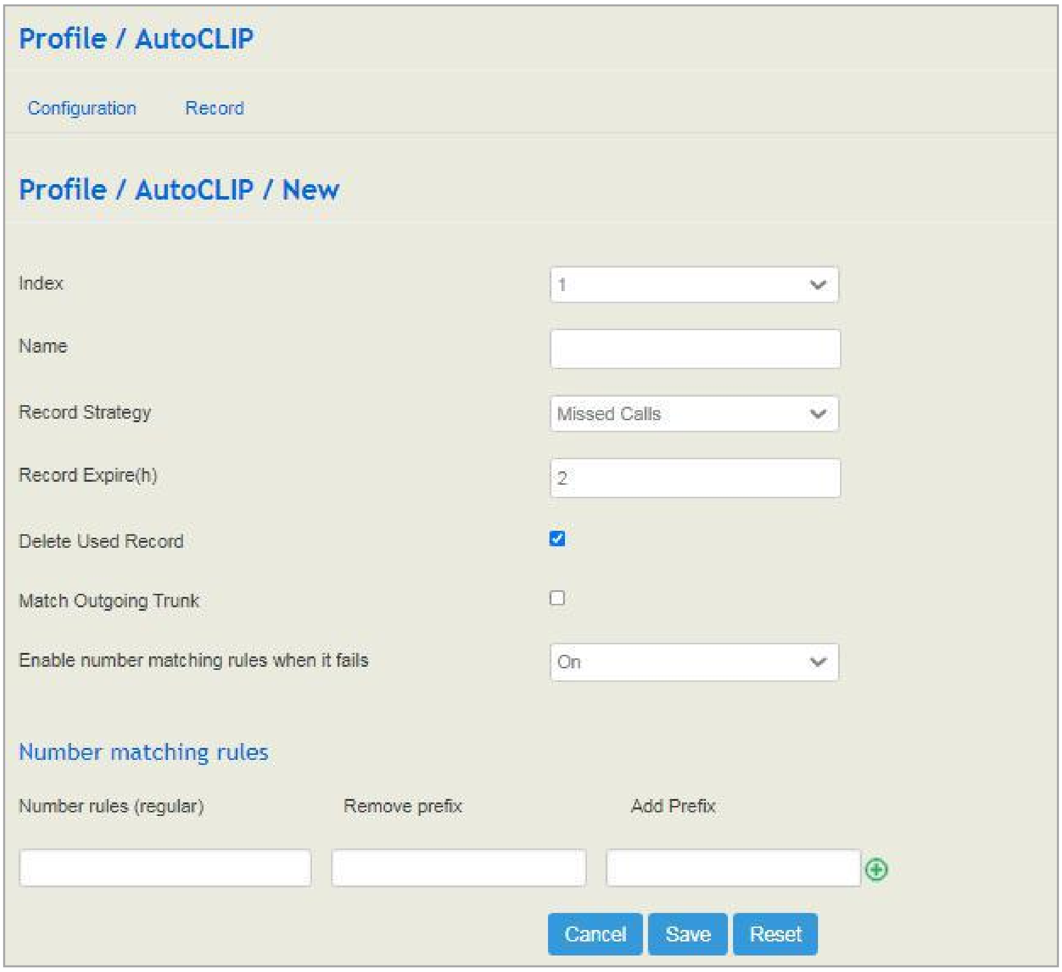


Table 5.5.9 Explanation of AutoCLIP Rule

Index	The index of AutoCLIP profile
Name	The name of AutoCLIP profile
Record Strategy	You can choose missed calls or all calls. If missed calls is selected, uc350 will record the missed calls of the trunk. If all calls are selected, all the calls going through the trunk will be recorded
Record Expire (hour)	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. If this parameter is selected, those records that have been used to match extension number or trunk will be deleted.
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 Fails	Enable number matching rules

5.5.10 Recording

How to Record Calls:

Configure a recording profile (or choose one of the two default recording profiles), and then add it to a SIP/FXS route. When there are calls going through the route and match the recording profile, the calls will be recorded.

Figure 5.5.10-1 Add Recording Profile

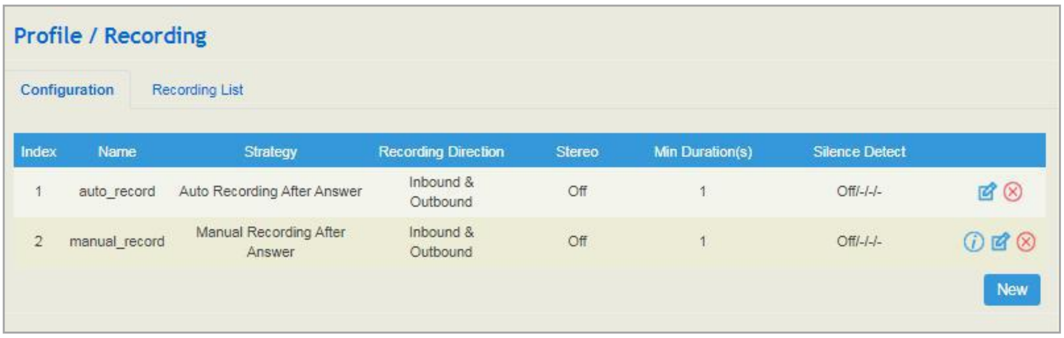


Figure 5.5.10-2 Recording Profile

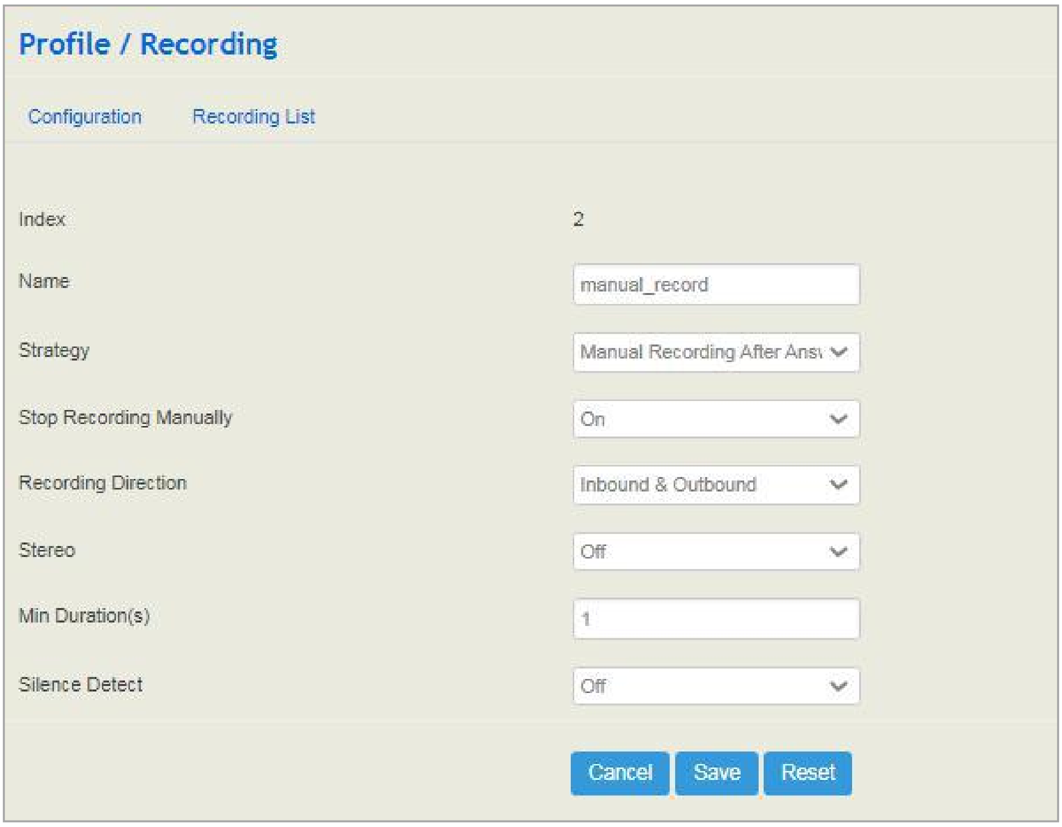
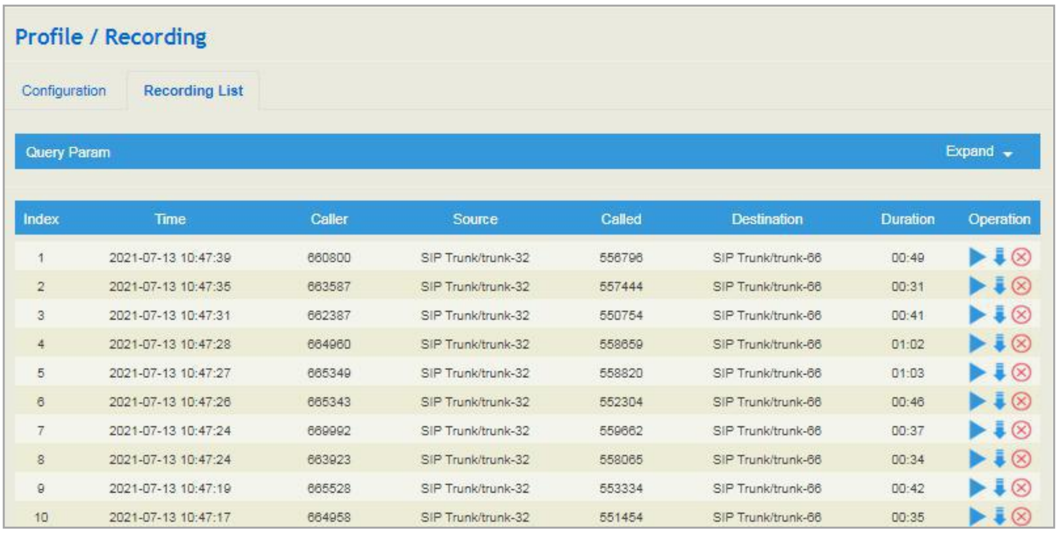


Table 5.5.10 Explanation of Recording Profile

Index	The index of the recording profile. Range: 1-32
Name	The name of the recording profile, used to identify the recording profile
Strategy	Auto Recording after Answer: start recording after the callee pick up the phone. Ban Recording: ether caller or callee enables his function, and then the call in both directions will not be recorded. Manual Recording after Answer: press *3 to start recording after the callee answers the call.
Recording Direction	Inbound & Outbound: If this recording profile is added to FXS/SIP extension, both inbound and outbound calls will be recorded. Inbound: If this recording profile is added to FXS/SIP extension, only inbound calls will be recorded. Outbound: If this recording profile is added to FXS/SIP extension, only outbound calls will be recorded. Note: If this recording profile is added to routing, this parameter is invalid and all calls going through the routing will be recorded.
Stereo	When enabled, the file size will be twice that of mono for the same call duration
Min Duration	If the actual recording time is shorter than this value, the recording file will not be saved.
Silence Detect	When silence is detected, no recording will be done during muting

You can click **Recording List** to view the recording files which show the caller/called number, recording duration and so on. You can also play, download or delete the recording files on this interface.

Figure 5.5.10-3 View Recording List



5.5.11 Voicemail

On the **Profile→ Voicemail** interface, you can configure the location, number and duration of a voice mail.

How to use voice mail:

Navigate to **Extension → SIP** or **Extension → FXS** interface, enable the voice mail function, and the voice mail will be activated when the call times out..

Figure 5.5.11-1 Voicemail Profile

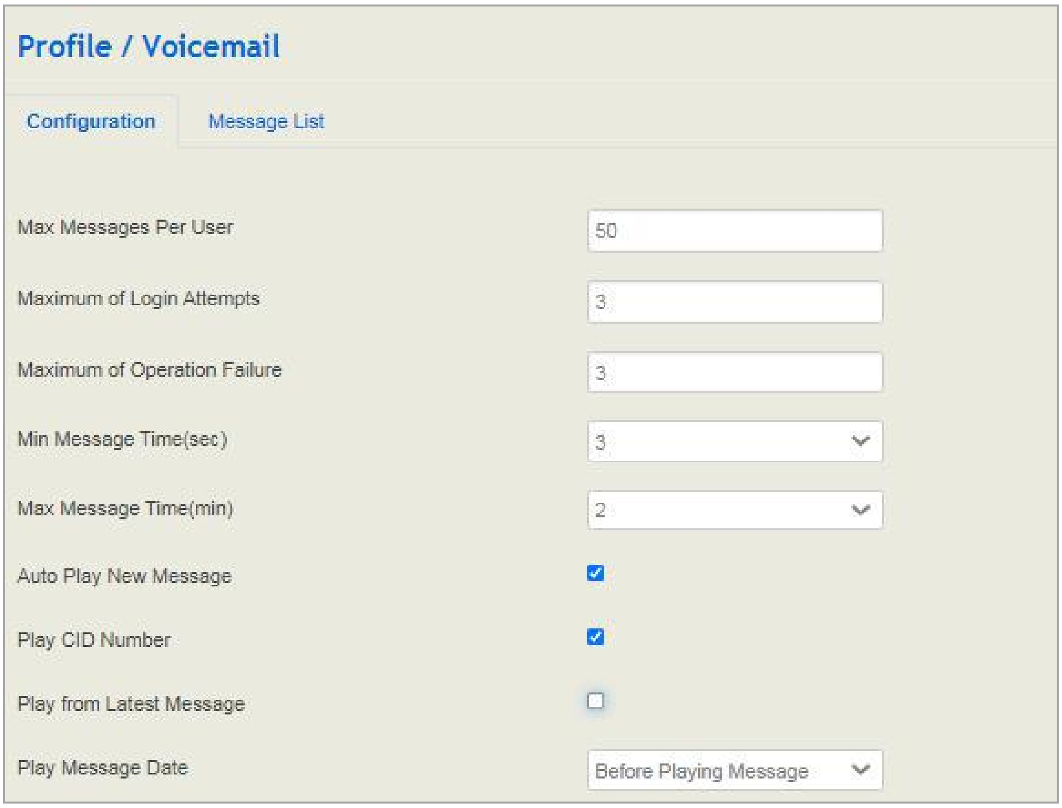


Table 5.5.11 Explanation of Voicemail Profile

Max Message 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 (second)	The minimum duration of a voice mail
Max Message Time (second)	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. You can choose ‘Before Playing Message’, ‘After Playing Message’ and ‘Never’.

You can click **Message List** to view the voicemail files which show the caller/called number, message duration and so on. You can also play, download or delete the message files on this interface.

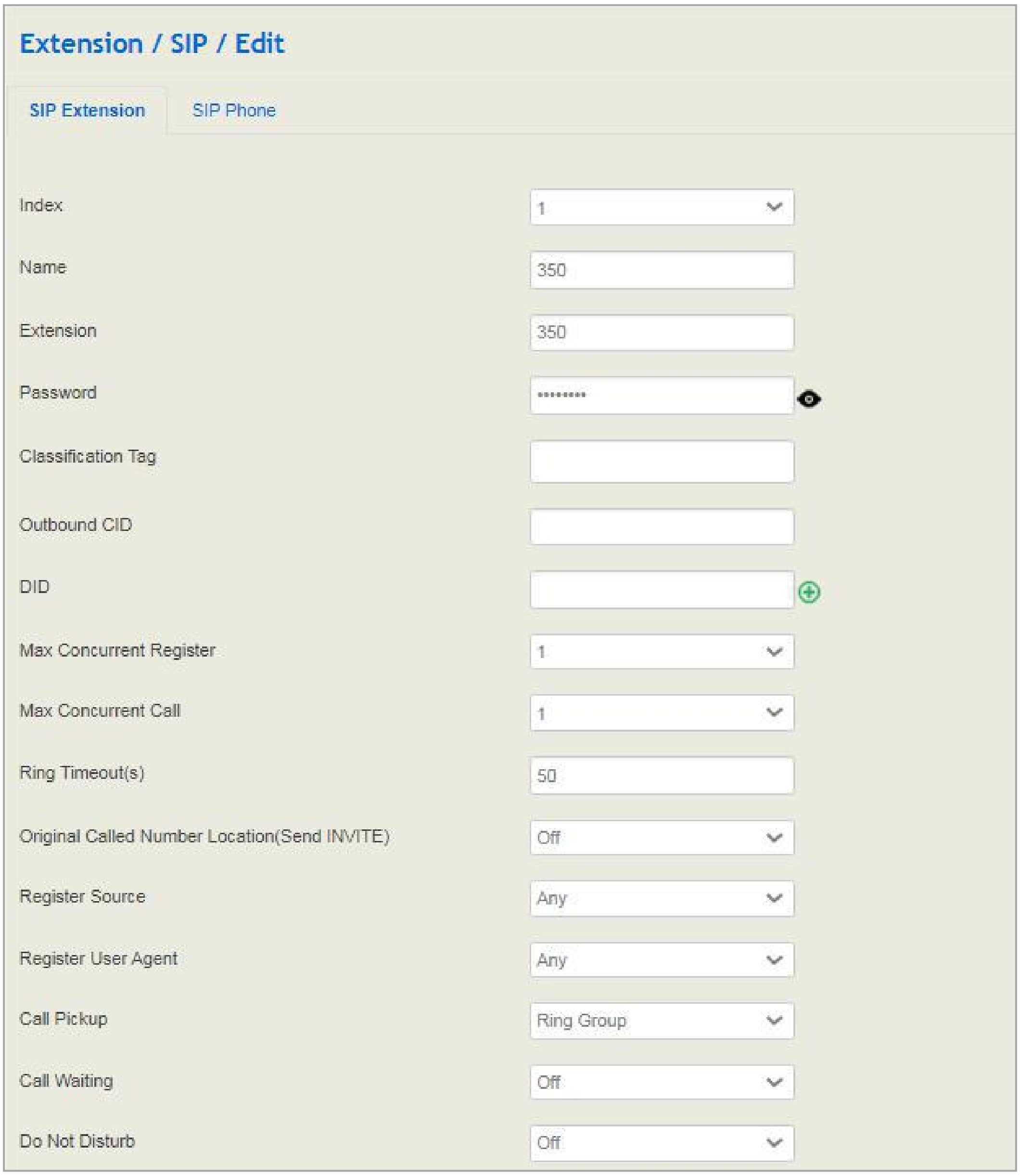
Figure 5.5.11-2 View Message List

5.6 Extension

5.6.1 SIP

On the **Extension → SIP** interface, you can configure the SIP accounts registered in the UC350 by SIP clients (hereby UC350 is regarded as a SIP server).

Figure 5.6-1 SIP Extension



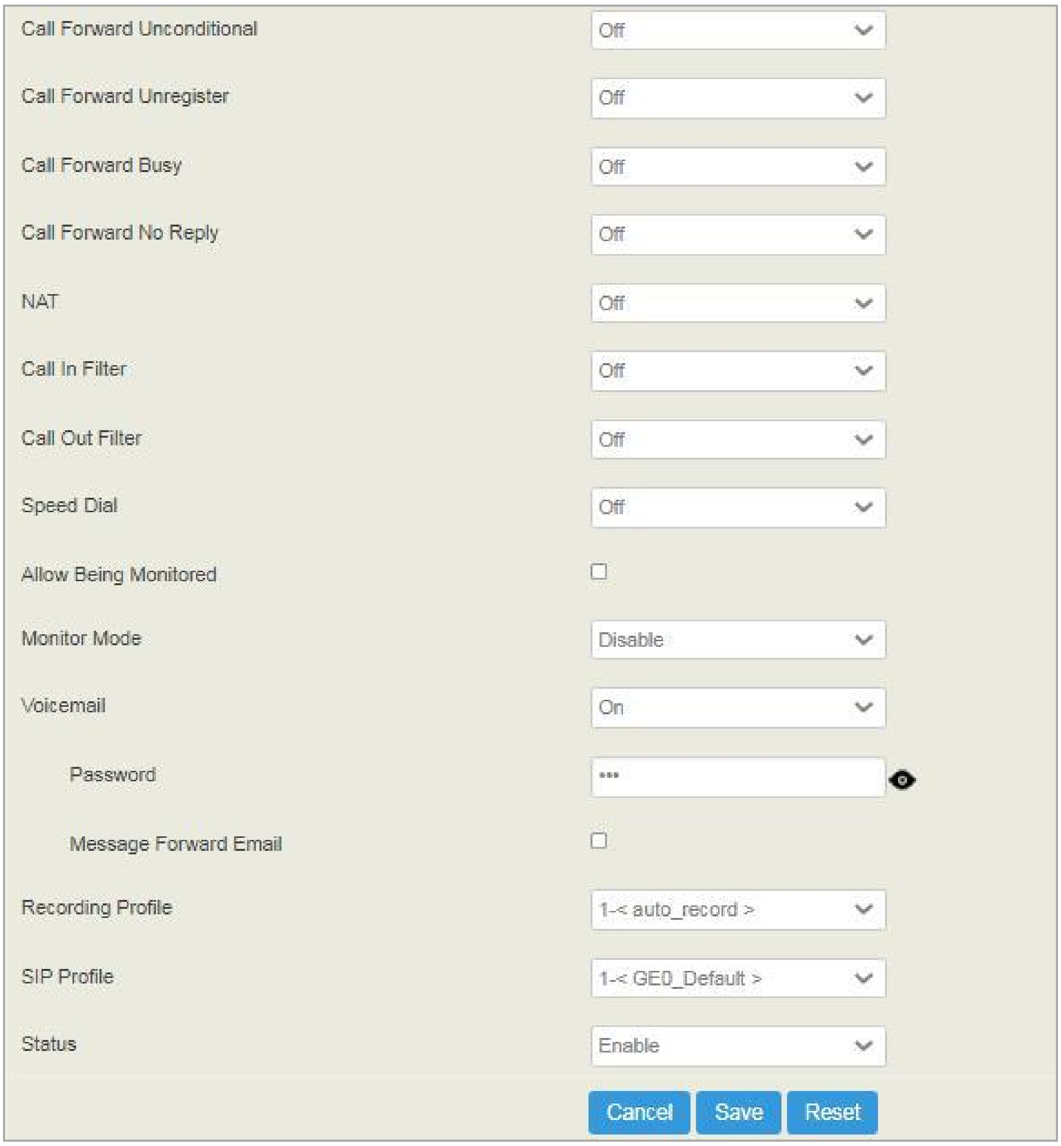


Table 5.6.1 Explanation of SIP Extension Parameters

Name	The name of this SIP extension
Extension	The SIP account of the extension registered in UC350 by a SIP client
Password	The password of the SIP account registered in UC350 by a SIP client
Classification Tag	Labels for extension classification
Outbound CID	After the outgoing caller number is configured, the caller number dialed from the SIP extension is replaced with the number configured here
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.

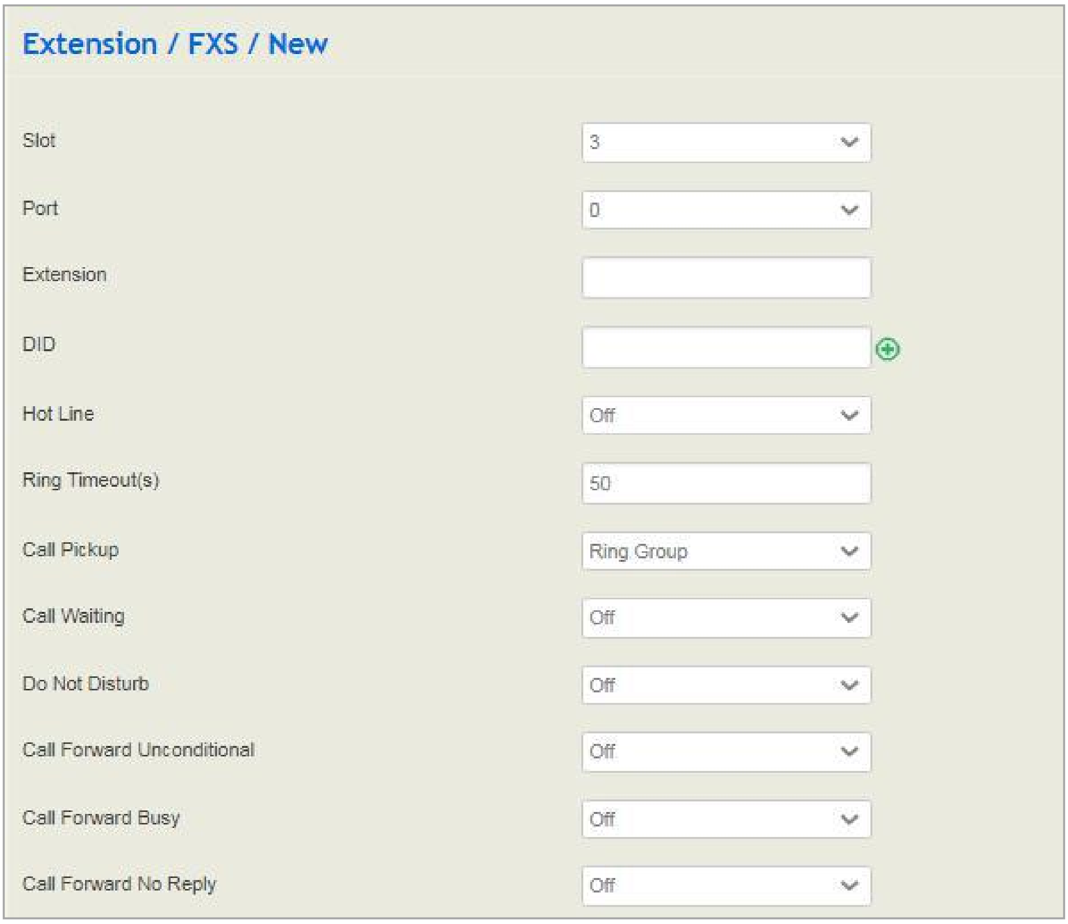
Max Concurrent Register	Number of clients that can register online at the same time
Max Concurrent Call	The number of concurrent calls that can be made at the same time
Ring Timeout(s)	The ringing timeout period for incoming calls to this extension, the default value is 50. If the extension does not go off-hook within 50s after ringing, the device will initiate disconnection
Original Called Number Location(Send INVITE)	When sending INVITE, configure the location of the original called number
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
Register User Agent	Filter the useragent field in the register message during registration
Call Pickup	After configuration, the designated call can be picked up（ring group/local extension, the default is the ring group）
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.
Do Not Disturb	If 'Do Not Disturb' feature is enabled, calls cannot reach the called party.
Call Forward Unconditional	If 'Call Forward Unconditional' feature is enabled, all coming calls will be forwarded to a preset number.
Call Forward Unregister	When the SIP extension is not registered, you 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.
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.
NAT	If NAT is enabled, the IP address of SIP extension in LAN will be turned into the outbound IP address of public network, thus making NAT traversal possible
Call In Filter	When you breathe in to SIP, you match the relevant filter conditions
Call Out Filter	When the SIP is called out,The filter conditions are matched
Speed Dial	Configuration for Speed dial
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
Recording Profile	When recording is enabled, FXS calls will be recorded according to the recording rules
SIP Profile	The SIP profile that is selected for the extension
Status	If it is enabled, this SIP extension is registered to UC350; Otherwise the SIP extension is not registered

5.6.2 FXS

On the **Extension → FXS** interface, you can configure the parameters of the FXS extension.

Figure 5.6-2 FXS Extension



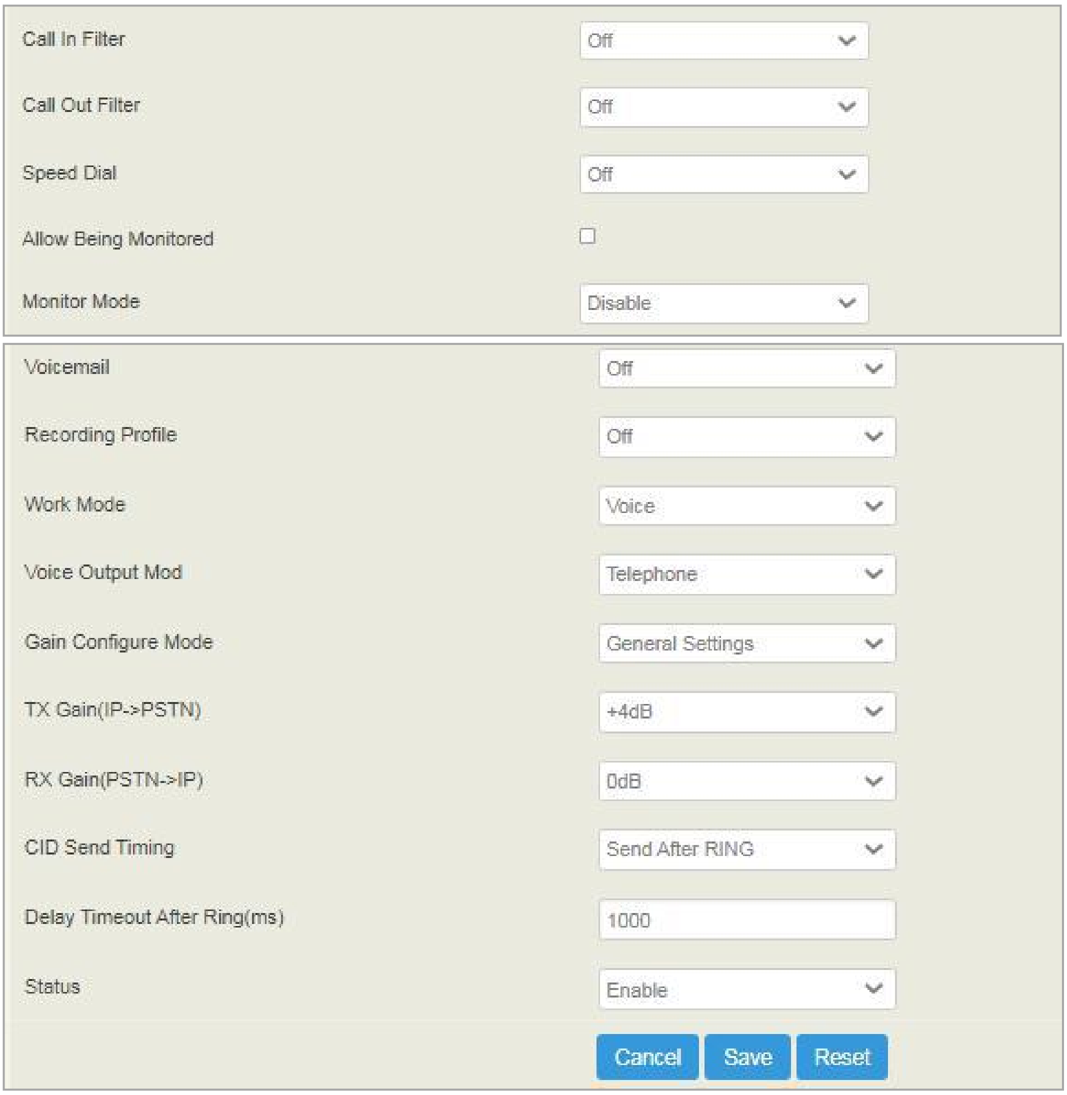


Table 5.6.2 Explanation of FXS Extension Parameters

Extension	The extension account of FXS port, which is used to register
DID	Direct Inward Dialing; if the called number is same with DID, the call will be directly forwarded to the extension, rather than choosing a route.
Hot line	If hotline is enabled, calls will directly go to the hotline number
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
Call Pickup	After configuration, the designated call can be picked up（ring group/ local extension, the default is the ring group）
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.

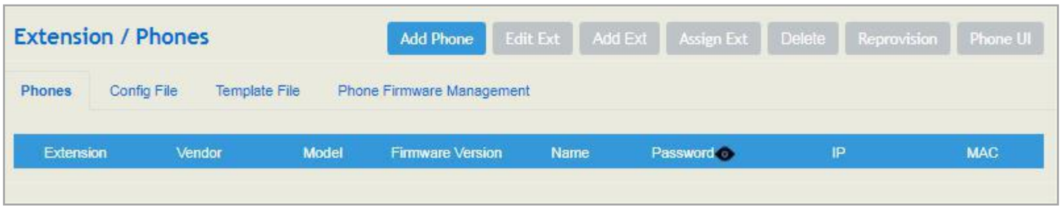
Do Not Disturb	If 'Do Not Disturb' feature is enabled, calls cannot reach the called party.
Call Forward Unconditional	If 'Call Forward Unconditional' feature is enabled, all coming calls will be forwarded to a preset number.
Call Forward Busy	If 'Call Forward Busy' feature is enabled, new coming call will be forwarded when the corresponding local port is busy.
Call Forward No Reply	If 'Call Forward No Reply' feature is enabled, calls will be forwarded when nobody answer the calls during a specified period.
Input Gain	The receiving gain of the FXS port
Output Gain	The sending gain of the FXS port
Work Mode	The working mode of the FXS port, including Voice and POS
Call In Filter	When a call is given to the FXS port of UC350, the call will not be connected to the FXO port if it is in the blacklist
Call Out Filter	When a call goes out from the FXS port of UC350, the call cannot go out if it is in the blacklist
Speed Dial	Configuration for Speed dial
Voicemail	Choose to on or off the voice mail
Recording Profile	When recording is enabled, FXS calls will be recorded according to the recording rules
Work Mode	Configure the working mode of the FXS port
Voice Output Mode	Configure the voice output mode of the FXS port
Gain Configure Mode	Select the gain configuration mode of the FXS port (general settings and advanced settings), TX gain and RX gain are newly added to advanced settings than general settings.
TX Gain(IP→PSTN)	The volume level of the remote end during a call; that is, adjusting the "TX gain" will affect the volume of the sound heard by the remote end.
RX Gain(PSTN→IP)	The volume level of the user during the call; that is, adjusting the "RX gain" will affect the level of sound you hear.
CID Send Timing	Set the phone caller ID display before ringing or after ringing
Delay Timeout After Ring(ms)	CID sending timing is configured when sending after ringing. Set how long the phone will ring before sending CID. When "CID send timing" is configured to "send after ring", you need to configure the delay timeout, that is, how long to ring before sending the CID
Status	If it is on, this FXS extension can be used, otherwise, the FXS extension is unavailable.

5.6.3 Phones

On the "**Extension -> Phone**" interface, the user can deliver the configuration to the phone according to the template file.

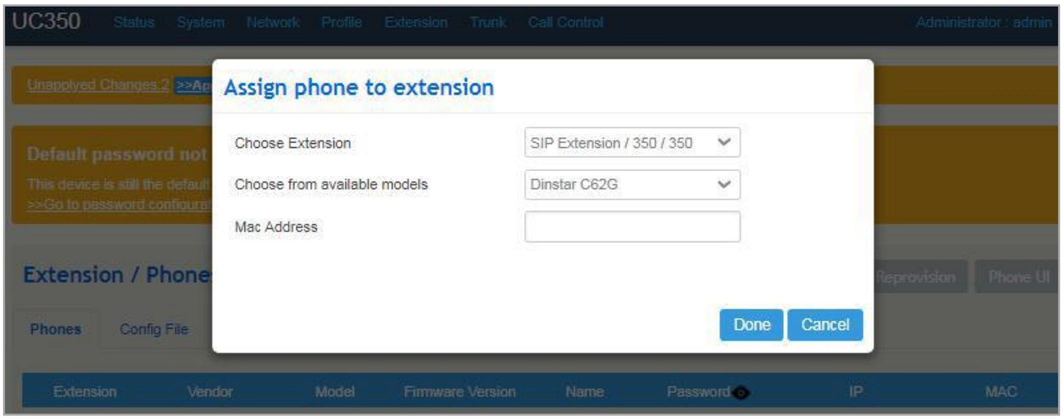
After enabling PNP, the phone will periodically send a Subscribe message to the multicast address. If the PBX receives the multicast message, it will list the phone models in the PBX configuration list.

Figure 5.6.3-1 Phones



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

Figure 5.6.3-2 Add phone



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, you can use the assigned extension to register.

5.6.4 Ring Group

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

Figure 5.6-4 Ring Group

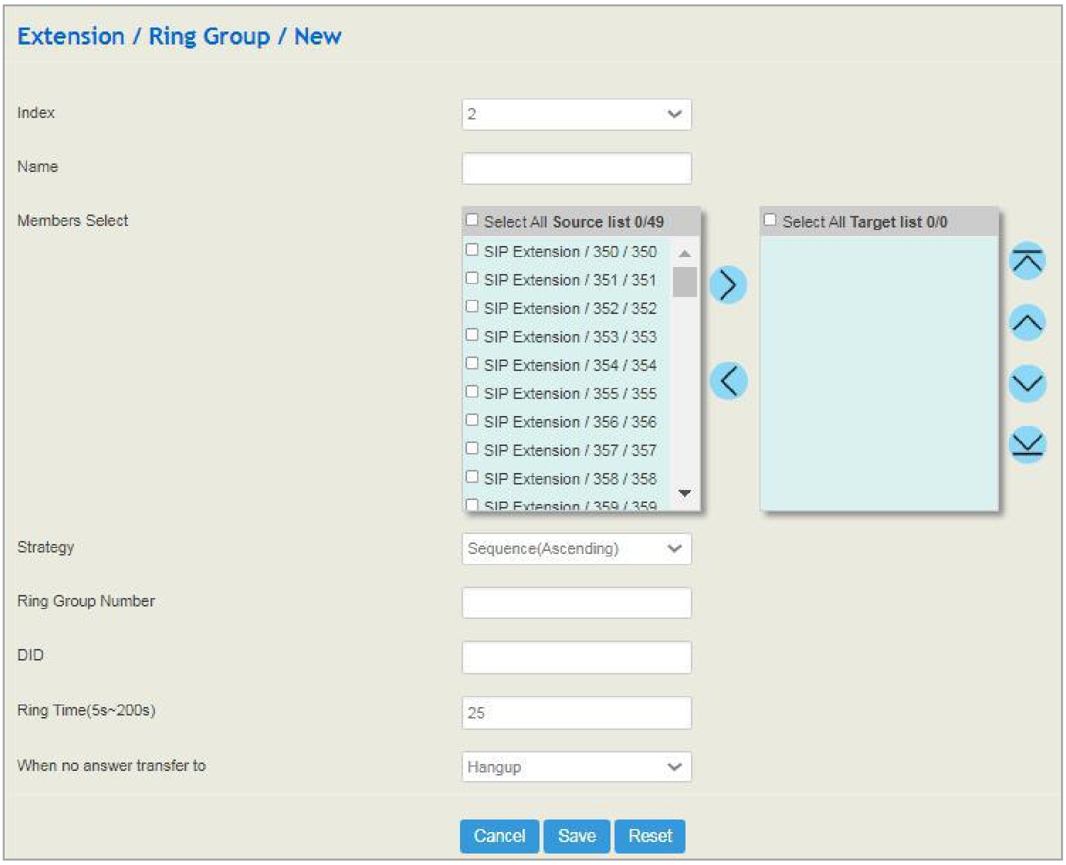


Table 5.6.4 Explanation of Ring Group Parameters

Name	The name of this ring group
Members Select	Select the FXS extension and an SIP extension or several SIP extensions; : Add an extension to the ring group : Delete an extension from the ring group
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（5-60s）	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, you 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.

5.6.5 Paging Group

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

Figure 5.6-5 Paging Group

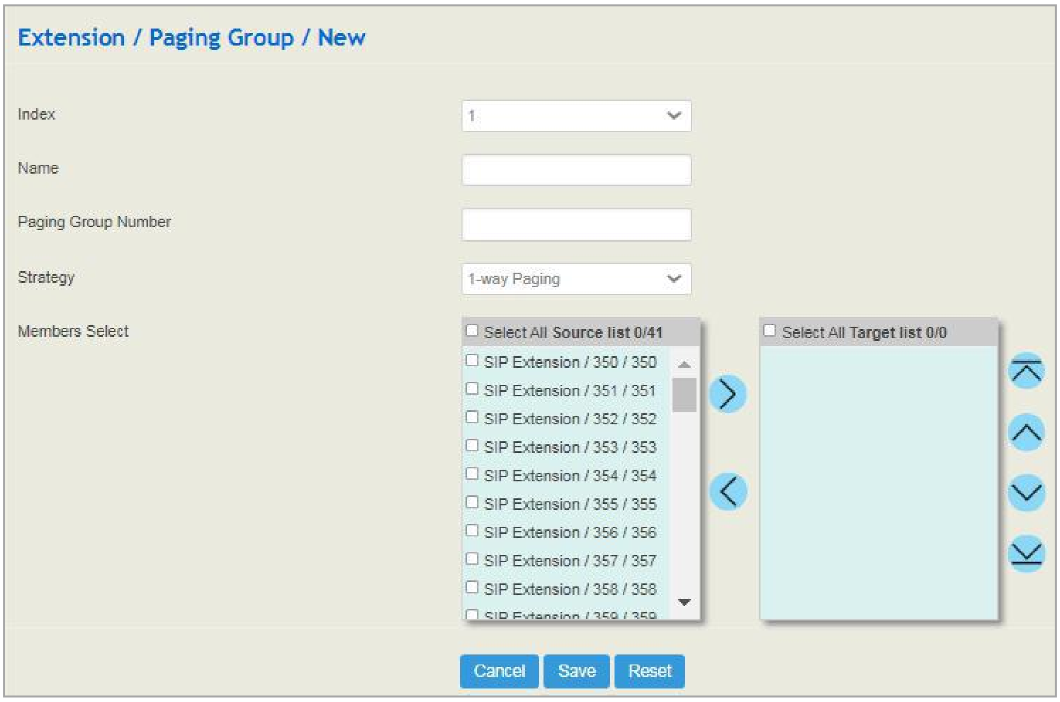


Table 5.6.5 Explanation of Paging Group Parameters

Name	The name of this paging group
Paging Group Number	The number of the paging group.When there calls given from FXS/ FXO/SIP to this number, the calls will be led to one extension of the 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. Click  to add an SIP extension to the paging group;

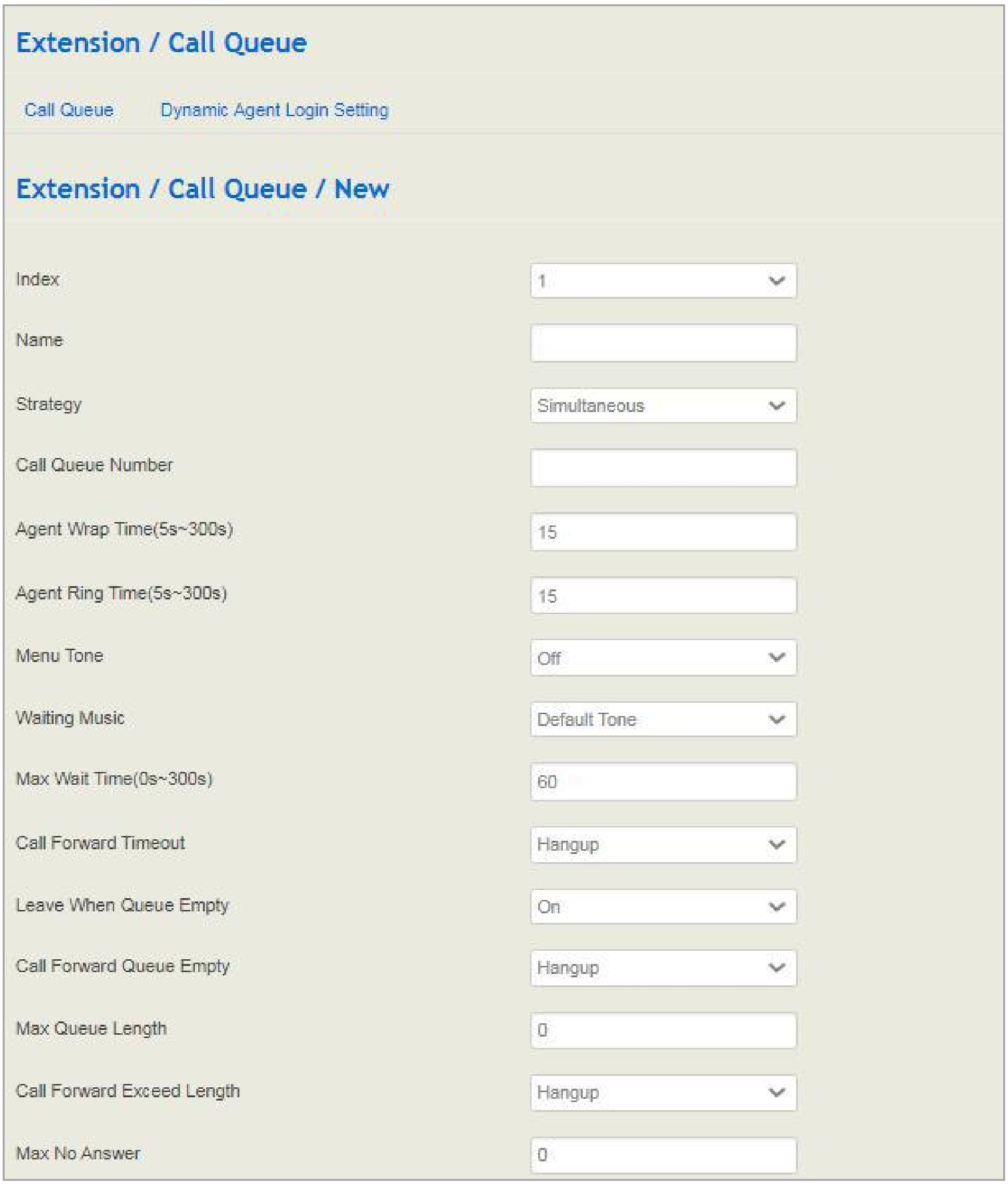
	Click to delete an SIP extension from the paging group.

5.6.6 Call Queue

On the "**Extension -> 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 5.6-6 New Call Queue



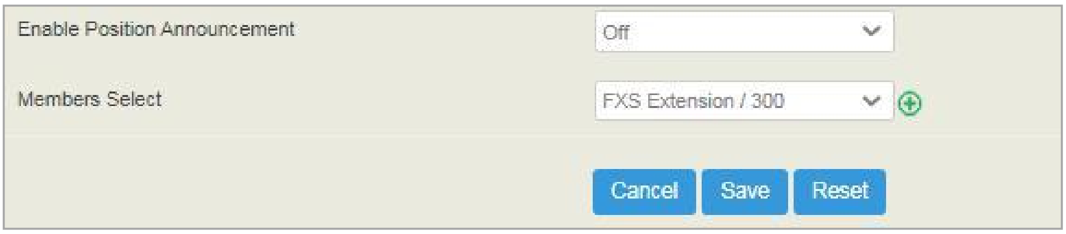


Table 5.6.6 Explanation of Call Queue Parameters

Name	The name of the call queue
Strategy	Calls into the queue, the agents ring according to the strategy. Simultaneous: The agents ring together Sequential Mode: 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 rotation mode : When there is no incoming call , a new user calls in, and the ringing starts from the next agent who hangs up last before Max idle time : Idle time, namely the time from the end of the agent's last call to the present; ringing in the order from longest to shortest time Min talk time :The ringing starts from the least to the most according to the times of calls
Call Queue Number	The number of the call queue can be called into the queue
Agent Wrap Time(5s-300s)	The interval time between the next ringing after the agent call hangs up
Agent Ring Time(5s-300s)	If the ringing exceeds the time, it will call to the next agent
Menu Tone	The first menu tone the remote end hears when calling in
Waiting Music	The remote end waits for the agent to answer the waiting tone after calling in
Max Wait Time(0s-300s)	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.
Call Forward Timeout	If the caller times out, other actions can be configured
Leave When Queue Empty	If there is no agent in the queue, it will exit the queue (if there is ON-Break, it is still an agent), and the call transfer will be performed

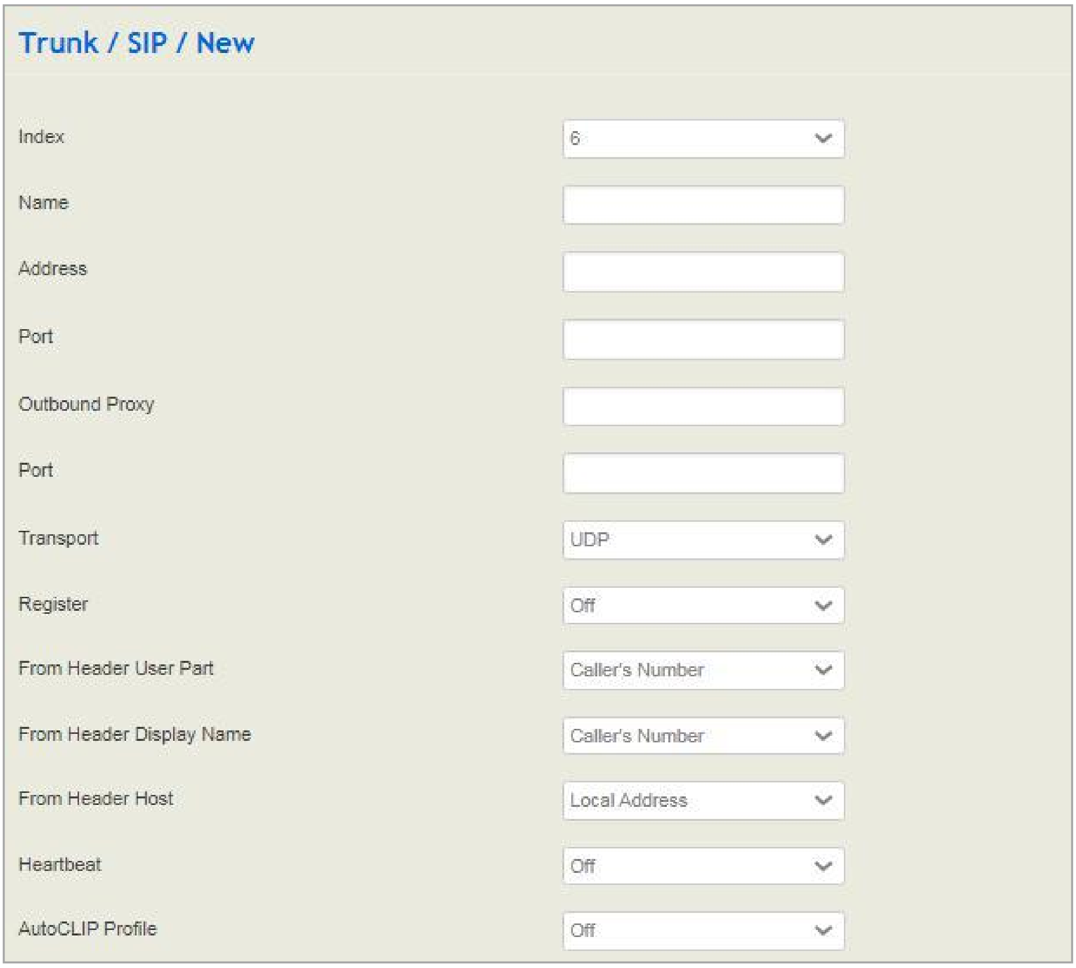
	when the queue is empty
Max Queue Length	How many users are waiting, those connected are not counted, 0 means no limit, hang up if the maximum number of queues is exceeded
Max No Answer	If the times that the agent does not answer is exceeded, it will enter On-Break state, in this state, it will not be ringing again until the agent answer
Enable Position Announcement	Timely notify the user of the waiting position in the queue, the first one does not notify

5.7 Trunk

5.7.1 SIP

SIP trunk can realize the connection between UC350 and IPPBX or SIP servers.

Figure 5.7-1 SIP Trunk



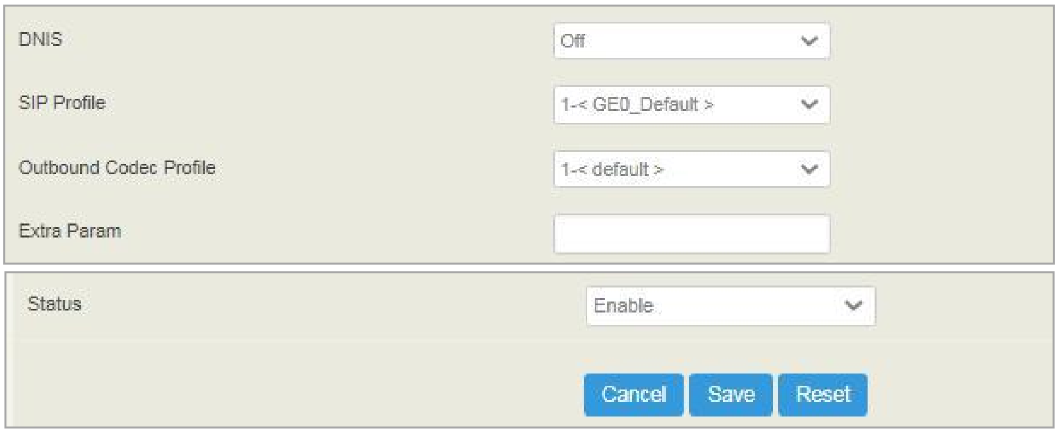


Table 5.7.1 Explanation of SIP Trunk Parameters

Name	The name of the SIP trunk
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
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
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.
Heartbeat	If heartbeat in on, heartbeat (options) messageswill be sent to examine the connection with servers; The default value is 'Off'
AutoCLIP Profile	AutoCLIP is mainly used to SIP trunks / FXO trunks and it helps record the outgoing and incoming calls of a trunk.
DNIS	When the SIP trunk calls in, the called number matches the DNIS, then the *from displayname* of the *invite* should be the display name of the called number
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

SIP Profile	The SIP profile of the SIP Trunk; make reference to Profile → SIP section
Status	If it is enabled, it means the SIP Trunk can be used; otherwise, the SIP trunk is unavailable

Note:

If UC350 is regarded as a terminal and intends to register to a server, you need to configure a SIP trunk connecting UC350 and the server, and then enable register for the SIP trunk.

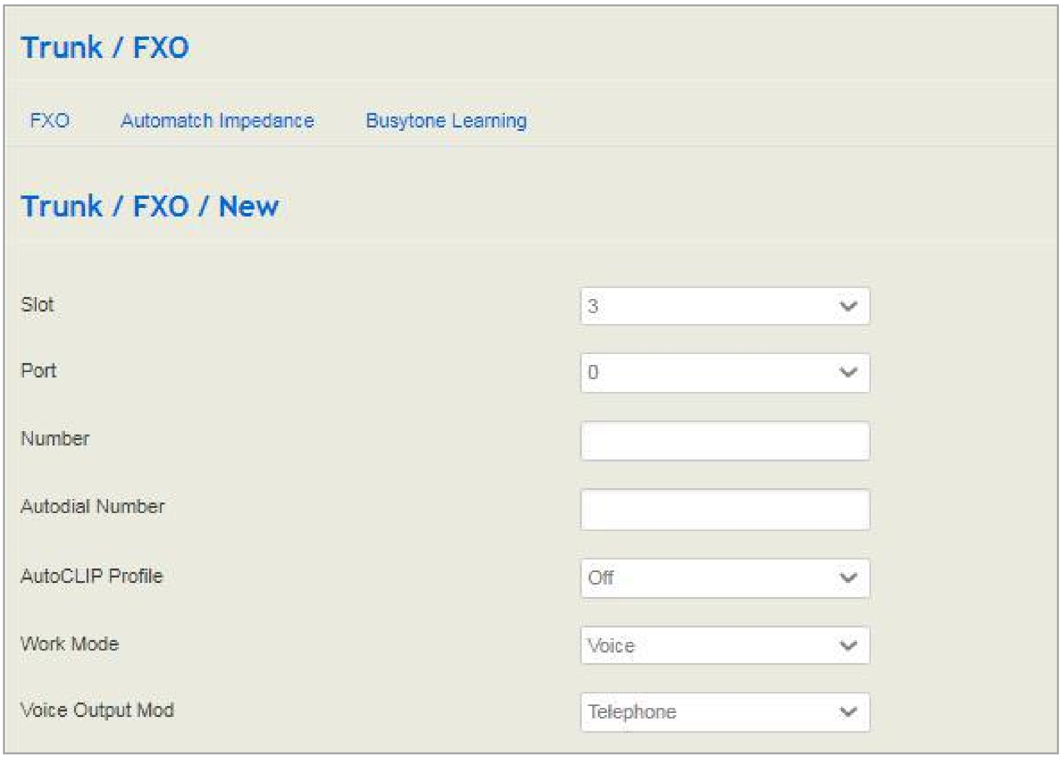
If the FXS of UC350 intends to register to a server, you need to configure a SIP trunk connecting UC350 and the server, then enable register for the port and designate the SIP trunk to it.

5.7.2 FXO

FXO Trunk interconnects the PSTN with UC350. Calls from the PSTN can come into the gateway and calls can go out from the gateway to search telephone numbers under the PSTN.

Different from the FXO ports of other gateways, the FXO port of UC350 only allows one-time dialing, which means called numbers needs to be dialed directly for calls that go out from the FXO port.

Figure 5.7.2-1 FXO Trunk



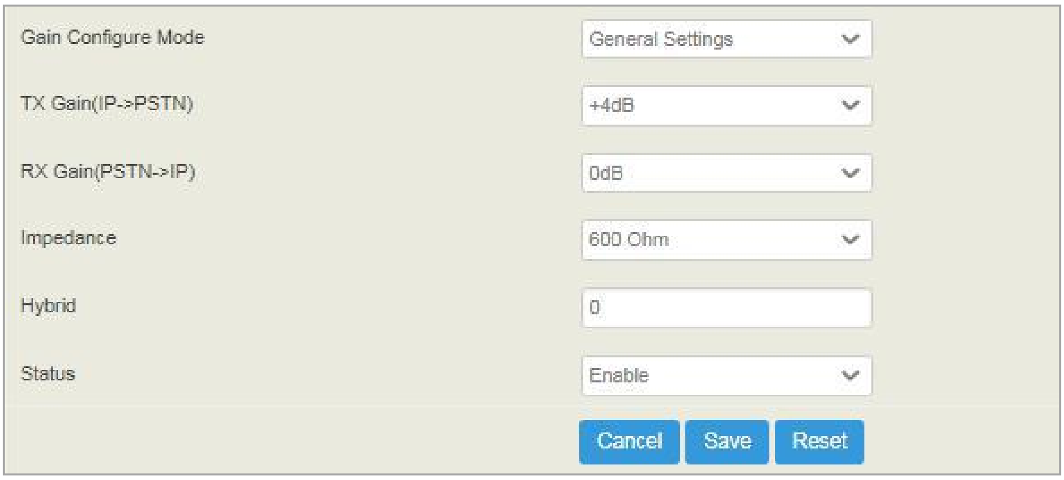


Table 5.7.2-1 Explanation of FXO Trunk Parameters

Port	The FXO port number
Autodial Number	The autodial number of the FXO port when there are incoming calls
AutoCLIP Profile	AutoCLIP is mainly used to SIP trunks, FXO trunks ans VoLTE trunks and it helps record the outgoing and incoming calls of a trunk.
Work Mode	Configure the working mode of the FXO port
Voice Output Mode	Configure the voice output mode of the FXO port
Gain Configure Mode	Select the gain configuration mode of the FXO port (general settings and advanced settings), TX gain and RX gain are newly added to advanced settings than general settings.
TX Gain(IP→PSTN)	The volume level of the remote end during a call; that is, adjusting the "TX gain" will affect the volume of the sound heard by the remote end.
RX Gain(PSTN→IP)	The volume level of the user during the call; that is, adjusting the "RX gain" will affect the level of sound you hear.
Impedance	Set the value of impedance(SLIC） for the remote FXS port
Hybrid	Set the value of hybrid for the remote FXS port
Status	If it is on, this FXO trunk can be used, otherwise, the FXO trunk is unavailable.

- **FXO Automatch Impedance：**

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

Conditions:

Only the ports in the online can be tested.

Steps:

1) Navigate to "**Trunk > FXO > Automatch Impedance**";
2) Configure the mode and parameters of automatch impedance , and click "Start";
3) After the test is completed, the Impedance and Hybrid values are displayed.

Figure 5.7.2-2 FXO Automatch Impedance

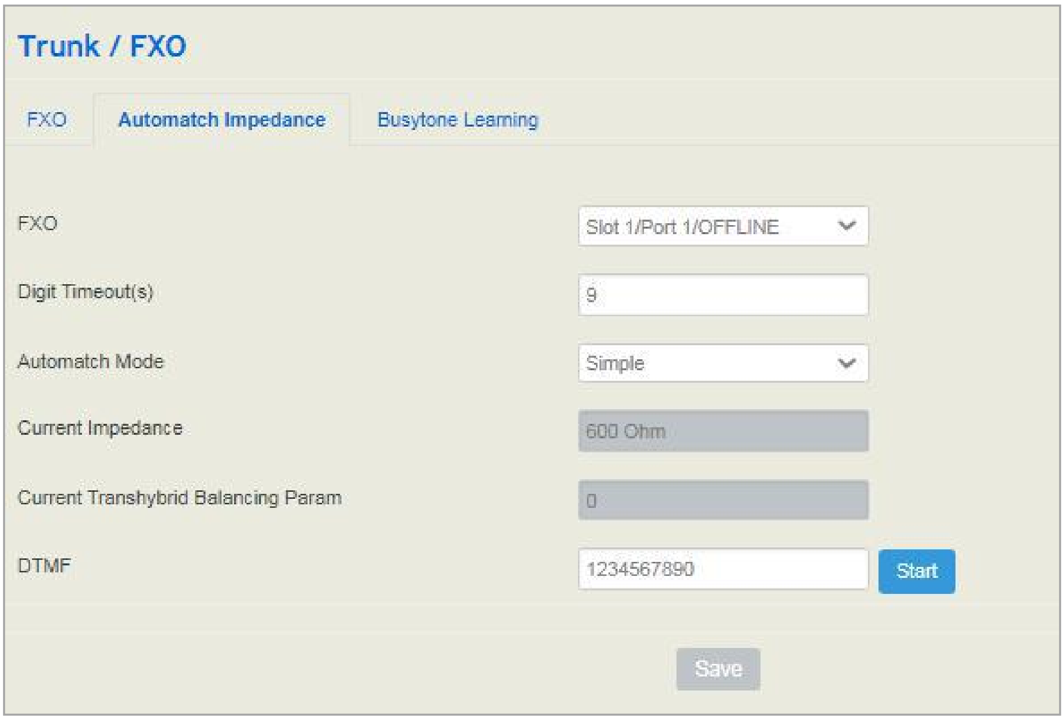


Table 5.7.2-2 Explanation of Automatch Impedance Parameters

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

- **Busytone Learning**

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

Conditions:

Only the ports in the online can be tested.

Steps:

1) Navigate to "**Trunk > FXO > Busytone Learning**";

Figure 5.7.2-3 FXO Busytone Learning

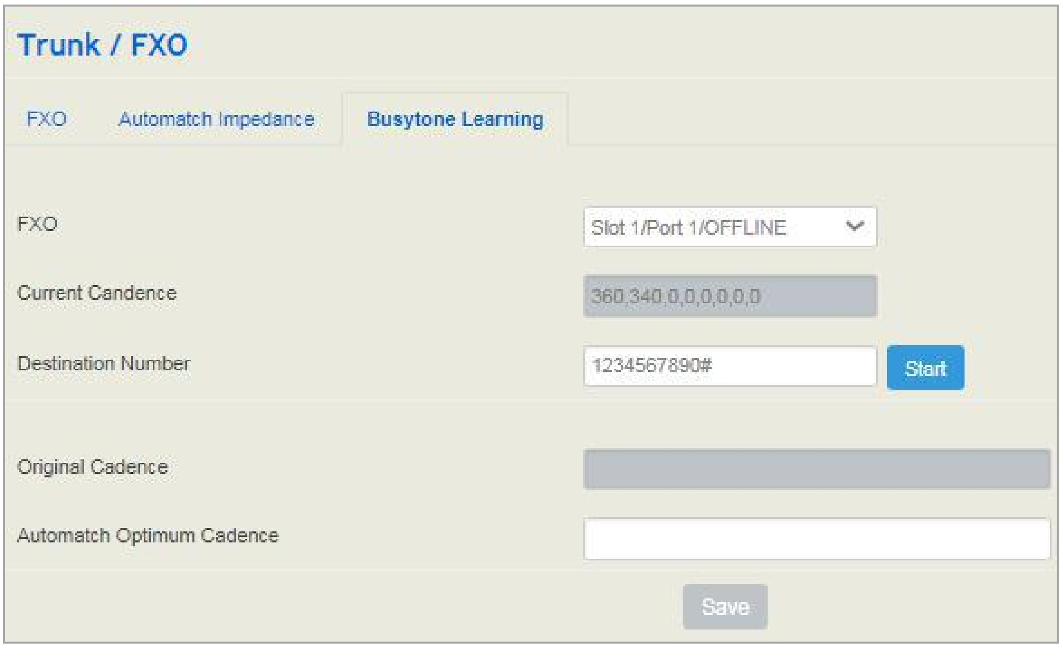


Table 5.7.2-3 Explanation of Busytone Learning Parameters

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

2) After selecting an online port, filling in the destination number, click "start";

3) Busy tone learning takes about 30-60s. After the learning is completed, the progress bar displays 100%.

4) Learning is completed, the Optimal cadence will be displayed. It will only take effect after saving.

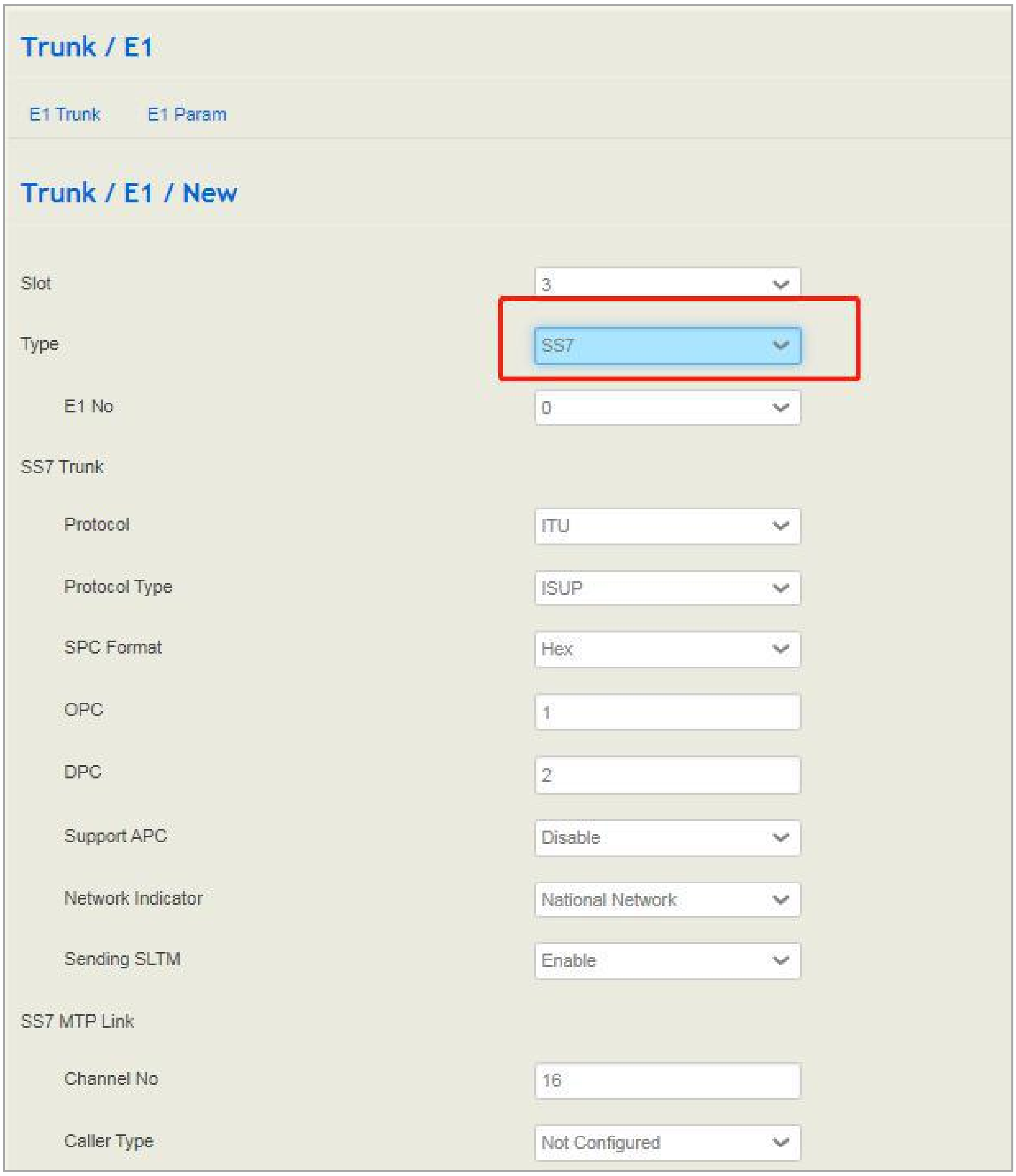
5.7.3 E1

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

- **SS7 Trunk**

Navigate to "Trunk >E1", select SS7 in the type bar.

Figure 5.7.3-1 SS7 Trunk



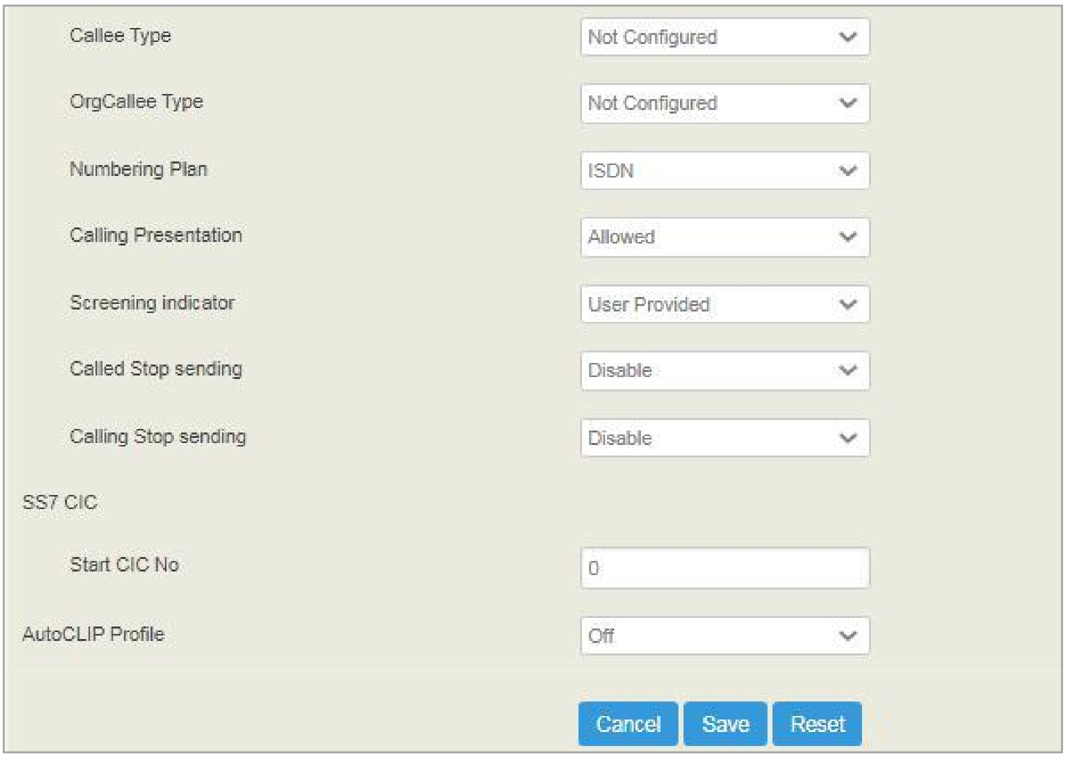


Table 5.7.3-1 Explanation of SS7 Trunk Parameters

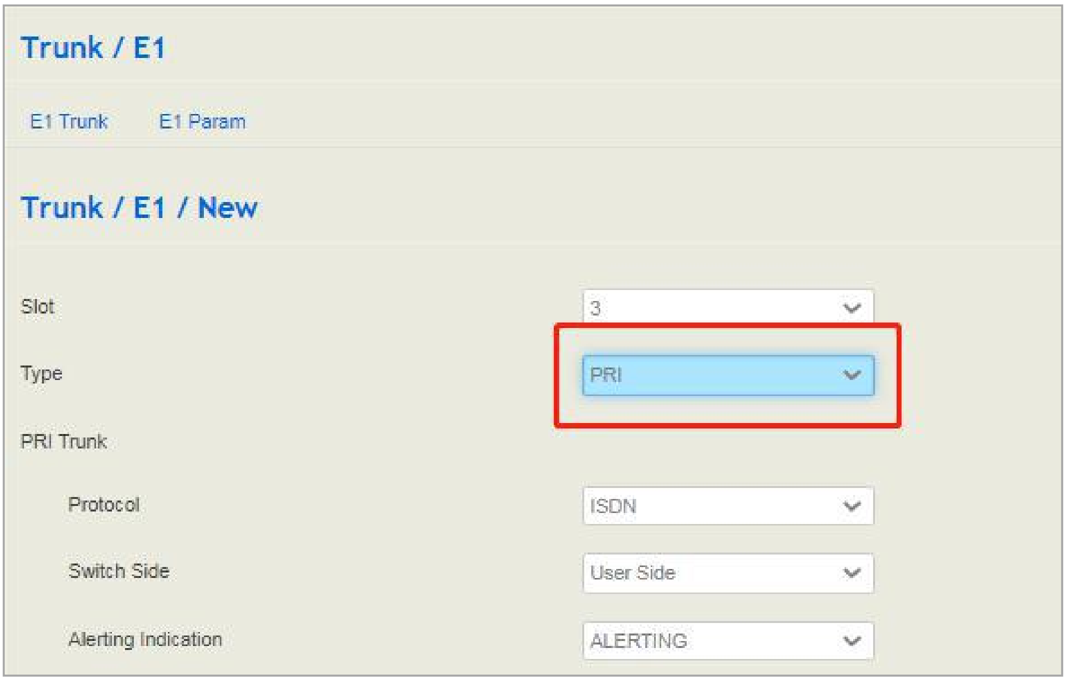
Type	Type: SS7 and PRI
Protocol	Protocol standard SPC types: ITU (14 bit), ANSI (24 bit), ITU-CHINA (24 bit)
Protocol Type	SS7 service types: ISUP (ISDN user side) and TUP (Telephone side)
SPC Format	SPC format: Hexadecimal system and 14bit (3-8-3)
OPC	OPC: Original Point Code, usually uniformly assigned by the operator
DPC	DPC: Destination Point Code, usually uniformly assigned by the operator
Support APC	APC is required when it is enabled, and the format is the same as the SPC format. Enter the STP code provided by the operator
Network Indicator	Display the network indicator of SS7, including :domestic network, domestic network backup, international network, and international network backup; the default is domestic network(mainly used in China, the United States and Japan), and "international network" is usually used exchange in the office, others are according to the environment.
Sending SLTM	SLTM: Signalling Link Test Message
Channel No	The channel for establishing link 7, usually channel No. 16 or No. 1, and the default channel No. 16

Caller Type	Configure the caller number type (not configured/ international/ domestic/ user)
Callee Type	Configure the callee number type (not configured/ international/ domestic/ user)
OrgCallee Type	Configure the original callee number type (not configured/ international/ domestic/ user)
Numbering Plan	Configure the number plan (ISDN/ data/ telegram/ special)
Calling Presentation	Calling presentation (allowed /limited/ invalid/ not configured)
Screening Indicator	Configure screening indicator (user-provided/ network-provided)
Called Stop Sending	After enabled, the called number with the suffix F
Calling Stop Sending	After enabled, the calling number with the suffix F
Start CIC No	The start CIC No of E1 port
AutoCLIP Profile	AutoCLIP is mainly used to SIP trunks, FXO trunks ans VoLTE trunks and it helps record the outgoing and incoming calls of a trunk.

- **PRI Trunk**

Navigate to "Trunk >E1", select PRI in the type bar.

Figure 5.7.3-2 PRI Trunk



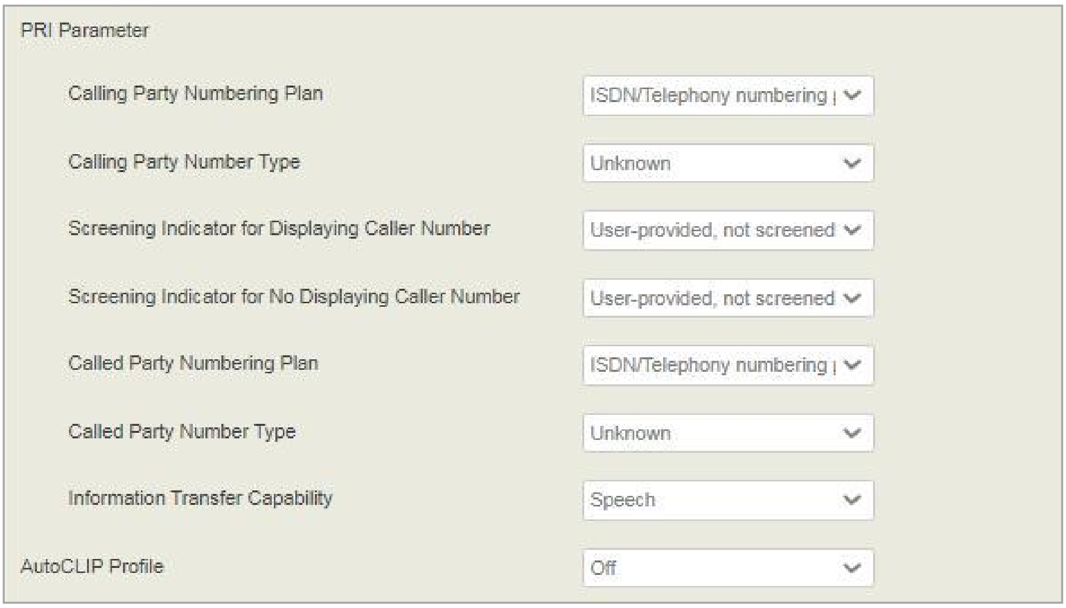


Table 5.7.3-2 Explanation of PRI Trunk Parameters

Type	Type: SS7 and PRI
Protocol	There are two types of PRI protocol: ISDN and QSIG
Switch Side	"User side" and "Network side" can be chosen. When implementing a PRI circuit, the nature of E1 in the network must be different on the receiving and sending sides
Alerting Indication	Configure alerting indication (Alerting and progressing)
Calling Party Numbering Plan	6 options, the default is "ISDN/telephone numbering plan"
Calling Party Number Type	6 types of calling party numbers can be selected
Screening Indicator for Displaying Caller Number	4 options, the default is "user-provided ,not screened"
Screening Indicator for No Displaying Caller Number	4 options, the default is "user-provided ,not screened"
Called Party Numbering Plan	6 options, the default is "ISDN/telephone numbering plan"
Called Party Number Type	6 options, the default is "unknown"
Information Transfer Capability	Support voice and 3.1khz voice
AutoCLIP Profile	AutoCLIP is mainly used to SIP trunks, FXO trunks ans VoLTE trunks and it helps record the outgoing and incoming calls of a trunk.

- **E1 Param**

Navigate to "Trunk >E1 **>E1 Param**".

Figure 5.7.3-3 E1 Trunk

Table 5.7.3-3 Explanation of E1 Trunk Parameters

PCM Mode	PCM (Pulse Code Modulation) Mode: ALAW or Mu LAW
Frame Format	The frame format of the E1 port is: DF, MF-CRC4, MF, the default is DF

5.8 Call Control

This section is to configure routes or route groups for incoming and outgoing calls through UC350, as well as IVR, SMS and so on.

5.8.1 Setting

Figure 5.8-1 Basic Setting of Call Control

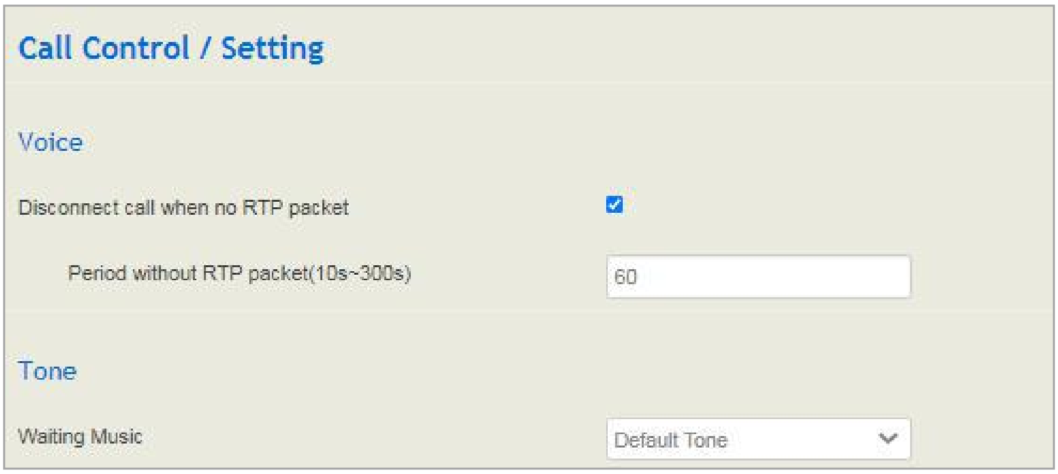


Table 5.8.1 Explanation of Call Control Parameters

Disconnect call when no RTP packet	If it is enabled, and no RTP packets are received within the preset time, calls will be disconnected
Waiting Music	Select the waiting music

5.8.2 Route Group

On the **Call Control →Route Group** interface, you can group SIP trunks, SIP extensions, FXS extension and FXO trunk together according to your needs and set strategy for choosing which trunk or extension as the destination route under a route group.

Figure 5.8-2 Create Route Group

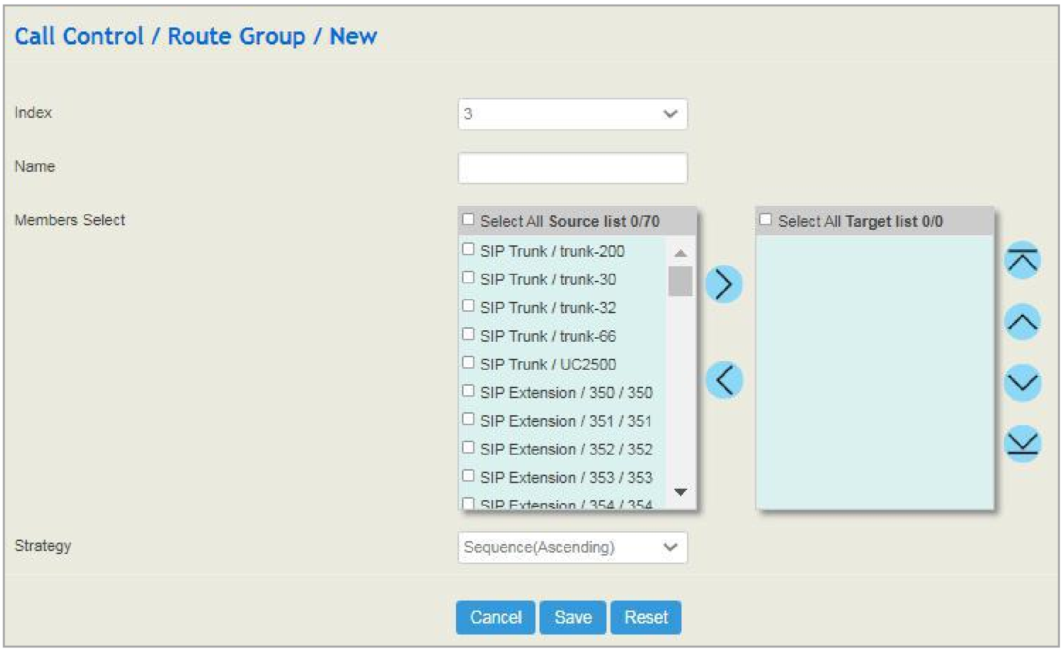


Table 5.8.2 Explanation of Route Group Parameters

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

5.8.3 Route

On the **Call Control → Route** interface, you can configure routes for incoming calls and outgoing calls.

Figure 5.8-3 Create a Route

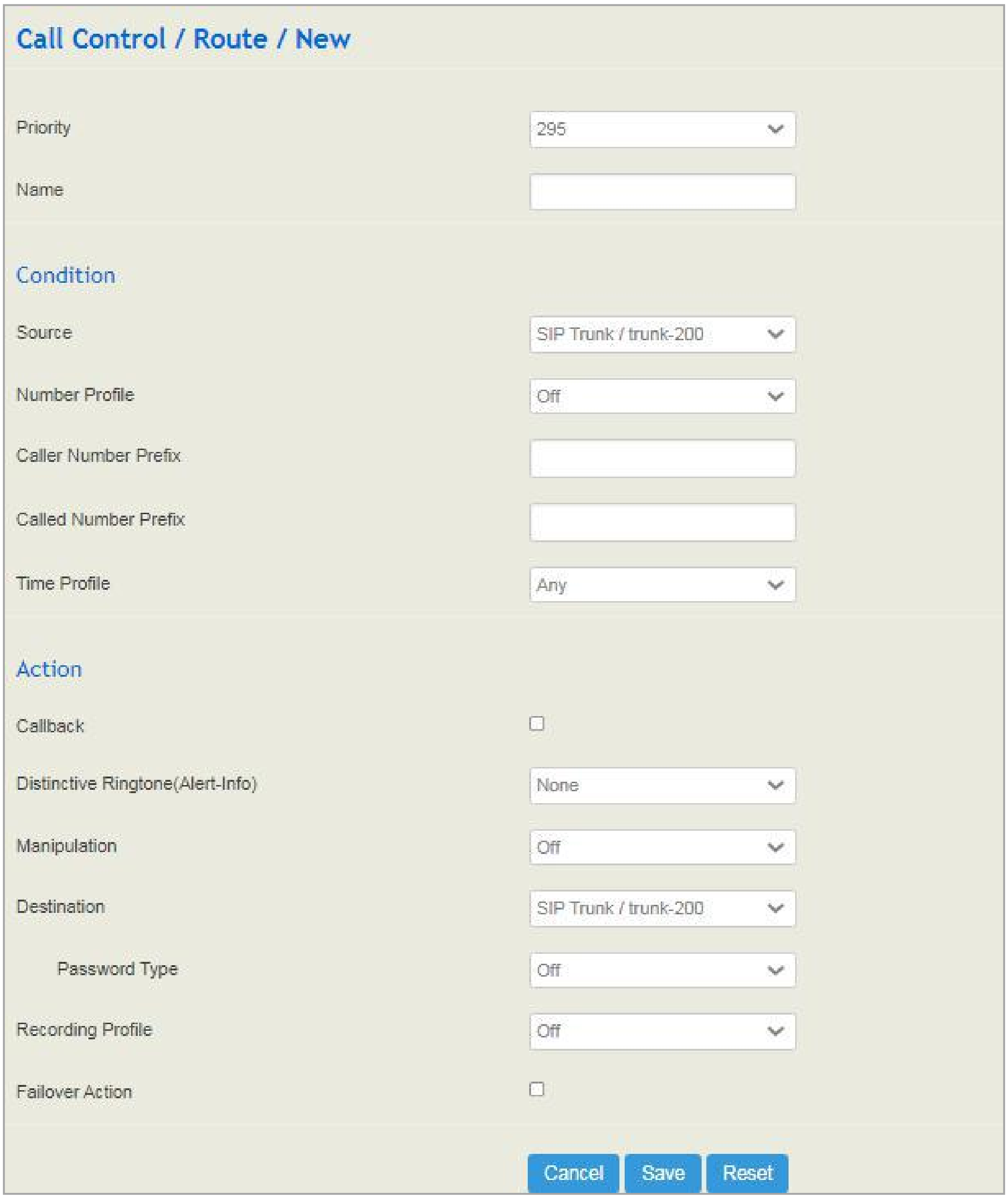


Table 5.8.3 Explanation of Route Parameters

Priority	The priority for choosing the route; the higher value, the lower priority
Name	The name of the route
Condition	The condition under which the route will be used
Source	The source of the call; it can be the FXS extension, SIP extension, FXO trunk ,GSM trunk, a customized source or any
Number Profile	The profile of the caller number and the called number; please make reference to the Profile → Number section.The default

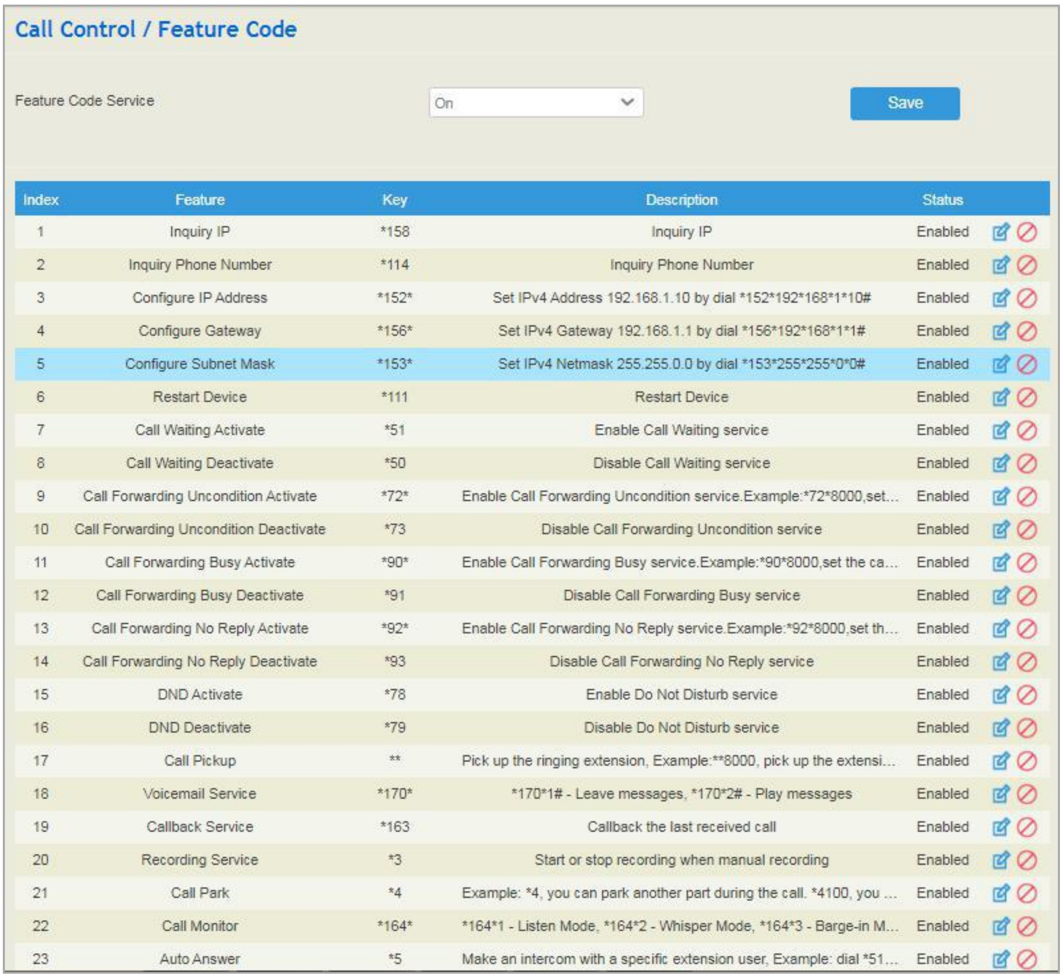
	value is ‘Off’ Note: it cannot be simultaneously used with the following parameters of ‘caller number prefix’ and ‘called number prefix’
Caller Number Prefix	The prefix of caller number; it supports regular expression
Called Number Prefix	The prefix of called number; it supports regular expression
Time Profile	The profile of time during which the route can be used; make reference to the Profile → 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
Manipulation	If it is on, the caller number or called number of the route will be manipulated; make reference to the Profile→ Manipulation section
Destination	The destination of the route
Recording Profile	Record according to the configured rules
Failover Action	The processing when a call through this route fails

5.8.4 Feature Code

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

The following is the corresponding function of each feature code:

Figure 5.8-4 Feature Code



Call Control / Feature Code

Feature Code Service | On | Save

Index	Feature	Key	Description	Status
1	Inquiry IP	*158	Inquiry IP	Enabled
2	Inquiry Phone Number	*114	Inquiry Phone Number	Enabled
3	Configure IP Address	*152*	Set IPv4 Address 192.168.1.10 by dial *152*192*168*1*10#	Enabled
4	Configure Gateway	*156*	Set IPv4 Gateway 192.168.1.1 by dial *156*192*168*1*1#	Enabled
5	Configure Subnet Mask	*153*	Set IPv4 Netmask 255.255.0.0 by dial *153*255*255*0*0#	Enabled
6	Restart Device	*111	Restart Device	Enabled
7	Call Waiting Activate	*51	Enable Call Waiting service	Enabled
8	Call Waiting Deactivate	*50	Disable Call Waiting service	Enabled
9	Call Forwarding Uncondition Activate	*72*	Enable Call Forwarding Uncondition service.Example:*72*8000,set...	Enabled
10	Call Forwarding Uncondition Deactivate	*73	Disable Call Forwarding Uncondition service	Enabled
11	Call Forwarding Busy Activate	*90*	Enable Call Forwarding Busy service.Example:*90*8000,set the ca...	Enabled
12	Call Forwarding Busy Deactivate	*91	Disable Call Forwarding Busy service	Enabled
13	Call Forwarding No Reply Activate	*92*	Enable Call Forwarding No Reply service.Example:*92*8000,set th...	Enabled
14	Call Forwarding No Reply Deactivate	*93	Disable Call Forwarding No Reply service	Enabled
15	DND Activate	*78	Enable Do Not Disturb service	Enabled
16	DND Deactivate	*79	Disable Do Not Disturb service	Enabled
17	Call Pickup	**	Pick up the ringing extension, Example:**8000, pick up the extensi...	Enabled
18	Voicemail Service	*170*	*170*1# - Leave messages, *170*2# - Play messages	Enabled
19	Callback Service	*163	Callback the last received call	Enabled
20	Recording Service	*3	Start or stop recording when manual recording	Enabled
21	Call Park	*4	Example: *4, you can park another part during the call. *4100, you ...	Enabled
22	Call Monitor	*164*	*164*1 - Listen Mode, *164*2 - Whisper Mode, *164*3 - Barge-in M...	Enabled
23	Auto Answer	*5	Make an intercom with a specific extension user, Example: dial *51...	Enabled

Note:

1) All feature codes are enabled by default.
2) Any feature code can be customized and edited

5.8.5 IVR

On the **Call Control → IVR** interface, you can carry out specific configurations for the IVR which has been uploaded from the **System → Voice** interface. IVR is often used for voice prompts in call centers.

Figure 5.8-5 IVR Setting

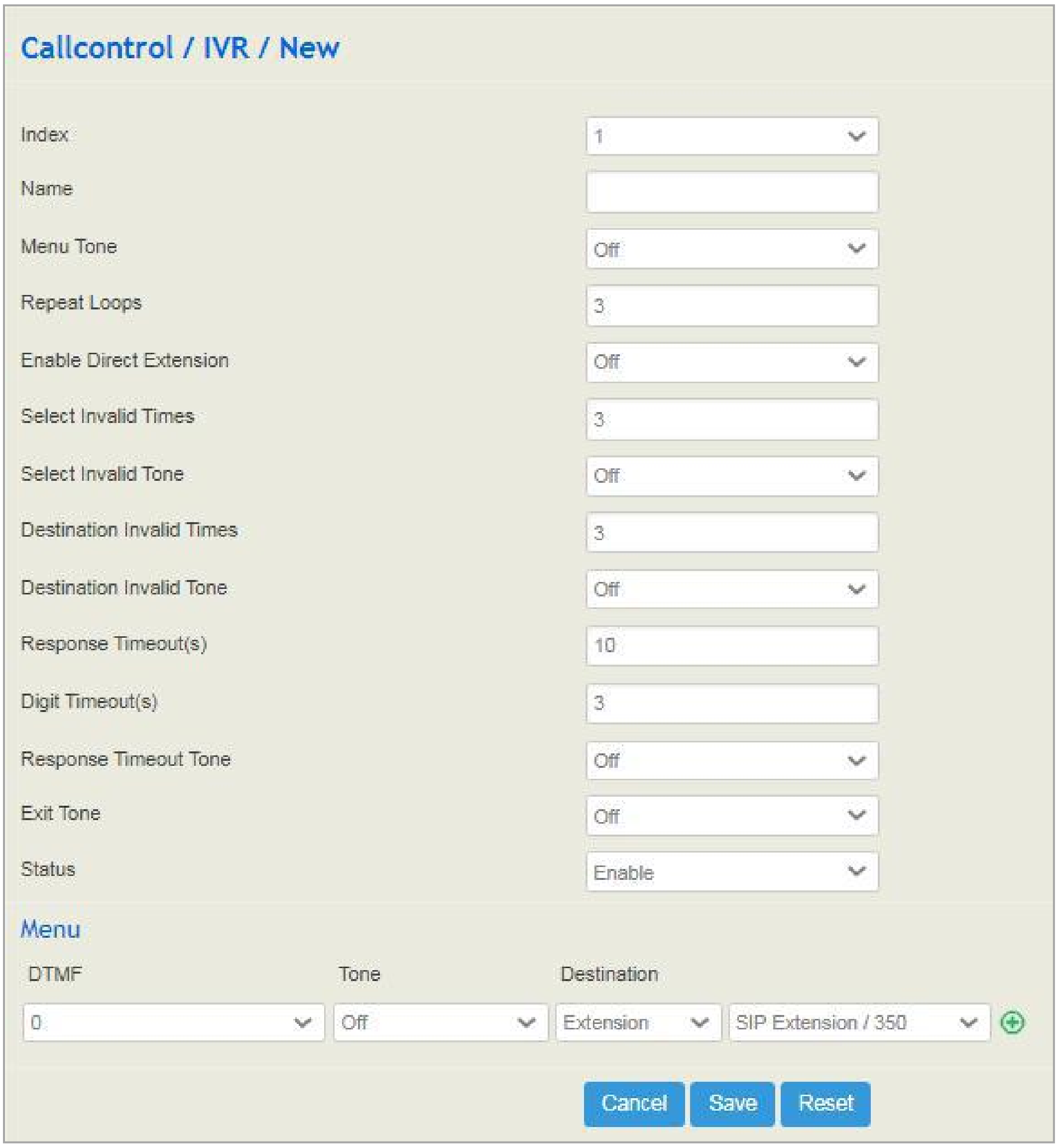


Table 5.8.5 Explanation of IVR Parameters

Name	The name of the IVR
Menu Tone	When a call comes to the IVR, the menu tone heard
Enable Direct Extension	Whether to allow direct dialing of extensions during the playing of IVR
Repeat Loops	If it is set as ‘3’, the call will be hanged up after the IVR has been repeated for three times during timeout.
Select Invalid Times	When a call comes to the IVR, according to the voice prompt, if you receive two dialings that do not match the DTMF , then the dialing is invalid, and the invalid prompt tone is played. When the invalid times is exceeded, the voice prompt: Goodbye

Select Invalid Tone	When an invalid dial is received, an invalid tone will be played
Destination Invalid Times	It takes effect when the direct extension is enabled. When you 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
Destination Invalid Tone	When receiving an invalid destination dial, the invalid tone will be played
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
Digit Timeout(s)	The timeout for dialing DTMF
Response Timeout Tone	When the second dialing timeout, the timeout will be played after being enabled
Exit Tone	When exiting IVR, the exit tone will be played
Status	If it is disabled, the IVR cannot be seen in the destination of route.
Menu	DTMF: It can be 0-9 quick-dial numbers, *, #, others or timeout. Destination: the destination of the IVR; it can be an extension or a trunk. For example, if DTMF is configured as 1,2,3 and others, and the telephone key that is pressed is not 1, 2 or 3, the IVR will choose the destination of ‘others’. When the the playing of the IVR times out, and user does not press any telephone key, the IVR will choose the destination of ‘timeout’. When the destination is a trunk, user does not need to pre-configure the called number, and the system will prompt the user to dial the called number.

5.8.6 Conference

On the "Call Control -> 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 5.8-6 Conference

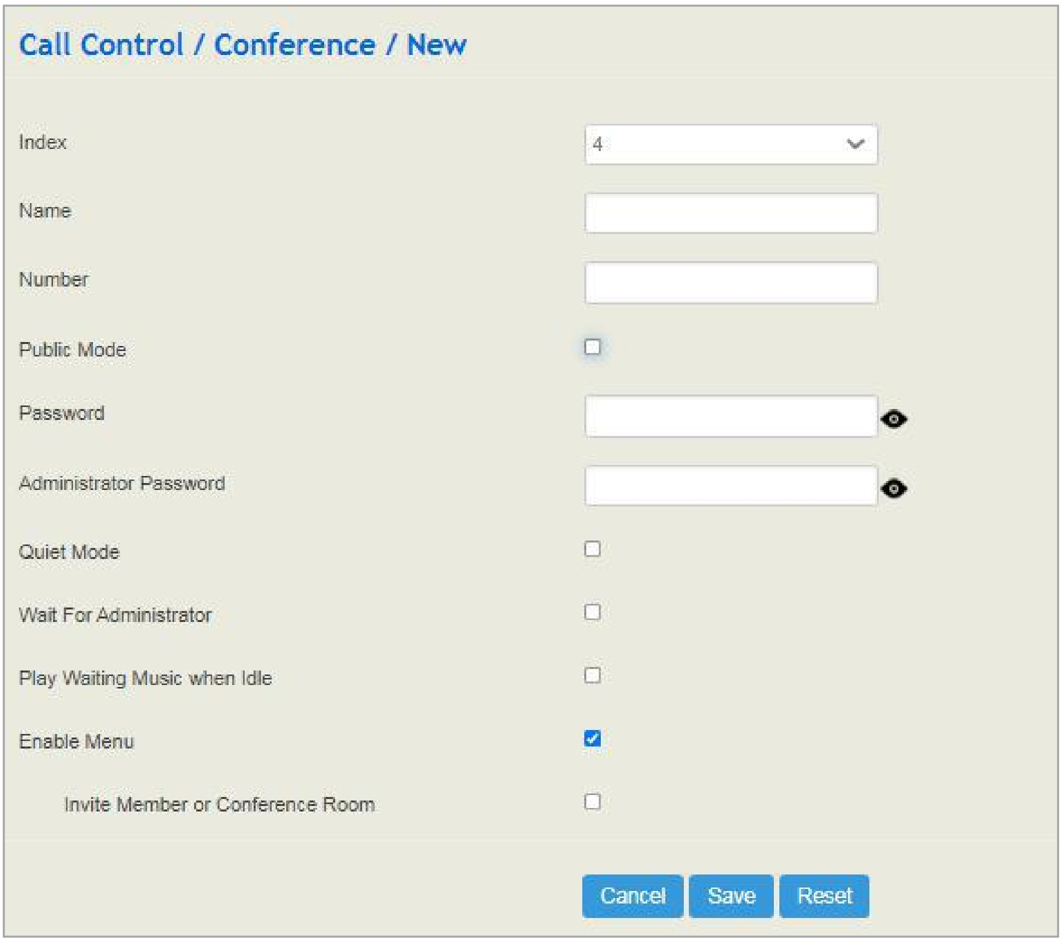


Table 5.8.6-1 Explanation of Conference Parameters

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

Table 5.8.6-2 Conference Menu

DTMF	Description	Notes
1	Invite members	Non-administrators need to enable configuration
2	Invite members, need to be confirmed by the invitee	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, you hear the prompt that you 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, you will be prompted to enter the conference room password;

3) Connect to the meeting.

5.8.7 SMS Route

UC350 allows SMS to be sent between SIP clients, On the **Call Control→ SMS Route** interface, you can establish route for these SMS.

Figure 5.8-7 Create SMS Route

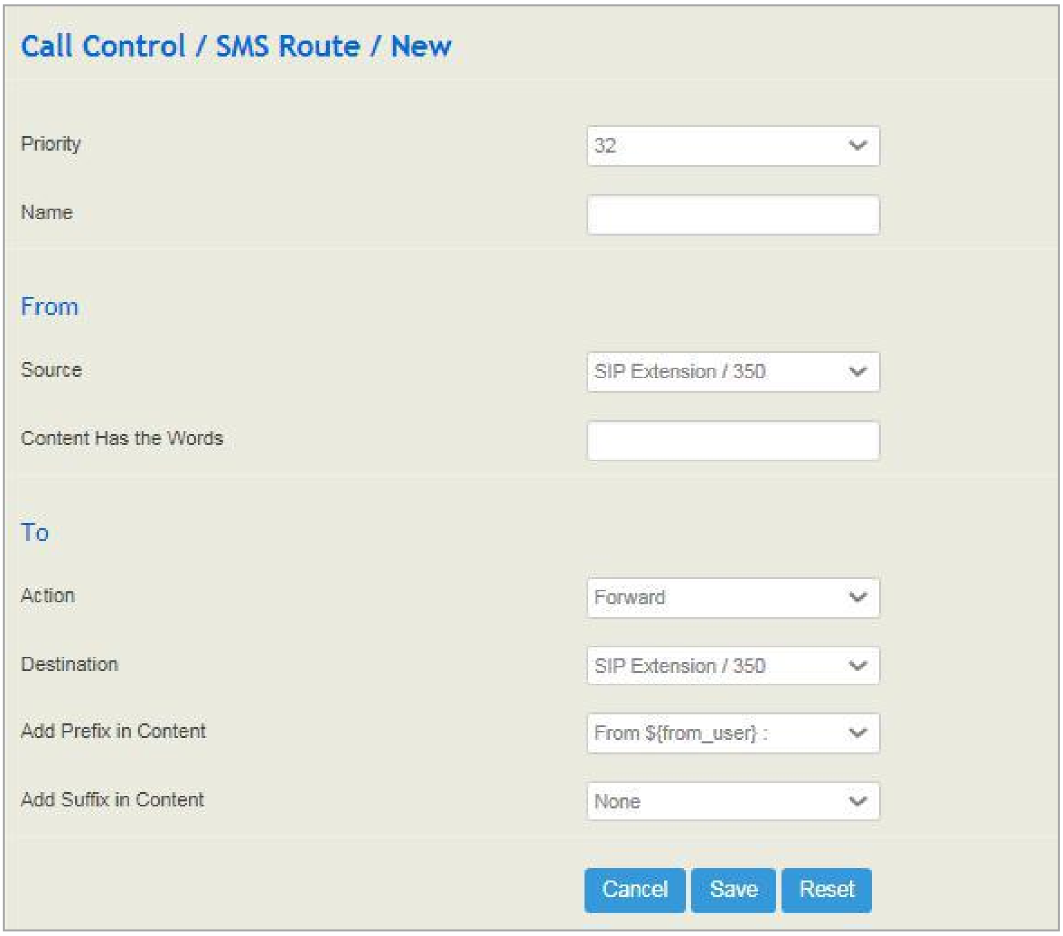


Table 5.8.7 Explanation of SMS Route Parameters

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.
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.

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.

5.8.8 Diagnostics

In case that call cannot be connected or voice has quality problem, you can enter the **Call Control →Diagnostics** interface to collect fault-related information and then send it to technical support to locate fault.

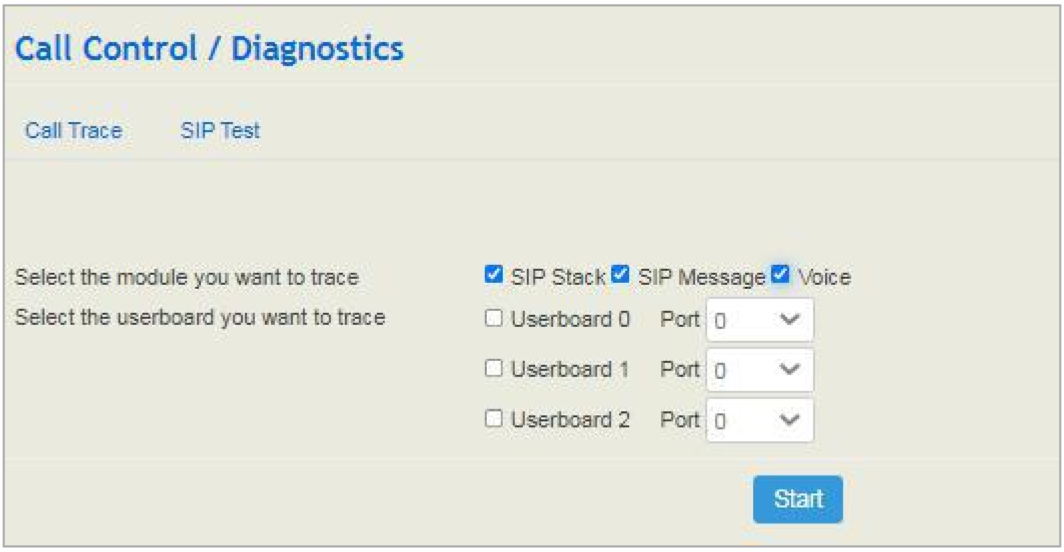
Operation Procedures:

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

2) Give a call, and come back to the **Call Control →Diagnostics** interface after the call ends. Then click Stop and download the tracing file.

3) In order to locate faults more quickly, you sometimes need to enter into the **System →Service Log** interface, click export, and then send this exported file and the tracing file to technical support,

Figure 5.8-8 Call Tracing for Diagnostics



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
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

Abbreviation	Explanation
NTP	Network Time Protocol
STUN	Simple Traversal of UDP over NAT
PSTN	Public Switched Telephone Network
WLAN	Wireless Local Area Network