UC2000 User Manual v1.0

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Revision Records

Document Name	UC2000 User Manual v1.0
Document version	V1.0
Firmware version	2.50.1.2
Revised by	Ivanka Yuan
Date	2014.6.3
Changes	Draft

Table of Contents

1 Product Description

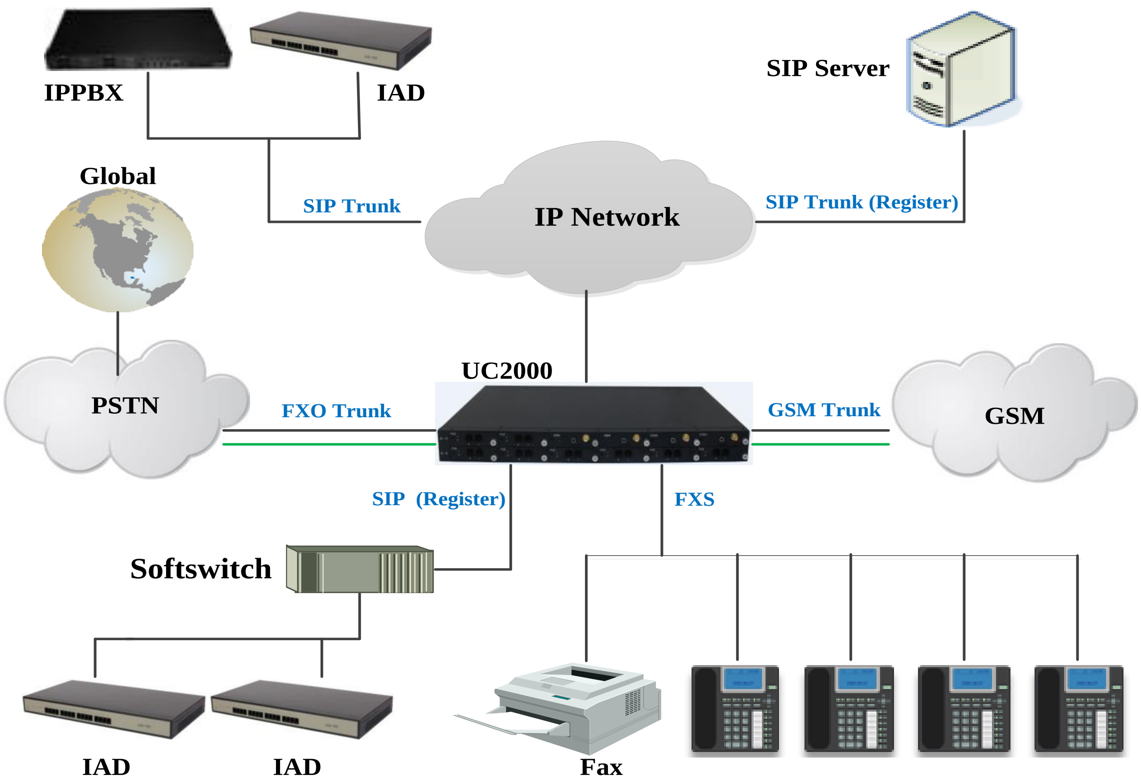
1.1 Overview

UC2000 is a multi-functional universal gateway, which is module-structured and provides connectivity to VoIP network, PLMN and PSTN via a standard 1U device. The gateway has a compact structure with fine appearance, featuring various ports and comprehensive application functions. Users can choose different modules based on their needs to construct a customized gateway.

UC2000 realizes the smooth migration from a single communication way (such as traditional PBX) to a comprehensive communication way which integrates VoIP, PLMN and PSTN. It also provides a united communication GUI for PLMN-based or PSTN-based voice service and messaging service. Offering unified communication service, the multi-functional gateway is perfect for those enterprises with different branches and can help them dramatically reduce communication fees.

1.2 Application Scenario

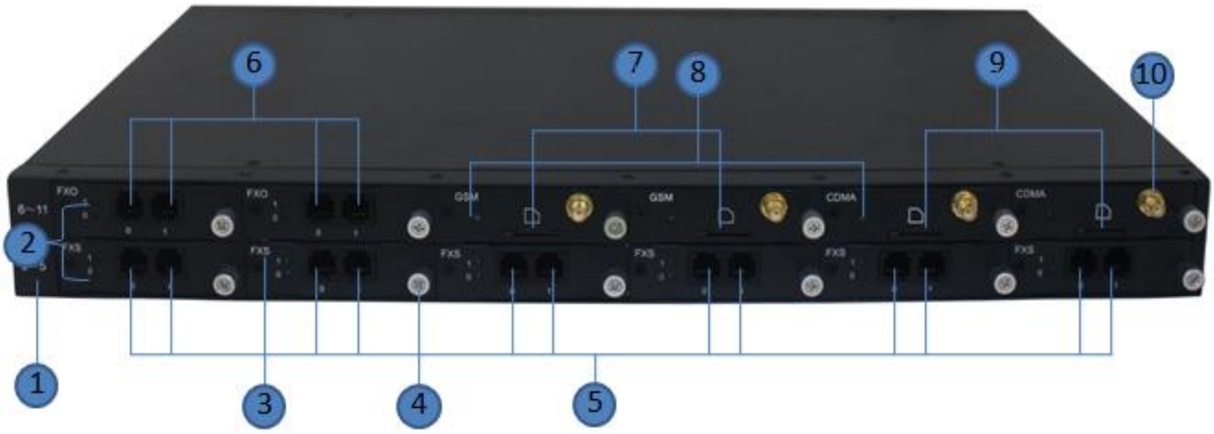
UC2000 provides high-quality and cost-effective VoIP solution. Its application scenario is shown as follows:



1.3 Product Appearance

1.3.1 Image of UC2000

1.3.2 Description of Ports and Indicators



Number	Name	Description	
1	Board Number	Board number: from 0 to11 (displayed on the left)	
2	Indicators of FXS/FXO Port（0,1）	There are two FXS/FXO ports for each FXS/FXO board; the No.0 port is on the left, while the No.1 port is on the right.	
2	Indicators of FXS/FXO Port（0,1）	On green	The FXS/FXO port is in off-hook state.
2	Indicators of FXS/FXO Port（0,1）	Flash slowly	The FXS/FXO port is in on-hook state.
2	Indicators of FXS/FXO Port（0,1）	On dull	Exception occurs on the FXS/FXO port.
3	Board Type	There are four types of board, including FXS, FXO, GSM and CDMA.	
4	Screw	The screw is used to fix or unplug a board	
5	FXS Port	Port to connect UC2000 with analog phones（RJ11)	
6	FXO Port	Port to connect Telco fixed line or connect to the FXS port	
7	Slot for GSM SIM Card	Slot where GSM SIM card is inserted	
8	SIM Card Indicator	Flash slowly	SIM card has been inserted and usable
8	SIM Card Indicator	On dull	SIM card is not inserted or SIM card is unavailable
9	Slot for CDMA SIM Card	Slot where CDMA SIM card is inserted	
10	Antenna Port	Used to connect UC2000 with the antenna for enhancing wireless signal	
11	Power Switch	ON	Turn on power
11	Power Switch	OFF	Turn off power
12	Power Interface	Used to input 100-240VAC	
13	Device Information	The Mac address and S/N of the UC2000	
14	Ethernet Port	Ethernet ports, including FE0 and FE1. IP addresses of both are 192.168.11.1 by default.	
15	Screw for Master Control Board	Used to fix or unplug the master control board	
16	Master Control Board	The core of UC2000	
17	Console Port	RS232 standard; Baud Rate: 115200bps	
18	RUN	On green	UC2000 is booting up
18	RUN	On dull	UC2000 has malfunctioned
18	RUN	Flash fast (Flash every 0.5s)	UC2000 is running normally
19	Grounding Port	Used for ground connection	

20	RST	When the RST button is pressed for 3 seconds to 6 seconds, the IP address of UC2000, the username and password of the Web of UC2000 will be restored to factory defaults.	
21	POWER	On green	UC2000 is powered on
		On dull	UC2000 is not powered on

1.4 Features & Functions

1.4.1 Key Features

- Multiple physical interfaces; modular design; customized for your needs.
- Local calls via extension numbers; easy communication between headquarters and branches.
- Flexible dial plan, via time, numbers, source IP etc.
- Simple and easy deployment; integrate seamlessly with your current system.
- CDRs on local
- SMS sending and receiving
- User-friendly web interface, multiple management ways

1.4.2 Software Features

- Ring Group
- 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
- FXO Impedance Auto Match
- CDRs

1.4.3 Voice Capabilities

- VoIP Protocols: SIP over UDP/TCP/TLS, H.323, SDP, RTP/SRTP
- Codecs: G.711a/μ law,G.723.1, G.729A/B, iLBC,G.726
- Silence Suppression
- Comfort Noise Generator(CNG)
- Voice Activity Detection(VAD)
- Jitter Buffer
- Echo Cancellation: G.168 with up to 128ms

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

1.4.4 Supplementary Services

- Call Waiting
- Call Transfer (Blind & Attended)
- Call Forwarding （Unconditional/Busy/No Reply）
- Call Holding
- Call Pickup
- No Disturbing
- Hotline

1.4.5 Physical Interfaces

- **12 Expansion Slot**s
- **Expansion Boards**:
  2 FXS Ports per Board
  2 FXO Ports per Board
  1 GSM Channel per Board
  1 CDMA Channel per Board
  1 WCDMA Channel per Board
  1 BRI Port per Board
  2 E1/T1 Ports per Board
  1 Radio Port per Board
  1 E&M port per Board
- **Ethernet Interfaces**
  2*10/100 Based-T RJ45
- **Console**
  1*RS232, 115200bps

1.4.6 FXS/FXO

- Connector: RJ11
- Caller ID: Bellcore Type 1&2, ETSI,BT,NTT and DTMF
- Answer and Disconnect Signaling: Answer, Disconnect, Busy Tone, Detection
- Polarity Reversal
- Flash-hook Detection

1.4.7 Mobile

- GSM: 850/900/1800/1900MHz
- CDMA: 800MHZ
- WCDMA: 850/900/1900/2100MHZ
- SIM/UIM: 1 SIM/UIM per Channel
- SIM Card: 1.8V, 3.0V
- Antenna: Panel Mount or External
- Cable: 3.0dB, SMA Connector
- SMS/USSD
- Bulk SMS
- SMS Code/Decode: ASCII, Unicode
- IMEI/PIN Code Management

1.4.8 Hardware Specifications & Environment

- Power Supply: 100-240V AC, 50-60Hz
- Power Consumption: 50W
- Operating Temperature. 0 ℃ ~ 45 ℃
- Storage Temperature: -20 ℃ ~80 ℃
- Humidity: 10%-90%, Non-Condensing
- Dimensions (W/D/H): 440×270×75mm
- Unit Weight: 4.7kg

2 Quick Installation

2.1 Attentions for Installation

The attentions for installing UC2000 include:

- The adapter of UC2000 accepts AC input voltage of 110- 220 V and converts it to 12V DC； Please ensure stable and safe power supply;
- Network interface should be standard RJ45 with 10Mbps or 100Mbps;
- Make sure the antenna of UC2000 is well-connected;
- If you want UC2000 to communicate with the GSM network or the CDMA network, please insert an SIM card.

Note: Please check whether power supply is up to the above requirement; otherwise, UC2000 and its power adapter may be damaged.

2.2 Installation Procedures

2.2.1 Installation of Userboards and Master Control Board

Images of Userboards:

FXO FXS GSM CDMA

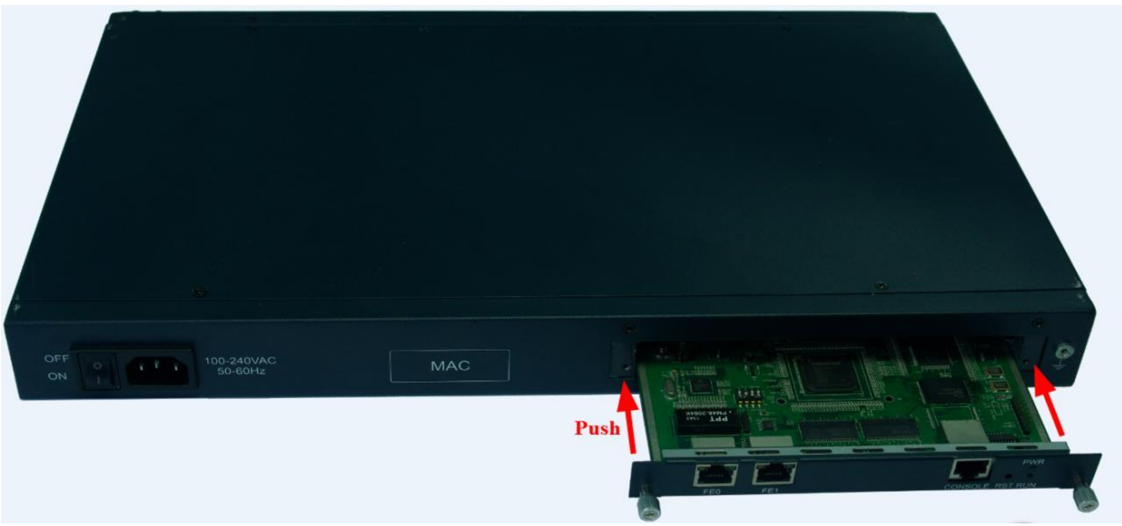
Installation of Userboards:

Please insert a userboard into the corresponding slot on the front panel of UC2000 according to the following indication.

Image of Master Control Board:

Installation of Master Control Board:

Please insert the master control board into the slot on the back panel of UC2000. Ensure that the master control board matches well with the guiding tracks on the left and the right of the slot.



2.2.2 How to insert SIM Card

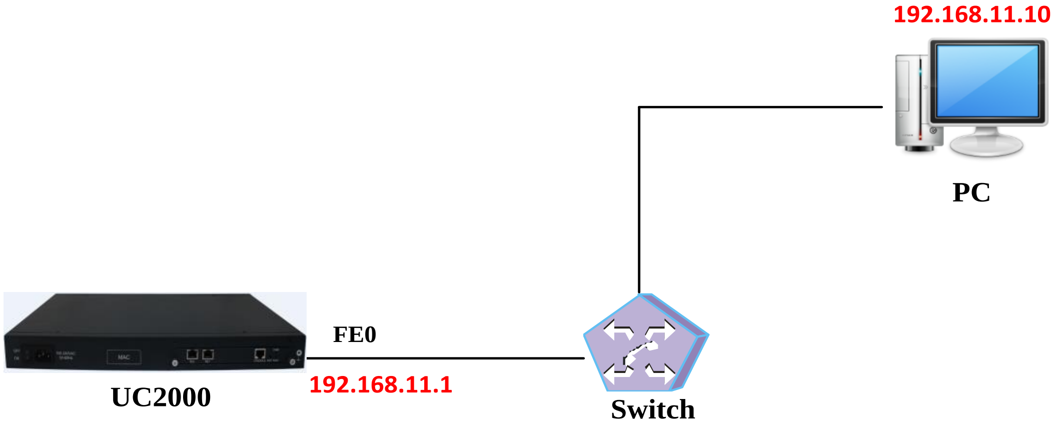
Please insert the SIM card according to the following indication:

2.2.3 Installation of Antenna

It is necessary to install the antenna on UC2000 for the GSM and CDMA userboards.

2.2.4 Network Connection

The UC2000 universal gateway has two 10/100M Ethernet interfaces, namely FE0 and FE1.



Note: The IP address of FE0 port and the IP address of PC should be at the same network segment.

As for the basic configurations about the network, SIP trunk, route and FXS/FXO/CDMA port, please make reference to the Quick Installation Guide of UC2000.

3 Basic Operation

3.1 Methods to Number Dialing

There are two methods to dial telephone number 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 the Digitmap dialplan format).
- Dial the called number and press #.

3.2 Call Holding

If a calling party places a call to a called party which is otherwise engaged, and the called party has the call holding feature enabled, the called party is able to switch to the new incoming call while keeping the current call holding on by pressing the flash button or the flash hook.

When the called party presses the flash button or the flash hook once again, he or she will switch back to the first call.

3.3 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 a IVR voice 'Please hold on, the subscriber you dialed is busy' and the called party will hear three beeps.

By pressing the flash button or the flash hook, the called party is able to switch between the new incoming call and the current call.

3.4 Call Transfer

When a calling party is in conversation with the called party, call transfer allows one of them to shift the call connection to a third party.

3.4.1 Blind Transfer

Blind transfer is a call transfer in which the transferring party connects the call to a destination extension before ringback begins (the transferring party will not hear any ringback).

Example: A and B are in conversation and B wants to transfer the conversation to C (the extension of C is 8000). Operation instructions are as follows:

1. B dials *18000 (*1 is the feature code for blind transfer, 8000 is the extension of C);
2. B hangs up the call;
3. After C picks up the phone, A and C go into conversation.

3.4.2 Attended Transfer

Attended transfer is a call transfer in which the transferring party either connects the call to a ringing phone (ringback heard) or speaks with the third party before connecting the call to the third party.

Example: A and B are in conversation and B wants to consultatively transfer the conversation to C (the extension of C is 8000). Operation instructions are as follows:

1. B dials *28000 (*2 is the feature code for attended transfer, 8000 is the extension of C);
2. The extension of C rings;
3. If C answers the call, C and B go into conversation;
4. If C hangs up the phone, B and A continue to be in conversation;
5. If B hangs up the phone and C picks up the phone, C and A go into conversation.

3.4.3 Call Transfer via Pressing Flash-hook

Example: A and B are in conversation and B wants to transfer the call to C via pressing the flash-hook (the extension of C is 8000). Operation instructions are as follows:

1. B presses the flash-hook and dials 8000;
2. The extension of C rings;
3. If C answers the call, B and C go into the conversation while the conversation between B and A is still held on.
4. If B presses the flash-hook again and dials 1, conversation is switched back between B and A;
5. If B presses the flash-hook again and dials 2, conversation is switched between B and C.

3.5 Description of Feature Codes

Code	Corresponding Function
*158	Dial *158 to inquiry IP address of UC2000
*114	Dial *114 to inquiry phone number
150	Dial *150*1 to set IP address as static IP address; dial *150*2 to set IP address as DHCP IP address
152	Dial *152*192*168*1*10# to set IPv4 address as 192.168.1.10
156	Dial *156*192*168*1*1# to set IPv4 gateway as 192.168.1.1
153	Dial *153*255*255*0*0*# to set IPv4 netmask as 255.255.0.0
*51	Dial *51 to enable the call waiting service
*50	Dial *50 to disable the call waiting service
*1	Example: Dial *18000, and you can blind transfer to the extension number 8000
*2	Example: Dial *28000#, and you can attended transfer to the extension number 8000
72	Enable unconditional call forwarding service. Example: Dial *72*8000, and calls will be unconditionally forwarded to extension number 8000
*73	Disable unconditional call forwarding service
90	Enable the call forwarding on busy service. Example: Dial *90*8000, and calls will be forwarded to extension number 8000 when the called number is on busy
*91	Disable the call forwarding on busy service
92	Enable the call forwarding on no reply service. Example: Dial *92*8000, and calls will be forwarded to extension number 8000 when there is no reply from the called number
*93	Disable the call forwarding on no reply service
*78	Enable the 'Do Not Disturb' service
*79	Disable the 'Do Not Disturb' service
**	Pick up the ringing extension which is in the same ringgroup. Example: Dial**8000, and you can take the incoming call of extension number 8000
Flash Button/ Flash Hook	By pressing the flash button or flash hook, a person can switch between the current call and the new incoming call.

Note:

- As for the GSM userboard and the CDMA userboard, use a mobile phone to call the corresponding SIM card number first (aimed to establish connection between the mobile phone and the UC2000), and then dial feature codes to do configurations after hearing an IVR voice “please dial the extension number’.

- As for the FXO userboard, use a phone to call an phone number under the PSTN connected to the FXO port of UC2000, and then dial feature codes to do configurations after hearing an IVR voice “please dial the extension number’.

- A voice prompt indicating successful configuration will be given after each configuration procedure. Please do not hang up until listening to this voice prompt.

3.6 Send or Receive Fax

3.6.1 Fax Mode Supported

- T.38（IP-based）
- T.30（Pass-through）

3.6.2 Explanation of T.38 and Pass-through

T.38: T.38 is used to transfer faxes over IP networks in real time. It could convert the analog fax signal into digital fax signal and could transform it back from T.38 into analog signal. Under the T.38 mode, fax traffic is carried in T.38 packets.

Pass-through: Under the pass-through mode, fax signal is not converted and fax traffic is carried in RTP packets. It uses the G.711 A or G711U codec in order to reduce the damage to fax signal.

3.7 Restore Default IP and Password

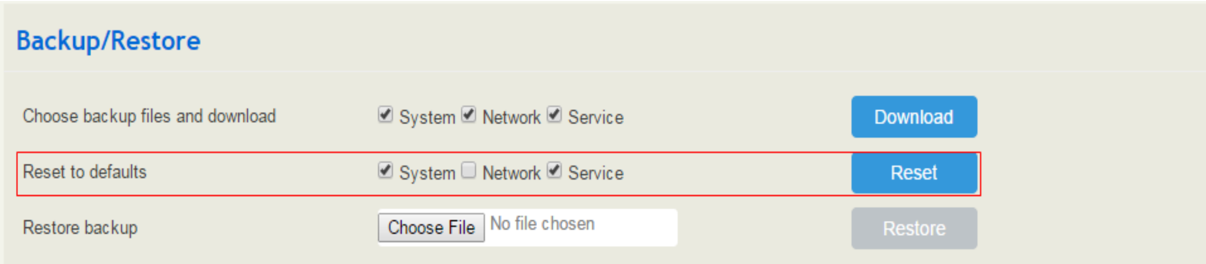
Press the **RST** button of UC2000 for 3 seconds to 6 seconds, the IP address, username and password of the device will be restored to factory defaults.

Press the **RST** button of UC2000 for more than 6 seconds, and all configurations of the device will be restored to the default settings.

3.8 Restore Default Setting

If you want to restore UC2000 to default settings, you can press the **RST** button of UC2000 for more than 6 seconds or you can configure it on the Web interface.

Click **System → Backup/Restore/Upgrade** on the Web of UC2000, and select the parts (system, network or service) that need to be restored to defaults, click **Reset**, restart the UC2000 device, and the selected parts will be restored to defaults.



3.9 Local Maintenance

To ensure easy maintenance, the UC2000 gateway provides a standard RS232 console port, which has a Baud rate of 115200bps. Users can log in the UC2000 to carry out maintenance-related configurations through the console port.

➢ **Example: Log in UC2000 via Console Port**

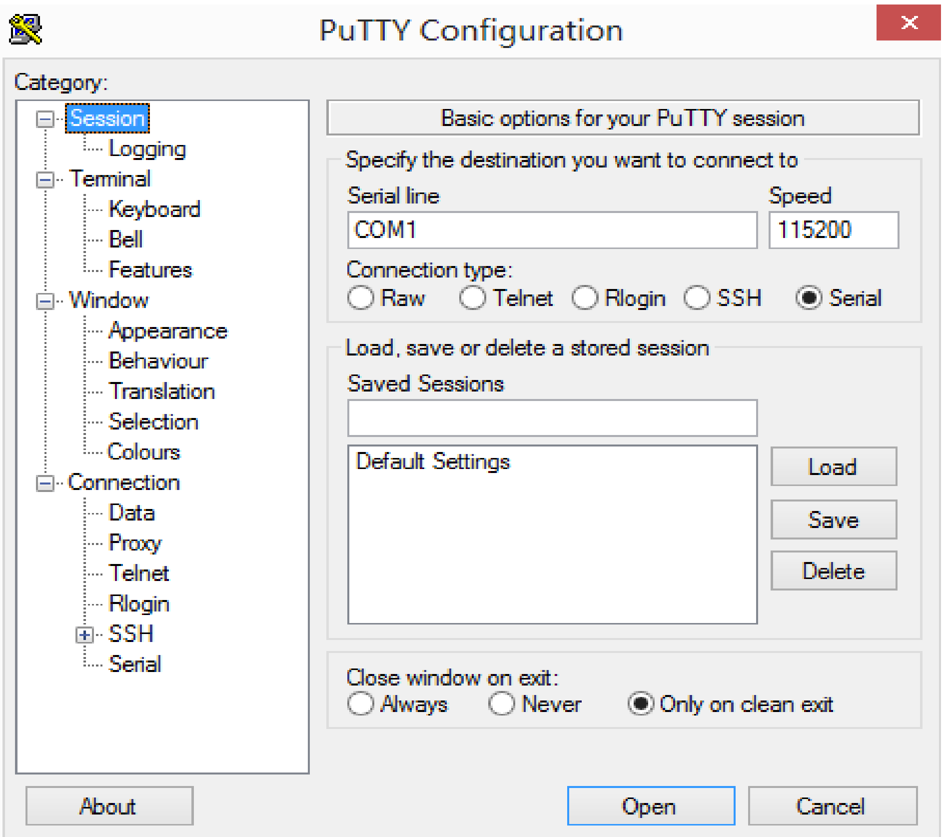
Step 1: Prepare a serial cable.

Step 2: Connect the F port of the serial cable with the COM port of PC. If the PC does not have a COM port, please use a USB-to-COM converting tool to connect the serial cable with the PC.

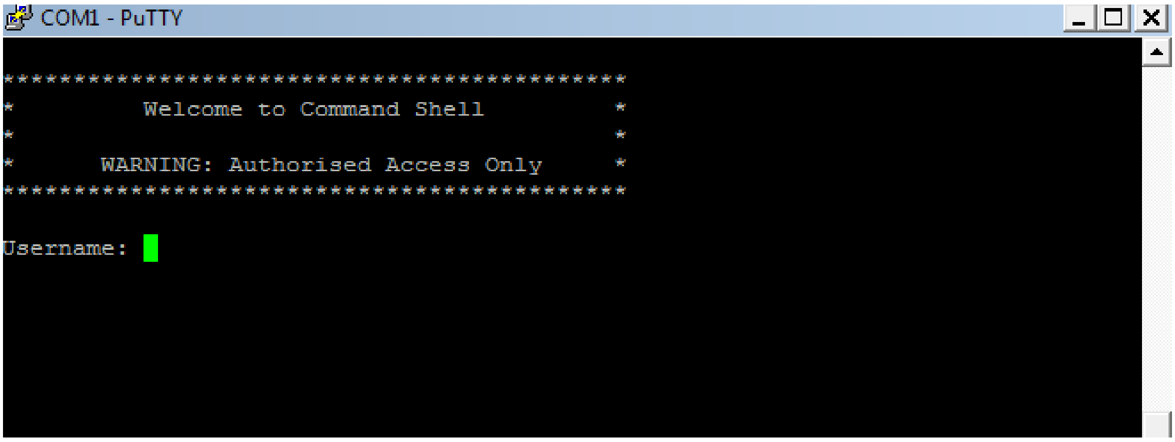
Step 3: Connect the M port of the serial cable with the console port of UC2000.

Step 4: Conduct configurations on login software.

Herein we take the PuTTY sofeware as an example. Detailed configurations are as follows:



After finishing the above configurations, click the **Open** button to enter the maintenance interface of the console port. The username and password are the same with those of the Web interface of UC2000.



Commands for configuring the IP address of UC2000:

(In the following example, IP address of UC2000 needs to be configured as 172.30.66.100, and netmask is 255.255.0.0)

> enable

enable# configure

config# interface ethernet

config-if-br-lan# ip address 172.30.66.100 255.255.0.0

config-if-br-lan# exit

config# ip default-gateway 172.30.0.1

Commands for inquiring the IP address of UC2000:

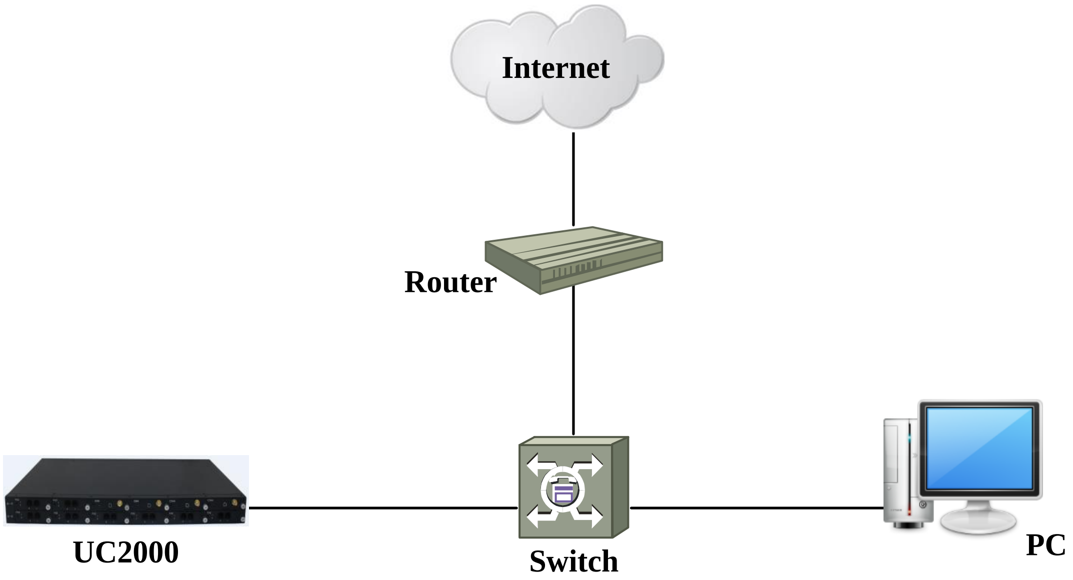
> enable

enable#show interface

4 Configurations on Web Interface

4.1 How to Log in Web Interface

Connect UC2000 to the network according to the following network topology, and dial *158 to query the IP address of the UC2000 device.



4.1.1 Preparations for Login

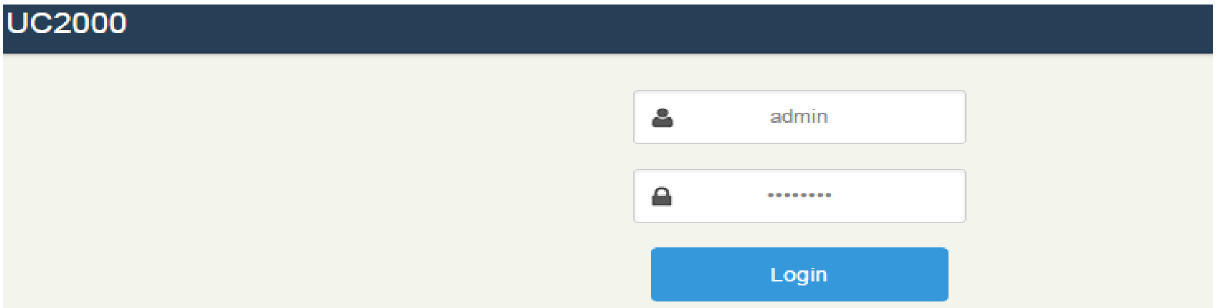
Modify the IP address of the PC to make it at the same network segment with the IP address of UC2000 device, since the default IP address of the UC2000 device is 192.168.11.1.

Check the connectivity between the PC and the UC2000. Click **Start → Run** of PC and enter cmd to execute 'ping 192.168.11.1' to check whether the IP address of the UC2000 runs normally.

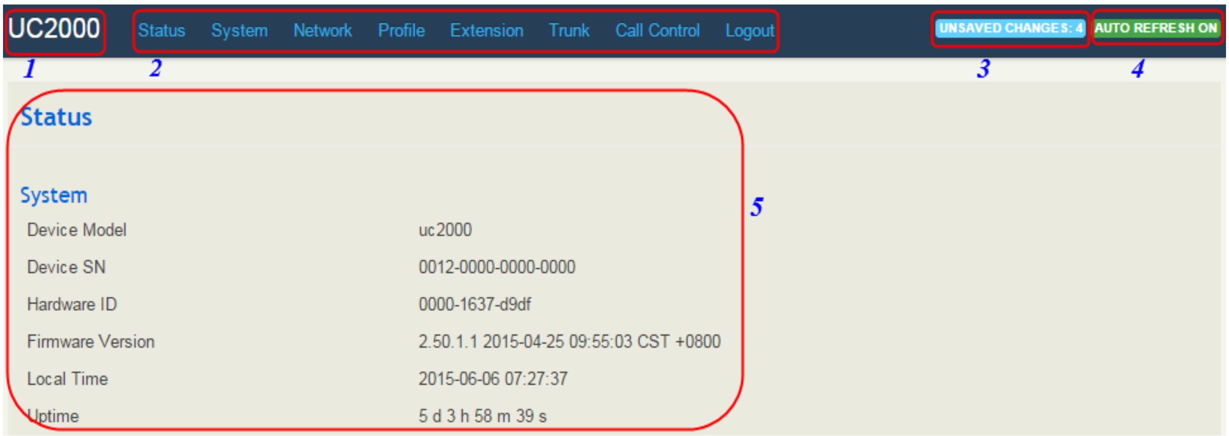
4.1.2 Log in Web Interface

Open a web browser and enter the IP address of the UC2000 (the default IP is 192.168.11.1). Then the login GUI will be displayed. Both the default username and password are admin.

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



Then the following interface will be displayed.

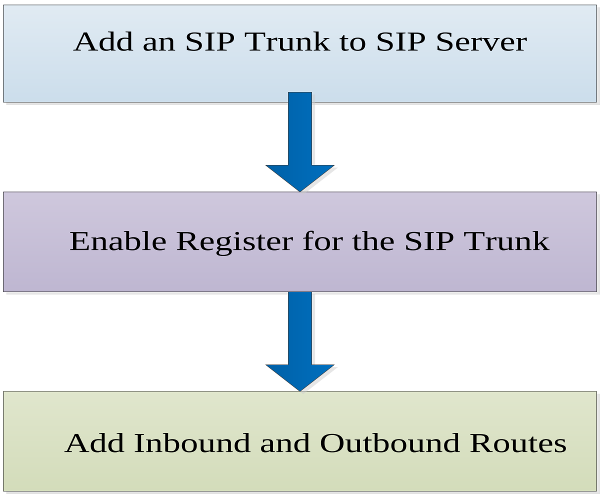


Index	Item	Description
1	UC2000	The name of the gateway; it can be edited on the **System → Setting** interface
2	Menu Bar	The menu bar of UC2000
3	'Unsaved Changes' Button	All configurations or modifications should be saved. Click the button, and you can see a log of all changes. Those changes won't take effect until they are saved.
4	'Auto Refresh' Button	The button can be enabled or disabled. If it is enabled (on green), the information on the **Status → Overview/SIP/PSTN/Current Call** interfaces will be refreshed automatically
5	Detailed Interface	The detailed configuration interface or display interface

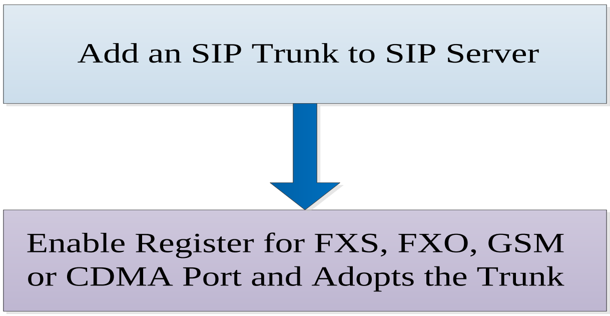
4.2 Configuration Wizard

The following are the common ways to configure the UC2000 device.

4.2.1 UC2000 Regarded as Terminal and Registered to SIP Server



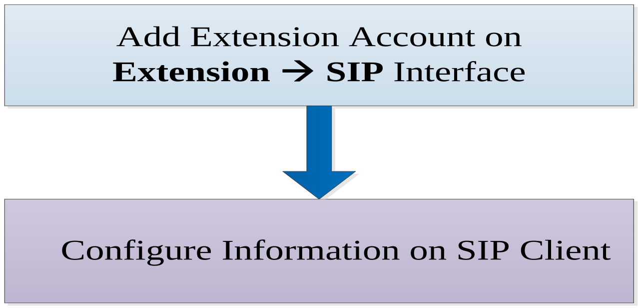
4.2.2 FXS, FXO, GSM or CDMA Port Registered to SIP Server



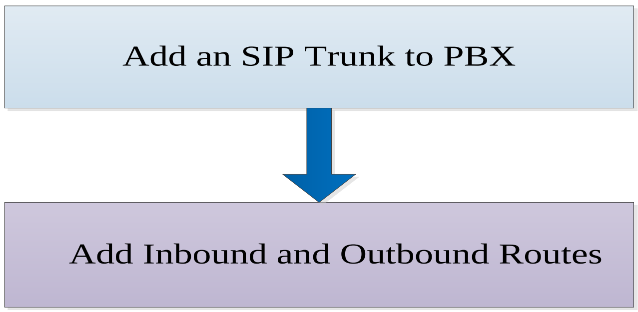
Note: If Register is enabled for the FXS, FXO, GSM or CDMA port, calls through the port will take inbound and outbound routes as first priority. For outgoing calls, if outbound route cannot be matched, then the registered SIP trunk will be selected. For incoming calls, if inbound route cannot be matched, then the registered FXS, FXO, GSM or CDMA port will be selected.

4.2.3 Other SIP Clients registered to UC2000

Under this mode, UC2000 is regarded as an SIP Server. Create an extension account first on the **Extension →SIP** interface of UC2000, and configure listening port on the **Profile → SIP** interface. Then, configure the UC IP address, extension account and listening port on SIP Clients.



4.2.4 UC2000 Connected to PBX through Trunking



4.3 Status

4.3.1 Overview

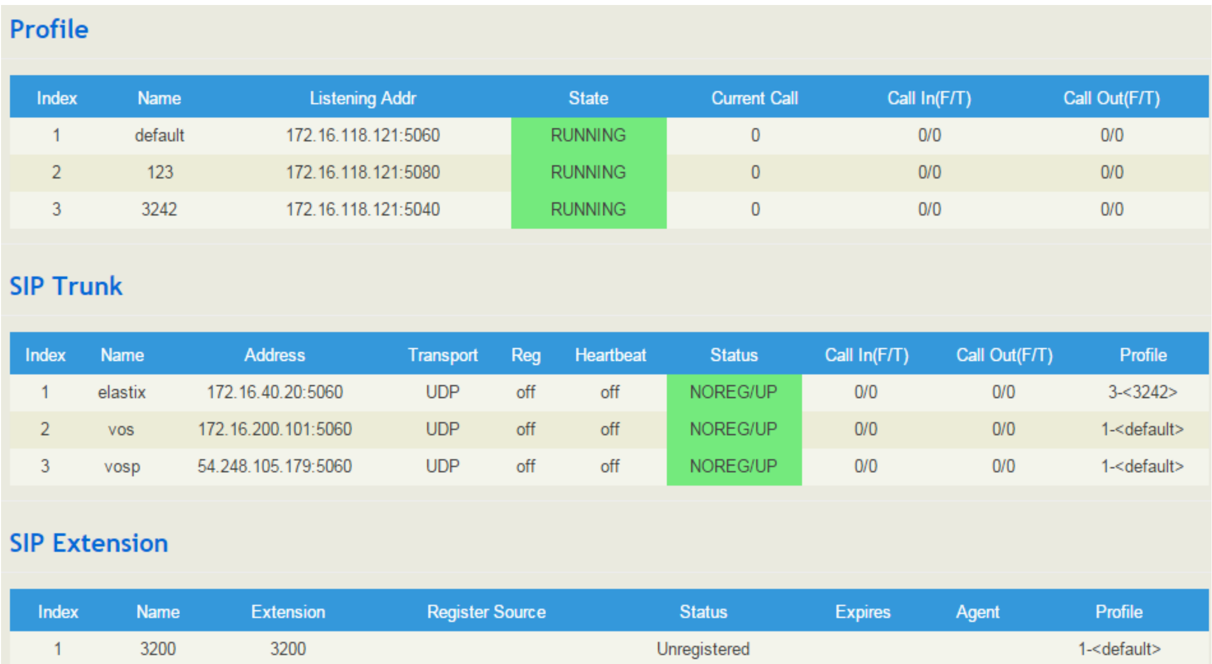
Log in the Web interface of UC2000, click **Status** → **Overview**, and the following interface will be displayed. On the interface, information about the system, performance and network is shown.

Parameter	Explanation
Device Model	The name of the device, which can be self-defined by users on the **System -> Setting** interface.
Device SN	The SN of the device
Hardware ID	The hardware ID of the device, which cannot be modified.
Firmware Version	The version number of the firmware and the time when the firmware is compiled.
Local Time	The local time, which can synchronize with NTP
Uptime	The running time of the device since it is booted up
CPU	The current CPU occupancy rate

Filesystem	The current utilization rate of the file system
Memory	The current utilization rate of the memory
Mac Address	The Mac address of the network port, which is used for network communication
Type	The type of IP address, which can be Static IP, PPPoE or DHCP
IP Address	The IP address of the network port
Netmask	The netmask of the network port
Gateway	The gateway address of the network port
DNS	The DNS address of the network port
RX/TX (Per Second)	The bit rate of receiving and sending messages per second over the network port
RX/TX (Total)	The total amount of messages received and sent over the network port since the device is booted up
Expires	The time remaining until expiry for DHCP IP address
Connected	The running time of the network port since the device is booted up

4.3.2 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.


Profile

Index	Name	Listening Addr	State	Current Call	Call In(F/T)	Call Out(F/T)
1	default	172.16.118.121:5060	RUNNING	0	0/0	0/0
2	123	172.16.118.121:5080	RUNNING	0	0/0	0/0
3	3242	172.16.118.121:5040	RUNNING	0	0/0	0/0

SIP Trunk

Index	Name	Address	Transport	Reg	Heartbeat	Status	Call In(F/T)	Call Out(F/T)	Profile
1	elastix	172.16.40.20:5060	UDP	off	off	NOREG/UP	0/0	0/0	3-<3242>
2	vos	172.16.200.101:5060	UDP	off	off	NOREG/UP	0/0	0/0	1-<default>
3	vosp	54.248.105.179:5060	UDP	off	off	NOREG/UP	0/0	0/0	1-<default>

SIP Extension

Index	Name	Extension	Register Source	Status	Expires	Agent	Profile
1	3200	3200		Unregistered			1-<default>

Belong To	Parameter	Explanation
Profile	Name	The name of the corresponding SIP profile
	Listening Address	The current listening address and the 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	Name	The name of the corresponding SIP trunk
	Address	The address and the port of peer device under the SIP trunk
	Reg	Whether the SIP trunk is registered to the peer device
	Transport	Transmission protocol
	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	Name	The name of the SIP extension
	Extension	The extension number
	Status	SIP extension is registered or not. There are two statuses: Registered. Unregistered
	Profile	The profile that is used by the SIP extension

4.3.3 PSTN

Click **Status → PSTN**, and the following interface will be displayed. On the top is the front view of UC2000, followed by the real-time information of FXS userboard, FXO userboard, GSM userboard and CDMA userboard. Green color means available or registered, while red color means abnormal, unregistered or prohibited.

FXS

Slot	Type	State	Config Status	Port	Number	SIP Register Status	Hook State
4	FXS	READY	OK	0	9600	Reged(All)	ONHOOK
				1	9601	Reged(All)	ONHOOK
5	FXS	READY	OK	0	8010	Unregistered	ONHOOK
				1	8011	Unregistered	ONHOOK

FXO

Slot	Type	State	Config Status	Port	Number	SIP Register Status	Hook State	Line State
7	FXO	READY	OK	0	5600	Reged(All)	ONHOOK	OFFLINE
				1	8015	Unregistered	ONHOOK	OFFLINE
8	FXO	READY	OK	0	8016	Unregistered	ONHOOK	OFFLINE
				1	8017	Unregistered	ONHOOK	OFFLINE

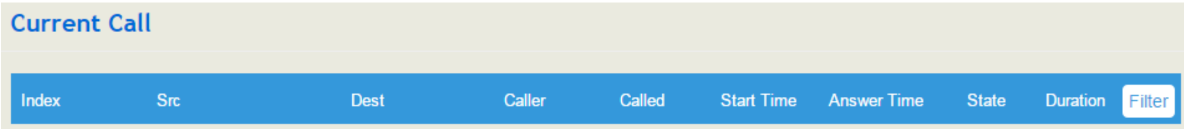
GSM/CDMA

Slot	Type	State	Channel State	Number	SIP Register Status	Signal	Talking State
6	GSM	READY	OK	8012	Unregistered		IDLE
9	GSM	READY	OK	8018	Unregistered		IDLE
10	CDMA	READY	SIMPIN_NOT_INSERTED	5601	Unregistered		IDLE
11	CDMA	READY	OK	8022	Unregistered		IDLE

Belong to	Parameter	Include
FXS	State	Ready, Unready
	Config Status	OK, Config Failed
	SIP Register Status	Registered, Unregistered
FXO	State	Ready, Unready
	Config Status	OK, Config Failed
	SIP Register Status	Registered, Unregistered
	Hook State	Onhook, Offhook
	Line State	Online, Offline
GSM	State	Ready, Unready
	Channel State	OK, Config Failed, SIMPIN_Not_INSERTED
	SIP Register Status	Registered, Unregistered
	Signal	: No SIM card is inserted : Full signal strength

4.3.4 Current Call

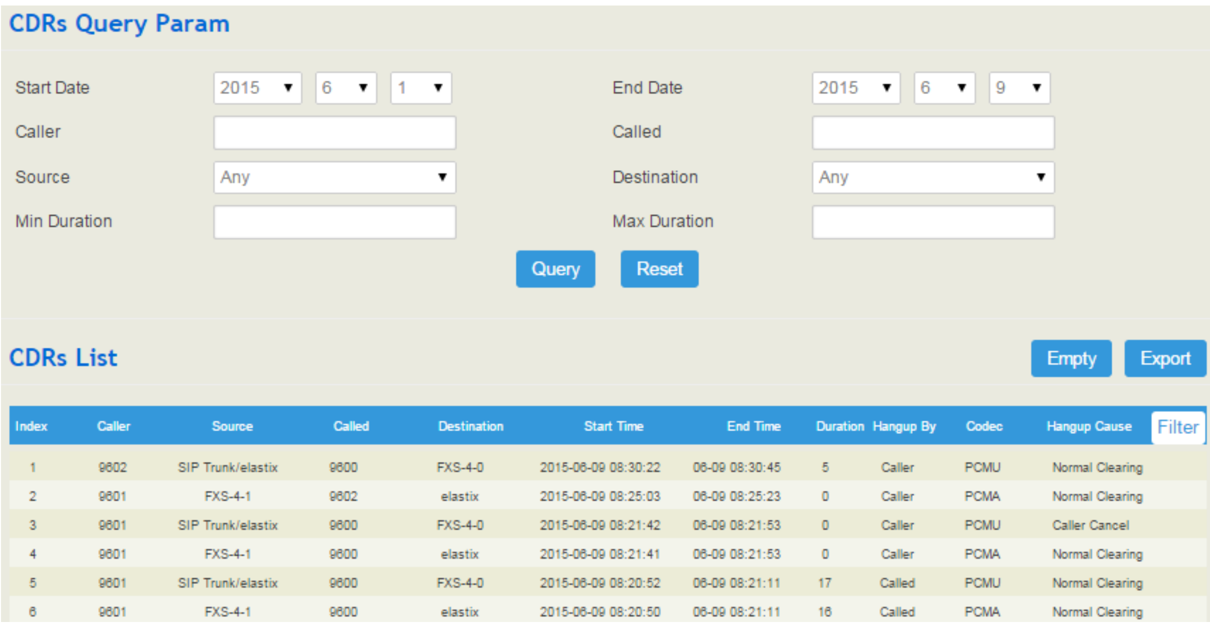
Click **Status →Current Call**, and the following interface will be displayed. On the interface, the source, destination, calling number, called number, start time, answer time, state and duration of the current real-time call are shown. If there is no current call, no information will be shown.



Parameter	Explanation
Src	The source of the current call
Dest	The destination of the current call
State	There are three states: Active: it means the caller and the called party is on conversation Ringing: it means the phone of the called party is ringing Early: It means the ring-back tone of the current call is manipulated

4.3.5 CDRs

Click **Status → CDRs,** and you can set query criteria to query the CDRs that you want on the displayed interface. Meanwhile, you are allowed to clear CDRs or export CDRs through pressing the **Empty** or **Export** button. The maximum number of CDRs that can be saved is 1000.



Hangup causes include normal clearing, no answer, caller cancel, user busy, circuit congestion, exchange routing error, recovery on timer expire, and none.

4.3.6 Service

Click **Status → Service**, and the service status of UC2000 is displayed. The function is enabled by default. The Web, SSH and Telnet service can be disabled and their ports can be modified on the **System → Access Control** interface. If no running status is shown, it means exception has occurred on the UC2000.

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

4.3.7 About

Click **Status → About**, copyright, device model, hardware version and firmware version are displayed.

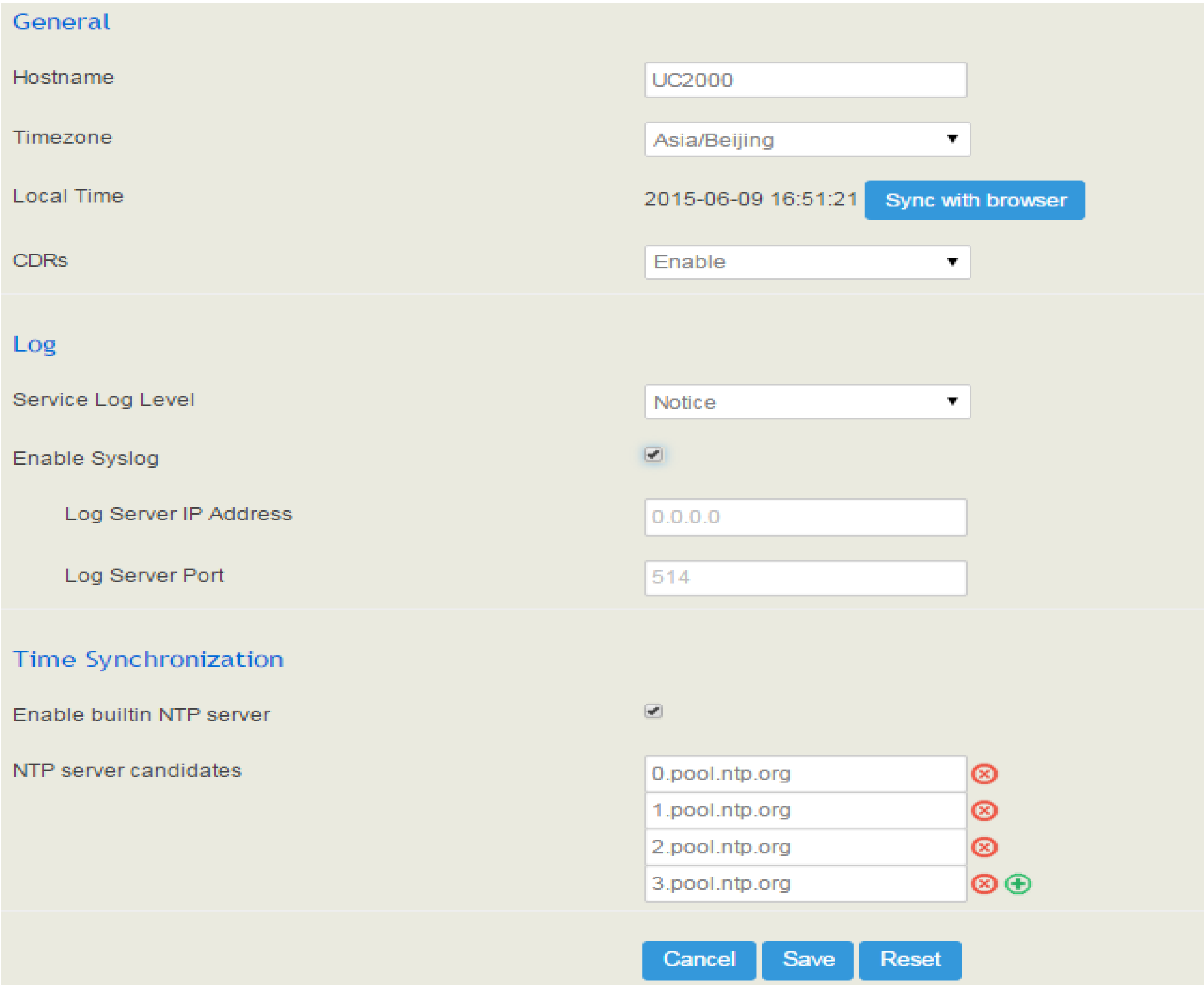
Userboard

Slot	Type	Hardware Version	Firmware Version
3	FXS	1	1.1.3
4	FXS	1	1.1.3
5	FXS	1	1.1.3
6	GSM	1	1.1.3
7	FXO	1	1.1.3
8	FXO	1	1.1.3
9	GSM	1	1.1.3
10	CDMA	1	1.1.3
11	CDMA	1	1.1.3

4.4 System

Configurations for language, time zone, NTP, login password, access control, provision, operation log, service log, upgrade, backup, restore, automatch impedance, IVR upload and device reboot can be carried out in the System section.

4.4.1 Setting

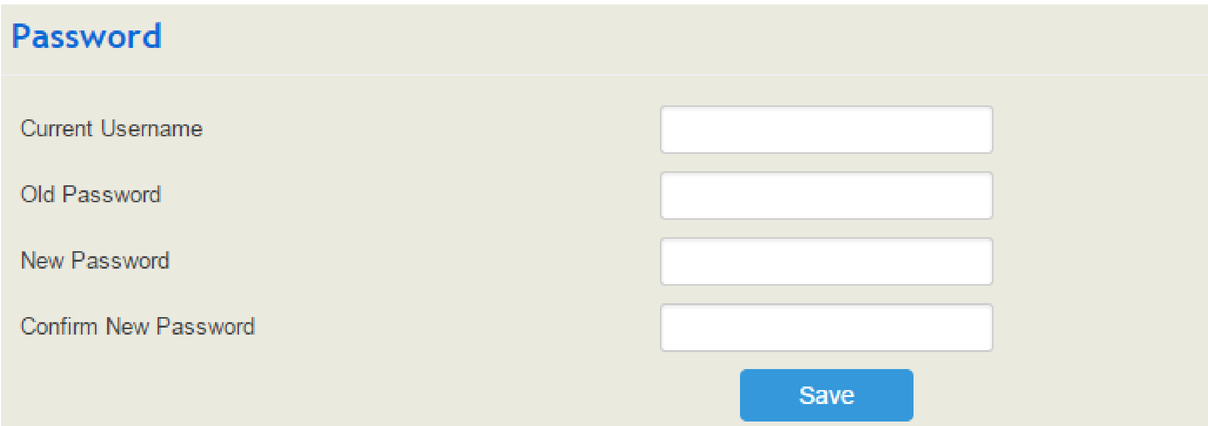


Parameter	Explanation
Hostname	The name of the gateway. After it is configured, the name will be displayed on the left of the menu bar.
Time Zone	You can choose a time zone you want. The default value is UTC (Universal Time Coordinated)
Local Time	The current time based on current time zone. It is synchronized with NTP.
CDRs	If it is enabled, CDRs will be saved automatically. 1000 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 built-in NTP server	If built-in NTP server is enabled, the UC2000 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 addresses of the NTP servers.
	Delete a NTP Server
	Add a NTP Server

4.4.2 User Manager

Click **System → User Manager**, and you can modify the username name and password for logging in the UC2000 device. Factory defaults for username name and password are both admin, so it is advised to modify them for security consideration.

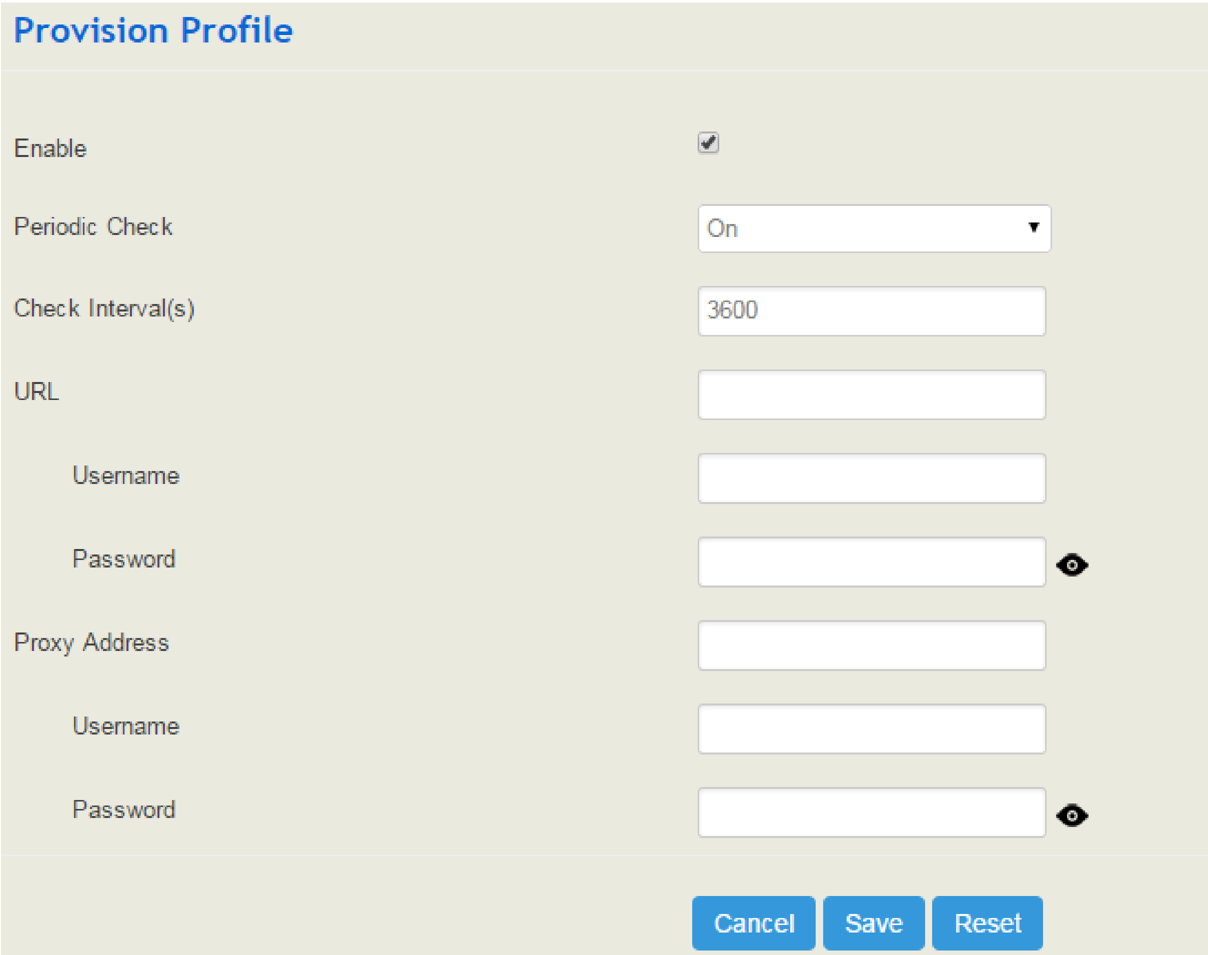
The abovementioned username and password are also used to log in Web Interface, Telnet and SSH.



4.4.3 Provision

Provision is used to make UC2000 automatically upgrade with the latest firmware stored on an http server an ftp server or a tftp server.

Select the checkbox on the right of **Enable**, and you will see the following interface:

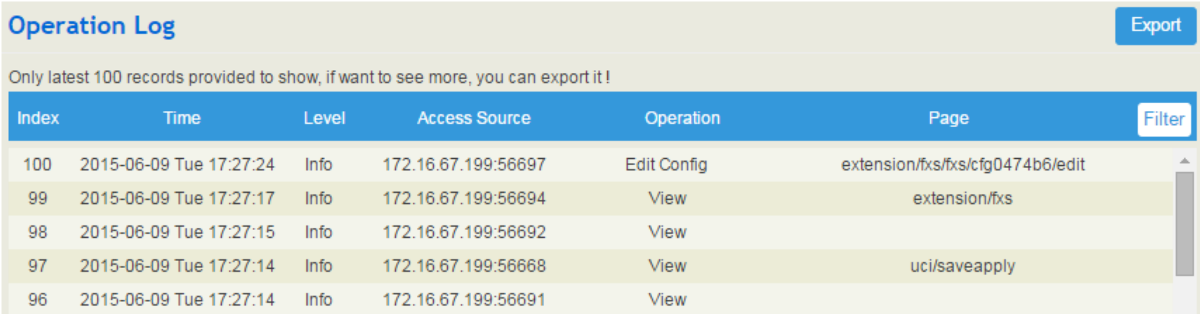


Parameter	Explanation
URL	The URL of the http/ftp/tftp server for example: ftp://172.16.77.200/home tftp://172.16.77.200/provision.xml http://test.domain.com/test
username	The login username of the http/ftp/tftp server
Password	The login password of the http/ftp/tftp server

Note: Proxy Address, Proxy Username and Proxy Password are optional to be configured.

4.4.4 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.

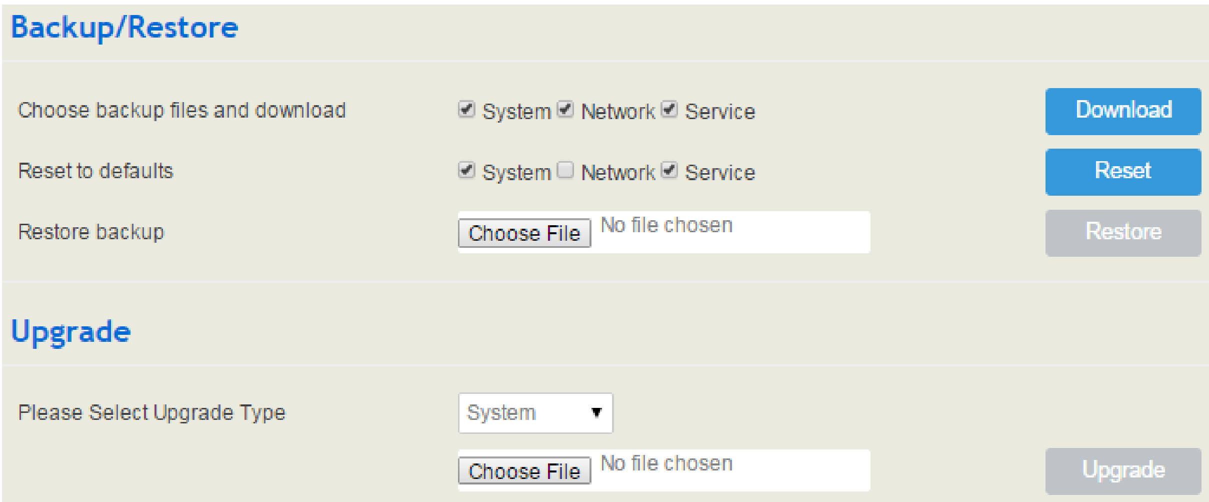


4.4.5 Service Log

Service logs (the running logs of UC2000) can be exported on the **System → Service Log** interface. Those logs are used for analyzing where a problem has occurred on UC2000.

4.4.6 Backup/Restore/Upgrade

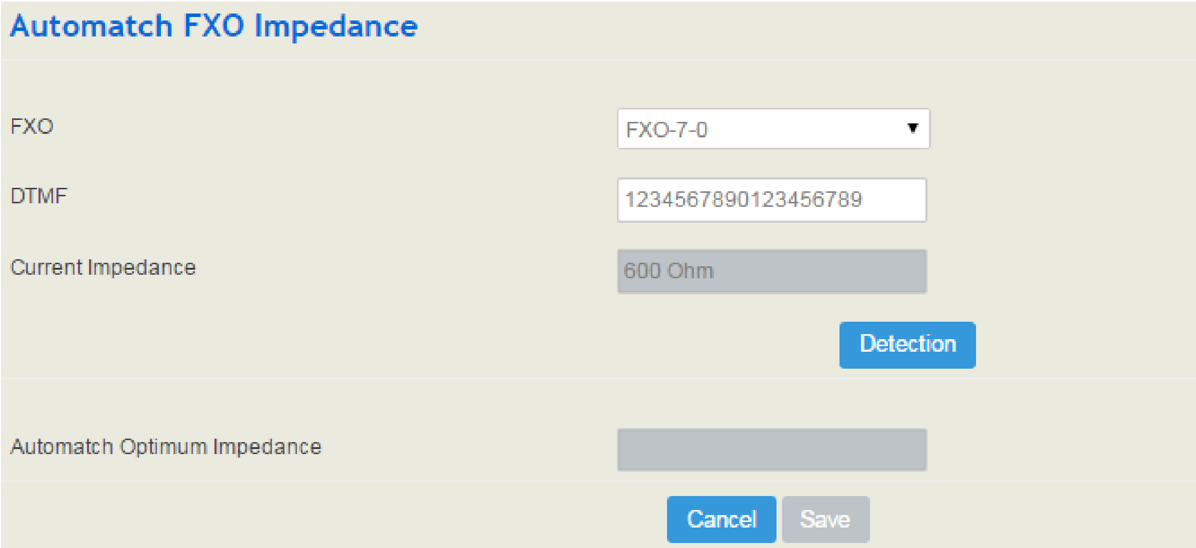
On the **System → Backup/Restore/Upgrade** interface, you can back up or restore configuration data, and can upgrade UC2000 to a new version. But you need to restart the device for the change to take effect after executing restore or upgrade.



Parameter	Explanation
Reset	Click **Reset**, and all configuration will be restored to factory defaults. Note: press the RST button on the back panel of UC2000, only the login username, password and the Web port can be restored to defaults.
Restore	Choose a backup file, and then click **Restore**.
Upgrade	Choose an upgrade file (which is provided by Shenzhen Dinstar Technologies), and then click **Upgrade**.

4.4.7 Automatch Impedance

Automatch impedance is used to improve the interoperability of the FXO userboards with other devices.



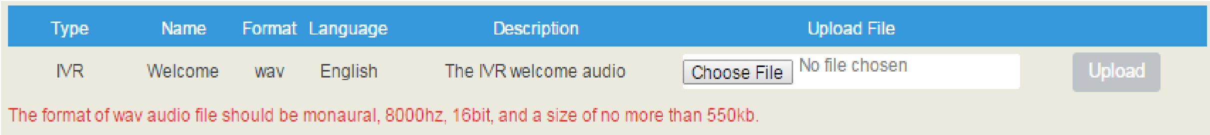
How to use automatch impedance:

1. Select the to-be-matched FXO port in the drop-down box;
2. Connect a telephone cable to the FXO port and ensure the connection is successful;
3. Click **Detection**, and the UC2000 device will automatically detect the optimum impedance (It takes a period of time to carry out the detection).
4. Save the optimum impedance.

Note: You can enter any digits for DTMF number; the default DTMF number 1234567890123456789.

4.4.8 Voice

On the **System → Voice** interface, you can upload an English IVR file or a Chinese IVR file according to your needs. At present, only those IVR files in wav format are allowed.



Note:

- The format of the wav audio file should be monaural, 8000hz, 16bit, play time of less than 30s, and size of no more than 1M bytes;
- If the following yellow bars appear, it means the UC2000 device lacks an IVR file and you need to upload one.

4.4.9 Reboot

On the **System → Reboot** interface, you can click **Perform Reboot** to reboot the UC2000 device. After the device is rebooted, those configurations that have been saved will remain unchanged.

4.5 Network

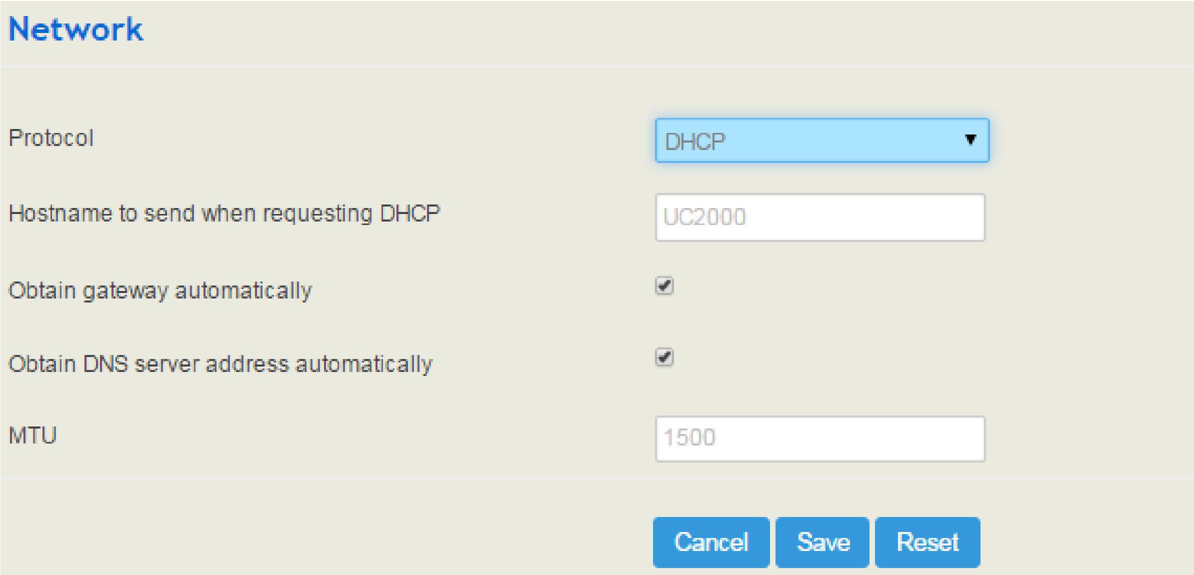
UC2000 only works in bridge mode, with FE0 port and FE1 port functioning identically

4.5.1 Setting

There are three kinds of IP addresses for selection for the FE0 and FE1 ports, including Static IP address, DHCP IP address and PPPOE IP address.

DHCP: Obtain IP address automatically.

UC2000 is regarded as a DHCP client, which sends a broadcast request and looks for a DHCP server to answer. Then the DHCP server automatically assigns an IP address to the UC2000 from a defined range of numbers.

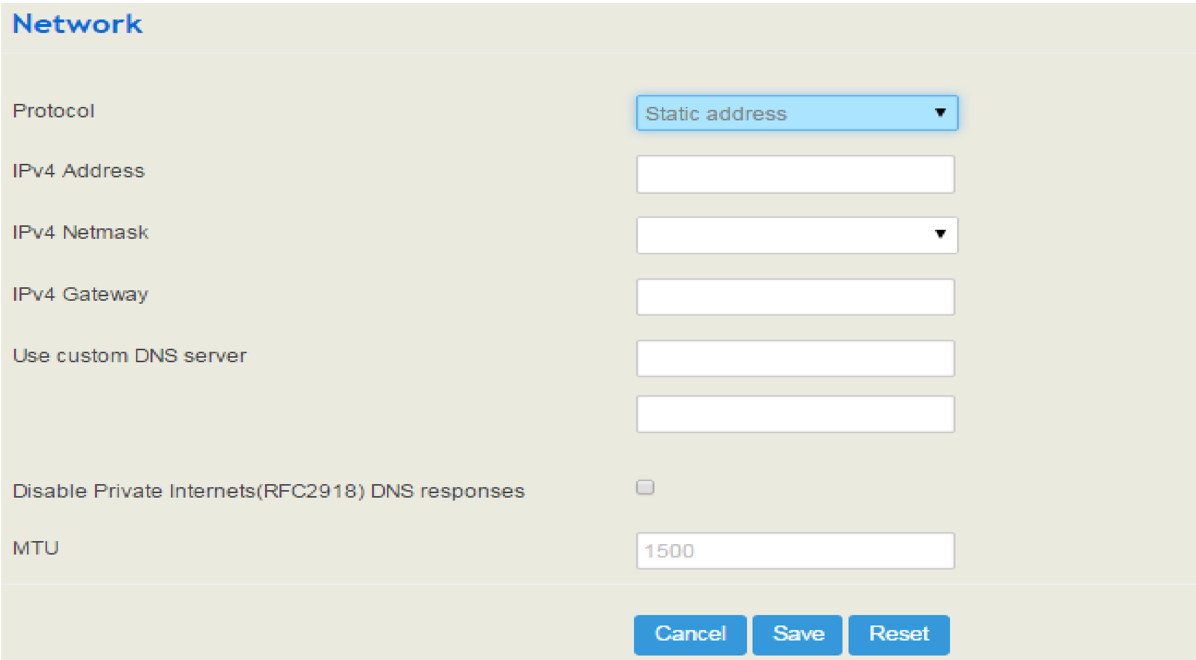


Static IP Address:

Static IP address is a semi-permanent IP address and remains associated with a single computer or other device 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:

- IPv4 Address: the IP address of the FE0/FE1 port of the UC2000 ;
- IPv4 Netmask: the netmask of the router connected the UC2000;
- IPv4 Gateway: the IP address of the router connected the UC2000;
- Use custom DNS server: the IP address of the DNS server

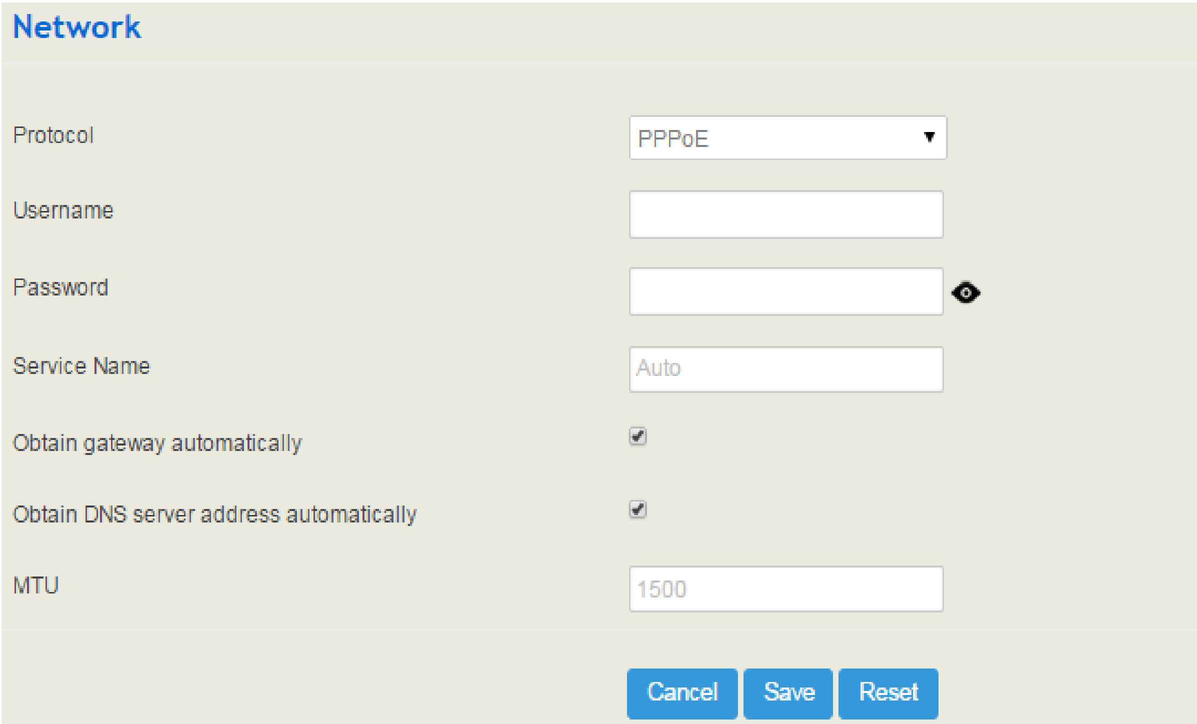


PPPoE:

PPPoE is an acronym for point-to-point protocol over Ethernet, which relies on two widely accepted standards: PPP and Ethernet. PPPoE is a specification for connecting the users on an Ethernet to the Internet through a common broadband medium, such as a single DSL line, wireless device or cable modem. PPPOE IP address refers to IP address assigned through the PPPoE protocol.

If you choose PPPoE, you need to fill in to fill in the following information:

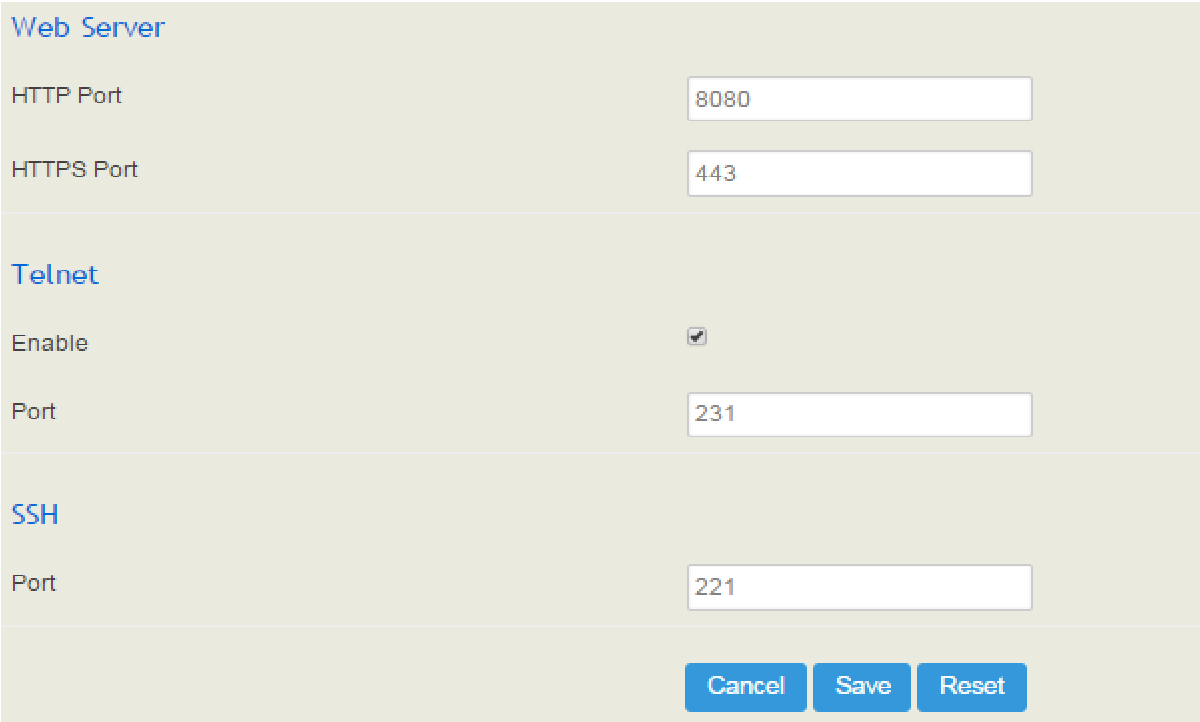
- Username: the account name of PPPoE
- Password: the password of PPPoE
- Server Name: the name of the server where PPPoE is placed



Note: After the configurations are completed, you need to reboot the UC2000 device for the network changes to take effect.

4.5.2 Access Control

The access ports of Web, Telnet and SSH can be configured on the Access Control interface. Web supports http and https, while SSH supports OAuth 2.0 protocol.



4.5.3 Diagnostics

There are three utilities to diagnose the network, including Ping, Traceroute and Nslookup.

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 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.

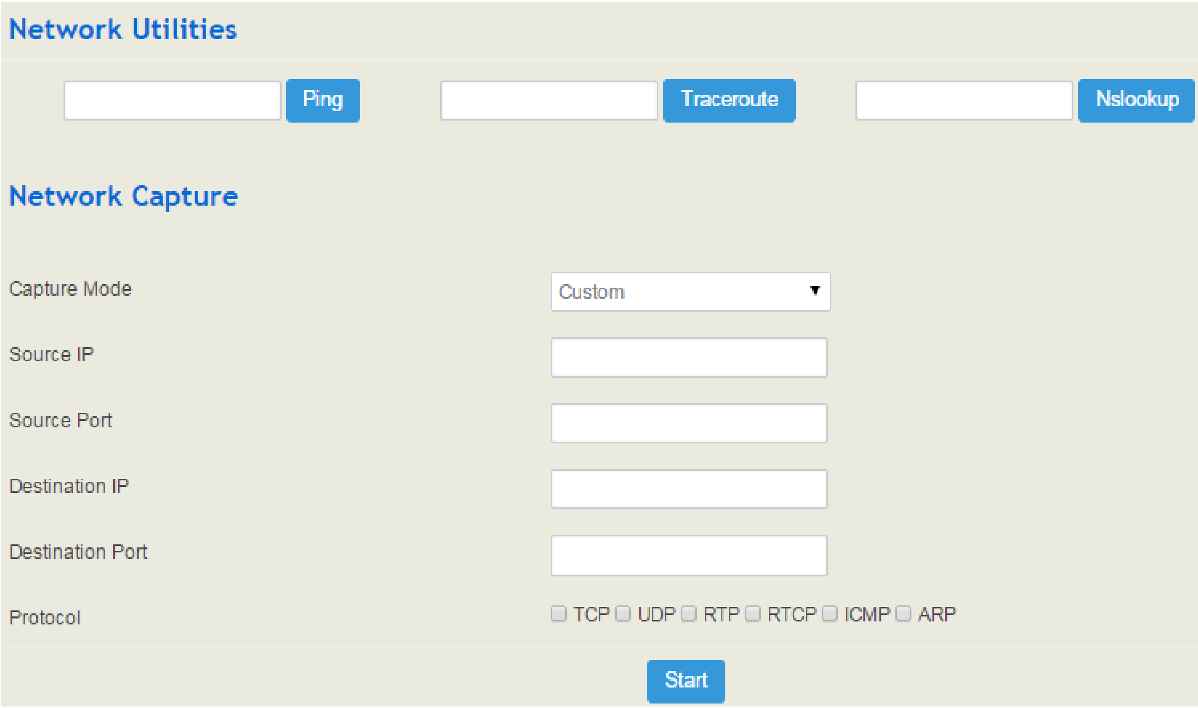
Nslookup (Name Server Lookup) is a network command-line tool to obtain domain name of internet or to diagnose the problems of DNS.

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.

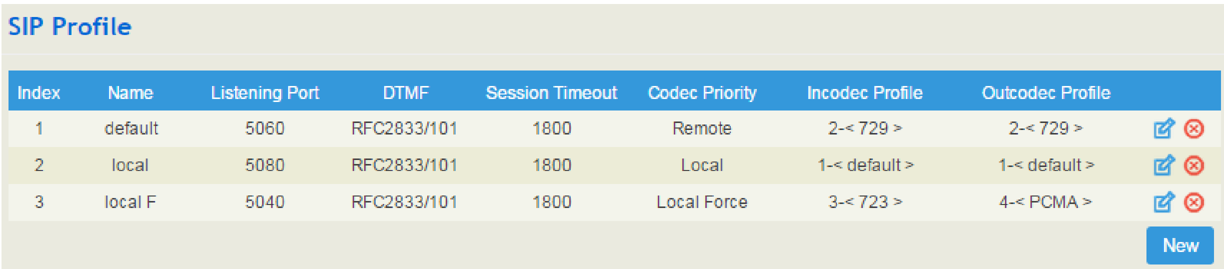


Note: If there are multiple IP addresses, please use | to separate them.

4.6 Profile

4.6.1 SIP Profile

On the **Profile → SIP** interface, you can set SIP information such as listening port, which will be used in FXO/FXS, extension and SIP trunk. Multiple SIP profiles can be configured for one UC2000 device, so you can choose different SIP profiles according to different needs.

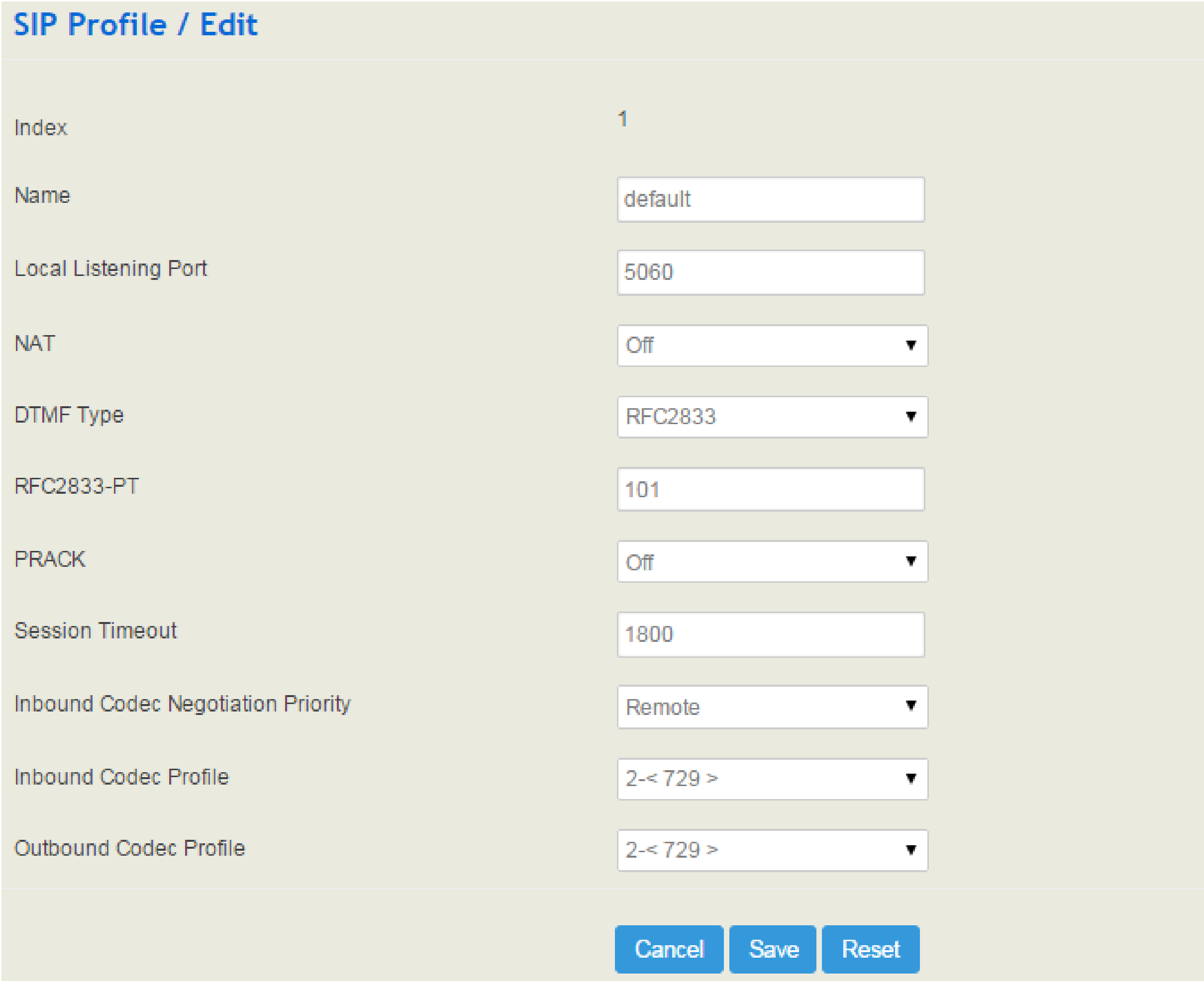


SIP Profile

Index	Name	Listening Port	DTMF	Session Timeout	Codec Priority	Incodec Profile	Outcodec Profile
1	default	5060	RFC2833/101	1800	Remote	2-< 729 >	2-< 729 >
2	local	5080	RFC2833/101	1800	Local	1-< default >	1-< default >
3	local F	5040	RFC2833/101	1800	Local Force	3-< 723 >	4-< PCMA >

New

Click **New** to create a new SIP profile, or click to edit the information of an existing SIP profile.



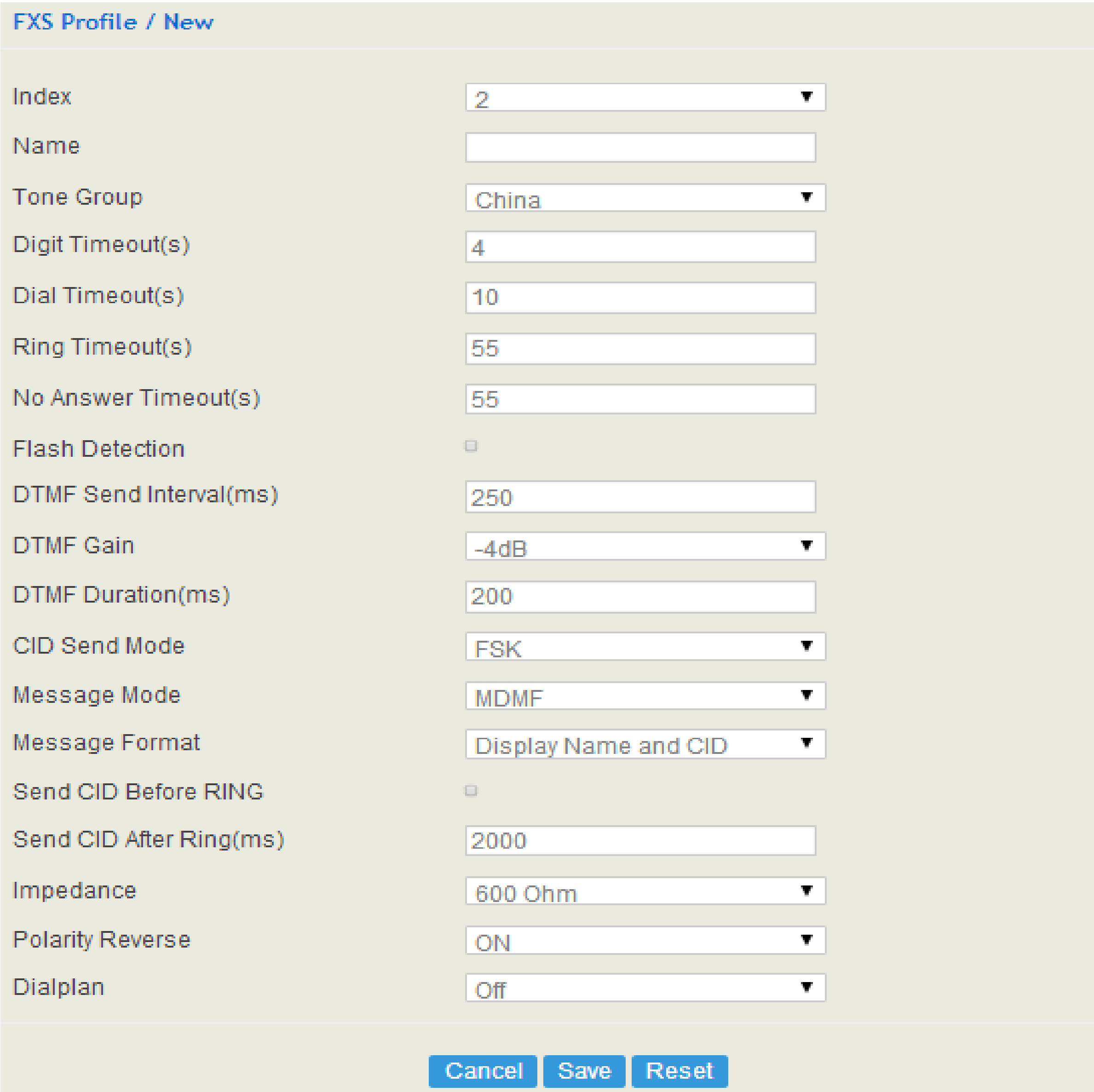
Parameter	**Explanation**
Name	The name of the SIP profile
Local Listening Port	The local listening port of SIP. If the SIP profile is used by a SIP trunk, the port filled in here is the listening port for the SIP trunk.
NAT	Methods for NAT traversal, including static NAT (IP Address and Host), dynamic NAT (uPNP/NAT-PMP), and STUN. NAT can also be disabled (off).
DTMF Type	There are three modes, including SIP Info, INBAND, RFC2833.
RFC2833-PT	RFC2833 payload coding. The default value is 101.
PRACK	Provisional Response ACKnowledgement
Session Timeout	The validity period of current registration. It is 1800 seconds by default.
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.
Inbound Codec Profile	The inbound codec profile supported by SIP of UC2000. Make reference to **Profile →Codec**.
Outbound Codec Profile	The outbound codec profile supported by SIP of UC2000. Make reference to **Profile →Codec**.

4.6.2 FXS/FXO

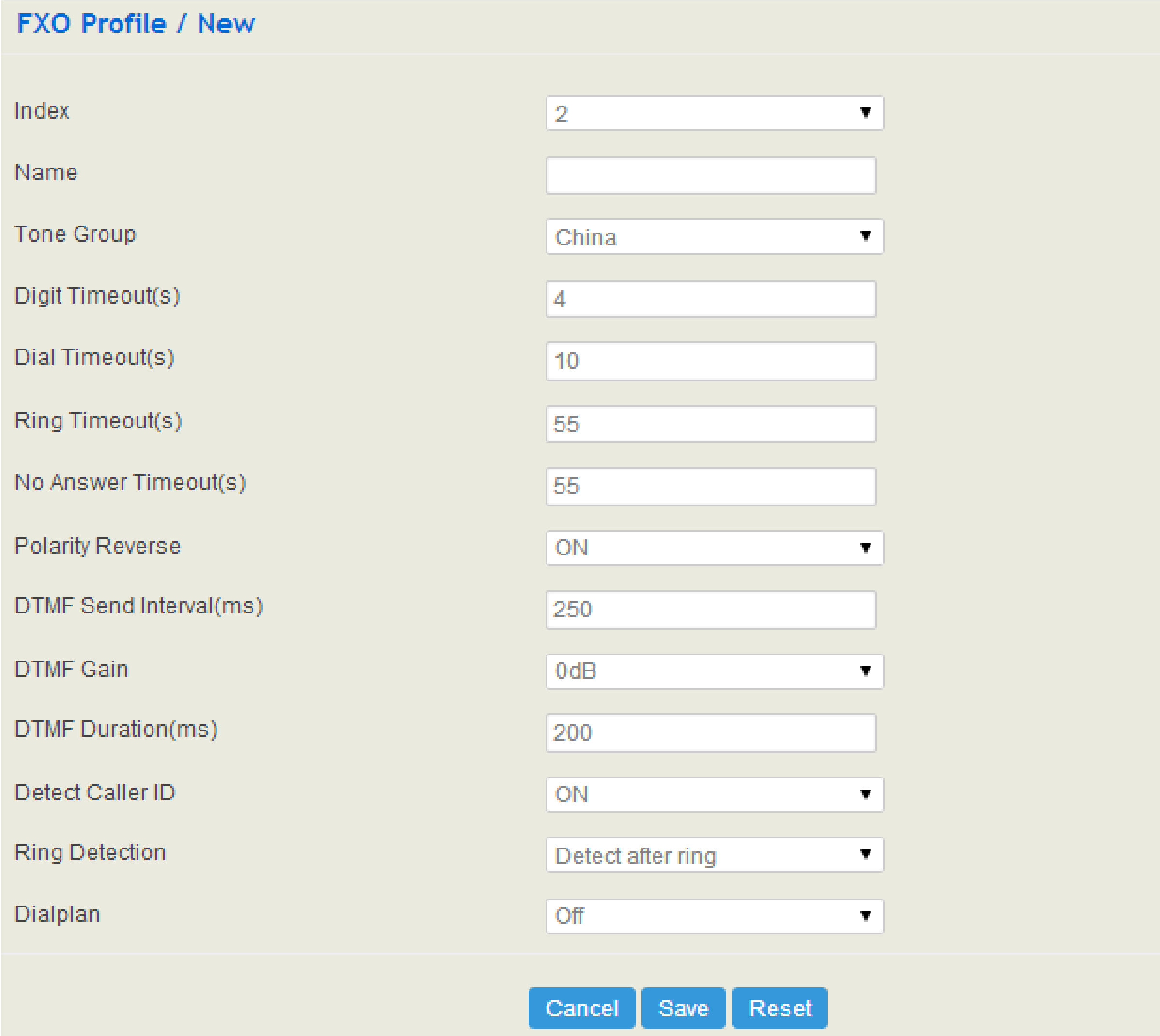
On the **Profile → FXS/FXO** interface, you can configure the driving parameters of FXS userboards and FXO userboards.

Profile → FXS Interface:



Parameter	Explanation
Name	The name of the FXS profile
Tone Group	The national standard of dialing tone, busy tone and ring tone
Digit Timeout (s)	The timeout value for dialing a digit of a telephone number. The default is 4 seconds.
Dial Timeout (s)	The timeout value for dialing the first telephone number after off-hook. The default is 10 seconds.
Ring Timeout (s)	The timeout value for the ringing of the analog phones of the FXS port when there are incoming calls
No Answer Timeout (s)	The timeout value for ending calls which go out through FXS port. When the value is exceeded, the call will end and the caller will hear busy tone.
Flash Detection	Whether to execute flash detection; If flash detection is not executed, the press on flash-hook won't be processed.
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
CID Send Mode	Include FSK and DTMF FSK: Frequency-shift keying CID: Caller ID
Message Mode	Include SDMF and MDMF
Message Format	Include Display Name and CID, Only display Name, Only display CID
Send CID Before Ring	If it is enabled, the CID will be shown before ringing; otherwise, CID will be displayed after ringing
Send CID After Ring(ms)	The interval between ringing and displaying of CID
Impedance	The impedance matched with analog phones
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 to delay offhook and call tolls will be calculated starting from the set time.
Dialplan	The rules for dialing

Profile → FXO Interface:

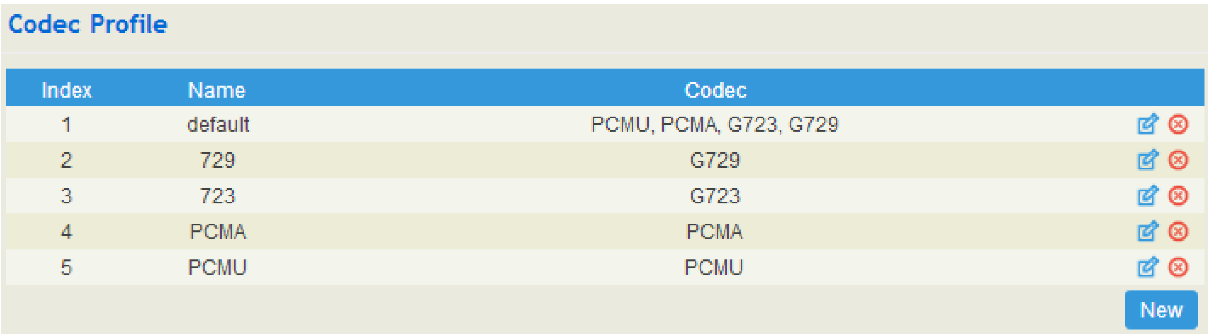


Parameter	Explanation
Name	The name of the FXO profile
Tone Group	The national standard of dialing tone, busy tone and ring tone
Digit Timeout (s)	The timeout value for dialing a digit of a telephone number; the default is 4 seconds.
Dial Timeout (s)	The timeout value for dialing the first telephone number after off-hook; the default is 10 seconds.
Ring Timeout (s)	The timeout value for the ringing of the phones through the FXO port when there are incoming calls
No Answer Timeout (s)	The timeout value for ending calls which go out through FXO port

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 delay and call tolls will be calculated starting from the set time.
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
Detect Caller ID	Whether to detect caller ID; default value is ‘On’
Ring Detection	Detect caller ID after ringing or detect caller ID before ringing
Dialplan	The rules for dialing

4.6.3 Codec

UC2000 supports four codec modes, including G729, G723, PCMU and PCMA. You can adjust the priority of these four modes according to you needs.



Codec Profile

Index	Name	Codec
1	default	PCMU, PCMA, G723, G729
2	729	G729
3	723	G723
4	PCMA	PCMA
5	PCMU	PCMU

New

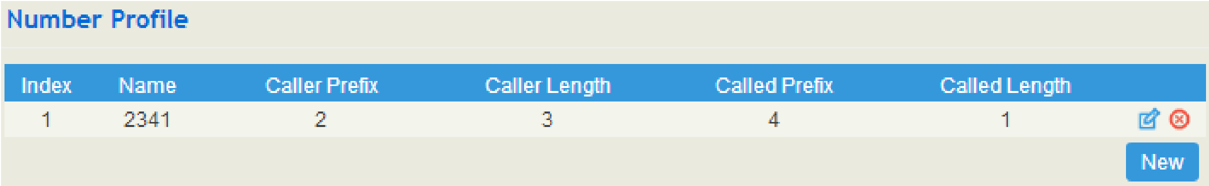
: Edit codec profile.

: Delete the corresponding codec profile or a codec mode.

New: Create a new codec profile.

4.6.4 Number

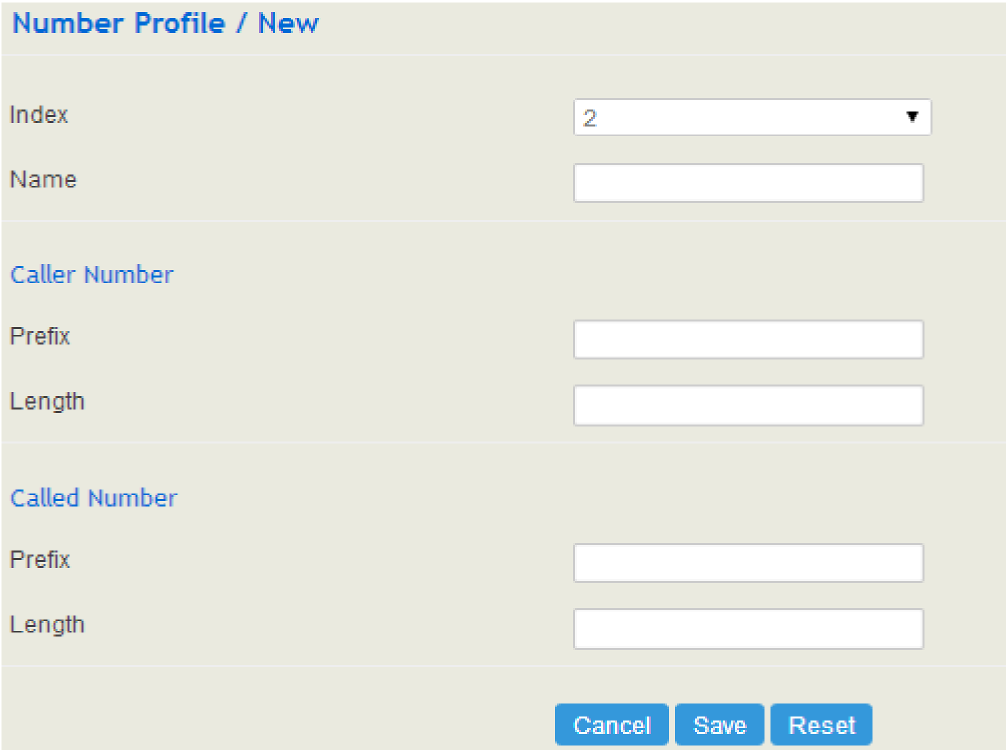
On the **Profile →Number** interface, you can set a prefix for calling numbers or called numbers. When the prefix of a calling number or a called number matches the set prefix, the call will be passed to choose a route.



: Edit codec profile.

: Delete the corresponding codec profile or a codec mode.

Click New, and you will see the following interface:

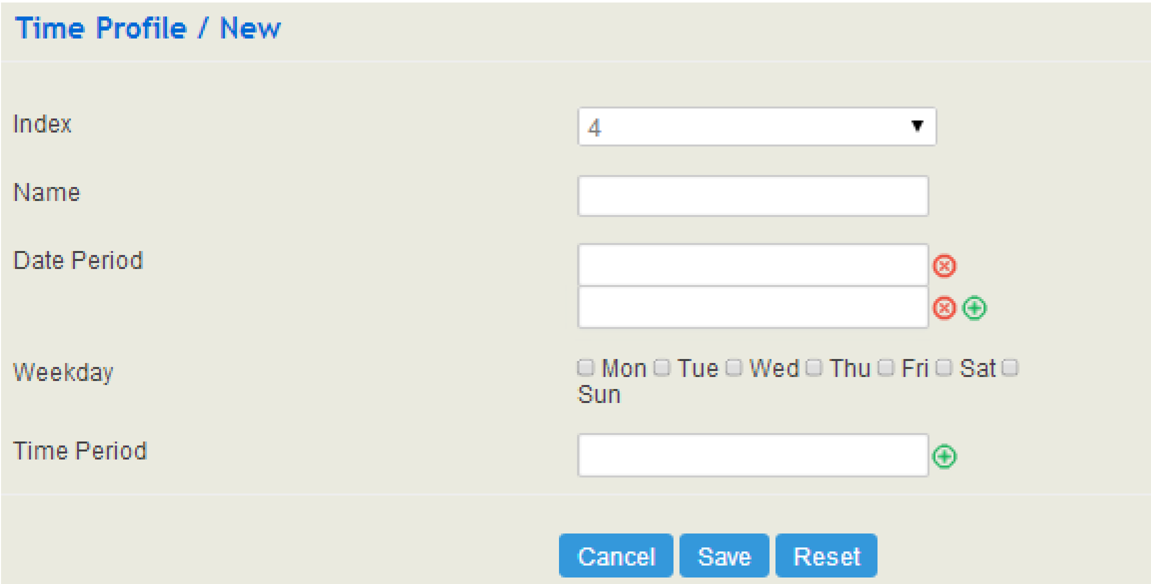


Parameter	**Explanation**
Name	The name of the number profile
Prefix of Caller Number	The prefix of the calling number. It supports regular expression.
Prefix of Called Number	The prefix of the called number. It supports regular expression.
Length	The length of the calling number or called number. For example, ： 4|6|7 means the calling number or called number must be 4 digits, 6 digits or 7 digits except the prefix.

4.6.5 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.

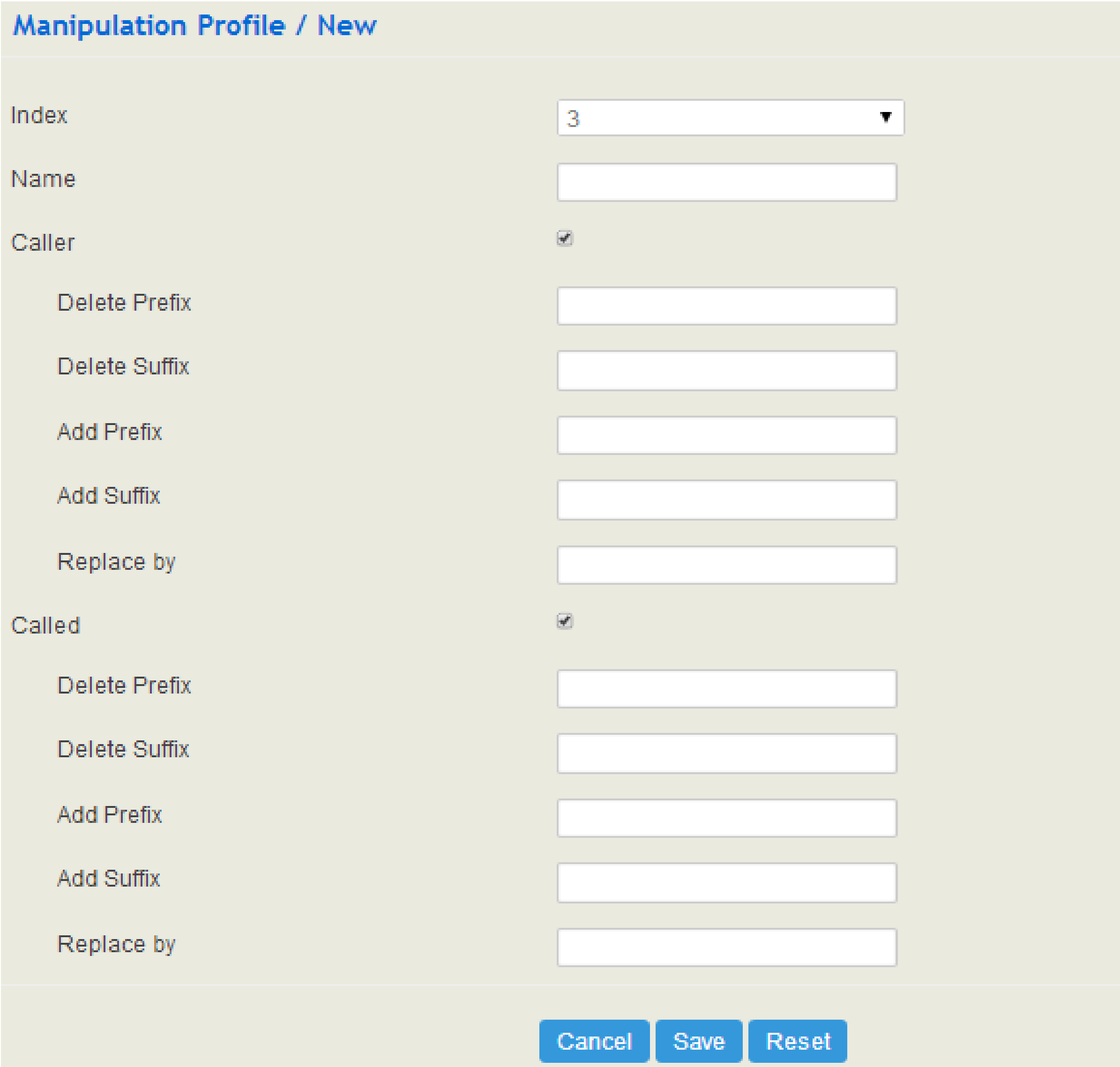
Click the **New** button, and you will see the following interface:



Parameter	Explanation
Name	The name of the time profile
Date Period	Choose a start date and an end date : Add a date period : Delete a date period
Weekday	Choose a weekday
Time Period	Choose start time and end time

4.6.6 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.



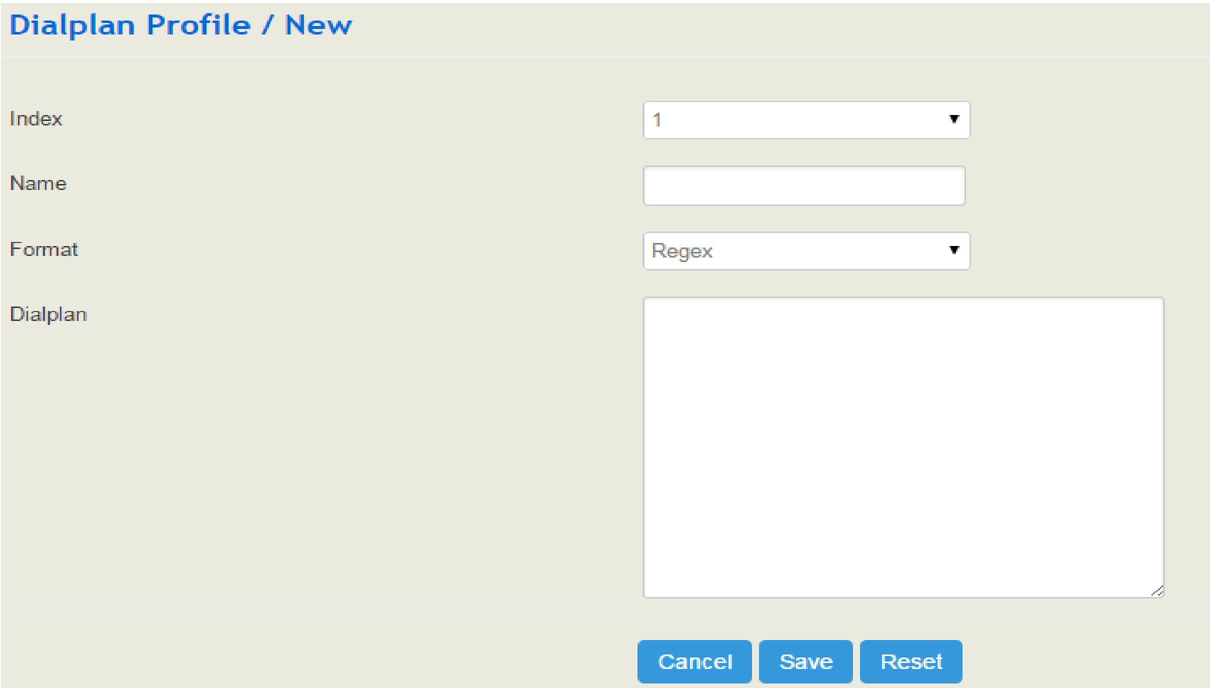
Parameter	Explanation
Delete Prefix	The number of digits that are deleted from the left of the caller number or the calling number
Delete Suffix	The number of digits that are deleted from the right of the caller number or the calling number
Add Prefix	The prefix added to the caller number or the calling 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.

4.6.7 Dialplan

Dialplan is used for number dialing of calls through FXS and FXO ports.



Regex (Regular Expression) Syntax

Explanation of frequently-used metacharacters in Regex:

^	Matches the starting position in a string. For example, ^134.
$	Matches the ending position of a string. For example, 2$.
|	Separates alternate possibilities. For example 2|3|4.
/	Quote the next metacharacter.
[ ]	Matches a single character that is contained within the bracket. For example, [123] matches 1, 2, or 3. [0-9] specifies a range which matches any lowercase letter from "0" to "9".
[^]	Matches any one character except those enclosed in [ ]. For example, [^9].

.	Matches a single character of any value, except end of line.
?	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.

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.

Digit Map 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	|	Separated expressions or DTMF symbols.
Subrange	-	Two digits separated by hyphen (-) which matches any digit between and including the two.
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

Example:

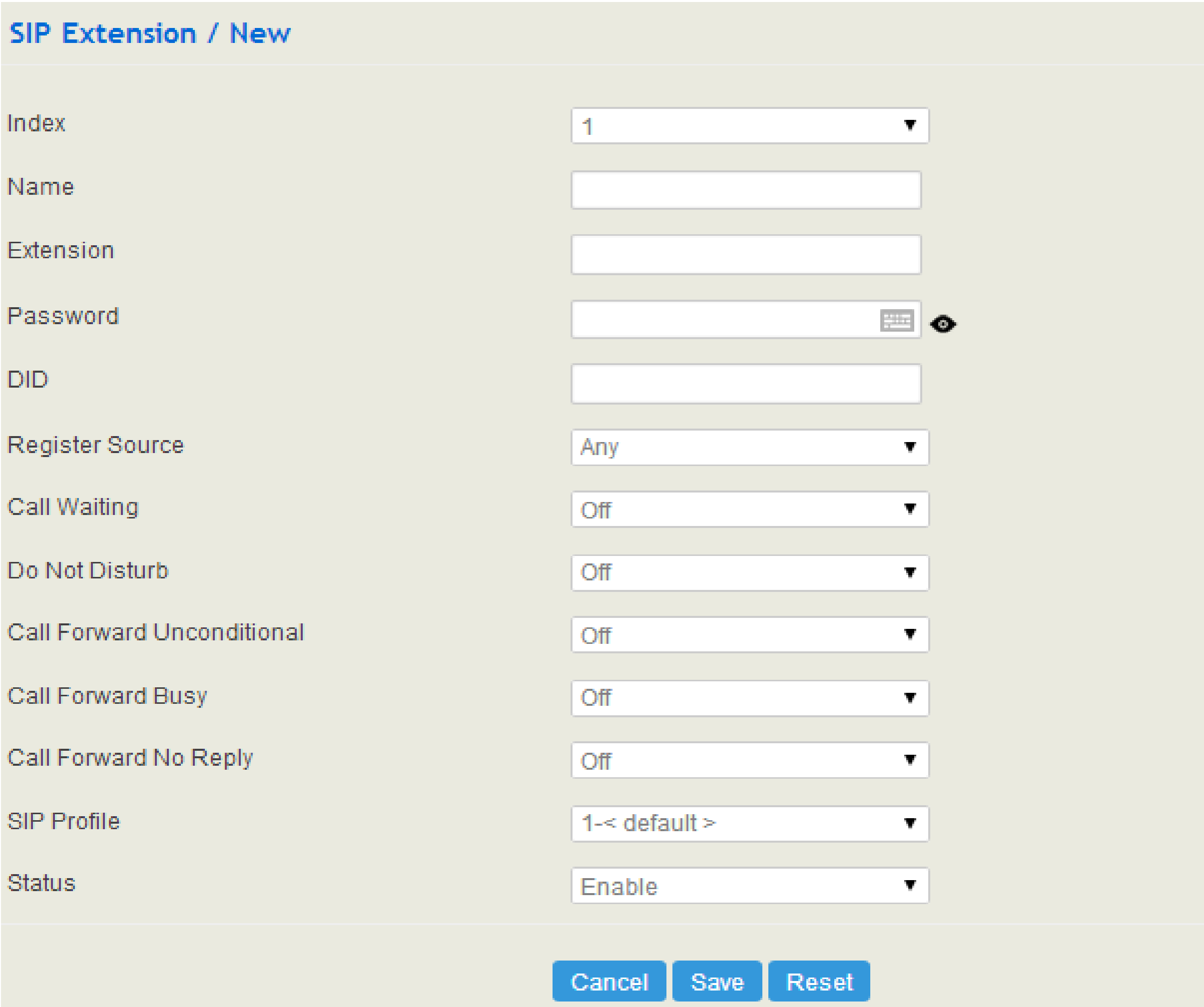
(13 | 15 | 18)xxxxxxxxx	Matches the phone numbers with stating digits as 13, 15 or 18 and the left nine digits as any of 0 to 9.

4.7 Extension

4.7.1 SIP Extension

On the **Extension → SIP** interface, you can configure the SIP accounts registered in UC2000 by SIP clients.

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

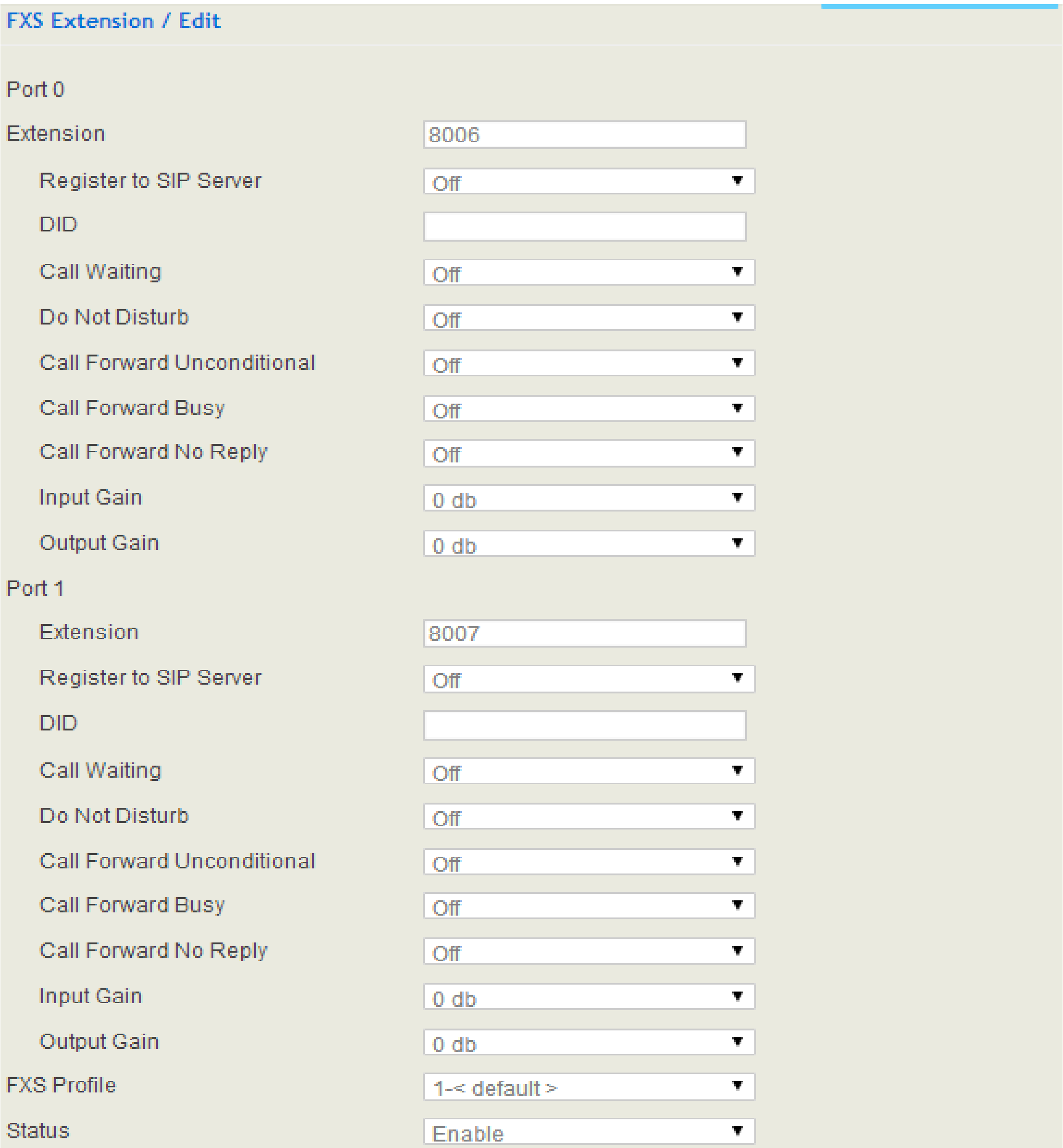


Parameter	Explanation
Name	The name of the SIP extension
Extension	The SIP account of the extension registered in UC2000 by a SIP client
Password	The password of the SIP account registered in UC2000 by a SIP client
DID	Direct Inward Dialing; if the called number is same with DID, the call will be directly forwarded to the extension, rather than choosing a route.

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 is allowed to register the SIP account of this extension.
SIP Profile	The SIP profile that is selected for the extension
Status	If it is enabled, the SIP account can be registered; Otherwise the SIP account cannot be registered.

4.7.2 FXS Extension

On the **Extension → FXS** interface, you can configure data for extensions of FXS userboards. If no FXS userboard is inserted into UC2000, there will no data on the interface.



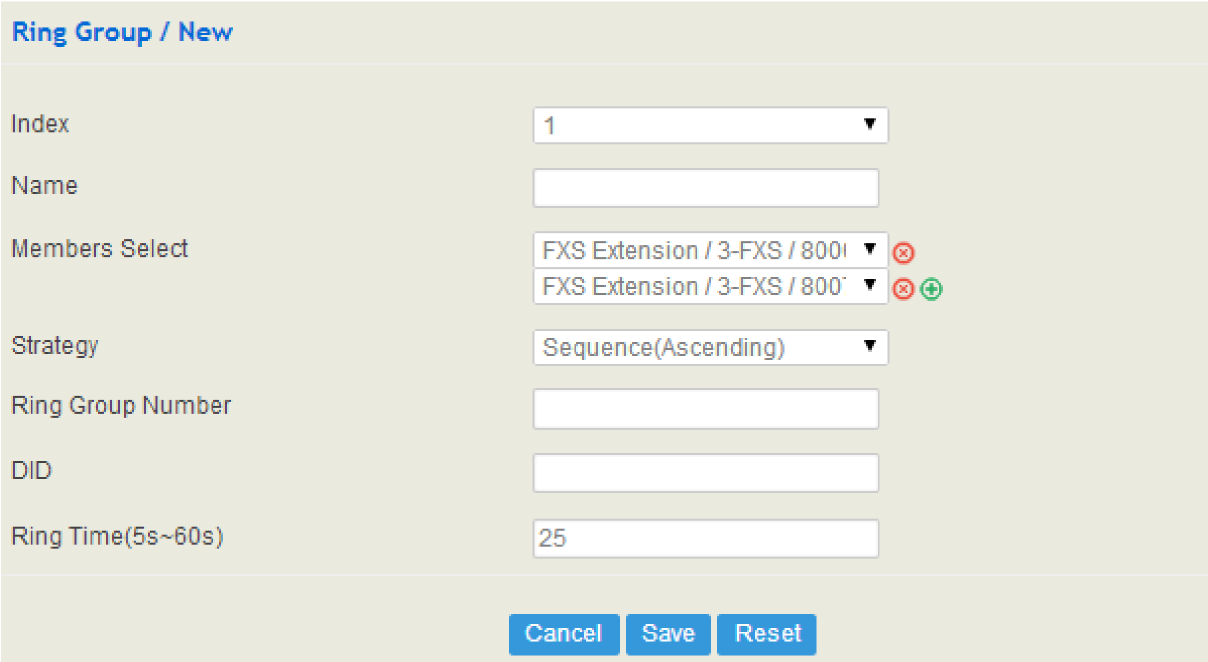
Parameter	Explanation
Extension	The extension account of the FXS port
Register to SIP Server	If it is enabled, the FXS extension account will be registered to a SIP server. Master Server: The address and port of the master SIP server; make reference to Trunk → SIP; Slave Server: The address and port of the slave SIP server; make reference to Trunk → SIP. The slave server will be in use when it is successfully registered but the master server fails to be registered Auth Username: the auth username of the SIP server; it is optional to fill in. Password: the password of the SIP server; it is optional to fill in.
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.
Input Gain	The receiving gain of the FXS port
Output Gain	The sending gain of the FXS port
FXS Profile	The FXS profile that is selected for the FXS extension; make reference to Profile →FXS/FXO
Status	If it is enabled, the FXS port is available; if it is disable, the FXS port is unavailable

Note:

- SIP server is a main component in a VoIP network and is responsible for establishing conversations for SIP calls. IPPBX, softswitch, UC2000 and UC100 can serve as a SIP server.
- SIP servers based on Linux include OpenSER, sipXecx, VoS and Mera.
- SIP servers based on Windows include miniSipServer, Brekeke and VoIPswitch.
- Softswitches of Cisco, Huawei and ZTE can also serve as SIP server.

4.7.3 Ring Group

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



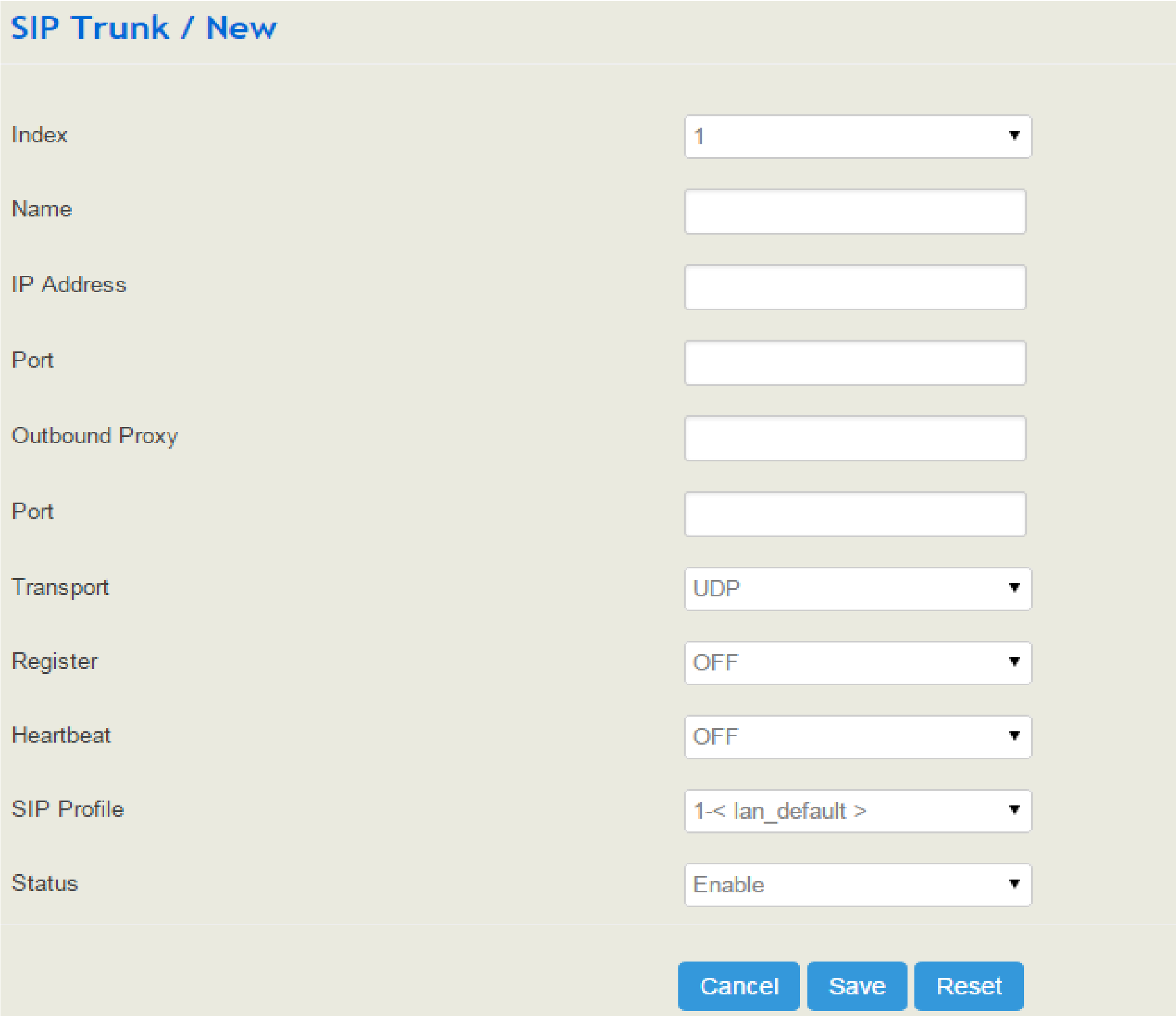
Parameter	Explanation
Name	The name of the ring group
Members Select	The extensions that are selected. A same extension cannot be selected for two ring groups. Before selecting extensions, you need to create extensions on the **Extension → SIP** interface or the **Extension → FXS** interface first. : Add an extension : Delete an extension
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
DID	Same with Ring Group Number; it is optional to fill in
Ring Time	The duration of ringing

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

4.8 Trunk

4.8.1 SIP Trunk

SIP trunk can realize the connection between UC2000 and IPPBX or SIP servers.



Parameter	Explanation
Name	The name of the SIP trunk
IP Address	The IP address or domain name of the peer devices or servers
Port	The SIP listening port of the peer devices or servers
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.
Heartbeat	If heartbeat in on, heartbeat messages (options) will be sent to examine the connection with servers. The default value is ‘Off’. Heartbeat period: the interval of two options

SIP Profile	The SIP profile used by the SIP Trunk; make reference to Profile → SIP.
Status	If it is enabled, it means the SIP Trunk is available; otherwise, the SIP trunk is unavailable. The default value is Enable.

Note:
If UC2000 is regarded as a terminal and intends to register to a server, you need to configure a SIP trunk connecting UC2000 and the server, and then enable register for the SIP trunk.

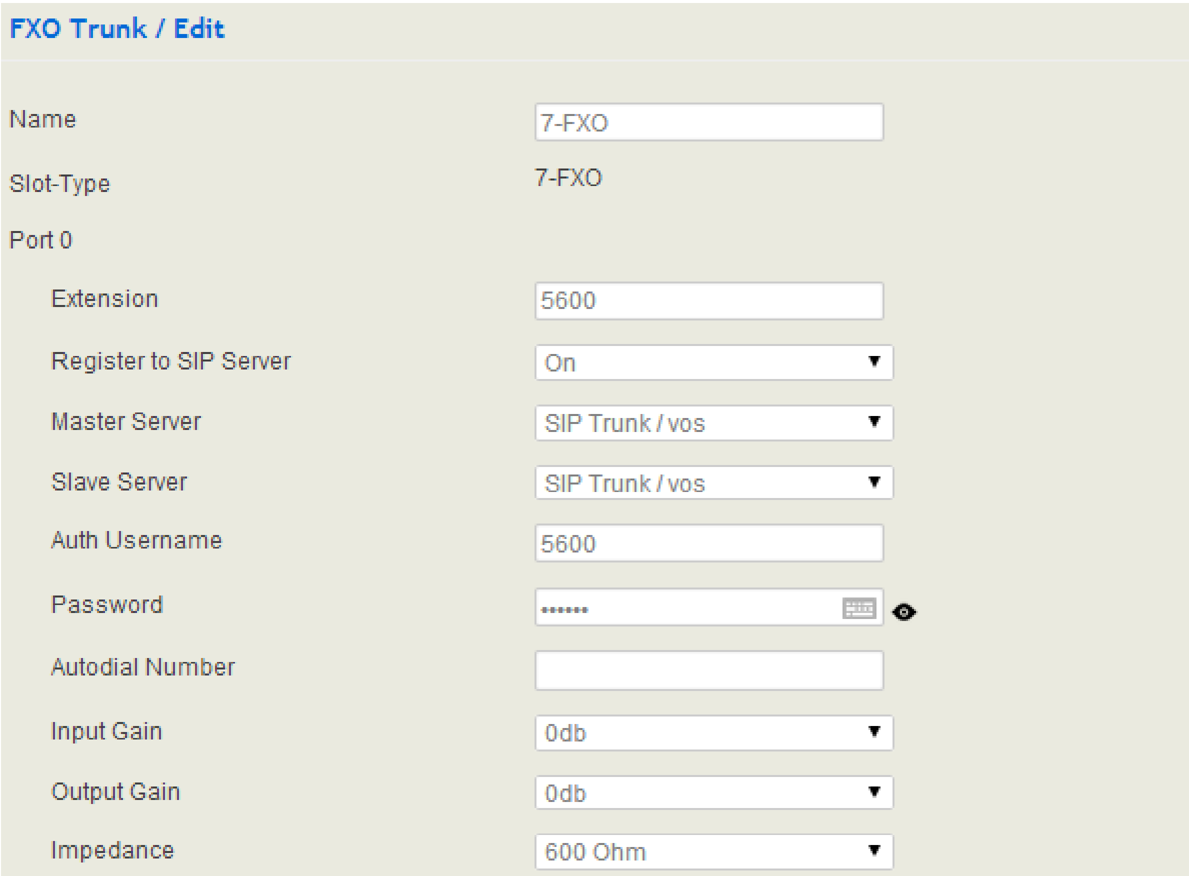
If all ports of UC2000 intend to register to a server, you need to configure a SIP trunk connecting UC2000 and the server, then enable register for the ports and designate the SIP trunk to them.

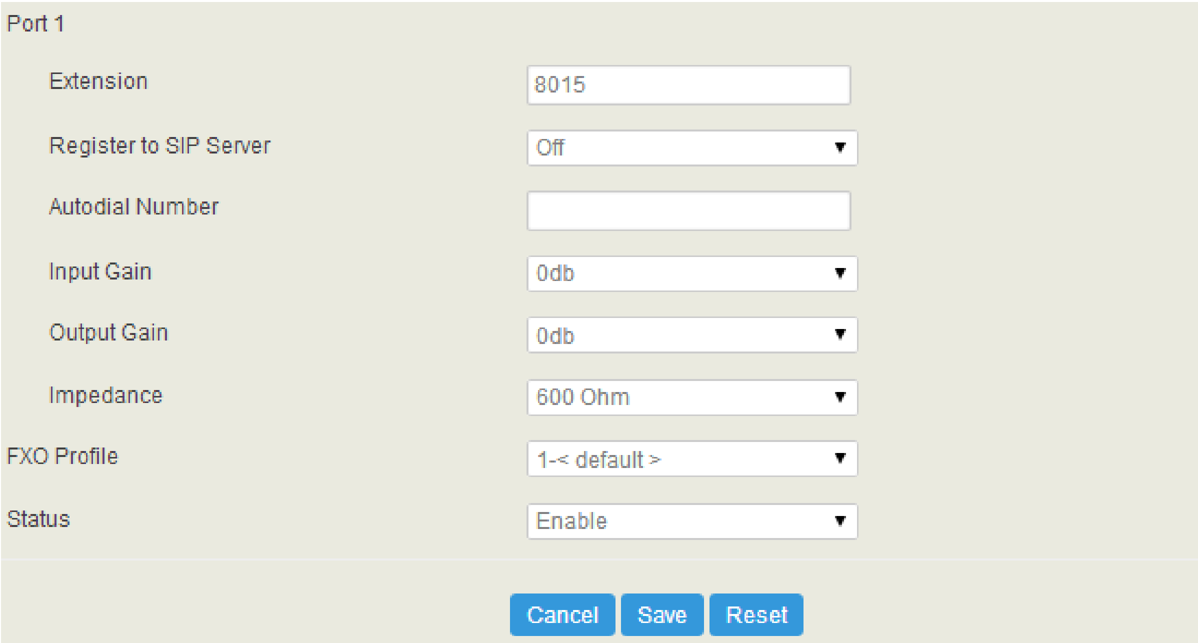
4.8.2 FXO Trunk

FXO Trunk interconnects the PSTN with UC2000. Calls from the PSTN can come into UC2000 and calls can go out from UC2000 to search telephone numbers under the PSTN.

Different from the FXO ports of other gateways, the FXO ports of UC2000 only allows one-time dialing, which means called numbers needs to be dialed directly for calls that go out from the FXO ports.

Click , and you will see the following interface:



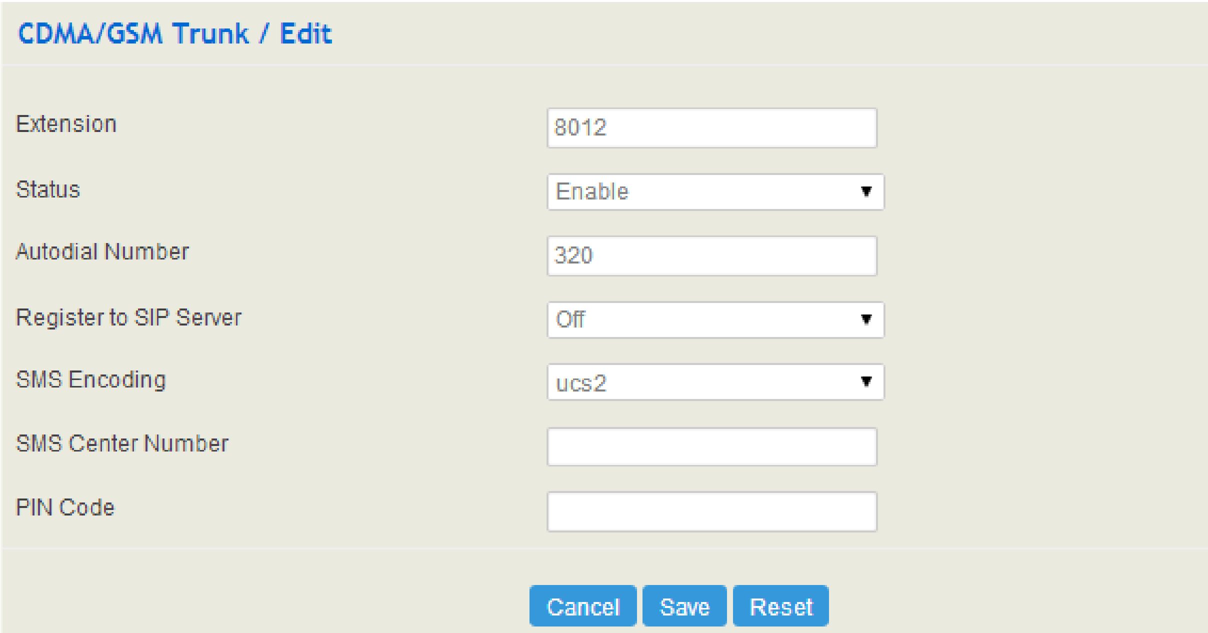


Parameter	**Explanation**
Extension	The extension account of the FXO port, which is used to register.
Register to SIP Server	If it is enabled, the FXO extension account will be registered to a SIP server. Master Server: The address and port of the master SIP server; make reference to Trunk → SIP; Slave Server: The address and port of the slave SIP server; make reference to Trunk → SIP. The slave server will be in use when it is successfully registered but the master server fails to be registered Auth Username: the auth username of the SIP server; it is optional to fill in. Password: the password of the SIP server; it is optional to fill in.
Autodial Number	The autodial number for incoming calls through FXO port
Input Gain	The receiving gain of the FXO port
Output Gain	The sending gain of the FXO port
Impedance	The impedance matched with analog phones
FXO Profile	The FXO profile that is selected for the FXO extension; make reference to Profile →FXS/FXO
Status	If it is enabled, the FXO port is available; if it is disable, the FXO port is unavailable

4.8.3 GSM/CDMA Trunk

GSM/CDMA trunk interconnects the GSM/CDMA wireless network with UC2000. Calls from the GSM/CDMA wireless network can come into UC2000 and calls can go out from UC2000 to search mobile numbers under the GSM/CDMA wireless network.

Click , and you will see the following interface:

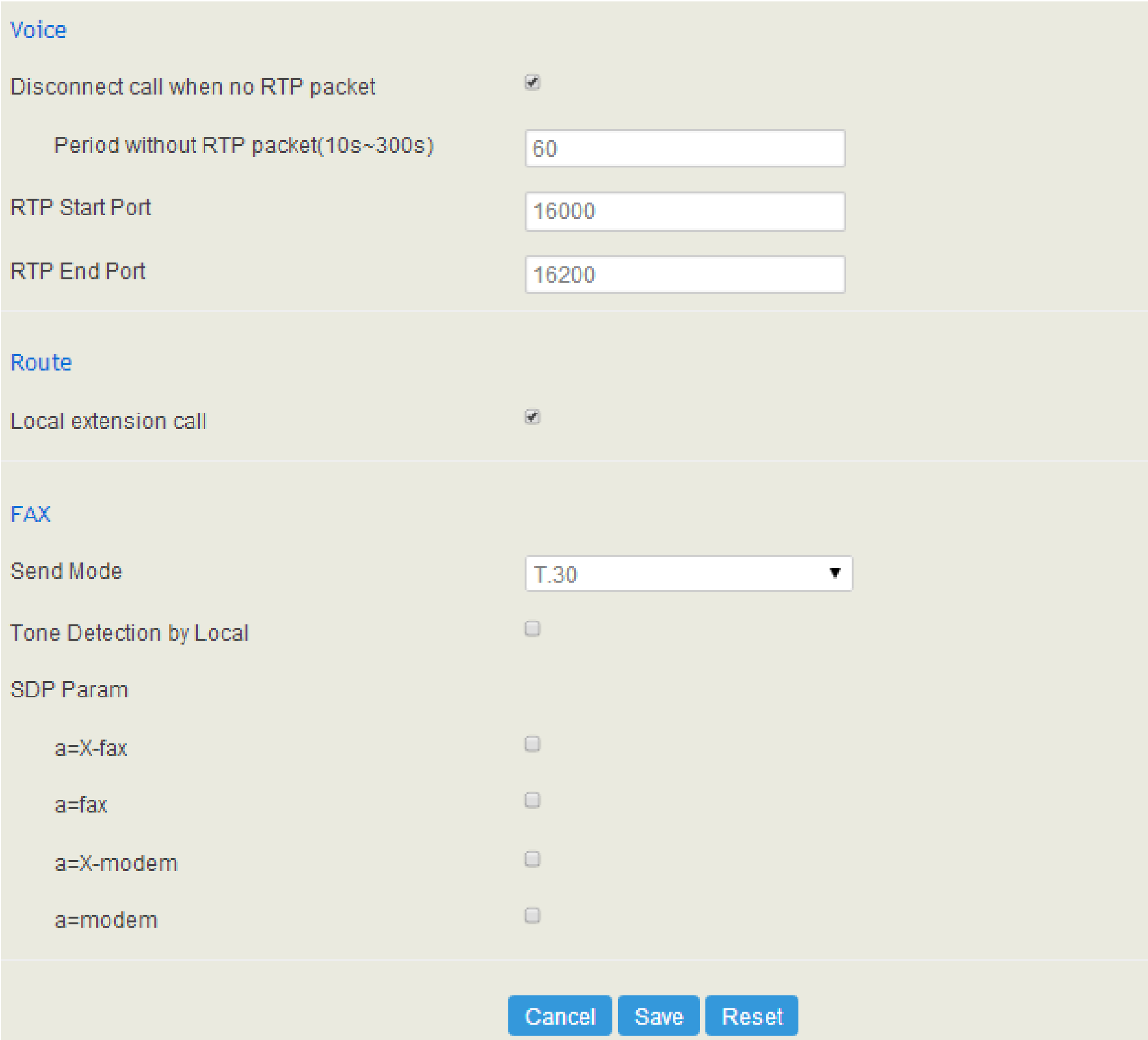


Parameter	Explanation
Extension	The extension account of the GSM/CDMA port, which is used to register
Status	If it is enabled, it means the GSM/CDMA trunk is available; otherwise, the GSM/CDMA trunk is unavailable
Autodial Number	The autodial number for incoming calls through GSM/CDMA port
Register to SIP Server	Whether to register the GSM/CDMA extension account to SIP server
SMS Encoding	uc s2 or 7bit
SMS Center Number	The SMS center number of SIM card provider
Pin Code	The Pin code of SIM card

4.9 Call Control

This section is to configure routes or route groups for incoming and outgoing calls through UC2000, as well as IVR, SMS, fax and call-related security.

4.9.1 Setting



Parameter	**Explanation**
Disconnect call when no RTP packet	If it is enabled, and no RTP packets are received within the set time, calls will be disconnected
RTP Start Port	The start port of RTP packets
RTP End Port	The end port of RTP packets
Local extension call	If it is enabled, calls between local extensions do not need routes.
Fax Mode	T38 or T30（Pass-through）
Tone Detection by Local	If it is enabled, UC2000 will detect fax tones automatically during a call and the call will be switched into fax mode after a fax tone is detected.
SDP Param ‘a=X-fax’	Attribute parameter ‘a=X-fax’ is carried in SDP

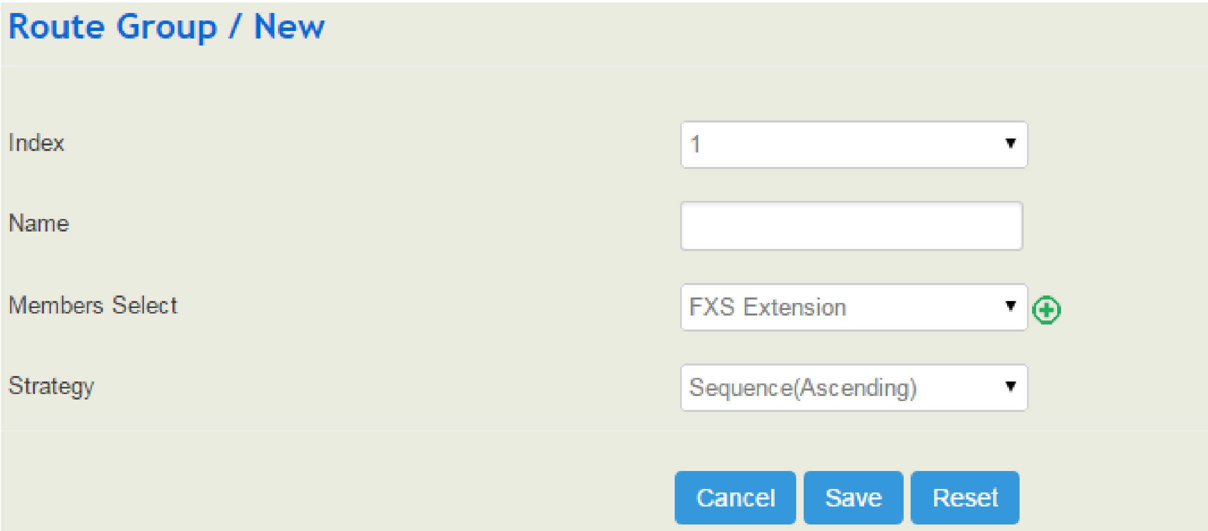
SDP Param ‘a=fax’	Attribute parameter ‘a=fax’ is carried in SDP
SDP Param ‘a=X-modem’	Attribute parameter ‘a=X-modem’ is carried in SDP
SDP Param ‘a=modem’	Attribute parameter ‘a=modem’ is carried in SDP

4.9.2 Route Group

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

Click 

, and you will see the following interface:

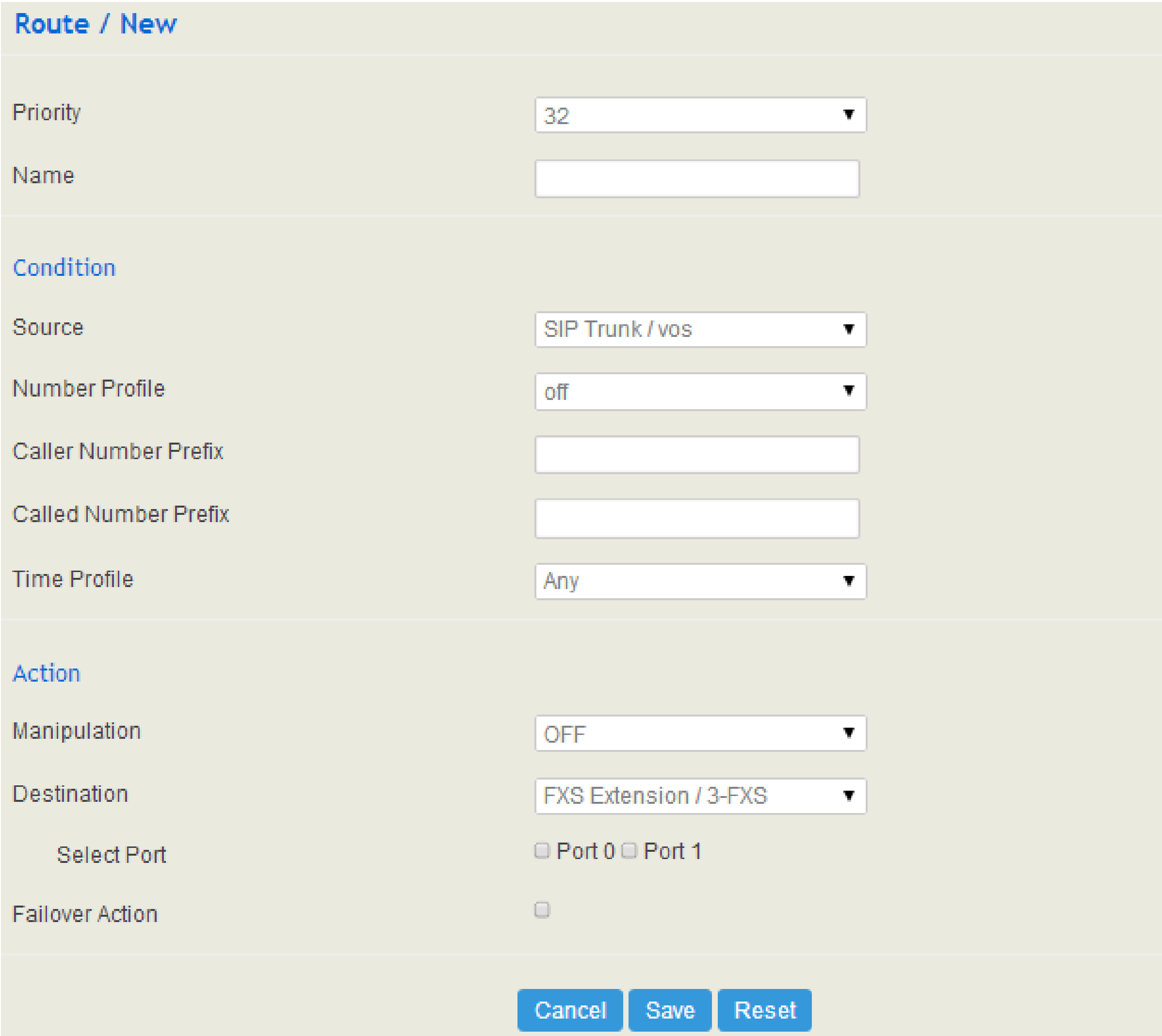


Parameter	Explanation
Name	The name of the route group
Members Select	Select extension(s) or trunk(s) in the drop-down box as the destination of the route group
Strategy	The strategies for choosing which trunk or extension as the destination route, including Sequence (Ascending), Sequence (Cyclic Ascending), Simultaneous and Random
	Add an extension or a trunk to the route group
	Delete an extension or a trunk from the route group

4.9.3 Route

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

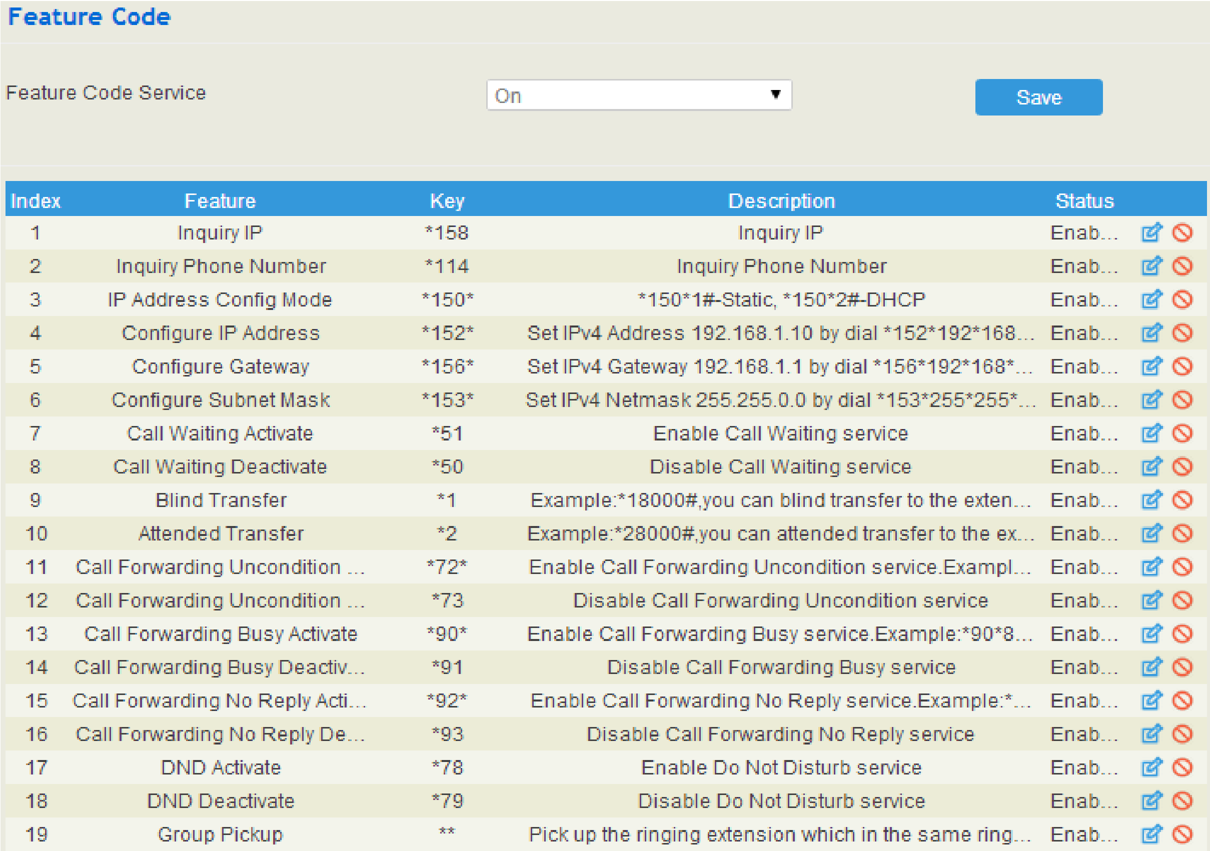
Click New, and you will see the following interface:



Parameter	Explanation
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 an extension, a trunk, a customized source or any source.
Number Profile	The profile of the caller number and the called number; please make reference to Profile → Number The default value is ‘Off’. Note: it is incompatible with 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 Profile → Time
Action	Include manipulating number and send call to destination
Manipulation	If it is on, the caller number of the route will be manipulated; make reference to Profile→ Manipulation
Destination	The destination of the route
Failover Action	The processing action when a call through this route fails Condition: If busy or timeout is selected, only the failed calls due to busyness or timeout will be processed. If both are not selected, all failed calls will be processed. Other Condition Code: the code of other conditions; please separate codes with ‘,’ Manipulation: If it is on, the caller number of the route will be manipulated Destination: the destination (userboard and extension) of the route Select Port: select a port for the route, since there are two ports in a userboard.

4.9.4 Feature Code



Feature Code

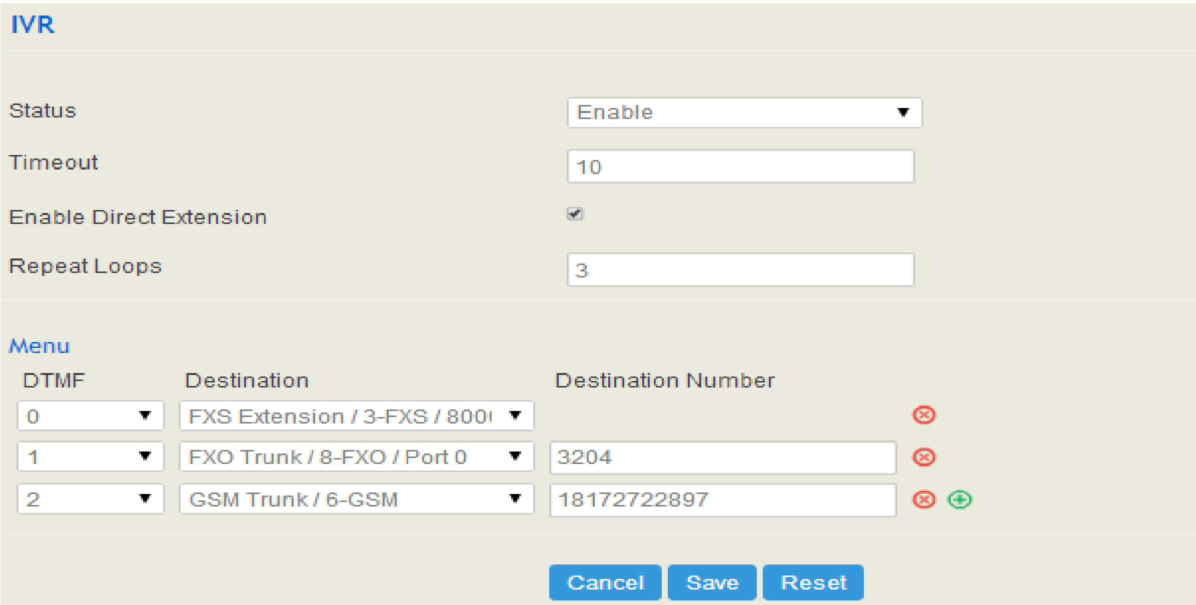
Feature Code Service: On — Save

Index	Feature	Key	Description	Status
1	Inquiry IP	*158	Inquiry IP	Enab...
2	Inquiry Phone Number	*114	Inquiry Phone Number	Enab...
3	IP Address Config Mode	*150*	*150*1#-Static, *150*2#-DHCP	Enab...
4	Configure IP Address	*152*	Set IPv4 Address 192.168.1.10 by dial *152*192*168...	Enab...
5	Configure Gateway	*156*	Set IPv4 Gateway 192.168.1.1 by dial *156*192*168*...	Enab...
6	Configure Subnet Mask	*153*	Set IPv4 Netmask 255.255.0.0 by dial *153*255*255*...	Enab...
7	Call Waiting Activate	*51	Enable Call Waiting service	Enab...
8	Call Waiting Deactivate	*50	Disable Call Waiting service	Enab...
9	Blind Transfer	*1	Example:*18000#,you can blind transfer to the exten...	Enab...
10	Attended Transfer	*2	Example:*28000#,you can attended transfer to the ex...	Enab...
11	Call Forwarding Uncondition ...	*72*	Enable Call Forwarding Uncondition service.Exampl...	Enab...
12	Call Forwarding Uncondition ...	*73	Disable Call Forwarding Uncondition service	Enab...
13	Call Forwarding Busy Activate	*90*	Enable Call Forwarding Busy service.Example:*90*8...	Enab...
14	Call Forwarding Busy Deactiv...	*91	Disable Call Forwarding Busy service	Enab...
15	Call Forwarding No Reply Acti...	*92*	Enable Call Forwarding No Reply service.Example:*...	Enab...
16	Call Forwarding No Reply De...	*93	Disable Call Forwarding No Reply service	Enab...
17	DND Activate	*78	Enable Do Not Disturb service	Enab...
18	DND Deactivate	*79	Disable Do Not Disturb service	Enab...
19	Group Pickup	**	Pick up the ringing extension which in the same ring...	Enab...

Note: All feature codes are enabled by default.

4.9.5 IVR

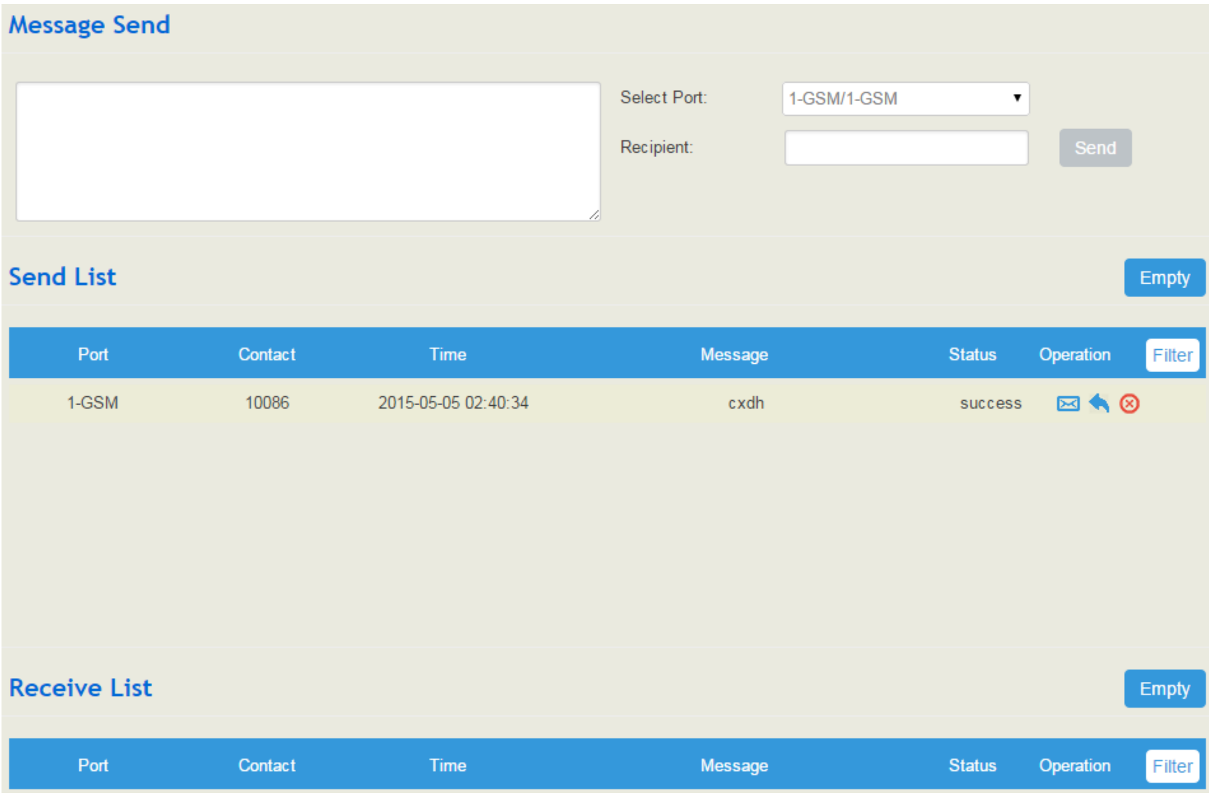
On the **IVR** interface, you can carry out specific configurations for the IVR which has been uploaded from the **System → Voice** interface.



Parameter	Explanation
Status	If it is disabled, the IVR cannot be seen in the destination of route.
Timeout	If it is set as '10', it means if no DTMF tone is received during 10 seconds, the IVR will be played repeatedly or the call will be hanged up. The default value is 20 seconds.
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.
Menu	It is the menu of quick-dial numbers for extensions or trunks. If it is a quick-dial for trunks, you need to configure the called number. Quick-dial numbers are 0 to 9.

4.9.6 SMS

If an SIM card has been inserted into a GSM/CDMA userboard, you can send or receive SMS on the **Call Control →SMS** interface.



Send Message

Enter contents into the box on the left, and then input the number of recipient . Click **Send** in the last.

Note: If there are mutilple recipients , use | to separate them, for example, 13151103146|18954405566.

Receive Message

All SMS received by UC2000 are displayed on the Receive List.

Read Message

Click on the Receive List to read SMS contents.

Reply Message

Click , and then enter SMS contents in the box on the left. Click Send in the last.

Delete Message

Click to delete an SMS.

Note: Group sending of message is not allowed.

➔ **End**