



UC100 User Manual v1.0



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1 Product Description

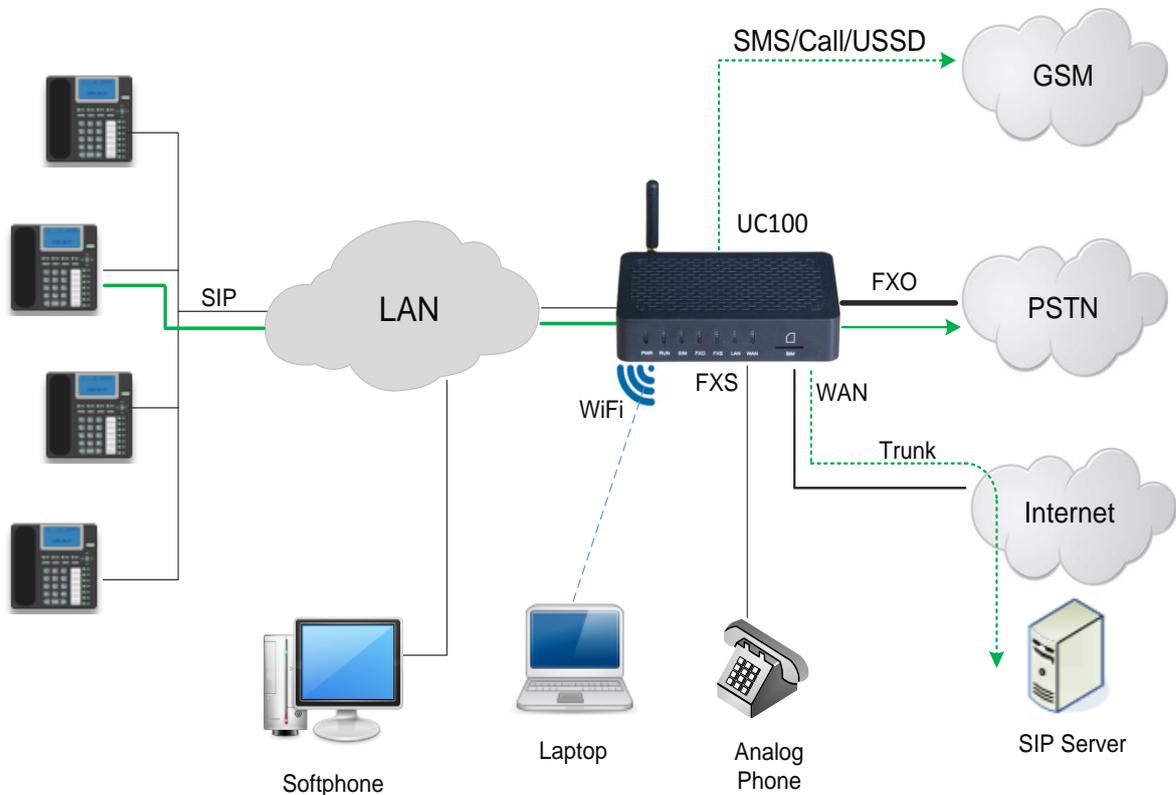
1.1 Overview

UC100 is a multi-functional and all-in-one gateway, which integrates voice service and data service. It provides three voice ports (including GSM, FXS and FXO), offering seamless connectivity to VoIP Network, PLMN and PSTN. Based on SIP, it not only can interact with VoIP network, but also supports four types of GSM frequency ranges, thus meeting the worldwide requirements about the mobile network. UC100 supports WiFi and has high-speed data-handling capacity, allowing users to enjoy high-speed internet surfing through WiFi or the LAN port.

With VPN transparent transfer function, UC100 is ideally suitable for personal use. Meanwhile, it is perfect for small and micro enterprises, providing high-speed internet access, good voice service as well as message service.

1.2 Application Scenario

UC100 provides high-quality and cost-effective VoIP solution. Its application scenario is shown as follows:



1.3 Product Appearance

1.3.1 Images of UC100



1.3.2 Description of Indicators

Indicator	Status	Description
PWR	Flash slowly	The system is initialized successfully and is in normal running.
	On green	The system is being initialized.
	On dull	There is no power supply or power supply is abnormal.

WiFi	Flash quickly	WiFi is in normal running
	On dull	WiFi is turned off.
GSM	Flash slowly	SIM card has not yet been inserted or SIM card is not registered.
	Flash quickly	SIM card is successfully registered.
	On dull	Fault occurs in GSM module
FXS/FXO	Flash slowly	FXS/FXO is idle or no off-hook is detected.
	On green	FXS/FXO is in off-hook status.
	On dull	Fault occurs in FXS/FXO module
WAN/LAN	Flash quickly	Network is successfully connected.
	On dull	Network does not work or network cable is not connected.

1.4 Features & Functions

1.4.1 Key Features

- FXS/FXO/GSM interface on a single device/gateway
- Send/receive calls from PSTN/PLMN via FXO/GSM
- Flexible dial plan, via time, numbers, source IP etc.
- IVR customization
- High speed NAT forwarding, support WIFI hotspot
- Built-in SIP server, support up to 8 SIP Extensions
- User-friendly web interface, multiple management ways

1.4.2 Software Features

- Ring Group
- Routing Groups
- Caller/Called Number Manipulation
- Routing Base on Time Period
- Routing Base on Caller/Called Prefixes
- Routing Base on Source Trunks
- Dial Rules

- Failover Routing
- FXO Impedance Auto Match
- IVR Customization
- Auto Attendant Function
- CDRs

1.4.3 Voice Capabilities

- VoIP Protocols: SIP over UDP/TCP/TLS,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)
- 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

1.4.4 Supplementary Services

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

1.4.5 Physical Interfaces

- FXO Port: 1
- FXS Port: 1
- SIM Slot: 1
- Ethernet Interfaces: 1WAN&1LAN 10/100 Base-T RJ45
- WIFI: 2.4GHz , 802.11n

1.4.6 FXS

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

1.4.7 FXO

- Connector: RJ11
- Caller ID: FSK, DTMF
- Polarity Reversal
- Answer Delay
- Busy Tone Detection
- No Current Detection

1.4.8 Mobile

- GSM: 850/900/1800/1900MHz
- SIM/UIM: 1 SIM/UIM per Channel
- SIM Card: 1.8V, 3.0V
- Antenna: 3.0dB, SMA Interface
- SMS/USSD
- Bulk SMS
- SMS Code/Decode: ASCII, Unicode
- IMEI/PIN Code Management

1.4.9 Hardware Specifications & Environment

- Power Supply: 12VDC, 1A
- Power Consumption: 10W
- Operating Temperature. 0 °C ~ 45 °C
- Storage Temperature: -20 °C ~80 °C
- Humidity: 10%-90%, Non-Condensing
- Dimensions (W/D/H): 126×75×25mm
- Unit Weight: 0.7kg

2 Quick Installation

2.1 Precautions for Installation

The precautions for installing UC100 include:

- The adapter of UC100 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 UC100 is well-connected;
- If you want UC100 to communicate with the GSM network, please insert an SIM card.

Note: Please check whether power supply is up to the above requirement; otherwise, UC100 and its power adapter may be damaged.

2.2 Installation Procedures

- Connect the antenna to the UC100 device
- Connect the power adapter to the power port;
- Connect network cable to the LAN port or the WAN port;
- Connect telephone wire to the FXO port and the FXS port;
- Insert SIM card to the SIM slot.

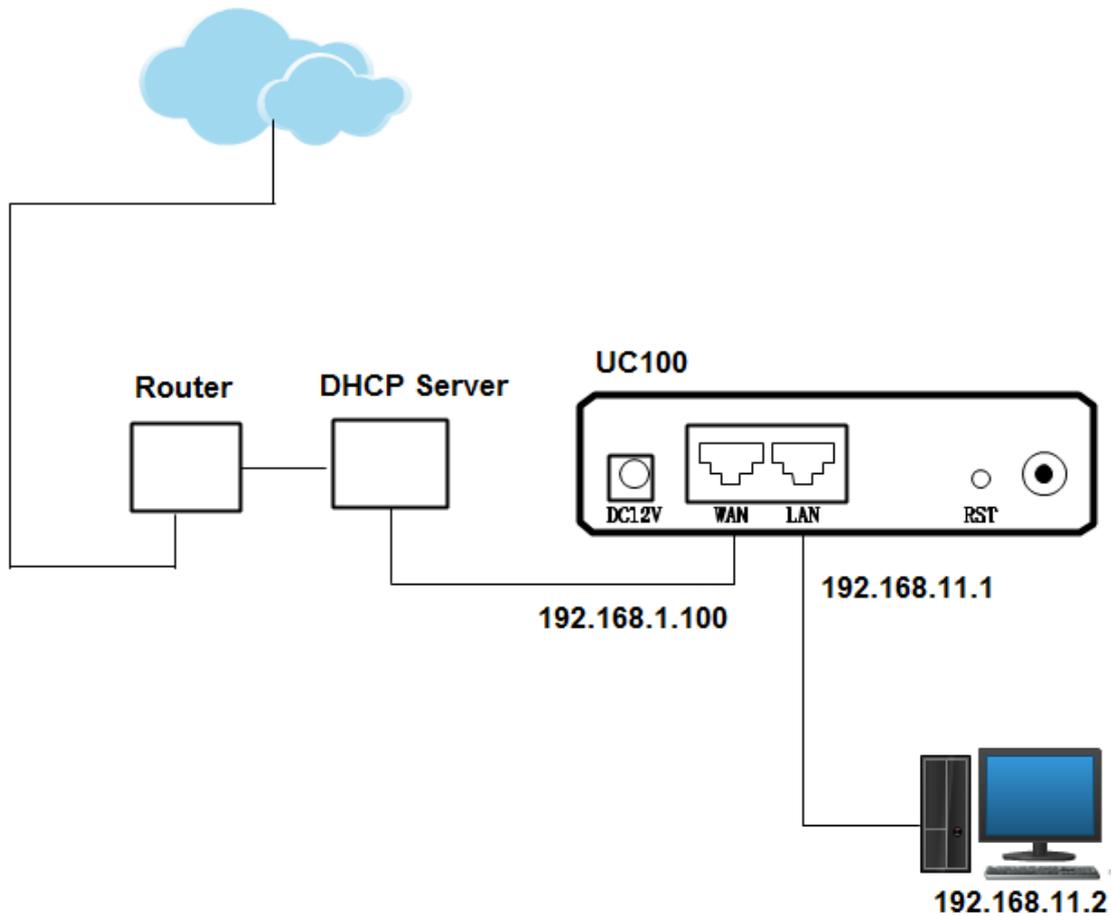


2.3 Network Connection

UC100 works in two modes: route mode and bridge mode. When it is under the route mode, the IP address of WAN port must be different from the IP address of LAN port. But when it is under the bridge mode, the IP address of WAN port and the IP address of LAN port are the same.

2.3.1 Network Connection Diagram under Route Mode

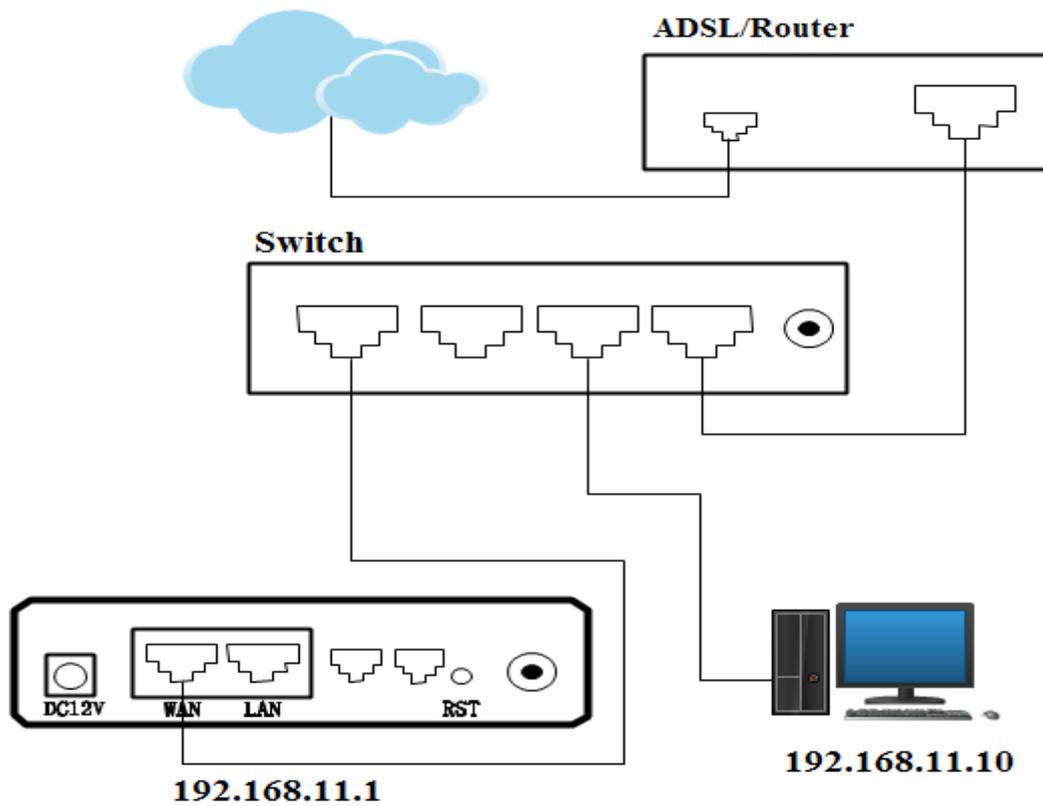
Under the route mode, the default IP address of WAN port is a DHCP IP address, while the default IP address of the LAN port is a static IP address, namely 192.168.11.1.



Note: The IP address of LAN port of UC100 and the IP address of PC are at the same network segment, while the IP address of WAN port is at a different network segment.

2.3.2 Network Connection Diagram under Bridge Mode

Under the Bridge mode, both the default IP address of WAN port and that of LAN port are 192.168.11.1.



Note: The IP address of PC and the IP address of WAN port of UC100 are at the same network segment.

For more details about installation, please make reference to Quick Installation Guide.

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 calling 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 one 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 LAN IP
*159	Dial *159 to inquiry WAN IP
*114	Dial *114 to inquiry phone number
157	Dial *157*0 to set route mode; dial *157*1 to set bride mode
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 *152*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
*111	Dial *111 to restart the UC100 device
*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 'No Disturbing' service
*79	Disable the 'No Disturbing' 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

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

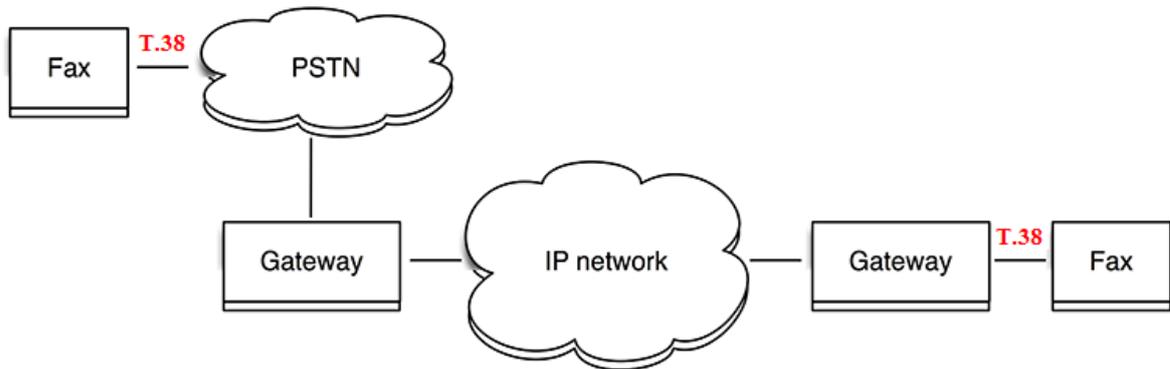
3.6 Send or Receive Fax

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. Fax data are transmitted. Under the T.38 mode, fax traffic is carried in T.38 packages.



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

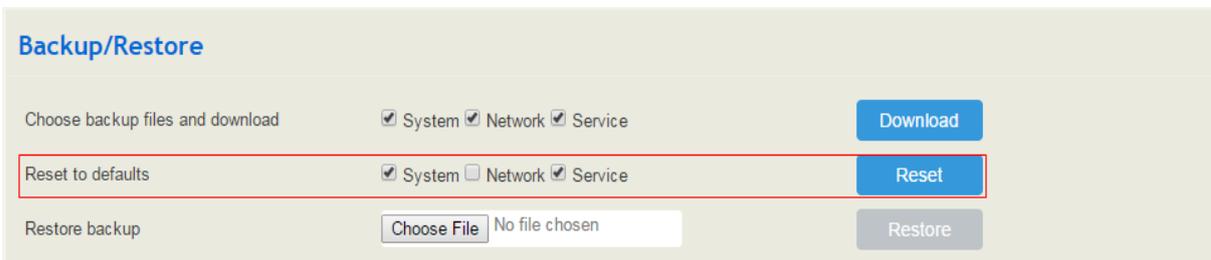
Press the **RST** button of UC100 for 3 seconds to 6 seconds, the IP address, username and password of the device will be restored to factory defaults.

Press the **RST** button of UC100 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 UC100 to default settings, you can press the **RST** button of UC100 for more than 6 seconds or you can configure it on the Web interface.

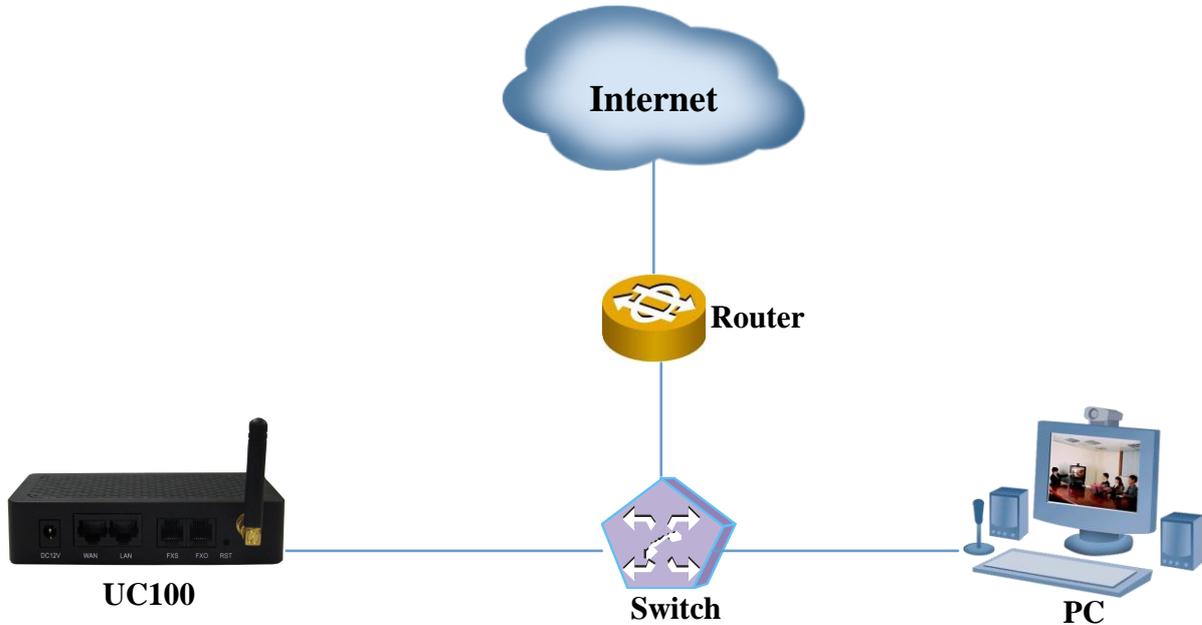
Click **System** → **Backup/Restore/Upgrade** on the Web of UC100, and select the parts (system, network or network) that need to be restored to defaults, click **Reset**, restart the UC100 device, and the selected parts will be restored to defaults.



4 Configurations on Web Interface

4.1 How to Log in Web Interface

Connect UC100 to the network according to the following network topology, and dial *158 to query the IP address of the UC100 device.



4.1.1 Preparations for Login

Modify the IP address of the PC to make it at the same network segment with the UC100 device, since the default IP address of the UC100 device is 192.168.11.1.

Check the connectivity between the PC and the UC100. Click **Start** → **Run** of PC and enter cmd to execute 'ping 192.168.11.1' to check whether the IP address of the UC100 runs normally.

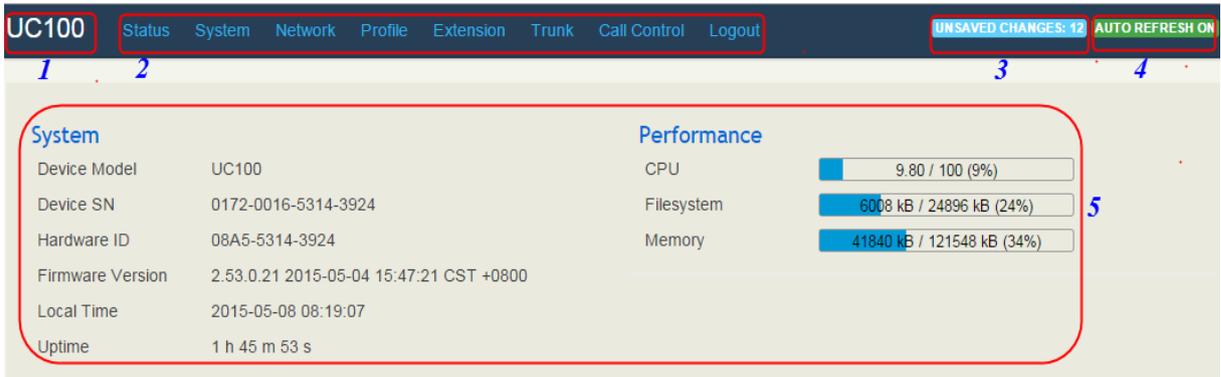
4.1.2 Log in Web Interface

Open a web browser and enter the IP address of the UC100 (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.

The screenshot shows a login form with three main components: a username field containing 'admin', a password field with masked characters '.....', and a blue 'Login' button.

Then the following interface will be displayed.

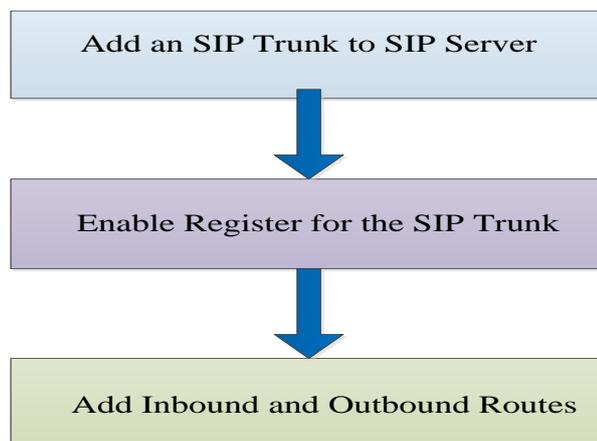


Index	Item	Description
1	UC100	The name of the gateway; it can be edited on the System → Setting interface
2	Menu Bar	The menu bar of UC100
3	Unsaved Changes	All configurations or modifications should be saved. Click the button, and you can see a log of all changes. Changes won't take effect until they are saved.
4	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
3	Detailed interface	The detailed configuration interface or display interface

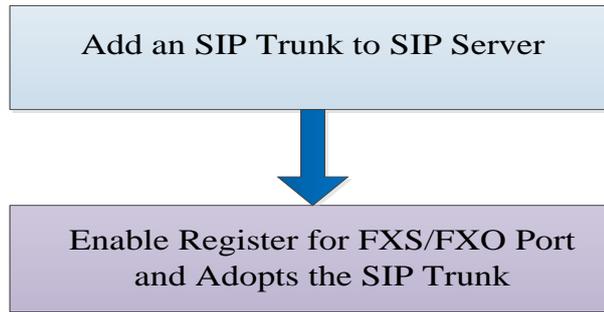
4.2 Configuration Wizard

The following are the common ways to configure the UC100 device.

4.2.1 UC100 regarded as Terminal and Registered to SIP Server



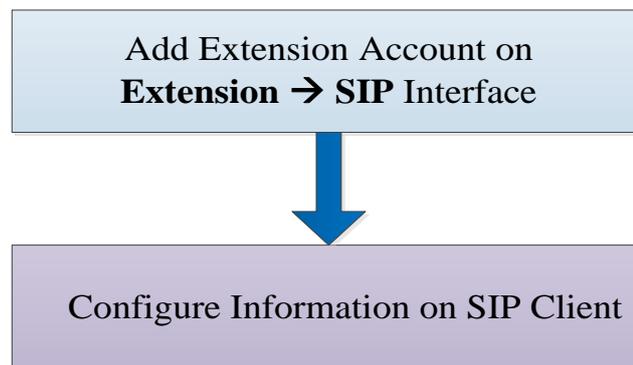
4.2.2 FXS/FXO Port Registered to SIP Server



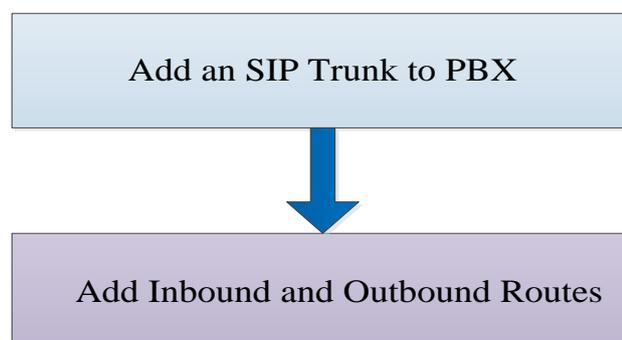
Note: If Register is enabled for FXS/FXO port, calls through FXS/FXO port will take inbound and outbound routes as first priority. For outgoing calls, if outbound route cannot be matched, then the registered SIP trunk will be selected. For incoming calls, if inbound route cannot be matched, then the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

4.2.3 Other SIP Clients registered to UC100

Under this mode, UC100 is regarded as an SIP Server. Create an extension account first on the **Extension** → **SIP** interface of UC100, 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 UC100 Connected to PBX through Trunking



4.3 Status

4.3.1 Overview

Log in the Web interface of UC100, click **Status** → **Overview**, and the following interface will be displayed. On the interface, information about the system, performance, WAN network, LAN network, WiFi and DHCP server is shown.

The screenshot displays the UC100 Status Overview page. The navigation bar includes 'uc100', 'Status', 'System', 'Network', 'Profile', 'Extension', 'Trunk', 'Call Control', and 'Logout'. There are buttons for 'UNSAVED CHANGES' and 'AUTO REFRESH ON'. The main content is organized into several sections:

- System:** Device Model (UC100), Device SN (0172-0016-5324-3924), Hardware ID (08A5-5324-3924), Firmware Version (2.53.0.20 2015-04-24 22:08:05 CST +0800), Local Time (2015-04-28 16:37:29), Uptime (3 d 17 h 53 m 41 s).
- Performance:** CPU (10.78 / 100 (10%)), Filesystem (6268 kB / 24896 kB (25%)), Memory (34024 kB / 121548 kB (27%)).
- WAN Network:** MAC Address (F8:A0:3D:53:24:25), Type (Static), IP Address (172.16.99.99), Netmask (255.255.0.0), Gateway (172.16.1.8), DNS (8.8.8.8 172.16.1.8), RX / TX (Per Second) (4.26 KB (66 Pkts.) / 289 Bytes (2 Pkts.)), RX / TX (Total) (1.83 GB (15245826 Pkts.) / 177.51 MB (1655269 Pkts.)).
- LAN Network:** MAC Address (F8:A0:3D:53:24:24), Type (Static), IP Address (192.168.11.1), Netmask (255.255.255.0), RX / TX (Per Second) (0 Bytes (0 Pkts.) / 0 Bytes (0 Pkts.)), RX / TX (Total) (0.00 B (0 Pkts.) / 1.64 MB (15411 Pkts.)).
- WiFi Network:** MAC Address (F8:A0:3D:53:24:39), SSID (James_UC100), Channel (1), Encryption (WPA2 PSK (CCMP)), RX / TX (Per Second) (537 Bytes (9 Pkts.) / 24 Bytes (0 Pkts.)), RX / TX (Total) (156.60 MB (1508707 Pkts.) / 766.15 MB (796251 Pkts.)).
- DHCP Server:** Status (Enabled), Start Address (192.168.11.99), End Address (192.168.11.199), Gateway, Expires (12 Hours), DNS.

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.

UC100 Status System Network Profile Extension Trunk Call Control Logout AUTO REFRESH ON

Profile

Index	Name	Listening Addr	State	Current Call	Call In(F/T)	Call Out(F/T)
1	lan_default	192.168.11.1:5060	RUNNING	0	0/0	0/0
2	wan_default	172.16.99.99:5080	RUNNING	0	0/0	0/0

SIP Trunk

Index	Name	Address	Transport	Reg	Heartbeat	Status	Call In(F/T)	Call Out(F/T)	Profile
-------	------	---------	-----------	-----	-----------	--------	--------------	---------------	---------

SIP Extension

Index	Name	Extension	Register Source	Status	Expires	Agent	Profile
-------	------	-----------	-----------------	--------	---------	-------	---------

Belong To	Parameter	Explanation
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. Unregistered

4.3.3 PSTN

Click **Status** → **PSTN**, and the following interface will be displayed. On the interface, information of FXS port, FXO port and GSM port is shown. Green color means available or registered, while red color means abnormal, unregistered or prohibited.

FXS						
State	Config Status	SIP Register Status	Hook State			
READY	OK	Unregistered	ONHOOK			

FXO				
State	Config Status	SIP Register Status	Hook State	Line State
READY	OK	Unregistered	ONHOOK	OFFLINE

GSM						
State	Channel State	SIP Register Status	Signal	Talking State	Call In(F/T)	Call Out(F/T)
READY	FAULT	Unregistered		IDLE	0/0	0/0

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
	SIP Register Status	Registered, Unregistered
	Signal	: No SIM card has been inserted : 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.

Current Call									
Index	Src	Dest	Caller	Called	Start Time	Answer Time	State	Duration	Filter

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.

CDRs Query Param

Start Date: 2015 ▾ 5 ▾ 1 ▾

Caller:

Source: OFF ▾

Min Duration:

End Date: 2015 ▾ 5 ▾ 11 ▾

Called:

Destination: OFF ▾

Max Duration:

CDRs List

Index	Caller	Source	Called	Destination	Start Time	End Time	Duration	Hangup By	Codec	Hangup Cause	Filter
1	8000	FXS	123456	SIP Trunk/lamon...	2015-05-11 14:45:07	05-11 14:45:08	0	Caller	PCMA	Normal Clearing	
2	8000	FXS	123456	SIP Trunk/lamon...	2015-05-11 14:43:26	05-11 14:43:28	0	Caller	PCMA	Normal Clearing	
3	8000	FXS	123456	SIP Trunk/lamon...	2015-05-11 14:43:06	05-11 14:43:13	0	Caller	PCMA	Normal Clearing	
4	8000	FXS	123456	SIP Trunk/lamon...	2015-05-11 14:42:24	05-11 14:42:41	3	Caller	PCMU	Normal Clearing	

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 UC100 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 UC100.

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

Service	
Running Status	
Msg Service	Running
Switch Kernel Service	Running
Log Service	Running
Upgrade Service	Running
Web	Running
SSH	Running
Telnet	Running

4.3.7 About

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

About

Copyright



DINSTAR
鼎信通达

www.dinstar.com

Tel: 86-755-26456664/61919966

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System

Device Model	UC100
Device SN	0172-0016-5324-3924
Hardware ID	08A5-5324-3924
Linux Version	3.14.18
Root Image	2.9
Boot Image	1.5
Firmware Version	2.53.0.20 2015-04-24 22:08:05 CST +0800

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

General

Hostname

Language

Timezone

Local Time 2015-04-29 16:48:49 Sync with browser

CDRs

Log

Service Log Level

Enable Syslog

Time Synchronization

Enable builtin NTP server

NTP server candidates

<input type="text" value="0.pool.ntp.org"/>	⊗
<input type="text" value="1.pool.ntp.org"/>	⊗
<input type="text" value="2.pool.ntp.org"/>	⊗
<input type="text" value="3.pool.ntp.org"/>	⊗ ⊕

Cancel
Save
Reset

Parameter	Explanation
Host Name	The name of the gateway. After it is configured, the name will be displayed on the left of the menu bar.
Language	Auto: the language of UC100 will be automatically adjusted into the language of the web browser. Auto is the default value. English: the language of UC100 is English. Chinese: the language of UC100 is Chinese.
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.
Time Synchronization	If NTP server is enabled, the UC100 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

4.4.2 User Manager

Click **System** → **User Manager**, and you can modify the username name and password for logging in the UC100 device. Factory defaults for username name and password are both admin, so it is advised that you should modify them for security consideration.

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

Password

Current Username

Old Password

New Password

Confirm New Password

4.4.3 Provision

Provision is used to make UC100 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:

Provision Profile

Enable	<input checked="" type="checkbox"/>
Periodic Check	On ▾
Check Interval(s)	3600
URL	<input type="text"/>
Username	<input type="text"/>
Password	<input type="password"/> 
Proxy Address	<input type="text"/>
Username	<input type="text"/>
Password	<input type="password"/> 

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.

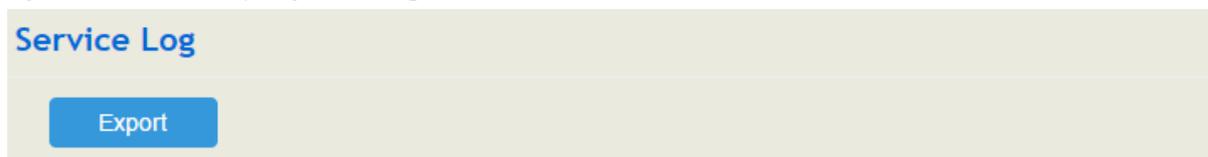
Operation Log Export

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

Index	Time	Level	Access Source	Operation	Page	Filter
100	2015-04-30 Thu 10:15:11	Info	172.16.188.123:58307	View	system/operationlog	
99	2015-04-30 Thu 09:41:55	Info	172.16.188.123:58034	Cancel	system/provision	

4.4.5 Service Log

Service logs (the running logs of UC100) can be exported on the **System → Service Log** interface. Those logs are used for analyzing where a problem has occurred on UC100.



4.4.6 Backup/Restore/Upgrade

On the **System → Backup/Restore/Upgrade** interface, you can back up or restore configuration data, and can upgrade UC100 to a new version. But you need to restart the device for the change to take effect after executing restore or upgrade.

Backup/Restore

Choose backup files and download System Network Service Download

Reset to defaults System Network Service Reset

Restore backup No file chosen Restore

Upgrade

Please Select Upgrade Type Upgrade

No file chosen

Parameter	Explanation
Reset	Click Reset , and all configuration will be restored to factory 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 port with other devices.

Automatch FXO Impedance

DTMF

Detection

Automatch Optimum Impedance

Cancel Save

How to use automatch impedance:

1. Connect a telephone cable to the FXO port;
2. Click **Detection**, and the UC100 device will automatically detect the optimum impedance (It takes a period of time to carry out the detection).
3. 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.

Please upload the English IVR welcome audio !

Please upload the Chinese IVR welcome audio !

Type	Name	Format	Language	Description	Upload File
IVR	Welcome	wav	English	The IVR welcome audio	Choose File No file chosen Upload

Attention: Upload of wav audio file format requirements, Monaural, 8000HZ, 16bit, less than 550kb.

Note:

- An IVR file has yet to be uploaded in the above figure;
- The format of the wav audio file must 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 UC100 device lacks an IVR file and you need to upload one.

Please upload the English IVR welcome audio !

Please upload the Chinese IVR welcome audio !

4.4.9 Reboot

On the **System** → **Reboot** interface, you can click **Perform Reboot** to reboot the UC100 device. After the device is rebooted, those configurations that have been saved will remain unchanged.

Reboot

Perform reboot

4.5 Network

UC100 works in two modes: route mode and bridge mode. When it is under the route mode, the IP of WAN must be different from the IP of LAN. But when it is under the bridge mode, the IP of WAN and the IP of LAN are the same.

4.5.1 Setting

Under the route mode, the default IP address of WAN port is a DHCP IP address, while the default IP address of the LAN port is 192.168.11.1.

In fact, there are three kinds of IP addresses for selection for the WAN port, including Static IP address, DHCP IP address and PPPOE IP address.

DHCP: Obtain IP address automatically.

UC100 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 UC100 from a defined range of numbers configured for a given network.

WAN

Protocol DHCP ▼

Obtain DNS server address automatically

MTU 1500

LAN

IP Address 192.168.11.1

Netmask 255.255.255.0 ▼

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 WAN port of the UC100 ;
- Netmask: the netmask of the router connected the UC100;
- Default Gateway: the IP address of the router connected the UC100;
- Use custom DNS server: the IP address of the DNS server

WAN

Protocol Static address ▼

IP Address 172.16.99.99

Netmask 255.255.255.0 ▼

Default Gateway 172.16.1.8

Use custom DNS server 8.8.8.8

172.16.1.8

MTU 1500

LAN

IP Address 192.168.11.1

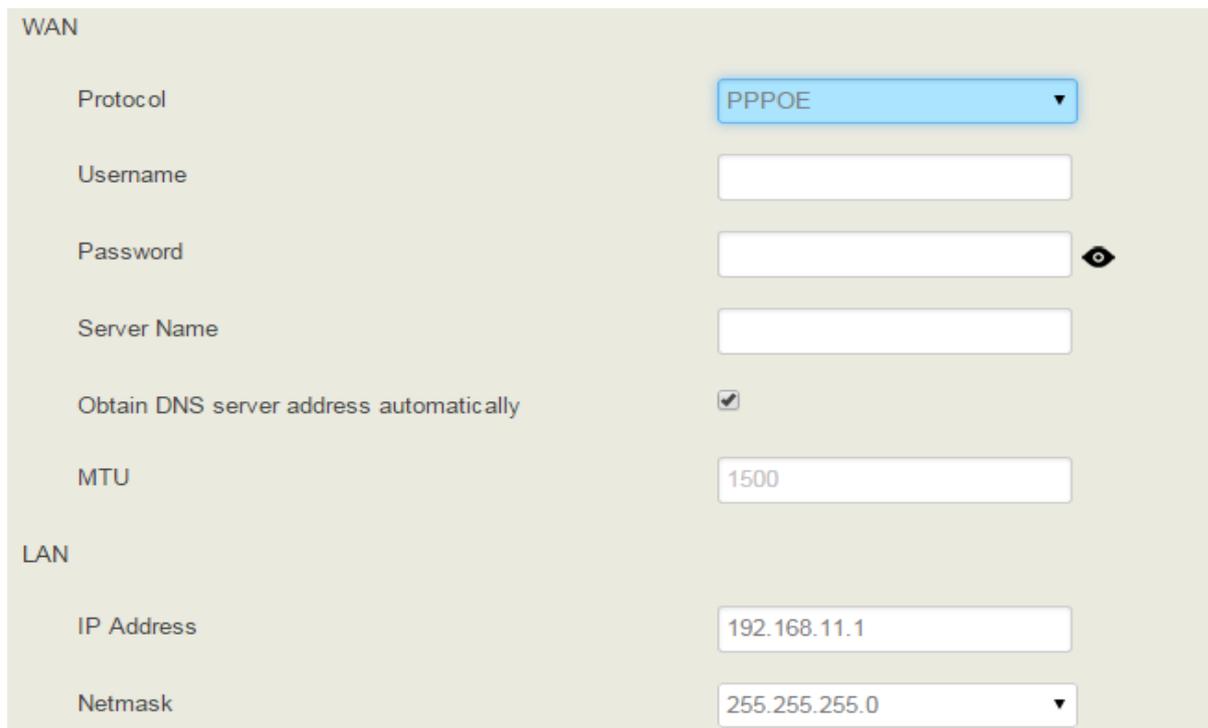
Netmask 255.255.255.0 ▼

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 mode.

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



The screenshot displays a configuration interface for WAN settings. Under the 'WAN' section, the 'Protocol' is set to 'PPPOE'. Below this, there are input fields for 'Username', 'Password', and 'Server Name'. The 'Password' field includes a toggle icon for visibility. The checkbox 'Obtain DNS server address automatically' is checked. The 'MTU' is set to '1500'. Under the 'LAN' section, the 'IP Address' is '192.168.11.1' and the 'Netmask' is '255.255.255.0'.

Note: The default IP address of WAN port is a DHCP IP address, but in actual conditions, the IP address of WAN port is more often set as a static IP address.

4.5.2 Access Control

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

Web Server

HTTP Port

Allow WAN access

HTTPS Port

Allow WAN access

Telnet

Enable

Port

Allow WAN access

SSH

Port

Allow WAN access

4.5.3 Firewall

If the UC100 works under the route mode, you can choose to enable the firewall and set filter rules to accept or reject certain destination IP addresses.

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.

Firewall

Filter Rules Control

Default filter action

Filter Rules

Index	Name	Protocol	Source	Destination	Action
2	test	TCP	192.168.11.140/**	/**	Reject  
3	test2	TCP	192.168.11.156/**	/**	Accept  

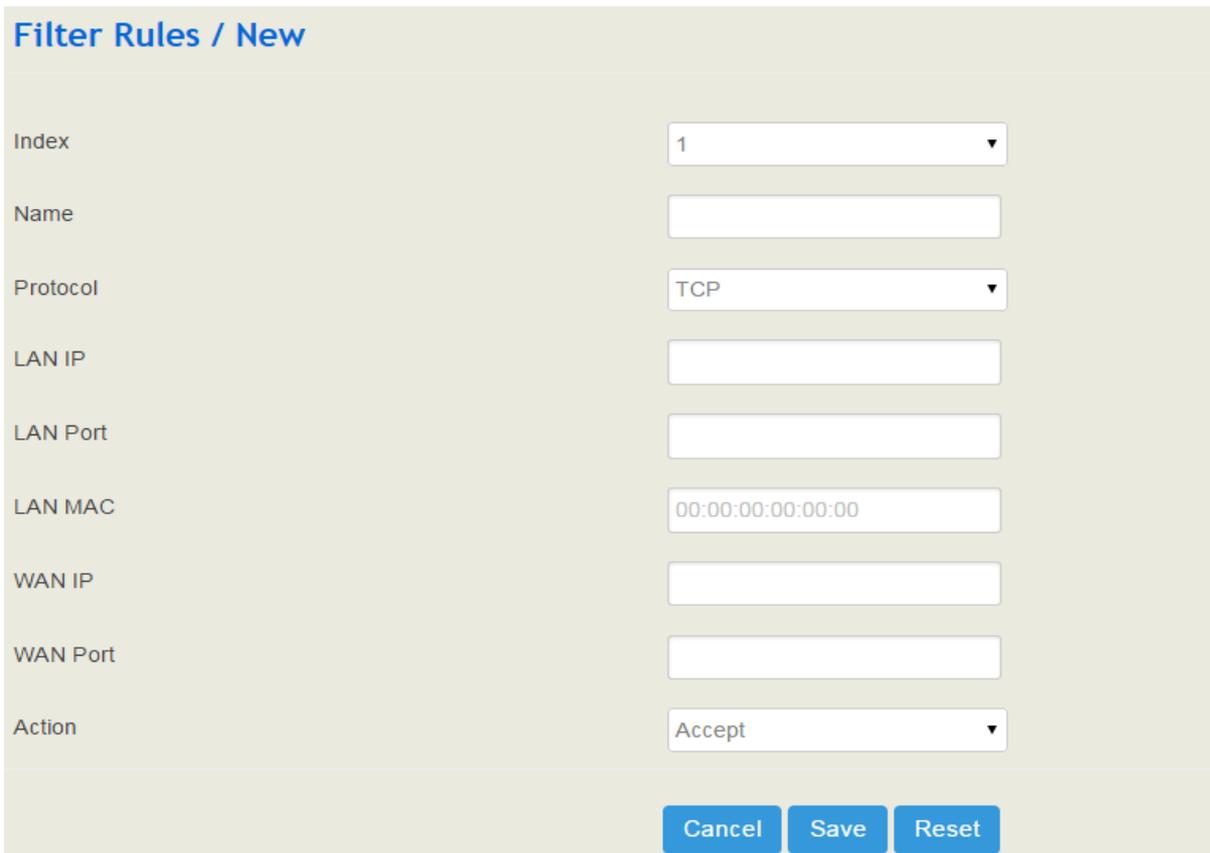
Note:

 : Edit information for the corresponding filter rule.

 : Delete the corresponding filter rule.

/*: Information of Source or Destination is not completely filled in.

4. Fill in relevant information in the following interface.



Filter Rules / New

Index: 1

Name: []

Protocol: TCP

LAN IP: []

LAN Port: []

LAN MAC: 00:00:00:00:00:00

WAN IP: []

WAN Port: []

Action: Accept

Buttons: Cancel, Save, Reset

Parameter	Explanation
LAN IP	The IP address of LAN port that you want UC100 to accept or reject
LAN Port	The LAN port which the accepted or rejected IP address belongs to
LAN Mac	The Mac of the LAN port that is accepted or rejected
WAN IP	The IP address of WAN port that you want UC100 to accept or reject
WAN Port	The WAN port which the accepted or rejected IP address belongs to
Action	Choose accept or reject

5. Click the **Save** button.

4.5.4 DHCP Server Setting

If there is a need, you can choose to enable the built-in DHCP server of UC100 to assign IP addresses to PC or other clients that are in the same local-area network with UC100.

DHCP Server Setting

DHCP Server: Enabled

Start Address: 192.168.11.99

End Address: 192.168.11.199

Leasetime(Hour): 12

Gateway:

Master DNS:

Slave DNS:

Buttons: Cancel, Save, Reset

Parameter	Explanation
Start Address	The start IP address to be assigned
End Address	The end IP address to be assigned
Lease Time	Period of validity
Gateway	The IP address of UC100; it is optional to fill in
Master DNS	The master DNS of the client whose IP address is assigned by the built-in DHCP server of UC100; it is optional to fill in
Slave DNS	The master DNS of the client whose IP address is assigned by the built-in DHCP server of UC100; it is optional to fill in

4.5.5 Client List

On the **Network** → **Client List** interface, information of the client devices whose IP addresses are assigned by the built-in DHCP server of UC100 is displayed.

Client List				
ID	Client Name	MAC Address	IP Address	Expiration
1	MacBook-Pro	60:03:08:A1:27:BA	192.168.11.177	2015-05-05 21:09:00
2	android-b47c31c0d0322ed2	A0:86:C6:8A:23:79	192.168.11.140	2015-05-05 21:05:37

4.5.6 Port Mapping

IF the UC100 works under the route mode, port mapping allows a client in the wide-area network to visit a client in the local-area network.

Configuration Procedures:

1. Click **Network** → **Port Mapping**, and the following interface will be shown.

Port Mapping					
Index	Name	Protocol	WAN Port	LAN IP	LAN Port
This section contains no values yet					
					New

2. Click the **New** button.

3. Fill in information on the following interface.

Port Mapping / New	
Index	<input type="text" value="1"/>
Name	<input type="text"/>
WAN Port	<input type="text"/>
Protocol	<input type="text" value="TCP"/>
LAN IP	<input type="text"/>
LAN Port	<input type="text"/>
<input type="button" value="Cancel"/> <input type="button" value="Save"/> <input type="button" value="Reset"/>	

Parameter	Explanation
Name	The name of the port mapping

WAN Port	The WAN port of the client in the wide-area network
Protocol	Choose TCP, UDP or TCP/UDP
LAN IP	The IP address of the LAN port of the to-be-visited client in local-area network
LAN Port	The LAN port of the to-be-visited client in local-area network

4. Click the **Save** button.

4.5.7 DMZ Setting

If the UC100 works under the route mode and the DMZ service is enabled, the clients in the wide-area network are allowed to have direct access to the clients in the DMZ (demilitarized zone).

4.5.8 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.

The screenshot shows a web interface with two main sections. The top section, titled "Network Utilities", contains three input fields with corresponding buttons: "Ping", "Traceroute", and "Nslookup". The bottom section, titled "Network Capture", contains several configuration options: "Capture Mode" (set to "Custom"), "Network Interface" (set to "WAN"), "Source IP", "Source Port", "Destination IP", and "Destination Port", each with an input field. Below these fields are checkboxes for "Protocol" (TCP, UDP, RTP, RTCP, ICMP, ARP). A "Start" button is located at the bottom right of the "Network Capture" section.

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 UC100 device, so you can choose different SIP profiles according to different needs.

SIP Profile / Edit

Index	1
Name	<input type="text" value="lan_default"/>
Local Listening Interface	<input type="text" value="LAN"/>
Local Listening Port	<input type="text" value="5060"/>
NAT	<input type="text" value="Off"/>
DTMF Type	<input type="text" value="RFC2833"/>
RFC2833-PT	<input type="text" value="101"/>
PRACK	<input type="text" value="Off"/>
Session Timeout	<input type="text" value="1800"/>
Inbound Codec Negotiation Priority	<input type="text" value="Remote"/>
Inbound Codec Profile	<input type="text" value="1-< default >"/>
Outbound Codec Profile	<input type="text" value="1-< default >"/>

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 uPNP/NAT-PMP, IP Address, Stun, and Host.
DTMF	There are three modes, including SIP Info, INBAND, RFC2833.
RFC2833-PT	RFC2833 payload coding
PRACK	Provisional Response ACKnowledgement
Session Timeout	The validity period of current registration. It is 1800 seconds by default
Inbound Codec	To take the remote device or the local device as priority for inbound codec

Negotiation Priority	<p>negotiation.</p> <p>Assume local device supports PCMA, PCMU, G.729 and G.723, while the remote device supports G.723 and G.729.</p> <p>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.</p>
----------------------	--

4.6.2 FXS/FXO

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

Profile → FXS Interface:

FXS Profile / New

Index	2 ▼
Name	<input type="text"/>
Tone Group	China ▼
Digit Timeout(s)	4
Dial Timeout(s)	10
Ring Timeout(s)	55
No Answer Timeout(s)	55
Flash Detection	<input type="checkbox"/>
DTMF Send Interval(ms)	250
DTMF Gain	-4dB ▼
DTMF Duration(ms)	200
CID Send Mode	FSK ▼
Message Mode	MDMF ▼
Message Format	Display Name and CID ▼
Send CID Before RING	<input type="checkbox"/>
Send CID After Ring(ms)	2000
Impedance	600 Ohm ▼
Polarity Reverse	ON ▼
Dialplan	Off ▼

Parameter	Explanation
Name	The name of the FXS profile
Tone Group	The state standard of dialing tone, busyness tone and ring tone; the default value is China
Digit Timeout (s)	The timeout value for dialing a digit of a telephone number
Dial Timeout (s)	The timeout value for dialing the first telephone number after off-hook
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
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; generally it is the default setting FSK: Frequency-shift keying CID: Caller ID
Message Mode	Include SDMF and MDMF; generally it is the default setting
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 ring; 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; Generally it is the default setting
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 preset time.
Dialplan	The rules for dialing

Profile → FXS Interface:

FXO Profile / New

Index	<input style="width: 90%;" type="text" value="2"/>
Name	<input style="width: 90%;" type="text"/>
Tone Group	<input style="width: 90%;" type="text" value="China"/>
Digit Timeout(s)	<input style="width: 90%;" type="text" value="4"/>
Dial Timeout(s)	<input style="width: 90%;" type="text" value="10"/>
Ring Timeout(s)	<input style="width: 90%;" type="text" value="55"/>
No Answer Timeout(s)	<input style="width: 90%;" type="text" value="55"/>
Polarity Reverse	<input style="width: 90%;" type="text" value="ON"/>
DTMF Send Interval(ms)	<input style="width: 90%;" type="text" value="250"/>
DTMF Gain	<input style="width: 90%;" type="text" value="0dB"/>
DTMF Duration(ms)	<input style="width: 90%;" type="text" value="200"/>
Detect Caller ID	<input style="width: 90%;" type="text" value="ON"/>
Ring Detection	<input style="width: 90%;" type="text" value="Detect after ring"/>
Dialplan	<input style="width: 90%;" type="text" value="Off"/>

Parameter	Explanation
Name	The name of the FXO profile
Tone Group	The state standard of dialing tone, busyness tone and ring tone; the default value is China
Digit Timeout (s)	The timeout value for dialing a digit of a telephone number

Dial Timeout (s)	The timeout value for dialing the first telephone number after off-hook
Ring Timeout (s)	The timeout value for the ringing of the phones of 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 to delay offhook and call tolls will be calculated starting from the preset 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

UC100 supports four codec modes, including G711A, G711u, G729 and G92. You can adjust the priority of these four modes according to you needs.

Codec Profile / New

Index	<input style="width: 90%;" type="text" value="2"/>
Name	<input style="width: 90%;" type="text"/>
Codec	<div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">G729 ▼ ⊗</div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">G723 ▼ ⊗</div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">PCMU ▼ ⊗</div> <div style="border: 1px solid #ccc; padding: 2px;">PCMA ▼ ⊗</div>

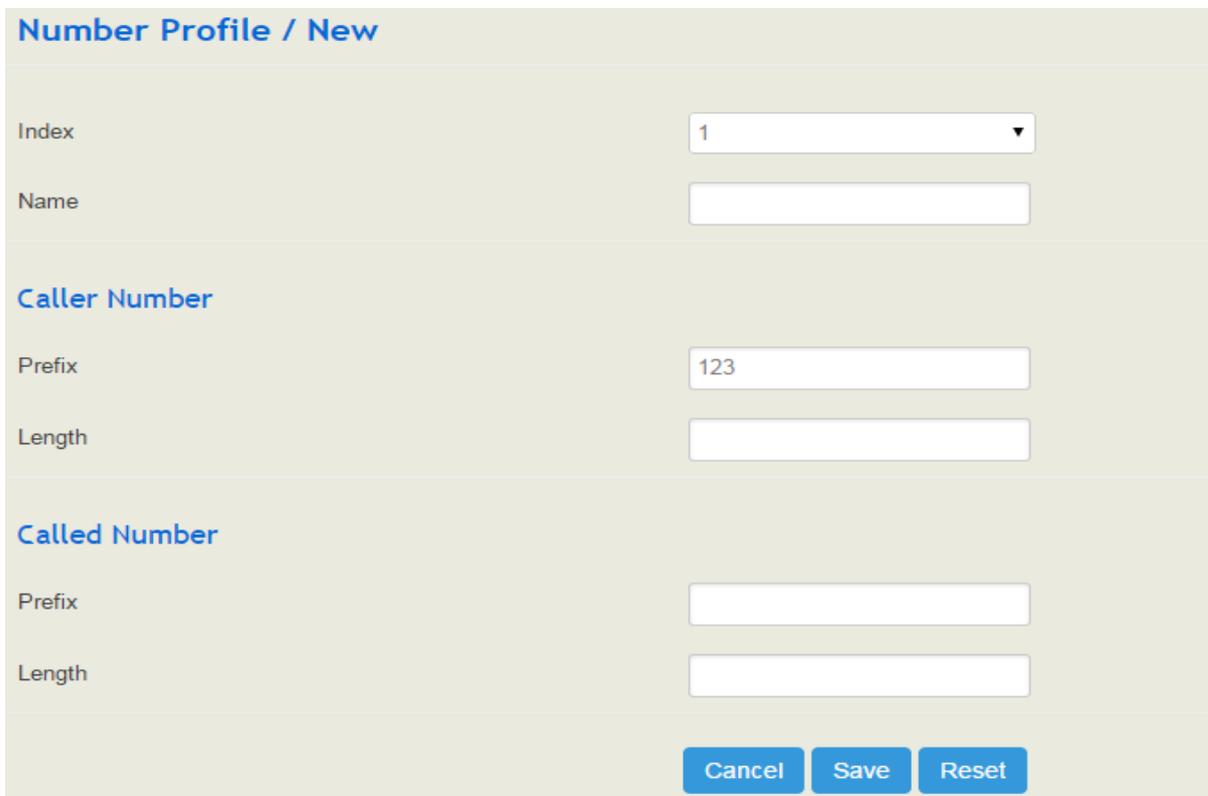
 : Edit codec profile.

 : Delete the corresponding codec profile or a codec mode.

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.

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



Number Profile / New

Index: 1

Name: [Text Input]

Caller Number

Prefix: 123

Length: [Text Input]

Called Number

Prefix: [Text Input]

Length: [Text Input]

Buttons: Cancel, Save, Reset

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. When the calling time of a call falls into the set time period, the call will be passed to choose a route.

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

Time Profile / New

Index	<input type="text" value="1"/>	
Name	<input type="text"/>	
Date Period	<input type="text"/>	+
Weekday	<input type="checkbox"/> Mon <input type="checkbox"/> Tue <input type="checkbox"/> Wed <input type="checkbox"/> Thu <input type="checkbox"/> Fri <input type="checkbox"/> Sat <input type="checkbox"/> Sun	
Time Period	<input type="text"/>	+

Parameter	Explanation
Date Period	Choose a start date and an end date
Weekday	Choose a weekday
Time Period	Choose start 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.

Manipulation Profile / New

Index	<input type="text" value="1"/>	
Name	<input type="text" value="123"/>	
Caller	<input checked="" type="checkbox"/>	
Delete Prefix	<input type="text"/>	
Delete Suffix	<input type="text"/>	
Add Prefix	<input type="text"/>	
Add Suffix	<input type="text"/>	
Replace by	<input type="text"/>	
Called	<input type="checkbox"/>	

Parameter	Explanation
Delete Prefix	The number of digits that are deleted from the left of the caller number or calling number
Delete Suffix	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
<input checked="" type="checkbox"/>	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 the dialing of calls through FXS and FXO ports.

Dialplan Profile / New

Index

Name

Format

Dialplan

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, colour?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

Examples:

(13 15 18)xxxxxxxx	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 the UC100 by SIP clients.

SIP Extension / New

Index: 1

Name:

Extension:

Password:

DID:

Register Source: Any

Call Waiting: Off

Do Not Disturb: Off

Call Forward Unconditional: Off

Call Forward Busy: Off

Call Forward No Reply: Off

SIP Profile: 1-< lan_default >

Status: Enable

Buttons: Cancel, Save, Reset

Parameter	Explanation
Name	The name of the SIP extension

Extension	The registered SIP account
Password	The password of the extension
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	Make reference to Profile → SIP
Status	If it is enabled, the SIP account can be registered; Otherwise the SIP account cannot be registered.

4.7.2 FXS

On the **Extension → FXS** interface, you can configure data for extensions of the FXS port.

FXS Extension / Edit

Extension	<input type="text" value="8000"/>
Status	<input type="text" value="Enable"/>
DID	<input type="text"/>
Register to SIP Server	<input type="text" value="On"/>
Master Server	<input type="text" value="Not Config"/>
Slave Server	<input type="text" value="Not Config"/>
Password	<input type="password"/> 
Call Waiting	<input type="text" value="On"/>
Do Not Disturb	<input type="text" value="Off"/>
Call Forward Unconditional	<input type="text" value="Off"/>
Call Forward No Reply	<input type="text" value="Off"/>
Input Gain	<input type="text" value="0 db"/>
Output Gain	<input type="text" value="0 db"/>
FXS Profile	<input type="text" value="1-< default >"/>

Parameter	Explanation
Extension	The extension number of FXS port
Status	If it is enabled, the FXS port is available; if it is disabled, the FXS port is unavailable
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 to SIP Server	If it is enabled, the FXS extension account will be registered to the SIP trunk that has been set.
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.
Input Gain	The receiving gain of the FXS port
Output Gain	The sending gain of the FXS port
FXS Profile	Make reference to Profile →FXS/FXO

4.7.3 Ring Group

On the **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.

Ring Group / New

Index

Name

Members Select

Strategy

Ring Group Number

DID

Ring Time(5s~60s)

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

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 UC100 and IPPBX or SIP servers.

SIP Trunk / New

Index	<input type="text" value="1"/>
Name	<input type="text"/>
IP Address	<input type="text"/>
Port	<input type="text"/>
Outbound Proxy	<input type="text"/>
Port	<input type="text"/>
Transport	<input type="text" value="UDP"/>
Register	<input type="text" value="OFF"/>
Heartbeat	<input type="text" value="OFF"/>
SIP Profile	<input type="text" value="1-< lan_default >"/>
Status	<input type="text" value="Enable"/>

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; 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	Whether the SIP trunk is registered or not
Heartbeat	If heartbeat is on, heartbeat messages (options) will be sent to examine the connection with servers. The default value is 'Off'.
SIP Profile	The SIP profile of 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.

Note: If UC100 is regarded as a terminal and registered to SIP server, and you want to register the UC100, you just need to register the corresponding SIP trunk.

4.8.2 FXO Trunk

FXO Trunk interconnects the PSTN with UC100. Calls from the PSTN can come into UC100 and calls can go out from UC100 to search telephone numbers under the PSTN.

Different from the FXO ports of other gateways, the FXO port of UC100 only allows one-time dialing, which means called numbers needs to be dialed directly for calls that go out from the FXO port.

FXO Trunk / Edit

Extension	<input type="text" value="8001"/>
Status	<input type="text" value="Enable"/>
Autodial Number	<input type="text"/>
Register to SIP Server	<input type="text" value="On"/>
Master Server	<input type="text" value="Not Config"/>
Slave Server	<input type="text" value="Not Config"/>
Password	<input type="text"/> 
Input Gain	<input type="text" value="0db"/>
Output Gain	<input type="text" value="0db"/>
Impedance	<input type="text" value="600 Ohm"/>
FXO Profile	<input type="text" value="1-< default >"/>

Parameter	Explanation
Extension	The registered SIP account of the FXO port
Status	If it is enabled, it means the FXO trunk is available; otherwise, the FXO trunk is unavailable
Autodial Number	The autodial number for incoming calls through FXO port
Register to SIP Server	Whether to register the FXO trunk to SIP server
Input Gain	Receiving gain of FXO port
Output Gain	Sending gain of FXO port
FXO Profile	Make reference to Profile → FXS/FXO

4.8.3 GSM Trunk

GSM trunk interconnects the GSM wireless network with UC100. Calls from the GSM wireless network can come into UC100 and calls can go out from UC100 to search mobile numbers under the GSM wireless

network.

GSM Trunk / Edit

Extension	<input type="text" value="8002"/>
Status	<input type="text" value="Enable"/>
Autodial Number	<input type="text"/>
Register to SIP Server	<input type="text" value="Off"/>
SMS Encoding	<input type="text" value="ucs2"/>
SMS Center Number	<input type="text"/>
PIN Code	<input type="text"/>

Parameter	Explanation
Extension	The registered SIP account of the GSM port
Status	If it is enabled, it means the GSM trunk is available; otherwise, the GSM trunk is unavailable
Autodial Number	The autodial number for incoming calls through GSM port
Register to SIP Server	Whether to register the GSM trunk 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 the routes and trunks for incoming and outgoing calls through UC100, as well as IVR, SMS, fax and call-related security.

4.9.1 Setting

Voice

Disconnect call when no RTP packet

Language of Tone English ▼

Start Port 16384

End Port 32768

Route

Local extension call

FAX

Send Mode T.30 ▼

Tone Detection by Local

SDP Param

a=X-fax

a=fax

a=X-modem

a=modem

Parameter	Explanation
Disconnect call when no RTP packet	If it is enabled, and no RTP packets are received within the preset time, calls will be disconnected
Language of Tone	English or Chinese; you can upload customized IVR on the System → Voice interface.
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, UC100 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.

Route Group / New

Index

Name

Members Select 

Strategy

Parameter	Explanation
Name	The name of the route group
Members Select	Select the FXS extension and a trunk or several trunks
Strategy	The strategies for choosing which trunk as the destination route, including Sequence (Ascending), Sequence (Cyclic Ascending), Simultaneous and Random

4.9.3 Route

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

Route / New

Priority

Name

Condition

Source

Number Profile

Caller Number Prefix

Called Number Prefix

Time Profile

Action

Manipulation

Destination

Destination

Failover Action

Condition Busy Timeout

Other Condition Code

Manipulation

Destination

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 the FXS extension, FXO trunk ,GSM trunk, a customized source or any
Number Profile	The profile of the caller 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→ Number Manipulation
Destination	The destination of the route
Failover Action	The processing 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 ','

4.9.4 Feature Code

Index	Feature	Key	Description	Status	
1	Inquiry LAN IP	*158	Inquiry LAN IP	Enabled	 
2	Inquiry WAN IP	*159	Inquiry WAN IP	Enabled	 
3	Inquiry Phone Number	*114	Inquiry Phone Number	Enabled	 
4	Network Work Mode	*157*	Dial *157*0 to set route mode.Dial *157*1 to set bridge mode	Enabled	 
5	IP Address Config Mode	*150*	*150*1#-Static, *150*2#-DHCP	Enabled	 
6	Configure IP Address	*152*	Set IPv4 Address 192.168.1.10 by dial *152*192*168*1*10#	Enabled	 
7	Configure Gateway	*156*	Set IPv4 Gateway 192.168.1.1 by dial *156*192*168*1*1#	Enabled	 
8	Configure Subnet Mask	*153*	Set IPv4 Netmask 255.255.0.0 by dial *153*255*255*0*0#	Enabled	 
9	Restart Device	*111	Restart Device	Enabled	 
10	Call Waiting Activate	*51	Enable call_waiting service	Enabled	 
11	Call Waiting Deactivate	*50	Disable call_waiting service	Enabled	 
12	Blind Transfer	*1	Example:*18000#,you can blind transfer to the extension number ...	Enabled	 
13	Attended Transfer	*2	Example:*28000#,you can attended transfer to the extension num...	Enabled	 
14	Call Forward Unconditional Activate	*72*	Enable Unconditional call_forward service.Example:*72*8000,set t...	Enabled	 
15	Call Forward Unconditional Deactivate	*73	Disable Unconditional call_forward service	Enabled	 
16	Call Forward Busy Activate	*90*	Enable USER_BUSY call_forward service.Example:*90*8000,set t...	Enabled	 
17	Call Forward Busy Deactivate	*91	Disable USER_BUSY call_forward service	Enabled	 
18	Call Forward No Reply Activate	*92*	Enable NO_REPLY call_forward service.Example:*92*8000,set th...	Enabled	 
19	Call Forward No Reply Deactivate	*93	Disable NO_REPLY call_forward service	Enabled	 
20	Do Not Disturb Activate	*78	Enable DND service	Enabled	 
21	Do Not Disturb Deactivate	*79	Disable DND service	Enabled	 
22	Group Pickup	**	Pick up the ringing extension which in the same ringgroup, Examp...	Enabled	 

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.

IVR

Status Enable 

Timeout 10

Enable Direct Extension

Repeat Loops 3

Menu

DTMF	Destination	Destination Number
<input type="text" value="0"/>	<input type="text" value="FXS Extension / FXS / 8000"/>	

Parameter	Explanation
Status	If it is disabled, the VCR 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 the GSM port, you can send or receive SMS on the **SMS** interface.

Message Send

Select Port: 1-GSM/1-GSM

Recipient: Send

Send List

Empty

Port	Contact	Time	Message	Status	Operation	Filter
1-GSM	10086	2015-05-05 02:40:34	cxdh	success	✉ ↶ ✖	

Receive List

Empty

Port	Contact	Time	Message	Status	Operation	Filter
------	---------	------	---------	--------	-----------	--------

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 mutiple recipients , use | to separate them, for example, 13151103146|18954405566.

Receive Message

All SMS received by UC100 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