

UC Basic Configuration Practice Guide

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- This course is mainly:
 - Introduce Dinstar UC series.
 - Guide we how to access the device
 - Guide we set network on UC
 - Introduce Dinstar UC common application scenario
 - Learn common application scenario configurations

Course Objective

Through this course
you will be able to






HOW To Access The Device



How To SET Network On Device



How To Make Calls On UC

-  Chapter One The Way To Access Device
-  Chapter Two The UC Call Configuration Instructions
-  Chapter Three Common Function Configuration



Chapter ONE

The Way To Access Device

01

1. Physical Connection
2. PC Settings
3. Access Device GUI

Physical Connection

- UC120/UC200

Support Router and Bridge network mode

Default network mode **Router**

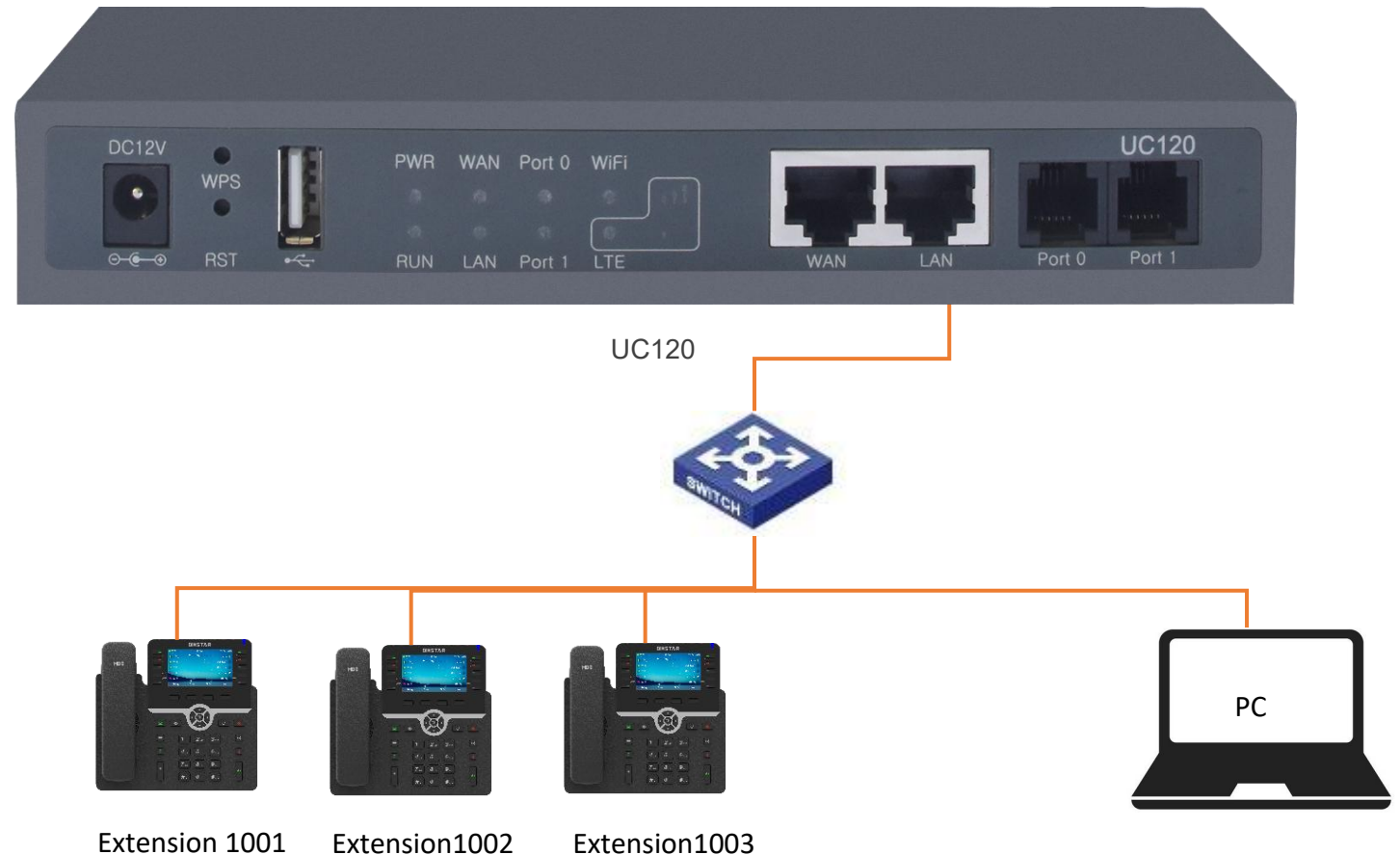
WAN Port DHCP, LAN Port---

192.168.11.1

Default username/password---

admin/admin@123#

Service Port WAN and LAN



Physical Connection

- UC200Pro/UC350

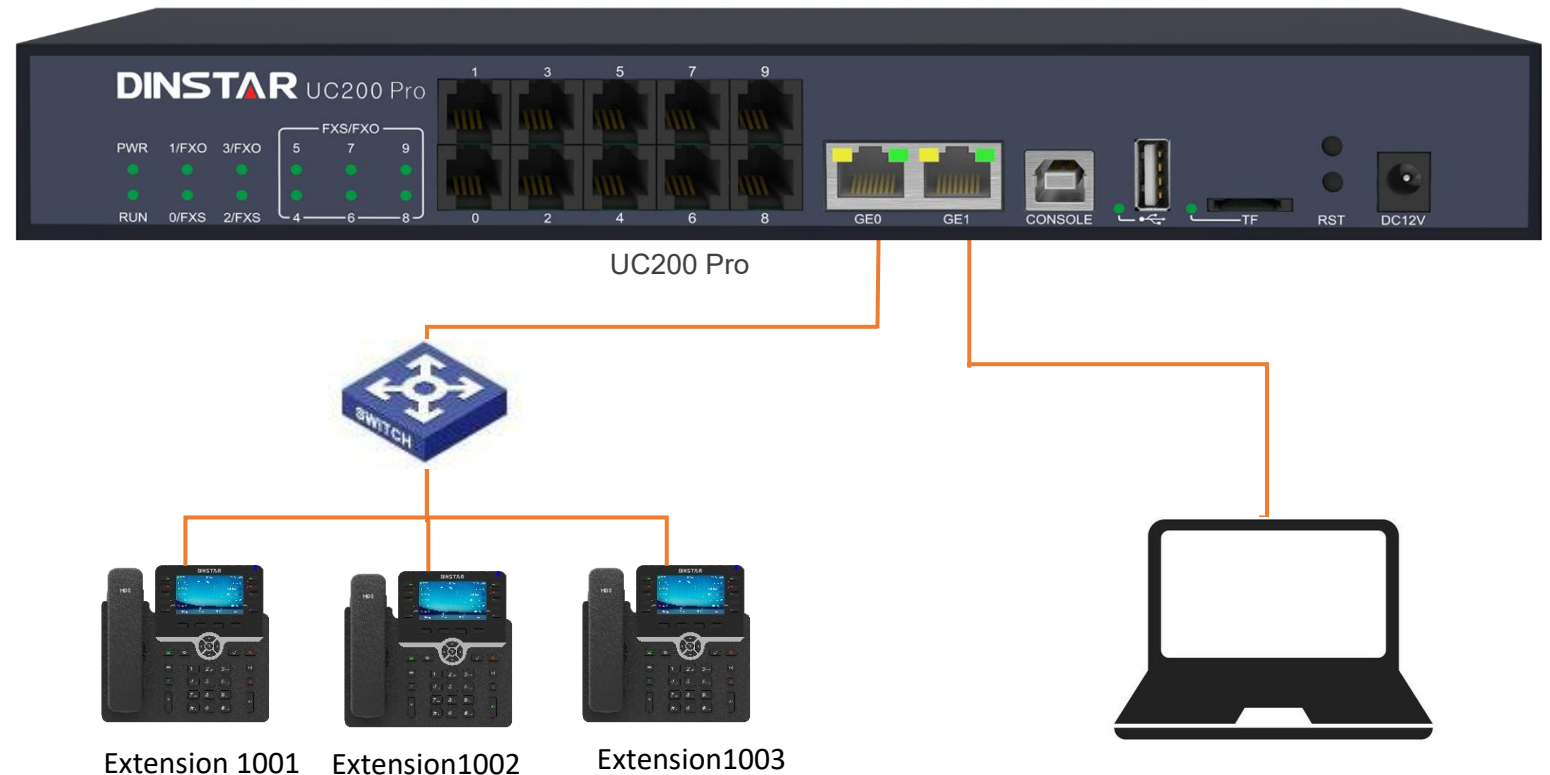
Support Gigabit Ethernet port

Management port GE1--192.168.11.1

Default username/password---

admin/admin@123#

Service Port GE0 and GE1



Physical Connection

- UC350Pro

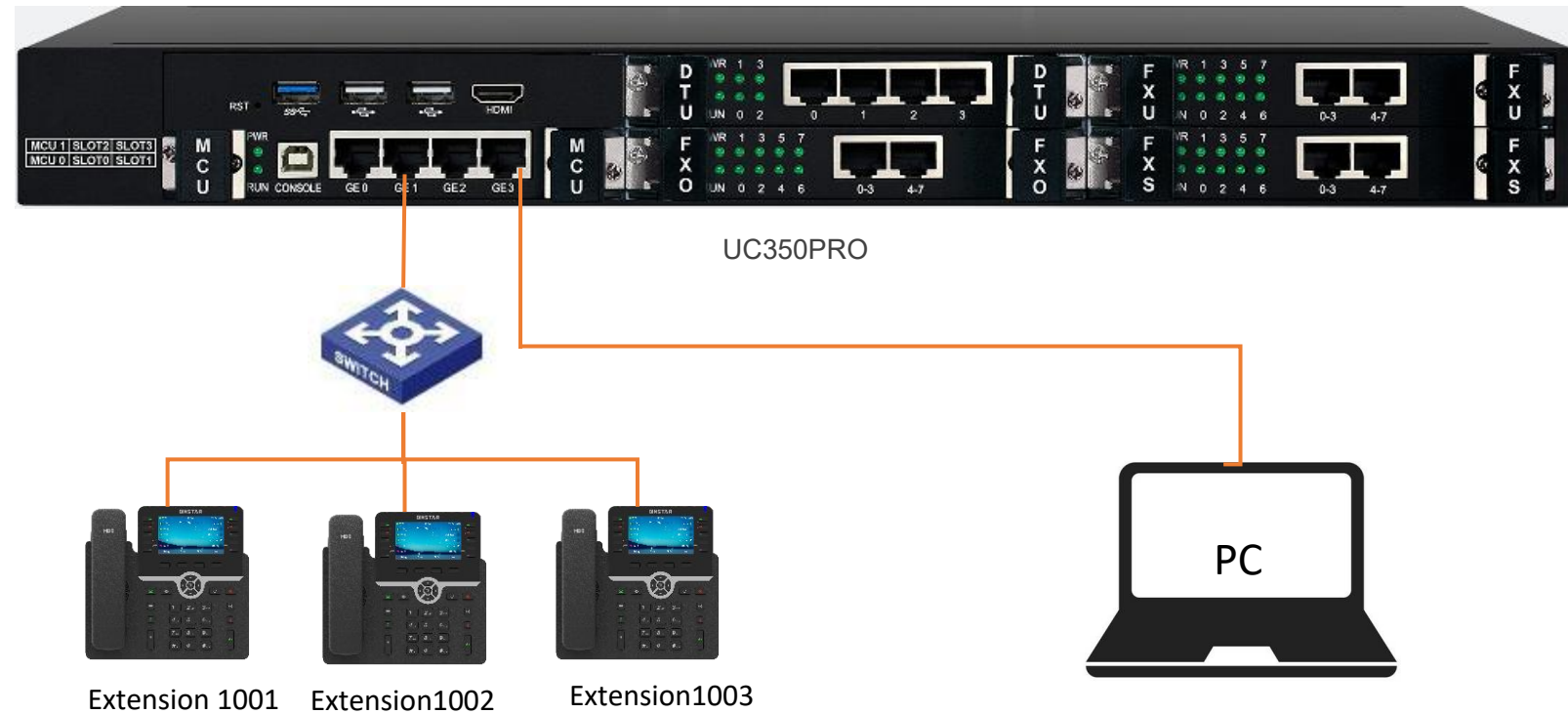
Support Gigabit Ethernet port

Management port GE3--192.168.11.1

Default username/password---

admin/admin@123#

Service Port GE0 to GE3



PC Settings

- Add IP

Add 192.168.11.xxx/255.255.255.0 to access the UC

Note:

1. Optional IP range: 192.168.11.2-

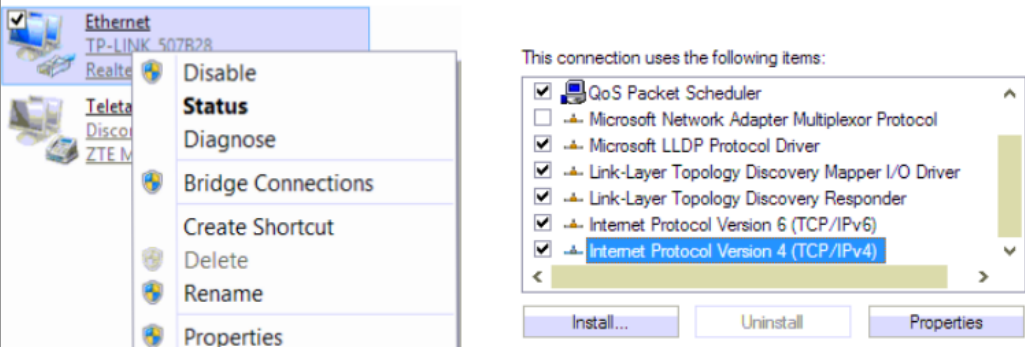
192.168.11.254

2. Set the IP address of the PC as shown in the figure on the right

1 On the PC, click 'Network (or Ethernet) → Properties'.

2 Double-click 'Internet Protocol Version 4 (TCP/IPv4)'.

3 Select 'Use the following IP address', and then enter an available IP address '192.168.11.XXX' which is at the same network segment with '192.168.11.1'.



The screenshot shows the Windows Network Connections window with the context menu open for the Ethernet adapter. The 'Properties' option is selected. The 'Internet Protocol Version 4 (TCP/IPv4)' property is highlighted in the list. The 'Internet Protocol Version 4 (TCP/IPv4) Properties' dialog is open, showing the 'General' tab. The 'Use the following IP address' radio button is selected. The IP address is set to 192.168.11.20, the subnet mask is 255.255.255.0, and the default gateway is 192.168.11.1.

Internet Protocol Version 4 (TCP/IPv4) Properties

General

You can get IP settings assigned automatically if your network supports this capability. Otherwise, you need to ask your network administrator for the appropriate IP settings.

☐ Obtain an IP address automatically

☒ Use the following IP address:

IP address: 192 . 168 . 11 . 20

Subnet mask: 255 . 255 . 255 . 0

Default gateway: 192 . 168 . 11 . 1

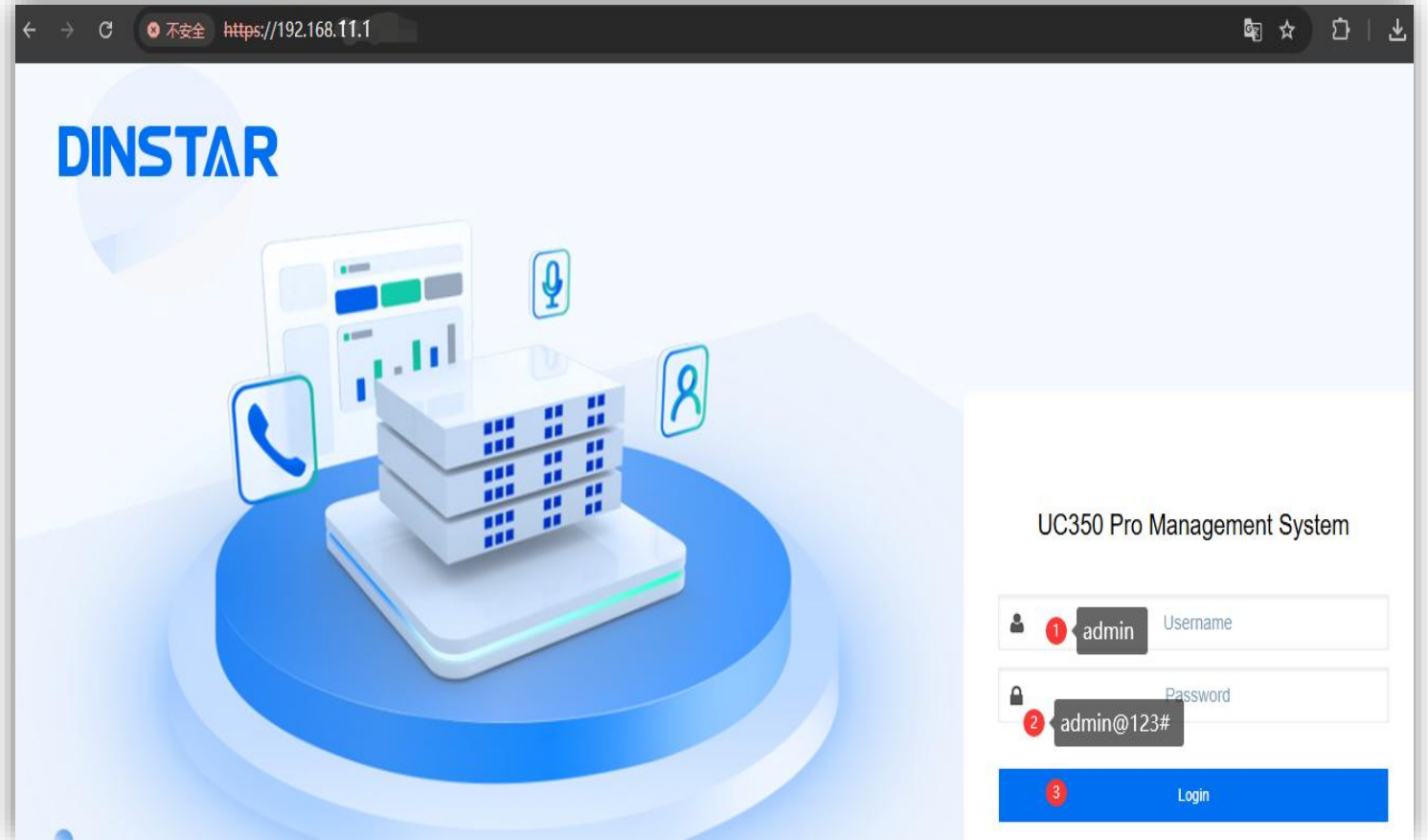
Access Device GUI

- Access Device

Browser input <https://192.168.11.1> to login interface .

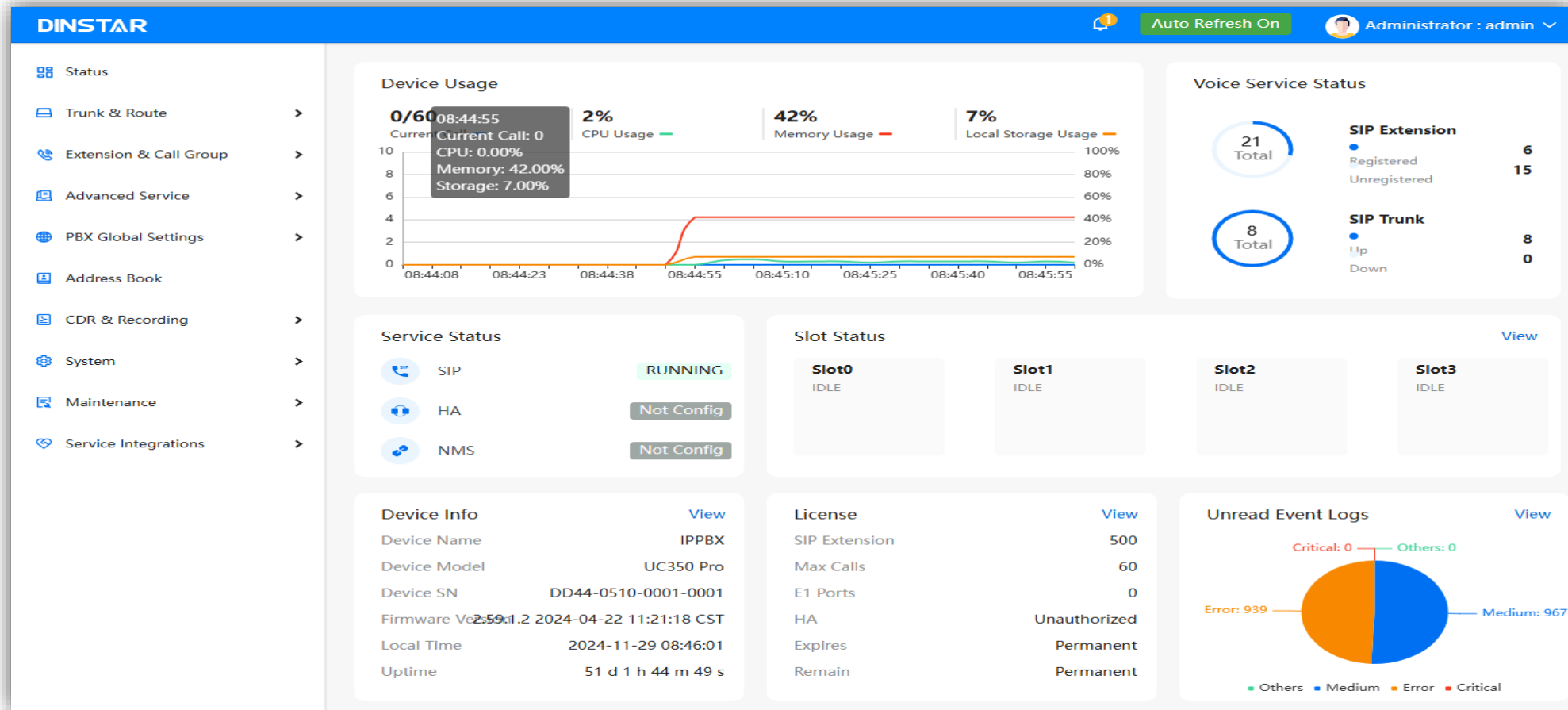
Input Username and Password:
admin/admin@123# ,




Click Login to access the device.



Access Device GUI

After logging in, you will see the status interface



- Chapter One The Way To Access Device
- Chapter Two The UC Call Configuration Instructions
- Chapter Three Common Function Configuration

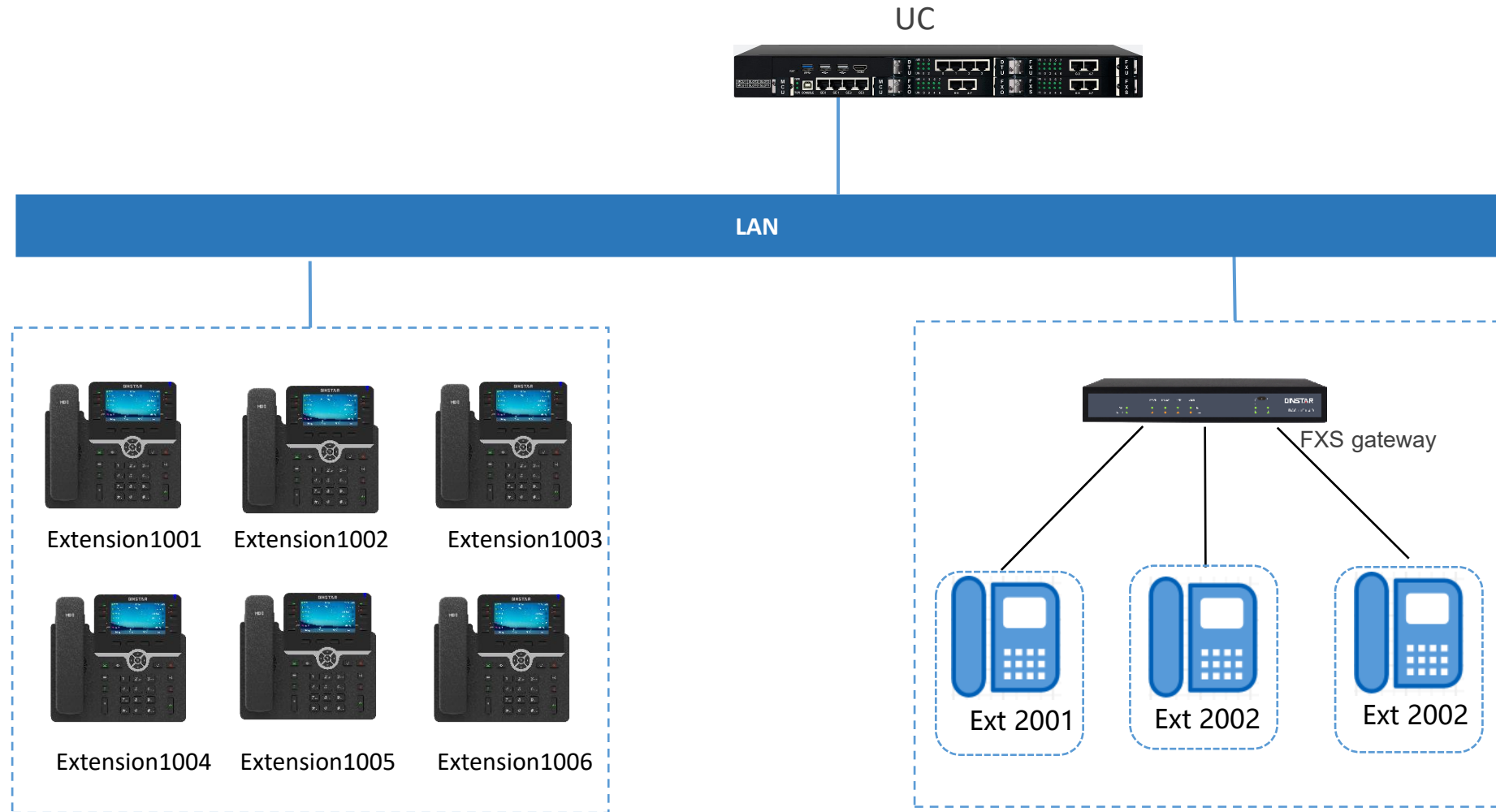
Chapter Two

UC Call Configuration Instructions

02

1. Local extension call
2. SIP Trunk Call Configuration
3. FXO Trunk Call Configuration
4. E1 Trunk Call Configuration

Common Application 1



Local extension call

IP Address Setting

SIP Stack Setting

SIP Extension
Setting

View Permissions

1. Click **System->Network**

2. Select the network port you want to connect to, and click "Edit" to set the IP address.

Tip:

After setting IP address, you need to save the application and reboot to take effect.

The screenshot displays the DINSTAR web interface for network configuration. On the left sidebar, the 'System' menu is expanded, and 'Network' is highlighted with a red box and a circled '1'. The main content area shows the 'Network' settings for two ports, GE0 and GE1. The 'Setting' tab is selected. For port GE0, the IPv4 status is 'Not Connect' and the IPv6 status is 'Not Connect'. For port GE1, the IPv4 status is 'Enabled' and the IPv6 status is 'Not Config'. The IP address for GE1 is set to 192.168.168.10, and the netmask is 255.255.255.0, both highlighted with red boxes. The 'Edit' button for GE1 is highlighted with a red box and a circled '2'.

Port	IPv4 Status	IPv4 Type	IPv4 IP Address	IPv4 Netmask	IPv4 Gateway	IPv4 Preferred DNS	IPv4 Alternate DNS	IPv6 Status	IPv6 Type	IPv6 IP Address	IPv6 Prefix Length	IPv6 Gateway	IPv6 Preferred DNS	IPv6 Alternate DNS
GE0	Not Connect	Static		0.0.0.0				Not Connect	Static					
GE1	Enabled	Static	192.168.168.10	255.255.255.0	192.168.168.1			Not Config	Static					

Local extension call

IP Address Setting

SIP Stack Setting

SIP Extension
Setting

View Permissions

1. Click **PBX Global Settings->SIP Stack->setting**

2. Select "Edit" to set the listening port

Tip:

Default listening port 5060,
modify as needed

SIP Stack

Status **Setting**

1

New

Index	Name	IPv4/IPv6	Interface	Listening Port	DTMF	Session Timeout	Codec Priority	Incodec Profile	Outcodec Profile	More
1	GE0_Default	IPv4	GE0	5060	RFC2833	Off	Remote	1-< default >	1-< default >	Edit Delete

2

Local extension call

IP Address Setting

SIP Stack Setting

SIP Extension
Setting

View Permissions

1. Click **Extension & Call Group**-
> **SIP Extension**

2. Create Batch new SIP extension,
configure the start extension
number and password

3. Select network port

Batch New SIP Extension

Basic Settings

Status ☒

Start Extension 1000

Extension Count 9

DID Off

Step 1

SIP Password Policy All Same

SIP Password 12345678

App Password Policy All Same

App Password 12345678

SIP Profile 1-< GE0_Default >

Local extension call

IP Address Setting

SIP Stack Setting

SIP Extension
Setting

View Permissions

1. Click **PBX Global Setting->Voice**

2. Check if the local extension call configuration is enabled

Tip:

Local extension call Permission is enabled by default

The screenshot displays the DINSTAR PBX Global Settings interface. On the left, a sidebar menu lists various settings categories: Status, Trunk & Route, Extension & Call Group, Advanced Service, PBX Global Settings (expanded), SIP Stack, Codec, FXS/FXO, Voice (highlighted with a red box and circled '1'), Feature Code, Address Book, CDR & Recording, System, Maintenance, and Service Integrations. The main content area is titled 'Voice' and contains two tabs: 'Setting' (selected) and 'File List'. Under the 'Setting' tab, there are several configuration items, each with a dropdown menu: Voice Language (English), Waiting Music (Default Tone), Timeout Tone (Off), Busy Tone (Off), Offline Tone (Off), Call Waiting Tone (Default Tone), Number Invalid Tone (Off), Reject Tone (Off), NotAuth Tone (Off), and Recording Prompt Tone (Off). Below these settings is a 'Route' section with a toggle switch for 'Area Call Auth' (disabled) and a toggle switch for 'Local extension call' (enabled, highlighted with a red box and circled '2').

IP Phone registration configuration

1. Click **Account-Basic**

2. Configure the account and password to match the SIP extension number and password of UC

3. Configure the IP and port of UC to be consistent with the UC SIP stack configuration

4. View registration results after submission

The screenshot shows the DINSTAR web interface for configuring SIP accounts. The top navigation bar includes links for Status, Account, Network, Phone, PhoneBook, Upgrade, and Security. The 'Account' tab is selected. The left sidebar has a 'Basic' tab highlighted with a red box and a circled '1'. The main content area is divided into two sections: 'SIP Account' and 'SIP Server'. The 'SIP Account' section contains fields for Status (Registered), Account (Account2: 6005), Active (Enabled), Display Label, Display Name (6005), Register Name (6005), Username (6005), and Password (masked with dots). A red box and a circled '2' highlight the 'Active', 'Display Label', 'Display Name', 'Register Name', 'Username', and 'Password' fields. The 'SIP Server' section has two sub-sections: 'SIP Server 1' and 'SIP Server 2'. 'SIP Server 1' has fields for Server IP (172.27.10.14), Port (5060), and Registration Expires (120). A red box and a circled '3' highlight the 'Server IP' and 'Port' fields. 'SIP Server 2' has fields for Server IP and Port (5060), and Registration Expires (1800). A red box and a circled '4' highlight the 'Submit' button in the bottom right corner. The right sidebar contains a 'Help' section with a 'Description' and a 'Warning' section.

SIP Account	
Status	Registered
Account	Account2: 6005
Active	Enabled
Display Label	
Display Name	6005
Register Name	6005
Username	6005
Password	*****

SIP Server 1		
Server IP	172.27.10.14	Port: 5060
Registration Expires	120	(30~65535s)

SIP Server 2		
Server IP		Port: 5060
Registration Expires	1800	(30~65535s)

FXS registration configuration

1. Click **SIP Server**

2. Configure the IP and port of UC to be consistent with the UC SIP stack configuration

3. Click **Port**

4. Configure the account and password to match the SIP extension number and password of UC

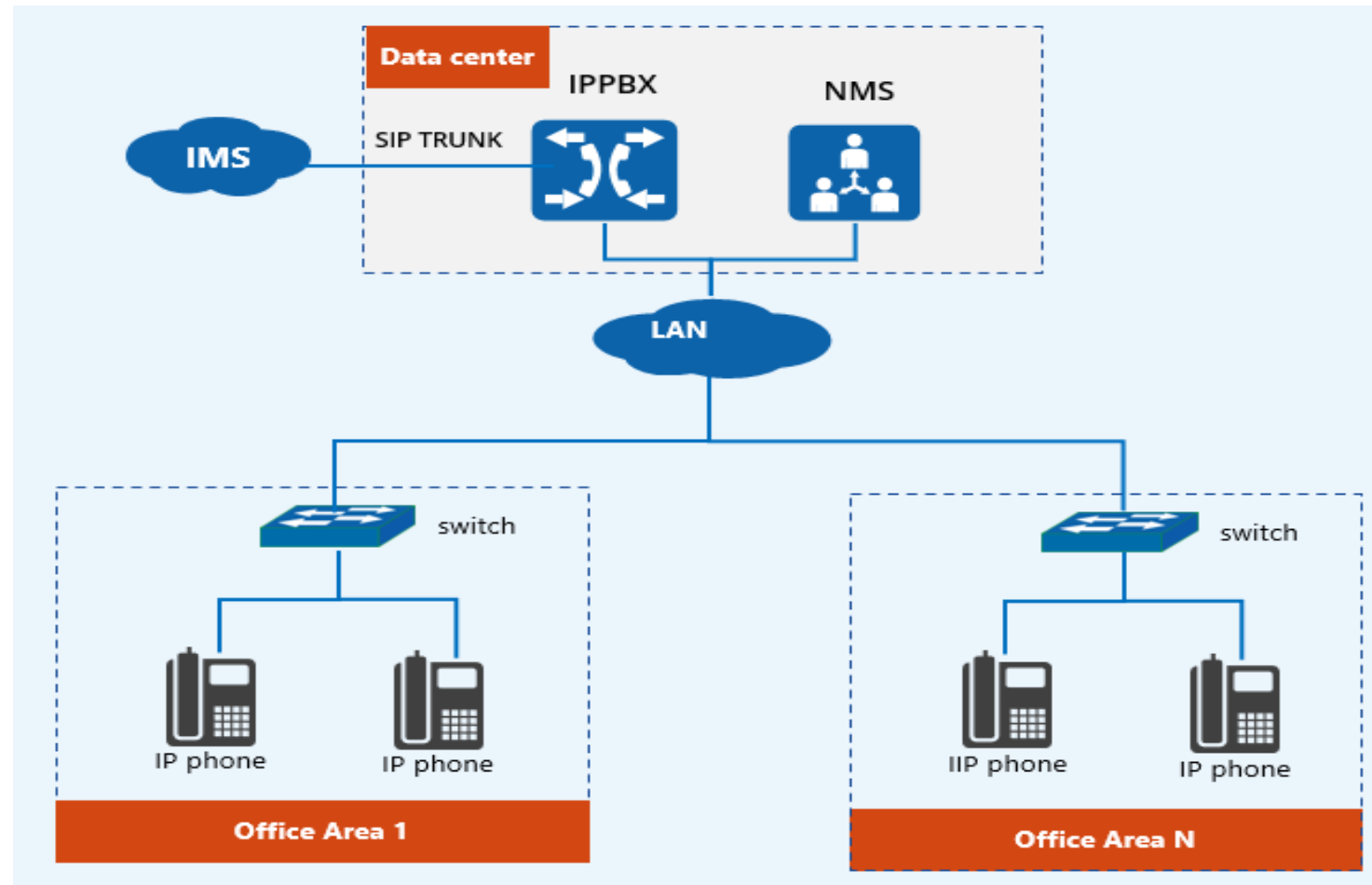
The screenshot shows the 'SIP Server' configuration page. On the left sidebar, 'SIP Server' is highlighted with a red box and a circled '1'. In the main content area, the 'SIP Server' section is highlighted with a red box and a circled '2'. The configuration fields are as follows:

SIP Server	
IP Protocol for SIP Stack	IPv4
SIP Server	172.27.10.14
SIP Server Port (Default: 5060)	5060
Registration Expires (Default: 300)	300 s
Heartbeat	<input type="checkbox"/> Enable
Primary Outbound Proxy	
Primary Outbound Proxy Address	
Primary Outbound Proxy Port	5060
Secondary Outbound Proxy	
Secondary Outbound Proxy Address	
Secondary Outbound Proxy Port	5060

The screenshot shows the 'Port Modify' configuration page. On the left sidebar, 'Port' is highlighted with a red box and a circled '3'. In the main content area, the 'Port' section is highlighted with a red box and a circled '4'. The configuration fields are as follows:

Port Modify	
Port	0
Disable Port	<input type="checkbox"/>
Registration	<input checked="" type="checkbox"/> Enable
IP Profile	0 <default>
Tel Profile	0 <default>
Display Name	6005
SIP User ID	6005
Authenticate ID	6005
Authenticate Password	*****
Offhook Auto-Dial	
Auto-Dial Delay Time	0 s
DND(Do Not Disturb)	<input type="checkbox"/> Enable

Common Application 2



SIP Trunk Call Configuration

SIP Extension Setting

SIP Trunk Setting

Route Setting

1. Click **Extension & Call Group->SIP Extension**
2. Create a new SIP extension, configure the extension number and password
3. Select network port

The screenshot displays the 'Edit SIP Extension' configuration page. The left sidebar contains a navigation menu with the following items: Status, Trunk & Route, Extension & Call Group (expanded), SIP Extension (circled with a red '1'), FXS, Phones, Ring Group, Intercom/Paging Group, Advanced Service, PBX Global Settings, Address Book, CDR & Recording, System, Maintenance, and Service Integrations. The main content area is titled 'Edit SIP Extension' and has three tabs: 'SIP Extension', 'User Info', and 'SIP Phone'. The 'SIP Extension' tab is selected, showing 'Basic Settings'. A red box labeled '2' highlights the following fields: 'Display Name' (value: 1000), 'Extension' (value: 1000), 'SIP Password' (masked with dots), and 'App Password' (value: 12345678). Below these are 'Classification Tag', 'DID', and 'Outbound CID' fields. At the bottom, the 'SIP Profile' dropdown is circled with a red '3' and shows the value '1-< GE0_Default >'. The 'Status' toggle at the top is turned on.

SIP Trunk Call Configuration

SIP Extension Setting

SIP Trunk Setting

Route Setting

1. Click **Trunk & Route**->**SIP Trunk**
2. Custom trunk name
3. Fill in the IP and port of the operator, select the protocol
4. Select network port

The screenshot displays the 'New SIP Trunk' configuration page. The left sidebar has a menu with 'SIP Trunk' highlighted. The main configuration area includes the following fields and settings:

- Status:** A toggle switch is turned on.
- Index:** A dropdown menu showing '12'.
- Name:** A text input field containing 'ims1'.
- Address:** A text input field containing '172.27.1.31'.
- Port:** A text input field containing '5060'.
- Outbound Proxy:** An empty text input field.
- Port:** An empty text input field.
- Transport:** A dropdown menu showing 'UDP'.
- Register:** A toggle switch is turned on.
- From Header User Part:** A dropdown menu showing 'Caller's Number'.
- From Header Display Name:** A dropdown menu showing 'Caller's Number'.
- From Header Host:** A dropdown menu showing 'Local Address'.
- Heartbeat:** A toggle switch is turned on.
- AutoCLIP Profile:** A dropdown menu showing 'Off'.
- DNIS:** A toggle switch is turned on.
- Local Ringtone Policy:** A dropdown menu showing 'Off'.
- SIP Profile:** A dropdown menu showing '1-< GE0_Default >'.

SIP Trunk Call Configuration

SIP Extension Setting

SIP Trunk Setting

Route Setting

1. Click **Trunk & Route**→**Route**

2. Custom route name

3. Select call source/SIP extension

4. Select call Destination/SIP Trunk

Priority	Name	Source	Num Match	Caller Prefix	Called Prefix	Time	Action: Manipulation/Dest	Followup: Manipulation/Dest
1	out	Local Extension	Off	Any	Any	Off	Off / ims1	Not Config
2	in	ims1	Off	Any	Any	Off	Off / Local Extension	Not Config
247	zhuce_IVR	Local Extension	Off	Any	Any	Off	Off / IVR / ->BIVR	Not Config
284	IVR	IVR	Off	Any	Any	Off	Off / IVR / ->BIVR	Not Config
285	FXO_IN	FXO	Off	Any	Any	Off	Off / Local Extension	Not Config

New Route

Priority: 288

Name:

Source:

Condition:

Number Matching:

Caller Number Prefix:

Called Number Prefix:

Time Profile:

Action:

Callback:

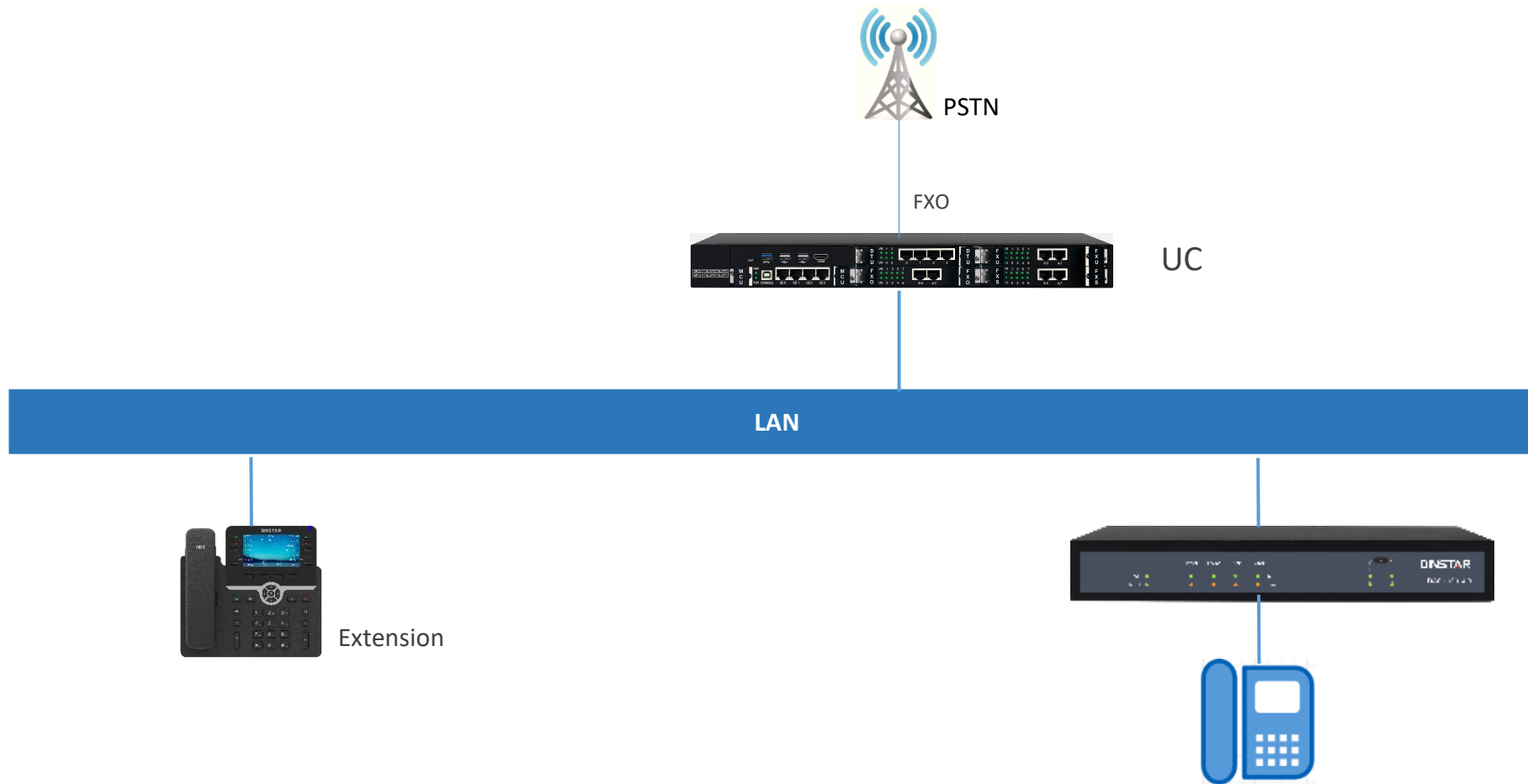
Distinctive Ringtone/Alert-Info:

Manipulation:

Destination:

Password Type:

Common Application 3



FXO Trunk Call Configuration

SIP Extension Setting

FXO Trunk Setting

Route Setting

1. Click **Extension & Call Group->SIP Extension**
2. Create a new SIP extension, configure the extension number and password
3. Select network port

The screenshot displays the 'Edit SIP Extension' configuration page in the DINSTAR web interface. The left sidebar shows the navigation menu with 'SIP Extension' selected, indicated by a red box and a circled '1'. The main content area is titled 'Edit SIP Extension' and has three tabs: 'SIP Extension', 'User Info', and 'SIP Phone'. The 'SIP Extension' tab is active, showing the 'Basic Settings' section. A red box highlights the 'Display Name', 'Extension', 'SIP Password', and 'App Password' fields, with a circled '2' next to it. The 'SIP Profile' dropdown is also highlighted with a red box and a circled '3', showing the selected value '1-< GE0_Default >'. Other fields include 'Status' (toggle), 'Index' (dropdown), 'Classification Tag', 'DID', and 'Outbound CID'.

FXO Trunk Call Configuration

SIP Extension Setting

FXO Trunk Setting

Route Setting

1. Click **Trunk & Route->FXO Trunk**
2. Select the slot and port where the FXO slot is located
3. Fill in the number and autodial number

Trunk & Route

- FXO
- E1/T1
- Number Matching
- Manipulation
- Route
- Emergency Number
- PIN List
- Blocked/Allowed Numbers
- AutoCLIP
- SMS Route

Extension & Call Group

Advanced Service

PBX Global Settings

Address Book

CDR & Recording

System

Maintenance

New FXO Trunk

Basic Settings

Status: ☒

Slot: 0

Port: 0

Name:

Number: 075561919966

Autodial Number: 9966

Advanced Settings

AutoCLIP Profile: Off

Work Mode: Voice

Voice Output Mode: Telephone

Gain Configure Mode: General Settings

TX Gain(P->PSTN): +4dB

RX Gain(PSTN->IP): 0dB

Impedance: 600 Ohm

Hybrid: 0

FXO Trunk Call Configuration

SIP Extension Setting

FXO Trunk Setting

Route Setting

1. Click **Trunk & Route**→**Route**

2. Custom route name

3. Select call source/SIP extension

4. Select call Destination/FXO Trunk

Route

Filter By Tag

Priority	Name	Source	Num Match	Caller Prefix	Called Prefix	Time	Action: Manipulation/Dest	Fa
1	out	Local Extension	Off	Any	Any	Off	Off / FXO / Slot 0 / Port 0	Ni
2	in	FXO / Slot 0 / Port 0	Off	Any	Any	Off	Off / Local Extension	Ni
247	zhuce_IVR	Local Extension	Off	Any	Any	Off	Off / IVR / —@IVR	Ni
284	IVR	IVR	Off	Any	Any	Off	Off / IVR / —@IVR	Ni
285	FXO IN	FXO	Off	Any	Any	Off	Off / Local Extension	Ni

Edit Route

Priority: 1

Name: out

Classification Tag:

Condition

Source: Local Extension

Number Matching: Off

Caller Number Prefix:

Called Number Prefix:

Time Profile: Any

Action

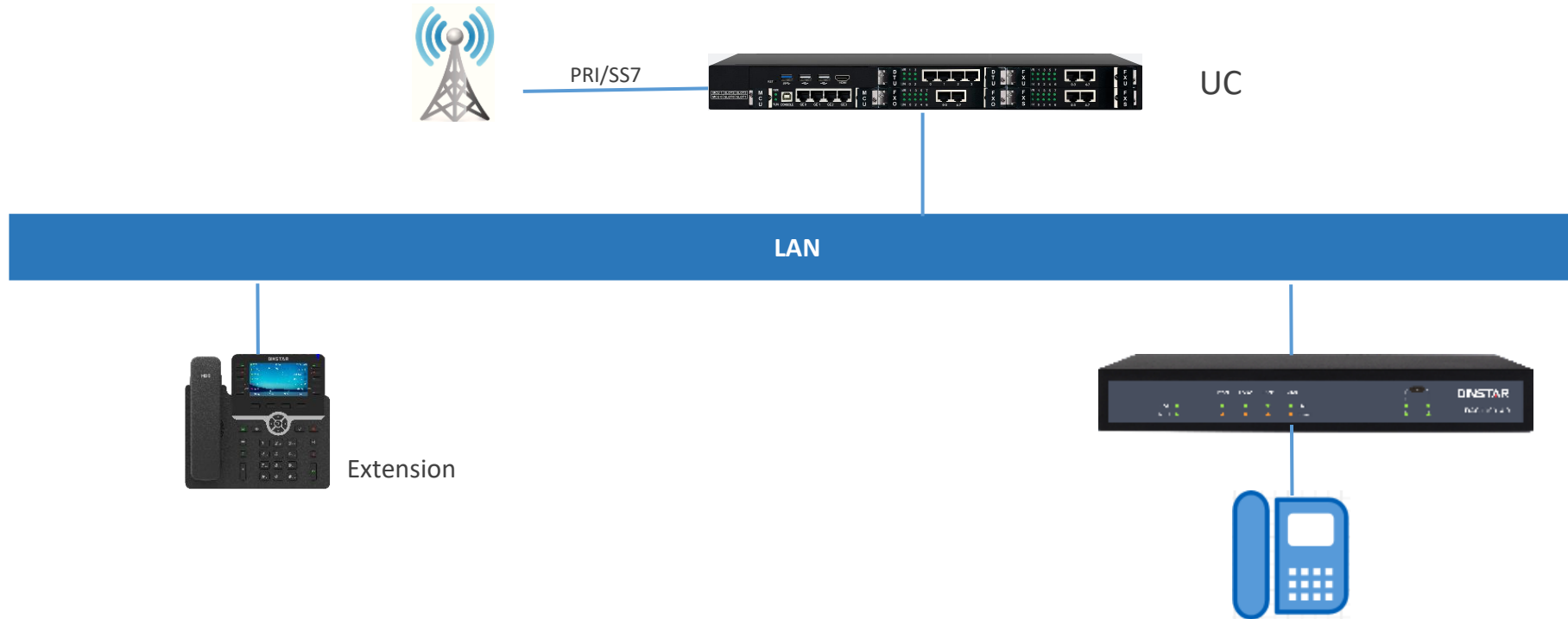
Callback: None

Distinctive Ringtone(Alert-Info):

Manipulation: Off

Destination: FXO Trunk / Slot 0 / Port 0

Common Application 4



E1 Trunk Call Configuration

SIP Extension
Setting

PRI Configuration
(optional)

SS7 Configuration
(optional)

Route Setting

1. Click **Extension & Call Group->SIP Extension**
2. Create a new SIP extension, configure the extension number and password
3. Select network port

The screenshot displays the 'Edit SIP Extension' configuration page in the DINSTAR web interface. The left sidebar contains a navigation menu with the following items: Status, Trunk & Route, Extension & Call Group, SIP Extension (highlighted with a red box and a circled '1'), FXS, Phones, Ring Group, Intercom/Paging Group, Advanced Service, PBX Global Settings, Address Book, CDR & Recording, System, Maintenance, and Service Integrations. The main content area is titled 'Edit SIP Extension' and has three tabs: SIP Extension (selected), User Info, and SIP Phone. Under the 'SIP Extension' tab, the 'Basic Settings' section includes the following fields: Status (a toggle switch), Index (a dropdown menu set to '1'), Display Name (a text input field), Extension (a text input field set to '1000', highlighted with a red box and a circled '2'), SIP Password (a masked text input field), App Password (a text input field set to '12345678'), Classification Tag (a text input field), DID (a text input field with a green plus icon), Outbound CID (a text input field), and SIP Profile (a dropdown menu set to '1-< GE0_Default >', highlighted with a red box and a circled '3').

E1 Trunk Call Configuration

SIP Extension
Setting

PRI Configuration
(optional)

SS7 Configuration
(optional)

Route Setting

1. Click **Trunk & Route** -> **E1/T1**

2. Custom name, select slot, type is
PRI

3. Select switch side

Trunk & Route

- SIP Trunk
- FXO
- E1/T1** ①
- Number Matching
- Manipulation
- Route
- Emergency Number
- PIN List
- Blocked/Allowed Numbers
- AutoCLIP
- SMS Route
- Extension & Call Group >
- Advanced Service >

Index 1

Name	1
Slot	1
Type	PRI

PRI Trunk

Port	0
Protocol	ISDN
Switch Side	User Side
Alerting Indication	ALERTING

Others

E1 Trunk Call Configuration

SIP Extension
Setting

PRI Configuration
(optional)

SS7 Configuration
(optional)

Route Setting

1. Click **Trunk & Route**->**E1/T1**

2. Custom name, select slot, type is SS7

3. Configure SS7 trunk

4. Configure ss7 MTP link

5. Configure CIC

Tip:

OPC、DPC、MTP link, CIC related
parameter configuration is consistent with
the operator

The screenshot shows the 'Trunk & Route' configuration page. The left sidebar has a menu with 'E1/T1' highlighted and circled with a red '1'. The main area is divided into three sections: 'SS7 Trunk', 'SS7 MTP Link', and 'SS7 MTP Link'. In the 'SS7 Trunk' section, 'Slot' is circled with a red '2' and 'Type' is circled with a red '3'. In the 'SS7 MTP Link' section, 'E1/T1 No' is circled with a red '4' and 'Start CIC No' is circled with a red '5'. The 'SS7 Trunk' section also includes fields for 'Protocol' (ITU-CHINA), 'Protocol Type' (ISUP), 'SPC Format' (24bits(8-8-8)), 'OPC' (8-8-8), 'DPC' (9-9-9), 'Support APC' (toggle), 'Network Indicator' (National Network), and 'Sending SLTM' (toggle).

E1 Trunk Call Configuration

PRI Configuration
(optional)

SS7 Configuration
(optional)

SIP Trunk Setting

Route Setting

1. Click **Trunk & Route** -> **Route**
2. Select call source/SIP extension
3. Select call Destination/ E1/T1 Trunk
4. Configure incoming routing using the same method

Trunk & Route

Route Group

Priority	Name	Source	Num Match	Caller Prefix	Called Prefix	Time	Action: Manipulation/Des
295	to_uc	Local Extension	Off	Any	Any	Off	Off / E1/T1 / Slot 2
296	uc_to	E1/T1 / Slot 2	Off	Any	Any	Off	Off / Local Extension
297	Bug	1108	Off	Any	Any	Off	Off / MTG200
298	to_uc	Custom	Off	Any	Any	Off	Off / to_uc
299	545	Local Extension	Off	Any	Any	Off	Off / 33322
300	outtoFXO	Local Extension	Off	Any	90	Off	0081 / 5000

Status

Trunk & Route

SIP Trunk

FXO

E1/T1

Number Matching

Manipulation

Route

Emergency Number

PIN List

Blocked/Allowed Numbers

AutoCLIP

SMS Route

Extension & Call Group

Advanced Service

PBX Global Settings

Address Book

CDR & Recording

System

Edit Route

Source

2

Select All

Source list 0/49

Any

SIP Trunk / to

SIP Trunk /

Trunk_ChinaTelecom

SIP Trunk /

Trunk_ChinaUnicom

SIP Trunk / to_uc

Select All Target list 0/1

Local Extension

Number Matching

Off

Caller Number Prefix

Called Number Prefix

Time Profile

Any

Action

Callback

Distinctive Ringtone(Alert-Info)

Manipulation

Off

Destination

3

E1/T1 Trunk / Slot 2

Password Type

Off

1

Chapter One The Way To Access Device

2

Chapter Two The UC Call Configuration Instructions

3

Chapter Three Common Function Configuration

Chapter Three

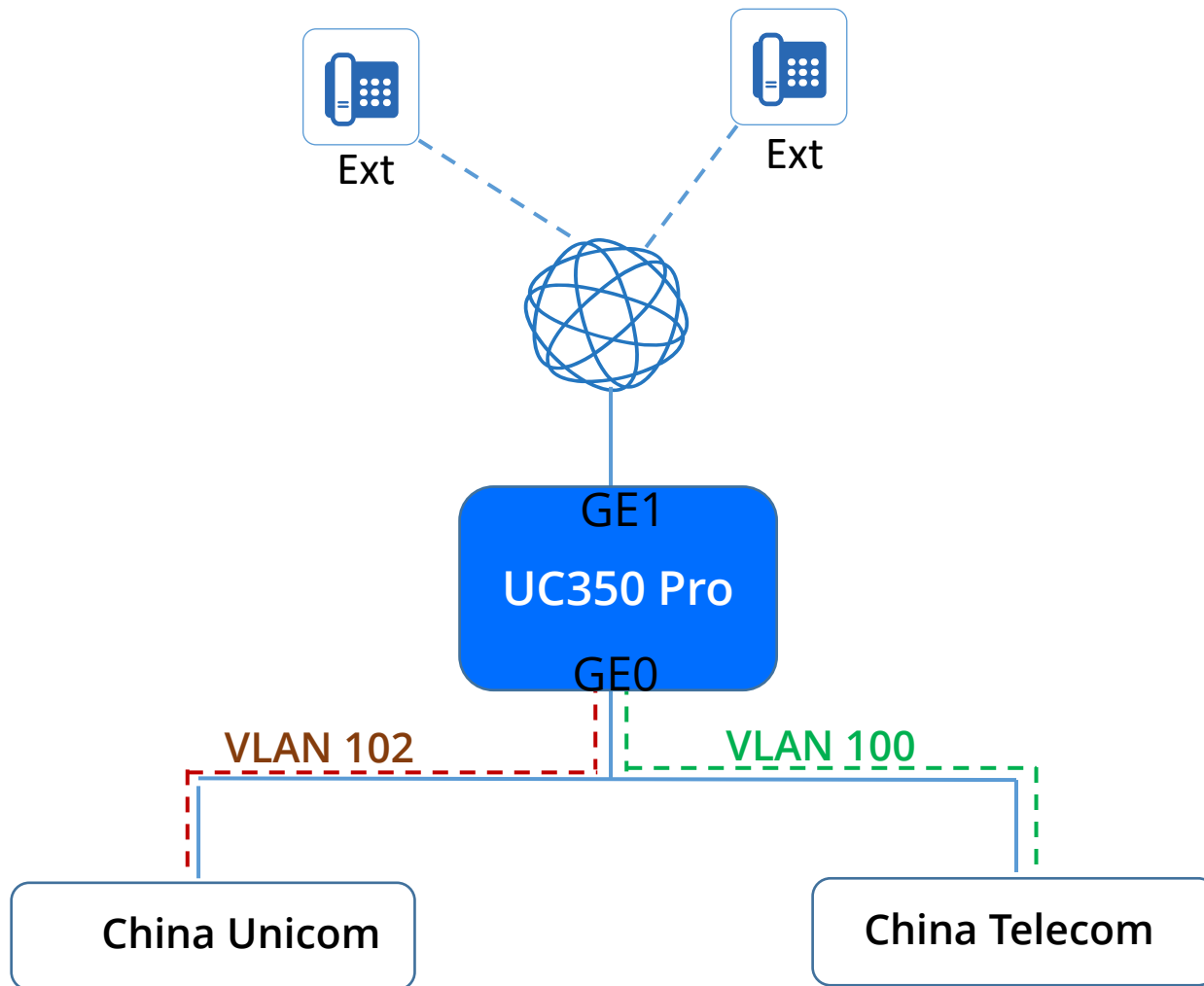
Common Function Configuration

03

1. VLAN
2. Call Queue
3. Paging
4. IVR
5. Recording

VLAN

- VLAN Scenario



VLAN

- Add VLAN Configuration

1. Click **System->Network**

2. Select VLAN Sub Interface

3. Click new

The screenshot shows the Dinstar web interface for network configuration. On the left, the 'System' menu is expanded, and 'Network' is highlighted with a red box and a circled '1'. The main content area shows the 'Network' section with the 'VLAN Sub Interface' tab selected, indicated by a red box and a circled '2'. In the top right corner of the main area, there is a 'New' button highlighted with a red box and a circled '3'. Below the tabs, a table lists the configured VLAN sub-interfaces.

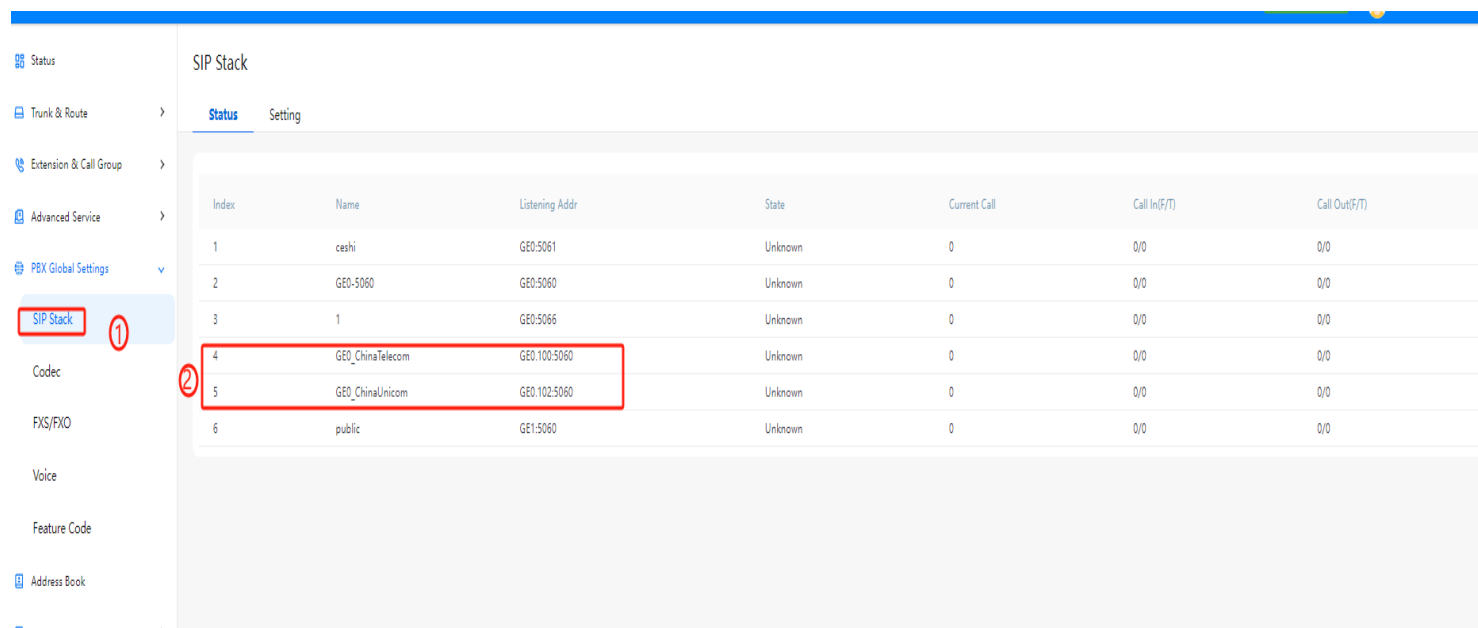
Name	MTU	IPv4 Address	Netmask	IPv4 Gateway	IPv4 DNS	
GE0.100	1500	52.115.191.57	255.255.255.252	52.115.191.58	8.8.8.8/4.4.4.4	More Edit Delete
GE0.102	1500	10.10.10.10	255.255.255.0	10.10.10.1	8.8.8.8/4.4.4.4	More Edit Delete

VLAN

- bonding VLAN

1. Click **PBX Global Settings -> SIP Stack**

2. Add the specific SIP Profile for each VLAN



Index	Name	Listening Addr	State	Current Call	Call In(F/T)	Call Out(F/T)
1	ceshi	GE0:5061	Unknown	0	0/0	0/0
2	GE0-5060	GE0-5060	Unknown	0	0/0	0/0
3	1	GE0-5066	Unknown	0	0/0	0/0
4	GE0_ChinaTelecom	GE0.100-5060	Unknown	0	0/0	0/0
5	GE0_ChinaUnicom	GE0.102-5060	Unknown	0	0/0	0/0
6	public	GE1-5060	Unknown	0	0/0	0/0

VLAN

- bonding VLAN

1. Click **Trunk & Route** -> **SIP Trunk**

2. Add the Trunk, SIP Profile choose the added Profile

SIP Trunk

Index	Name	Address	Transport	Register	Heartbeat	Status	Call In(F/T)	Call Out(F/T)	Profile
1	fio	8888@172.27.10.34:5060	UDP	On	Off	FAIL_WAIT	0/0	0/0	2-<GEO-5060>
2	Trunk_ChinaTelecom	52.115.191.1:5060	UDP	Off	Off	NOREG/UP	0/0	0/0	4-<GEO_ChinaTelecom>
3	Trunk_ChinaUnicom	10.10.10.8:5060	UDP	Off	Off	NOREG/UP	0/0	0/0	5-<GEO_ChinaUnicom>
4	to_uc	28@172.27.10.14:5060	UDP	On	Off	REGED/UP	0/2	2/2	2-<GEO-5060>
5	MTG2000	172.27.10.15:5060	UDP	Off	5	NOREG/DOWN	0/0	0/0	2-<GEO-5060>
6	test	8888@172.27.10.34:5060	UDP	On	Off	FAIL_WAIT	0/0	0/0	2-<GEO-5060>
7	33322	172.19.211.141:5062	UDP	Off	Off	NOREG/UP	0/0	1/1	2-<GEO-5060>
8	MTG200	172.27.10.31:5060	UDP	Off	Off	NOREG/UP	0/0	0/0	2-<GEO-5060>

Call Queue

- **Main Scenarios**

Limited number of customer service representatives

Seasonal and specific marketing activities bring about peak call periods

- **Main function**

When a customer calls into the queue, the call will be intelligently assigned to an extension. If all extensions are busy, the call will be queued and wait



Call Queue

- **Configure Call Queue**

1. Click **Advanced Service->Call Queue**

2. Configure queue number, select type

3. Select call allocation policy

4. Select agent members

Edit Call Queue

Index: 2

Queue Name: test

Queue Number: 555

Type: Common Queue

Queue Settings

Call Allocation Policy: Simultaneous

Menu Tone

Waiting Music

Enable Position Announcement

Maximum Queue Time(0,300): 60

Queue Timeout Policy: Hangup

Max Queue Length: 0

Quantity Overlimit Policy: Hangup

Agent Settings

Agent Members

Simultaneous

Linear

Random

Memory Round Robin

Least Recent

Fewest Calls

Select All

Source list 0/88

SIP Extension / 72144 /

72144

SIP Extension / 72145 /

72145

SIP Extension / 72147 /

72147

SIP Extension / 72148 /

Select All

Target list 0/4

SIP Extension / 72139 /

72139

SIP Extension / 72140 /

72140

SIP Extension / 72141 /

72141

SIP Extension / 72142 /

Call Queue

- Parameters-Type

Common queue: corresponding to the current call queue function, suitable for scenarios where the phone serves as the agent terminal

Attendant console queue: used to support call center business. To use a web call center, this type of call queue must be created first

Edit Call Queue

Index	2
Queue Name	<input type="text" value="test"/>
Queue Number	<input type="text" value="000"/>
Type	<div>Common Queue Common Queue Attendant Console Queue</div>

Queue Settings

Call Queue

- **Parameter-call allocation policy**

Simultaneous: The agents ring together.

Linear: When there is no incoming call, a new user calls in, each time it will ring sequentially from the first agent.

Random: one is randomly selected for ringing.

Memory round robin: When there is no incoming call, a new user calls in, and the ringing starts from the next agent who hangs up last before.

Least recent: namely the time from the end of the agent's last call to the present, ringing in the order from longest to shortest time.

Fewest calls: The ringing starts from the least to the most according to the times of calls.

Edit Call Queue

Index	2
Queue Name	<input type="text" value="test"/>
Queue Number	<input type="text" value="888"/>
Type	<input type="text" value="Common Queue"/>

Queue Settings

Call Allocation Policy

Menu Tone

Waiting Music

Enable Position Announcement

Simultaneous	▼
Simultaneous	
Linear	
Random	
Memory Round Robin	
Least Recent	
Fewest Calls	



Paging

- **Main Scenarios**

In hospitals, it can be used for emergency calls and cross departmental information synchronization

At school, it can be used for opening ceremonies and daily notifications

In shopping malls, used for promotional activities and safety management

- **Main function**

Implement one to many communication



Paging

- **Configure Paging**

1. Click **Extension & Call Group**
Group>Intercom/Paging Group

2. Configure name and number

3. Select Strategy

4. Select members

The screenshot displays the 'Edit Paging Group' interface. On the left is a sidebar menu with options: Status, Trunk & Route, Extension & Call Group (selected), SIP Extension, FXS, Phones, Ring Group, Intercom/Paging Group (highlighted with a red box and '1'), Advanced Service, PBX Global Settings, Address Book, CDR & Recording, System, Maintenance, and Service Integrations. The main area is titled 'Edit Paging Group' and contains the following fields and controls:

- Index:** A dropdown menu set to '1'.
- Name:** A text input field containing 'test'.
- Intercom/Paging Group Number:** A text input field containing '888', highlighted with a red box and '2'.
- Strategy:** A dropdown menu set to '1-way Paging', highlighted with a red box and '3'.
- Members Select:** A button to select members, highlighted with a red box and '4'.
- Members List:** Two panels showing a list of SIP extensions. The left panel is titled 'Source list 0/91' and the right panel is titled 'Target list 0/3'. Both lists contain entries like 'SIP Extension / 72141 / 72141', 'SIP Extension / 72143 / 72143', etc.
- Specifies Caller Number:** A toggle switch, currently turned on.
- Verify PIN Code:** A toggle switch, currently turned on.
- Media Play:** A dropdown menu set to 'Off'.
- Timing Trigger:** A toggle switch, currently turned on.

Paging

- Parameters-Strategy

1-way Paging: members of the paging group only can listen to the voice of presenter and cannot answer the call

2-way Intercom: members of the paging group can have conversation with the presenter, but members cannot talk to each other

Edit Paging Group

Index	1
Name	test
Intercom/Paging Group Number	888
Strategy	1-way Paging
Members Select	<input type="checkbox"/> Select All

IVR

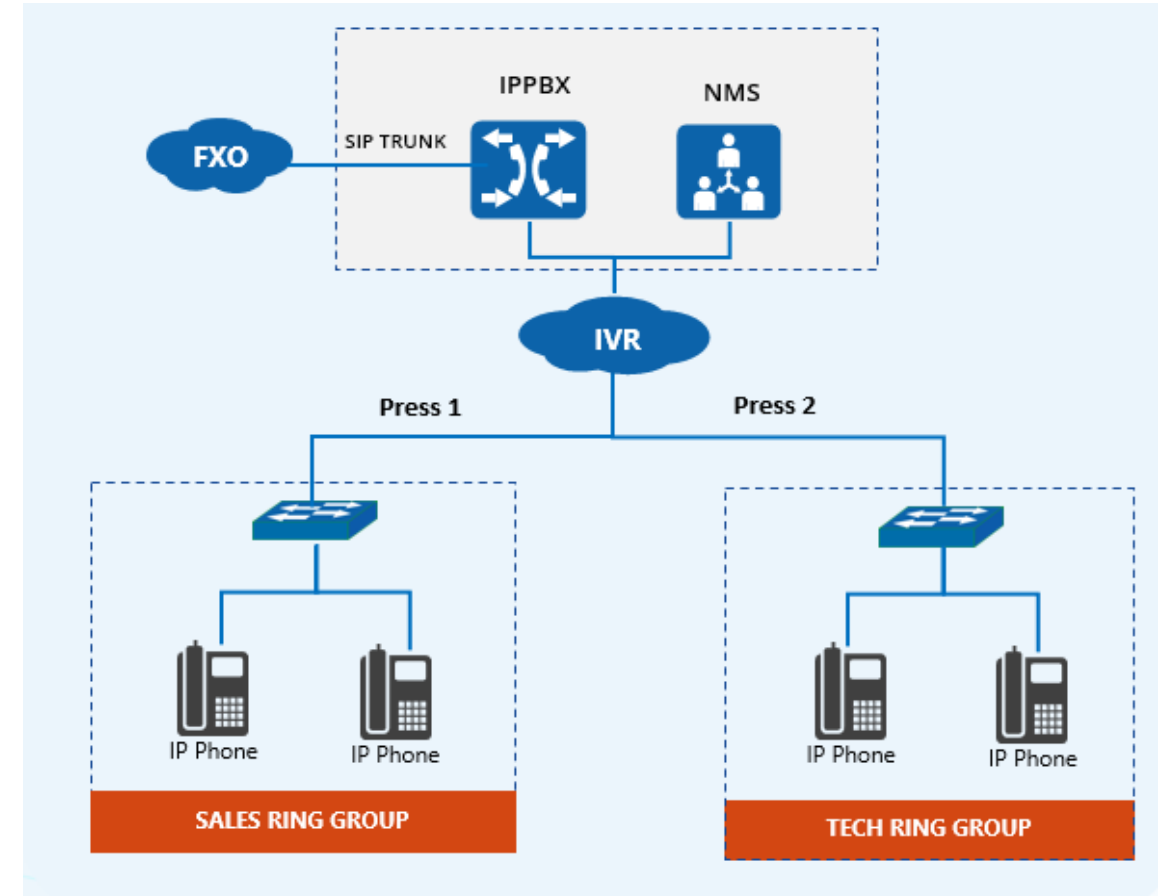
- Main function

Auto answer: The system automatically answers the call and provides a voice menu.

User interaction: The user selects menu options by voice or by pressing buttons.

Information query: Provide account balance, order status and other information.

Call forwarding: Transfers the call to the corresponding department or personnel as selected by the user.



- IVR Configuration

Ring group setting

Upload IVR file

IVR Menu Hints setting

IVR Menu setting

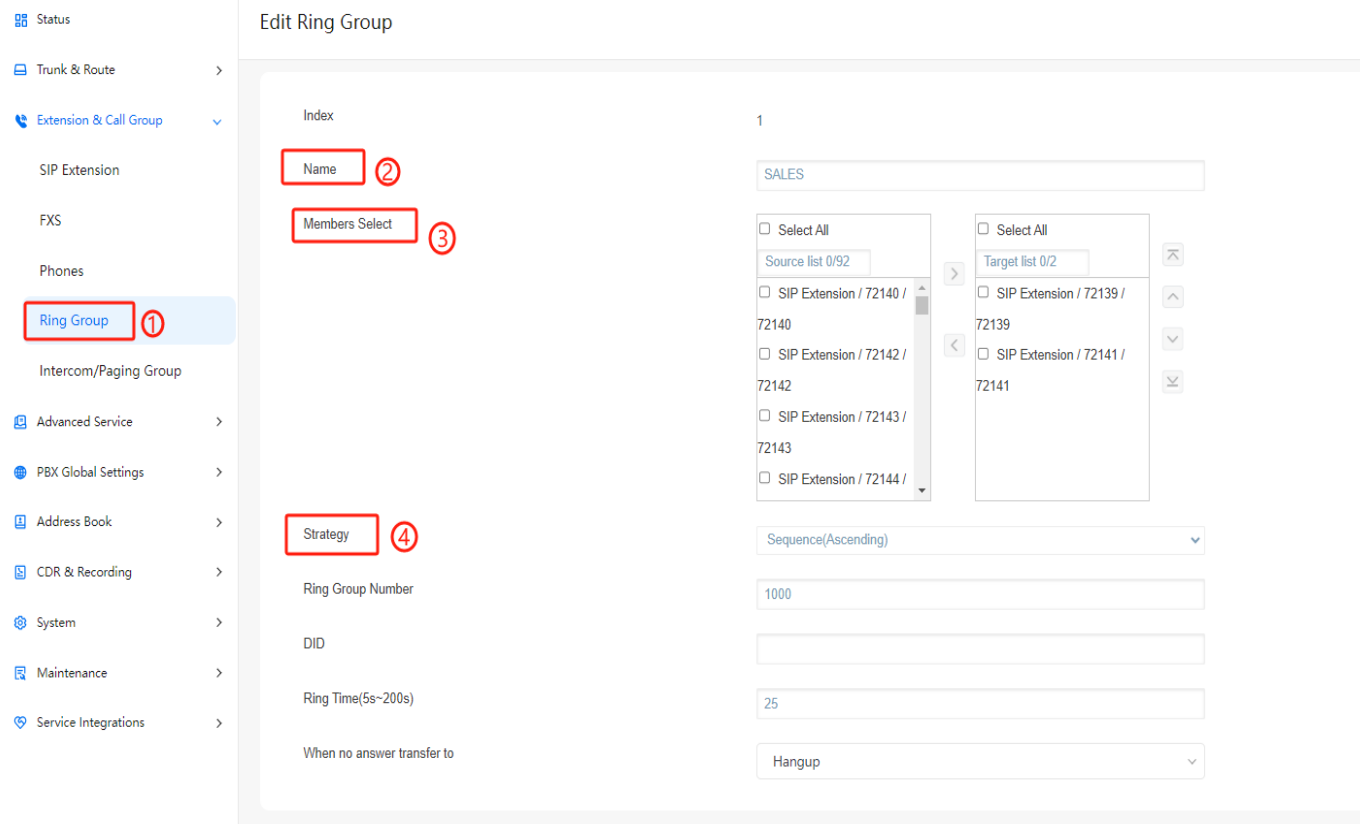
1. Click **Extension & Call Group->Ring Group**

2. Set ring group name

3. Select the extension numbers of the ringing group members

4. Select Strategy : Sequence (Ascending), Sequence (Cyclic Ascending), Simultaneous and Random

5. Create other ringing groups using the same method



Status

Trunk & Route

Extension & Call Group

SIP Extension

FXS

Phones

Ring Group

Intercom/Paging Group

Advanced Service

PBX Global Settings

Address Book

CDR & Recording

System

Maintenance

Service Integrations

Edit Ring Group

Index

Name

Members Select

Strategy

Ring Group Number

DID

Ring Time(5s~200s)

When no answer transfer to

SALES

Select All

Source list 0/92

SIP Extension / 72140 / 72140

SIP Extension / 72142 / 72142

SIP Extension / 72143 / 72143

SIP Extension / 72144 /

Select All

Target list 0/2

SIP Extension / 72139 / 72139

SIP Extension / 72141 / 72141

Sequence(Ascending)

1000

25

Hangup

- IVR Configuration



1. Click **PBX Global Settings->Voice**
2. Select the upload file type, set the name and description
3. Choose the prepared voice file and click the upload

Note: the format of the wav audio file uploaded must be : **monaural, 8000hz, 16bit, and size of no more than 3M**

①

②

③

The format of wav audio file should be monaural, 8000hz, 16bit, and a size of no more than 3MB.

Type	Name	Description	Storage Location	Operation
Waiting Music	default waiting music	Default waiting/hold music, will play repeatedly	Local	
IVR	default ivr	Default IVR welcome audio	Local	

IVR menu menu Local wav Upload

- IVR Configuration

Ring Group Setting

Upload IVR file

IVR Menu Hints setting

IVR Menu setting

1. Click **Advanced Service->IVR**

2. Customize the name of the IVR

3. Set greeting tone and menu tone: When a call comes to the IVR, play the greeting tone first and then the menu tone

4. Set Repeat Loops: it is 3 in default. the call will be hung up after the IVR has been repeated for three times during timeout

5. Set Repeat Policy : It can be configured with "Greeting Tone + Menu Tone" or "Menu Tone"

- IVR Configuration



1. Set DTMF: select the number of the destination
2. Set Tone: The tone that is played before the callee rings, Default is off
3. Select Destination and the corresponding ringing group

Menu			
DTMF	Tone	Destination	
1	Off	Ring Group	Ring Group / SALES
2	Off	Ring Group	Ring Group / TECH

Recording

- View & Set Recording Rules

Recording Mode: local recording or streaming recording

local recording : Select the storage location for the recording, which can be either local or Udisk

streaming recording: Configure the recording server address, and the recording will be uploaded to the server

Edit& New: can adjust parameters such as strategy, recording direction, recording object, time, etc

CDRs

CDRs **Recording**

Recording Mode: Local Recording

Master Storage Location: Local

Slave Storage Location: Local

Index	Name	Strategy	Recording Direction	Stereo	Min Duration(s)	Silence Detect
1	auto_record	Auto Recording After Answer	Inbound & Outbound	Off	1	Off/-/-/-
2	manual_record	Manual Recording After Answer	Inbound & Outbound	Off	1	Off/-/-/-

Recording

- Recording configuration

SIP extension/FXS extension: After enabling recording, extension calls will be recorded according to the selected recording rules

Routing: After activation, calls that match the routing are recorded according to the selected rules

The image displays two screenshots of the Dinstar PBX web interface, illustrating the configuration steps for recording.

Top Screenshot: Edit SIP Extension

- The left sidebar menu shows "SIP Extension" highlighted with a red box and a circled "1".
- The main content area is titled "Edit SIP Extension" and has tabs for "SIP Extension", "User Info", and "SIP Phone".
- Under the "SIP Extension" tab, various settings are listed. "Recording Profile" is highlighted with a red box and a circled "2". It is currently set to "1-< auto_record >".
- Other settings include "Call Forward Unconditional" (Off), "Call Forward Unregister" (Off), "Call Forward Busy" (Off), "Call Forward No Reply" (Off), "Call Back When Dest Ext Busy" (On), "Priority" (Normal), "Ringtone" (Off), "Ring Timeout(s)" (50), "Allow Being Monitored" (On), "Monitor Mode" (Disable), and "Voicemail" (On).




Bottom Screenshot: New Route

- The left sidebar menu shows "Route" highlighted with a red box and a circled "1".
- The main content area is titled "New Route".
- Under the "New Route" tab, settings for "Number Matching", "Caller Number Prefix", "Called Number Prefix", and "Time Profile" are shown. "Number Matching" is set to "Off".
- The "Action" section is expanded, showing "Recording Profile" highlighted with a red box and a circled "2". It is currently set to "1-< auto_record >".
- Other settings in the "Action" section include "Callback" (On), "Distinctive Ringtone(Alert-Info)" (None), "Manipulation" (Off), "Destination" (IVR / IVR), "Password Type" (Off), and "Failover Action" (On).

Recording

- Record viewing

View recording files on the
CDR & Recording->CDR

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

Status

Trunk & Route

Extension & Call Group

Advanced Service

PBX Global Settings

Address Book

CDR & Recording

CDRs

Recording

Query Param



Expand

CDRs List

Empty

Export

Export Queried

ID	Caller	Source	Called	Destination	Start Time	End Time	Dur...	Hangup By	Codec	Hangup Cause	Operation	Iter
1	500	SIP Extension/500	501	SIP Extensi...	2025-02-17 ...	2025-02-17 ...	00...	Called	PCMA	Normal Clean...		
2	500	SIP Extension/500	501	SIP Extensi...	2025-02-17 ...	2025-02-17 ...	00...	Caller	PCMA	Normal Clean...		
3	500	SIP Extension/500	501	SIP Extensi...	2025-02-17 ...	2025-02-17 ...	00...	Caller	PCMA	Normal Clean...		
4	500	SIP Extension/500	501	SIP Extensi...	2025-02-17 ...	2025-02-17 ...	00...	Caller	PCMA	Normal Clean...		

CDRs 1

2



THANKS



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