

# UC Common Function Configuration Guide

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# Foreword

This course is mainly:

- Describe common function of Dinstar IPPBX

- Introduce how to configure common function

- Introduce how to configure routing

- Introduce the NAT Public Proxy Server and share setting

# Course Objective

Through this course  
you will be able to



Understand and know common function



Learn how to configure routing



Understand how to use Nat Public Proxy Server

# Contents

- 1 Common function
- 2 Routing Configuration
- 3 Nat Public Proxy Server

# Common function

01

1. Call Queue

2. Paging

3. Follow Me

4. SCA

# Why Need 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



# How To Configure Call Queue

1. Click Advanced Service->Call Queue

2. Configure queue number, select type

3. Select call allocation policy

4. Select agent members

The screenshot displays the 'Edit Call Queue' interface in the DINSTAR system. The left sidebar contains a navigation menu with the following items: Status, Trunk & Route, Extension & Call Group, Advanced Service (highlighted with a red box and a circled '1'), IVR, Conference, Voicemail, Speed Dial, Dialplan, Follow Me, SCA, Alarm Clock, PBX Global Settings, Address Book, CDR & Recording, System, Maintenance, and Service Integrations. The main content area is titled 'Edit Call Queue' and includes an 'Index' section with a page number '2'. Below this, there are three main sections: 'Queue Settings', 'Agent Settings', and 'Agent Members'. The 'Queue Settings' section contains fields for 'Queue Name' (test), 'Queue Number' (888, highlighted with a red box and a circled '2'), 'Type' (Common Queue), 'Call Allocation Policy' (highlighted with a red box and a circled '3'), 'Menu Tone', 'Waiting Music', 'Enable Position Announcement', 'Maximum Queue Time(0,300)', 'Queue Timeout Policy', 'Max Queue Length', and 'Quantity Overlimit Policy'. The 'Agent Settings' section is currently empty. The 'Agent Members' section (highlighted with a red box and a circled '4') shows two lists of agents: 'Source list 0/88' and 'Target list 0/4'. The 'Source list' contains SIP extensions 72144, 72146, 72147, and 72148. The 'Target list' contains SIP extensions 72139, 72140, 72141, and 72142. A dropdown menu for 'Simultaneous' is also visible, showing options like Linear, Random, Memory Round Robin, Least Recent, and Fewest Calls.



# How To Configure 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="888"/>
Type	<div>Common Queue Common Queue Attendant Console Queue</div>

Queue Settings



# How To Configure 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	



# Why Need 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



# How To Configure Paging

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

2. Configure name and number

3. Select Strategy

4. Select members

The screenshot displays the 'Edit Paging Group' configuration page in the DINSTAR web interface. The left sidebar shows the navigation menu with 'Intercom/Paging Group' selected, indicated by a red box and a circled '1'. The main configuration area contains the following fields and options:

- Index:** A dropdown menu set to '1'.
- Name:** A text input field containing 'test', highlighted with a red box and a circled '2'.
- Intercom/Paging Group Number:** A text input field containing '888', highlighted with a red box and a circled '2'.
- Strategy:** A dropdown menu set to '1-way Paging', highlighted with a red box and a circled '3'.
- Members Select:** A section with two lists of SIP extensions, each with a 'Select All' checkbox and a list of individual extensions. The first list is labeled 'Source list 0/91' and the second is 'Target list 0/3'. Both lists are highlighted with red boxes and circled '4'.
- Specifies Caller Number:** A toggle switch set to 'On'.
- Verify PIN Code:** A toggle switch set to 'On'.
- Media Play:** A dropdown menu set to 'Off'.
- Timing Trigger:** A toggle switch set to 'On'.

# How To Configure 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

# Why Need follow me

- **Main Scenarios**

The user has multiple workstations

Users cannot receive calls from workstations during business trips or outings

- **Main function**

Unify various commonly used communication numbers (mobile phone, office phone, residential phone) into an extension number, so that anyone can find the user by simply dialing this number



# How To Configure follow me

1. Click Advanced Service- >Follow Me

2. Select extension number、ring strategy and ring time

3. Configure destination, It can be an extension number or a mobile phone number

**Edit Follow Me**

Status ☒

Index 1

Extension Number SIP Extension / 72140 / 72140

Ring Strategy **Simultaneous** 2

Ring Time(5s~200s) 5

**Destination List**

Time	Destination
Any	SIP Extension / 72139 / 72139
Any	FXO Trunk / Slot 0 / Port 0 1775267412852

Number only could use 0-9,a-Z or +/\*/# or /, Max length is 32

# How To Configure follow me

- Parameters-Ring Strategy

**simultaneous:** all numbers ring together

**sequence (ascending ):** starts from the extension and rings from top to bottom

The screenshot shows a configuration panel for 'follow me' with the following fields:

- Status:** A toggle switch that is currently turned on (blue).
- Index:** A text field containing the value '1'.
- Extension Number:** A dropdown menu showing 'SIP Extension / 72140 / 72140'.
- Ring Strategy:** A dropdown menu with a red rectangular highlight around its label. The menu is open, showing three options: 'Simultaneous' (highlighted in dark blue), 'Simultaneous', and 'Sequence(Ascending)'.
- Ring Time(5s~200s):** A text field that is currently empty.



# How To Configure follow me

- Parameters-Destination List

**Time:** Make calls within the specified time period after setting up

**Destination:** users can choose SIP extension, SIP Trunk, FXO Trunk, and E1/T1 Trunk. When selecting a Trunk, users can configure a phone number

The screenshot shows the 'Destination List' configuration window. It has two main sections: 'Time' and 'Destination'. The 'Time' section has two dropdown menus, both currently set to 'Any'. Below them is a red error message: 'Number only could use 0-9,a-z or +/\*/# or /, Max length is 32'. The 'Destination' section has a dropdown menu currently showing 'SIP Extension / 72139 / 72139'. A list of options is open, showing various SIP extensions (e.g., 72219, 72220, 72221, 72222, 72223, 72224, 72225, 72226, 72227, 72228, 72229, 72230, 72231, 72232, 72233, 72234, 72235), 'SIP Trunk / test', 'FXO Trunk / Slot 0 / Port 0', and 'E1/T1 Trunk / Slot 3'. The 'SIP Extension / 72230 / 72230' option is highlighted. To the right of the dropdown is a text input field containing '1775267412852', with a red 'X' icon and a green plus icon to its right.

# Why Need SCA

- **Main Scenarios**

Managers need to focus on more important tasks and delegate daily phone conversations to their secretaries, which improves their work efficiency

- **Main function**

All calls from the call manager will be forwarded to the secretary extension, filtered by the secretary extension, and then forwarded to the manager extension

Support multiple secretary extensions



# How To Configure SCA

1. Click Advanced Service- >SCA

2. Select Manager Number and customize manager private number

3. Select Secretary Number and customize Secretary private number

**Edit SCA**

Index: 1

Name	72140
Manager Number	SIP Extension / 72140 / 72140
Private Number	8
Enable Manager Ring	<input checked="" type="checkbox"/>
Enable Multiple Call	<input checked="" type="checkbox"/>
Status	Enable

**Secretary List**

Private Number	Secretary
0	SIP Extension / 72139 / 72139

Private number cannot be the same, Secretaries cannot be the same

# How To Configure SCA

- **Parameters-Manager Number**

The manager number must be a SIP extension number that enables the SCA function

## Edit SIP Extension

### SIP Extension

### User Info

### SIP Phone

Index	2
Display Name	72140
Extension	72140
SIP Password	.....
App Password	.....
Classification Tag	
DID	
Outbound CID	
SIP Profile	2-< GE0 >

### Extended Service

DinLink Client	<input checked="" type="checkbox"/>
Speed Dial	Off
SCA	<input checked="" type="checkbox"/>

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- 2 Routing Configuration
- 3 Nat Public Proxy Server

# Routing Configuration

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1.How To configure routing

2.Use Case

# How To Configure Routing

1. Click Trunk & Route->Route

2. Custom route name

3. Select call source

4. Select call Destination

The screenshot shows the 'New Route' configuration page in the DINSTAR interface. The left sidebar contains a menu with the following items: Status, Trunk & Route (expanded), SIP Trunk, FXO, E1/T1, Number Matching, Manipulation, Route (highlighted with a red box and circled '1'), Emergency Number, PIN List, Blocked/Allowed Numbers, AutoCLIP, SMS Route, Extension & Call Group, Advanced Service, PBX Global Settings, Address Book, CDR & Recording, System, Maintenance, and Service Integrations. The main content area is titled 'New Route' and contains several sections: 'Priority' (set to 298), 'Name' (highlighted with a red box and circled '2'), 'Condition' (containing 'Source' highlighted with a red box and circled '3'), 'Number Matching' (set to Off), 'Caller Number Prefix' (empty), 'Called Number Prefix' (empty), 'Time Profile' (set to Any), 'Action' (containing 'Destination' highlighted with a red box and circled '4'), 'Callback' (set to None), 'Distinctive Ringtone(Alert-Info)' (set to Off), 'Manipulation' (set to Off), 'SIP Trunk / test' (set to SIP Trunk / test), and 'Password Type' (set to Off). The 'Source' list on the right shows a list of extensions (72141 to 72144) with 'Local Extension' selected.



# How To Configure Routing

- **Parameters- Source**

users can choose from local extension, SIP extension, SIP trunk, FXO trunk, e1/t1 trunk, and any

New Route

Condition

Source

☐ Select All  
Source list 0/99

☐ Any  
☐ Local Extension  
☐ SIP Trunk / test  
☐ E1/T1 Trunk / Slot 3  
☐ FXO Trunk / Slot 0 /  
Port 0  
☐ SIP Extension / 72139 /

☐ Select All  
Target list 0/0

# How To Configure Routing

- Parameters- Destination

users can choose from local extension, SIP extension, SIP trunk, FXO trunk, e1/t1 trunk, ring group, call queue, intercom/paging group and IVR

Action

Callback

Distinctive Ringtone(Alert-Info)

Manipulation

Destination

Password Type

Search...

SIP Trunk / test

E1/T1 Trunk / Slot 3

FXO Trunk / Slot 0 / Port 0

Ring Group / ring

Call Queue / 队列一

SIP Trunk / test

Off

# Use Case 1

IPPBX combines IP phones and operator SIP dedicated lines to achieve full IP networking

## Outbound route:

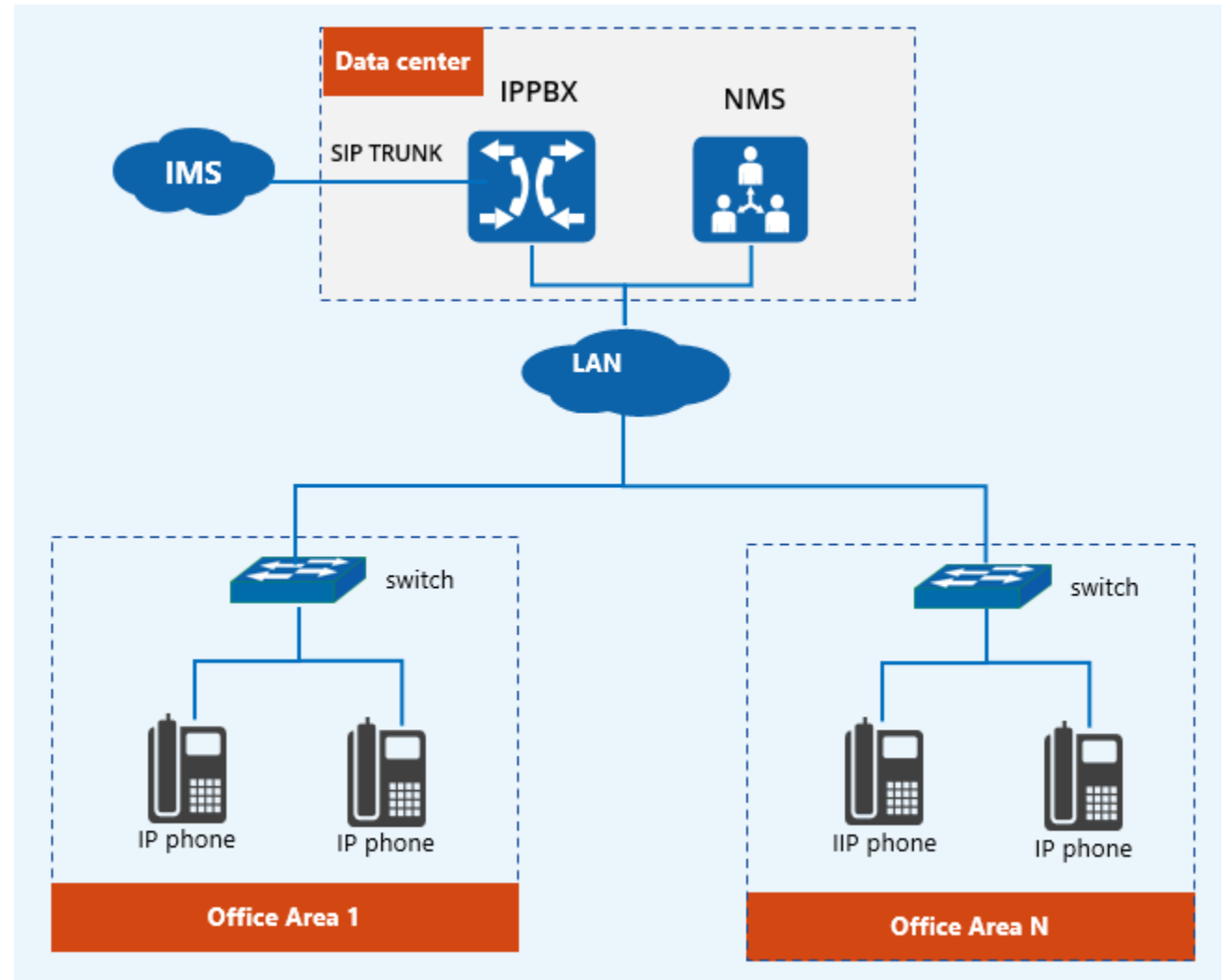
Route->Source: local extension

Route->Destination: SIP trunk/IMS

## Inbound Route:

Route->Source: SIP trunk/IMS

Route->Destination: local extension



# Use Case 2

Upgrade and renovate the existing traditional telephone system, retain the original telephones, and save costs

## Outbound route:

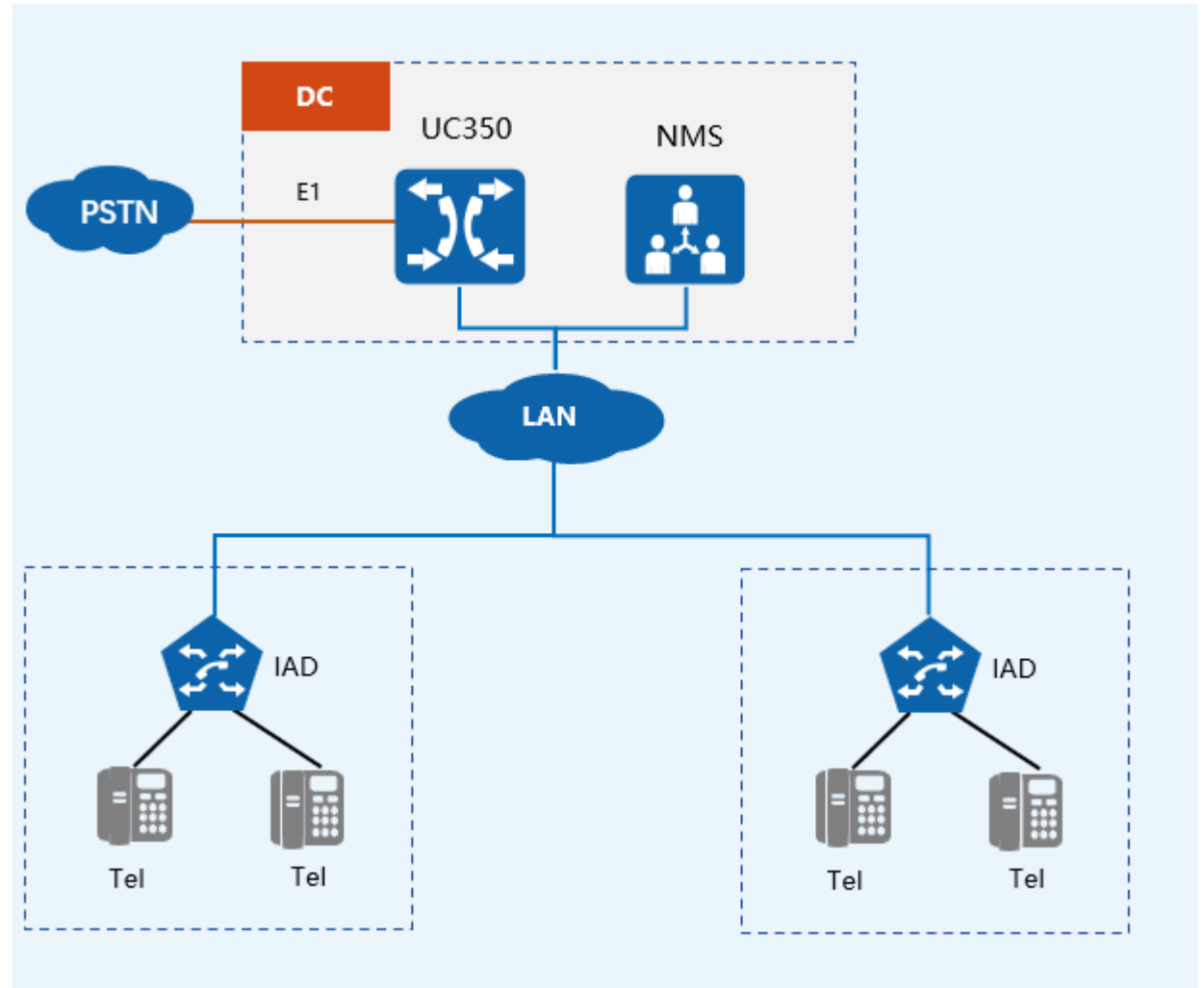
Route->Source: SIP trunk/IAD

Route->Destination: E1/T1 trunk

## Inbound Route:

Route->Source: E1/T1 trunk

Route->Destination: SIP trunk/IAD



# Use Case 3

Reasonably allocate incoming calls to operators to ensure high-quality service is provided to customers

## Outbound route:

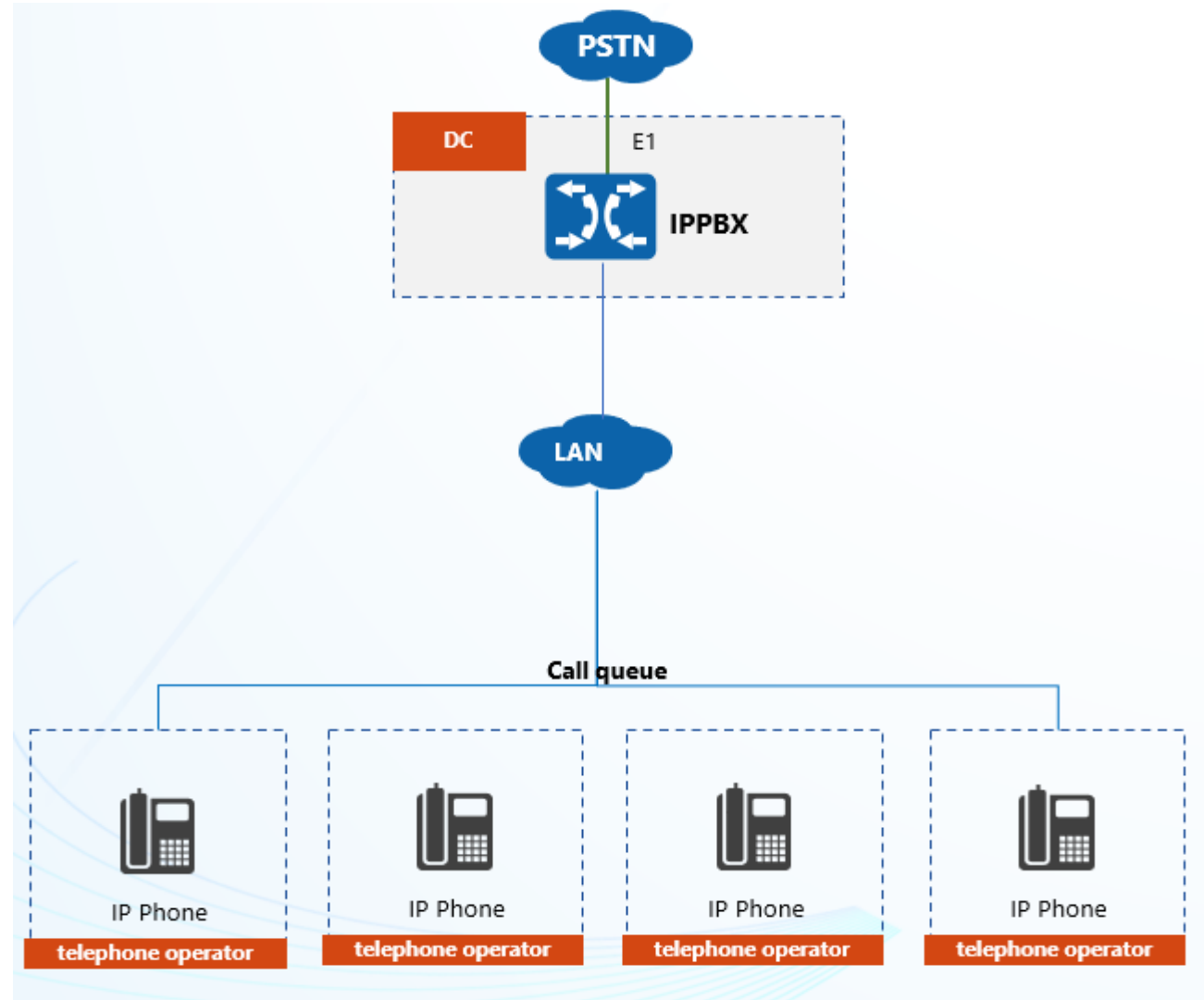
Route->Source: local extension

Route->Destination: E1/T1 trunk

## Inbound Route:

Route->Source: E1/T1 trunk

Route->Destination: Call Queue



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- 3 Nat Public Proxy Server

# Nat Public Proxy Server

03

1. Why Need Nat Public Proxy Server

2. How To Configure Nat Public Proxy Server



# Why Need Nat Public Proxy Server

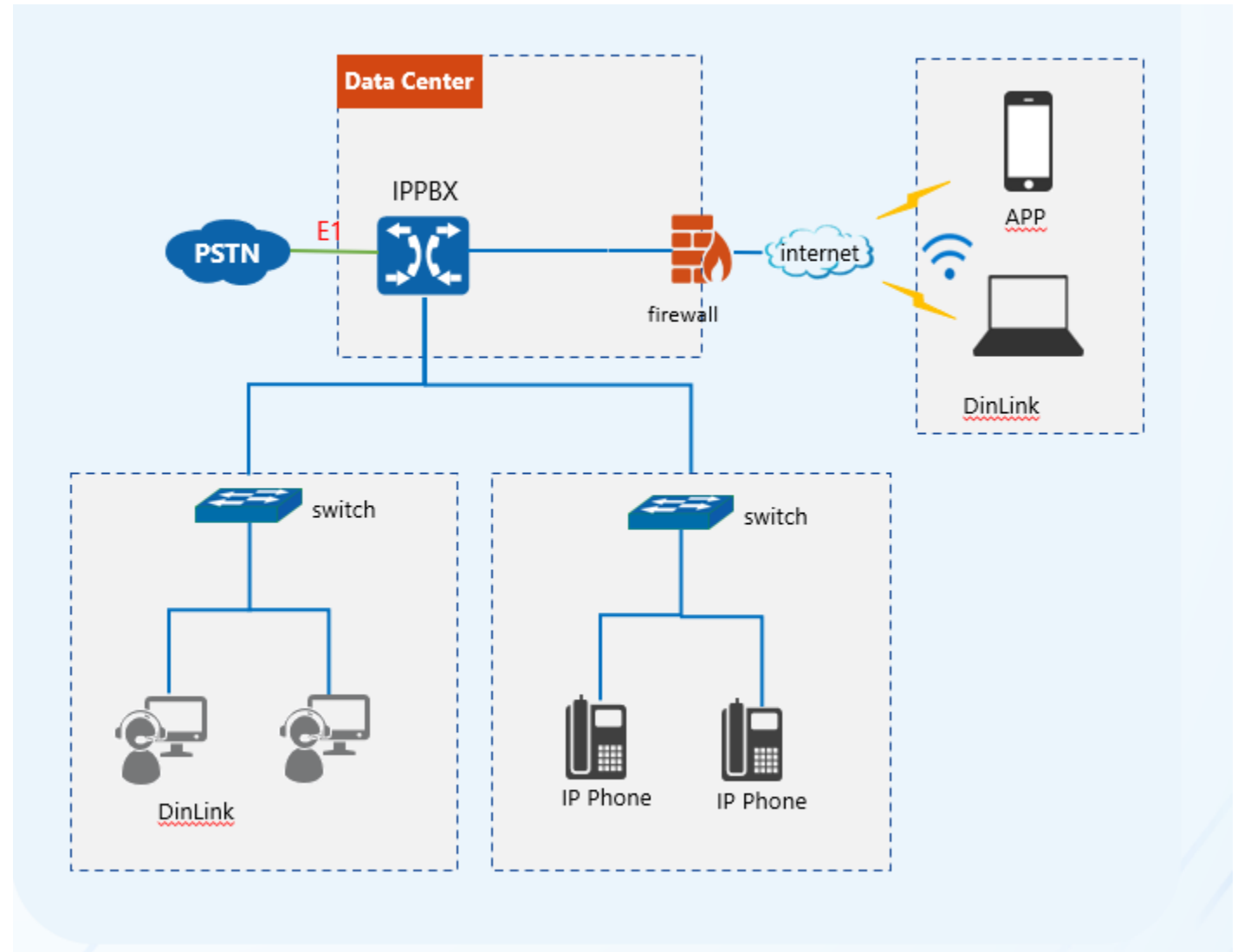
- **Main Scenarios**

1. UC deployment on enterprise intranet
2. Outward personnel may need to use a mobile app to access UC through external public WIFI or mobile data

- **Main function**

Signal penetration, using frpc and frps for forwarding

Media penetration, using Stun server to penetrate through ice



# How To Configure Nat Public Proxy Server

1. Click PBX Global Settings->SIP stack

2. Edit SIP Profile and enable public proxy function

3. Set the address and port of the public proxy server, and select the proxy options

The screenshot displays the 'Edit SIP Profile' configuration page in the DINSTAR PBX management system. On the left sidebar, the 'SIP Stack' option is highlighted with a red box and a circled '1'. The main content area shows various SIP settings. The 'Public Proxy' toggle switch is turned on, highlighted with a red box and a circled '2'. Below it, the 'Public Proxy Server Address' and 'Public Proxy Server Port' fields are empty, and the 'Proxy Options' section shows 'TCP', 'TLS', and 'WebRTC' checkboxes, with 'TCP' selected. A red box and a circled '3' encompass these three options.

Setting	Value
Detect Extension is Online	<input checked="" type="checkbox"/>
DTMF Send Type	RFC2833
RFC2833-PT	101
Detect Inband When Call in IVR	<input checked="" type="checkbox"/>
Process DTMF as Hold/Unhold	Off
PRACK	<input checked="" type="checkbox"/>
WebRTC	<input checked="" type="checkbox"/>
WebRTC Listening Port	7443
Public Proxy	<input checked="" type="checkbox"/>
Public Proxy Server Address	
Public Proxy Server Port	
Proxy Options	<input checked="" type="checkbox"/> TCP <input type="checkbox"/> TLS <input type="checkbox"/> WebRTC

# Summary

This course we already learn:

- What are the common functions

- How to configure common functions

- How to configure routing

- What is Nat Public Proxy Server



# THANKS



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