# UC200 IPPBX Test Scenarios

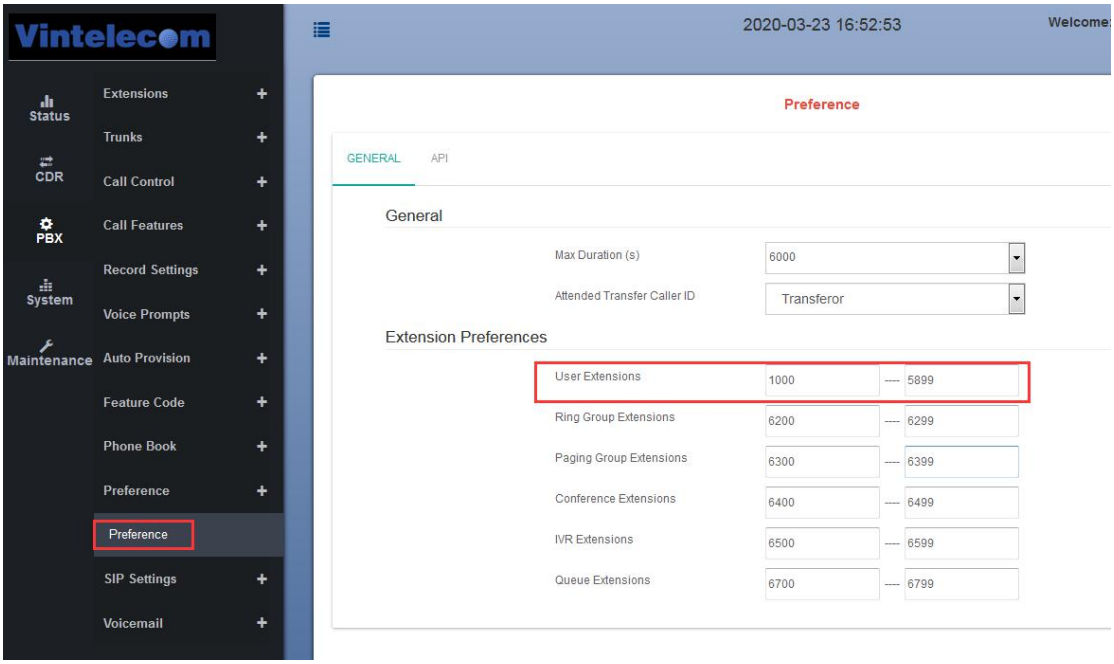
By XIAOYINDI

**Content**

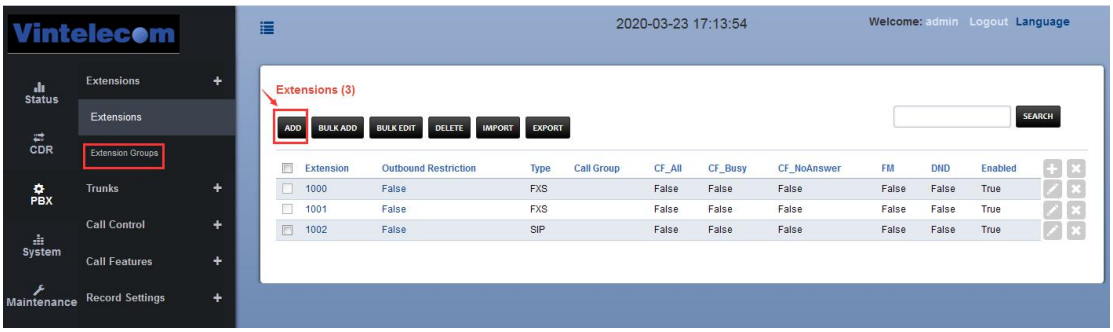
# 1. Register Extensions.

## 1.1. Create Extension

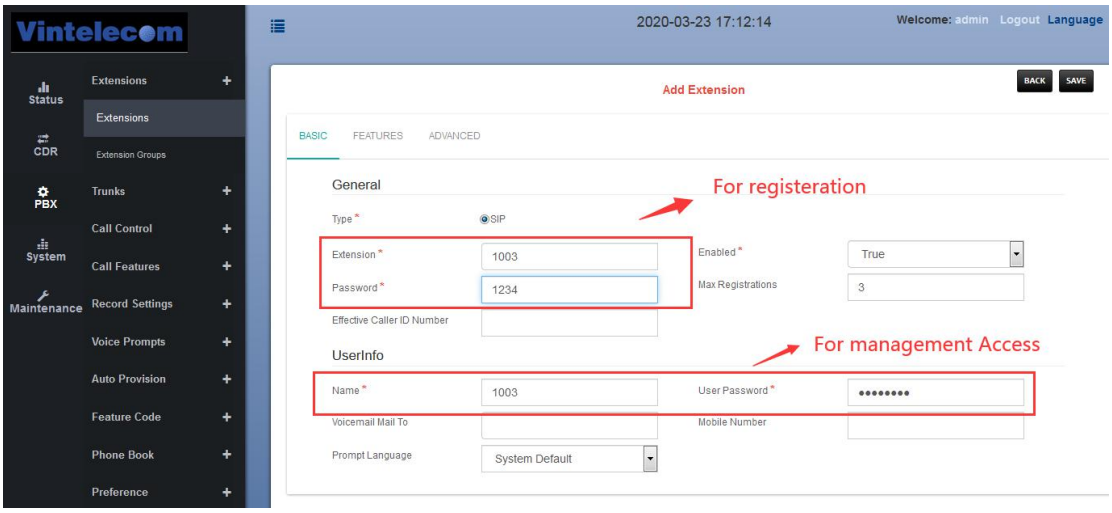
Default Extension Range：



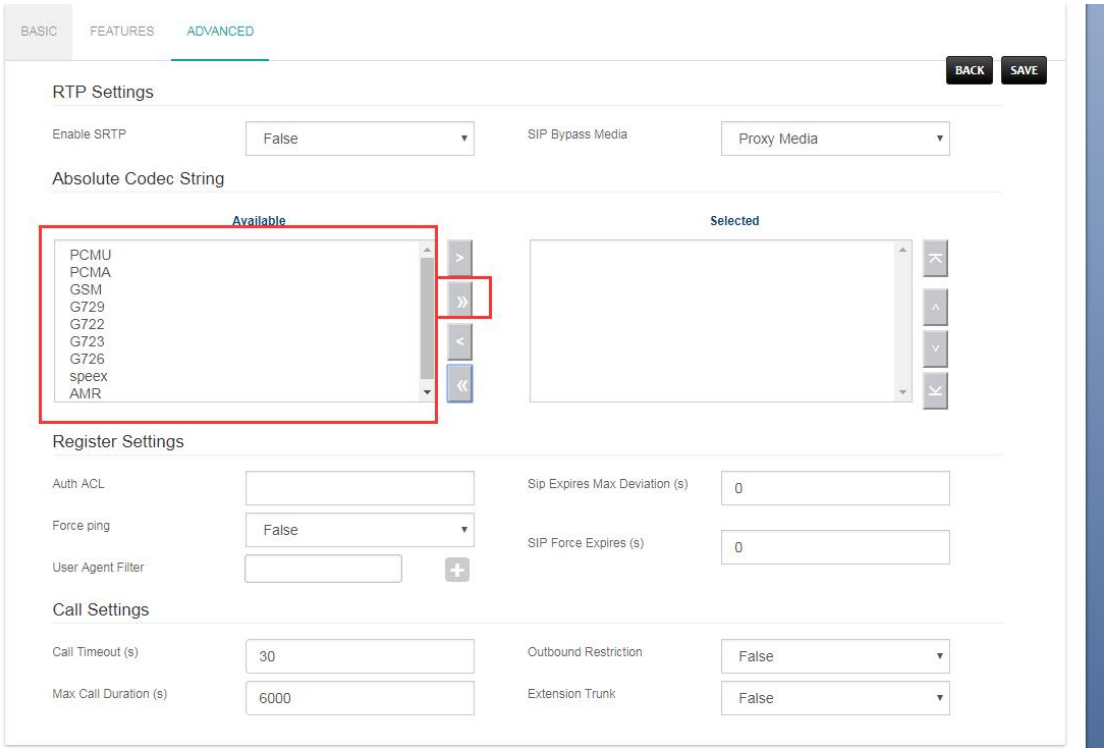

### 1.1.1. Create single extension




Create Extension with account and password.

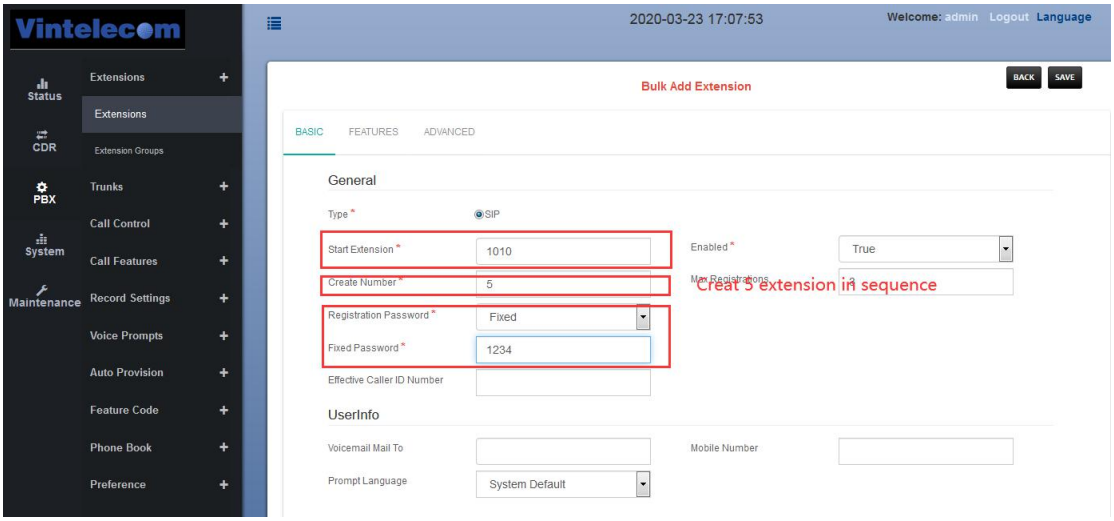



Add CODEC

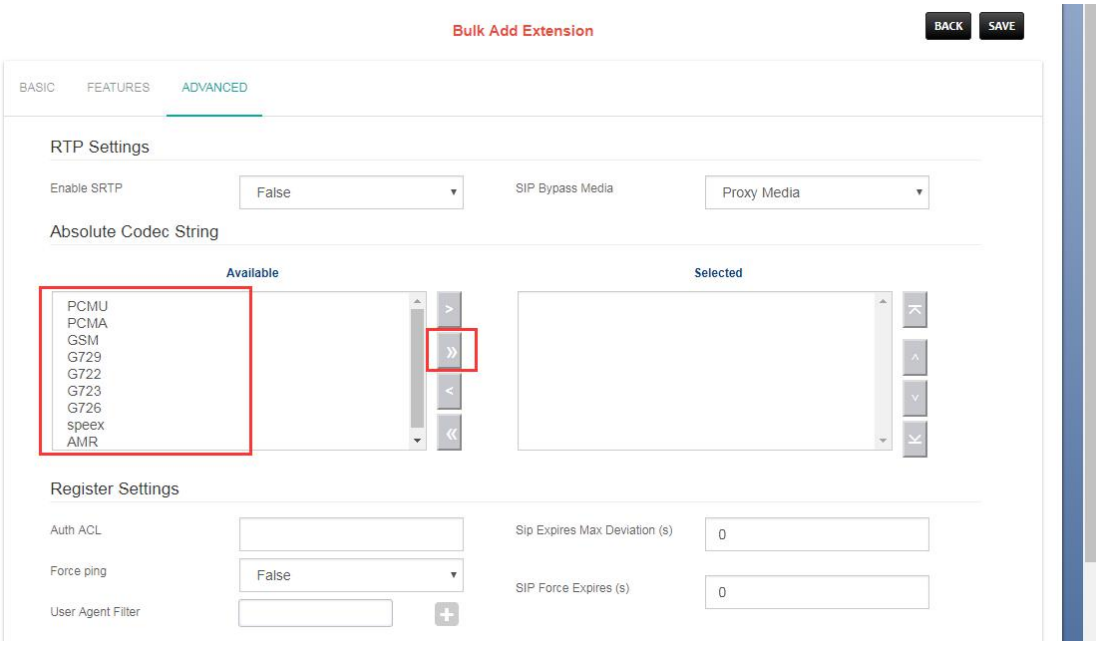



## 1.1.2. Mass/bulk create extensions

Start extension + numbers of extension to be created + fixed password.

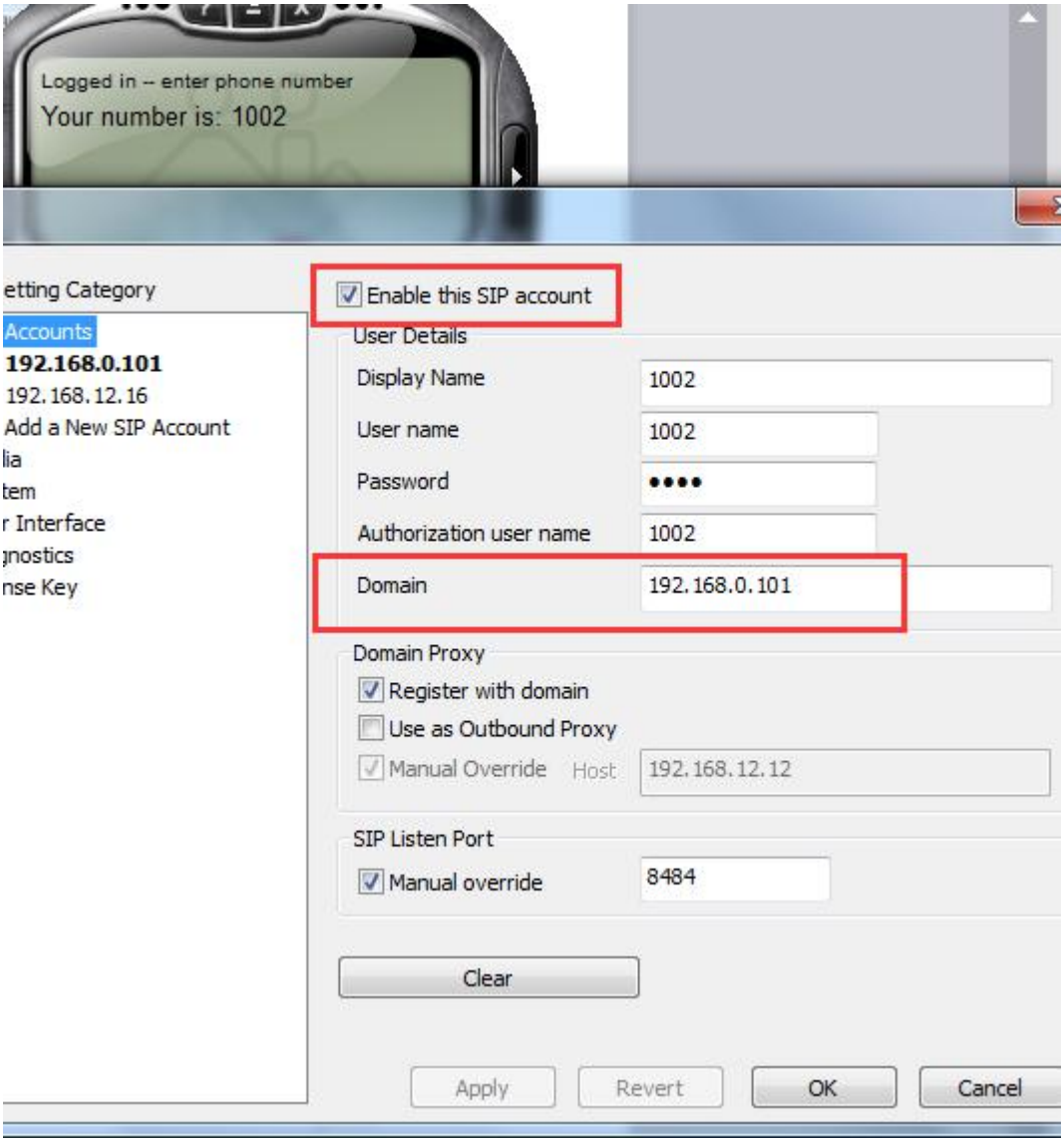



Select CODEC



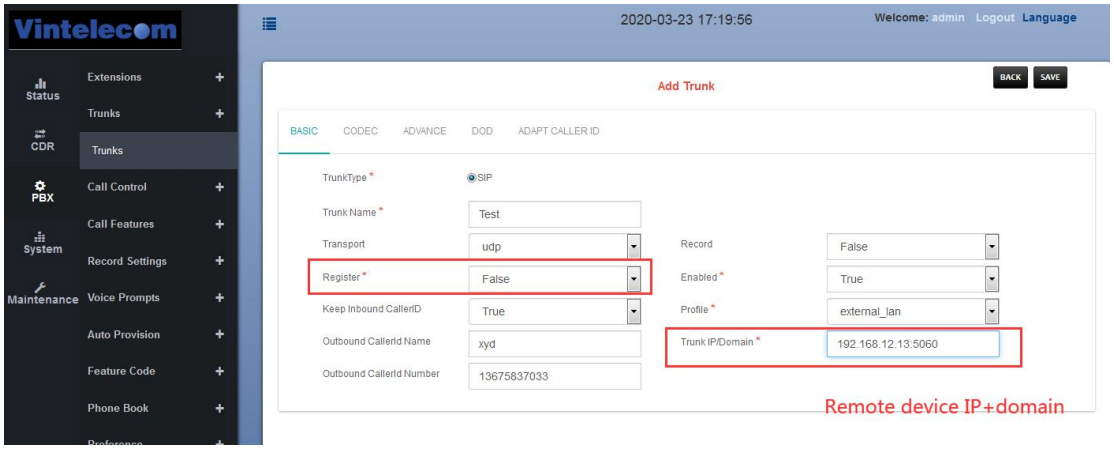

# 1.2. Registration

Fill up the Account Name,User name,Password and Authorization Name


Logged in -- enter phone number
Your number is: 1002
etting Category
Accounts
192.168.0.101
192.168.12.16
Add a New SIP Account
lia
tem
r Interface
gnostics
nse Key
Enable this SIP account
User Details
Display Name
1002
User name
1002
Password
Authorization user name
1002
Domain
192.168.0.101
Domain Proxy
Register with domain
Use as Outbound Proxy
Manual Override
Host
192.168.12.12
SIP Listen Port
Manual override
8484
Clear
Apply
Revert
OK
Cancel

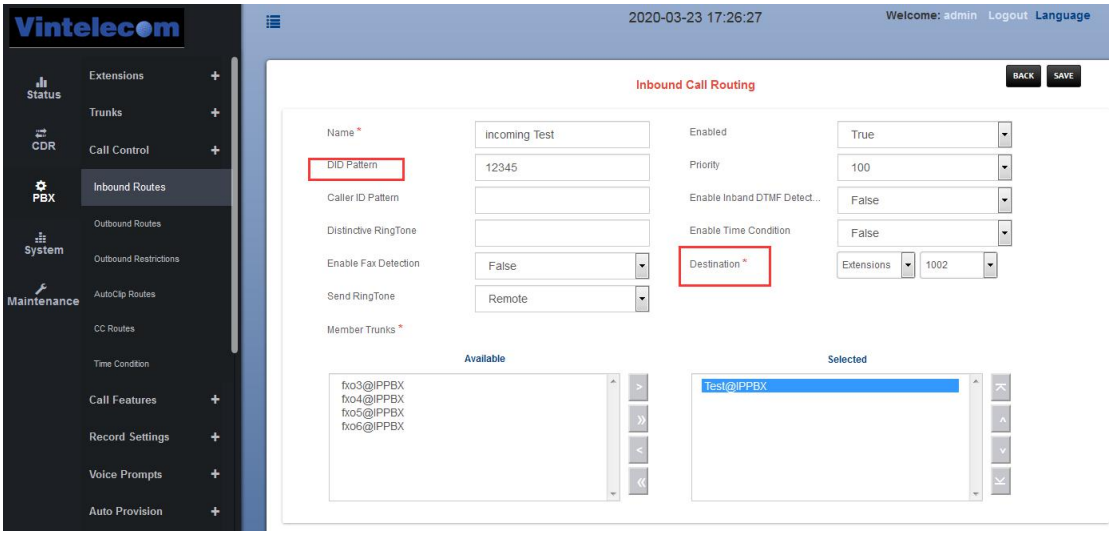
# 2. Peer to Peer SIP trunks

## 2.1. Create sip trunk

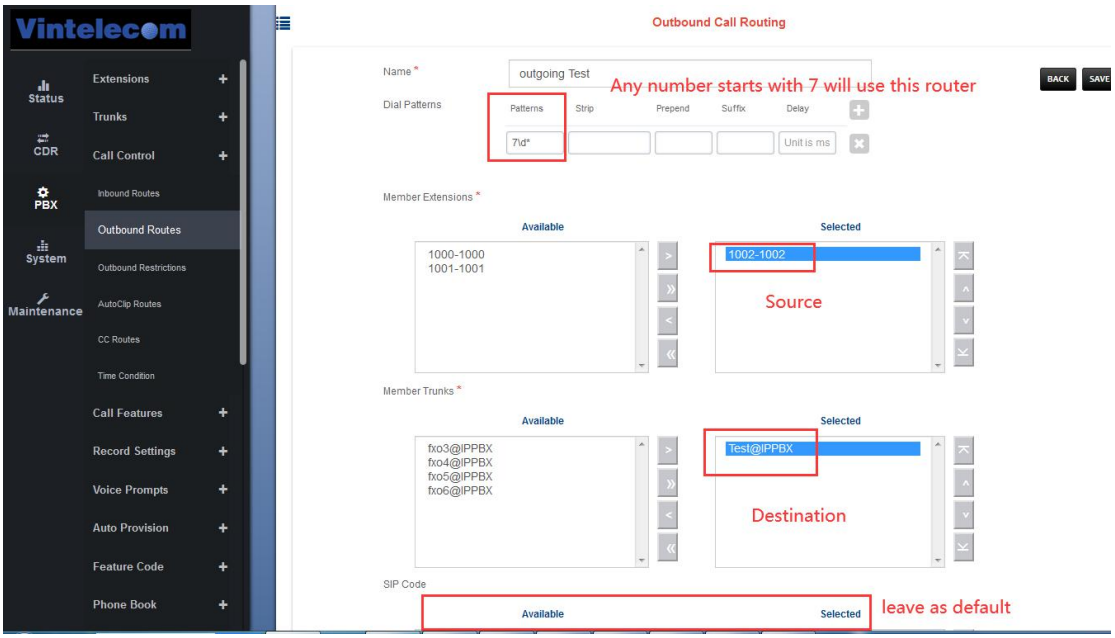



## 2.2. Inbound Route

Incoming call with DID 12345 will be lead to extension 1002



## 2.3. Outbound Route

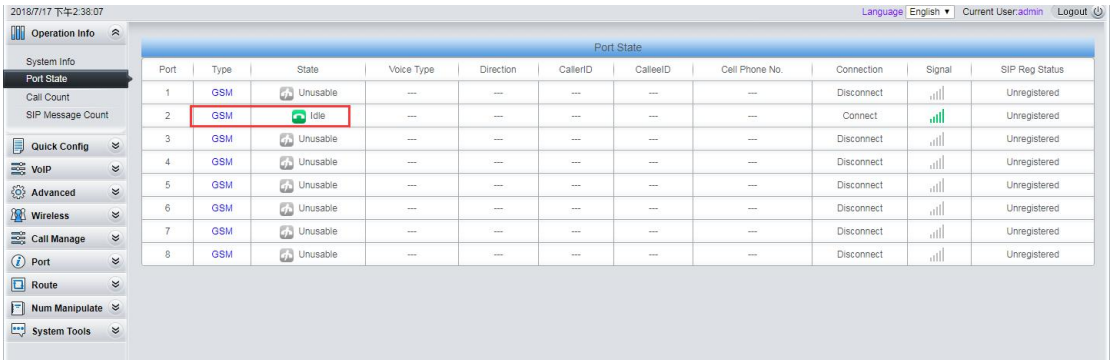



## 2.4. Connect to Goip gsm gateways

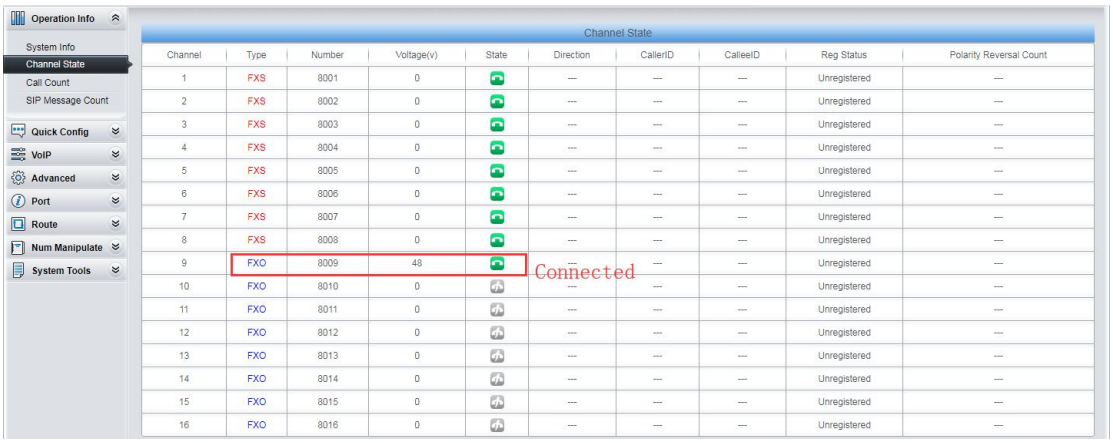
### 2.4.1. Analog, GSM gateway

First make sure the gateway channel are available

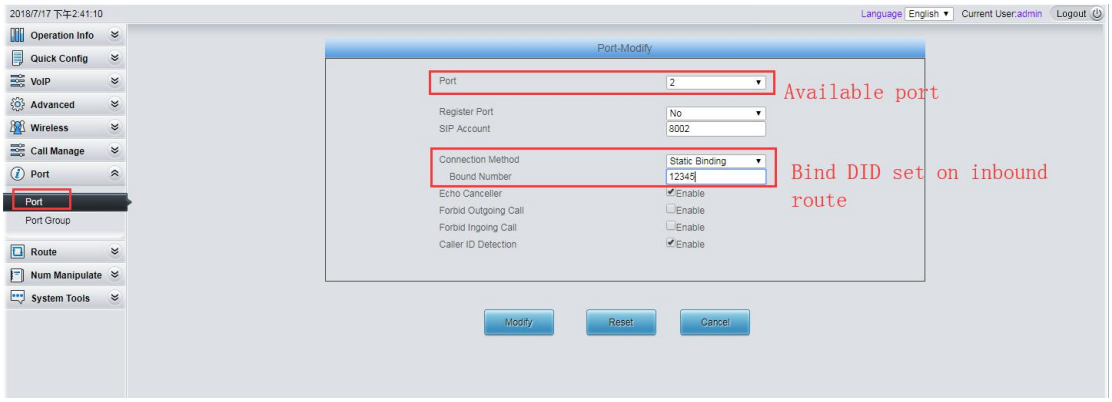
GSM:



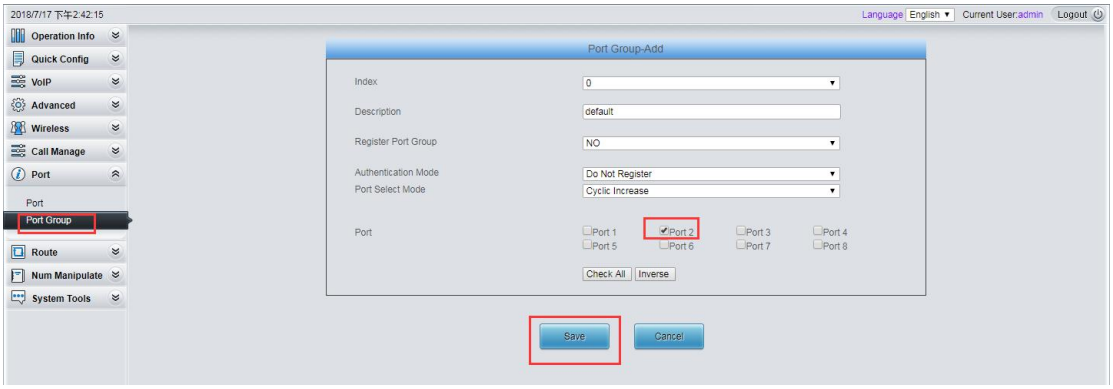

Analog:



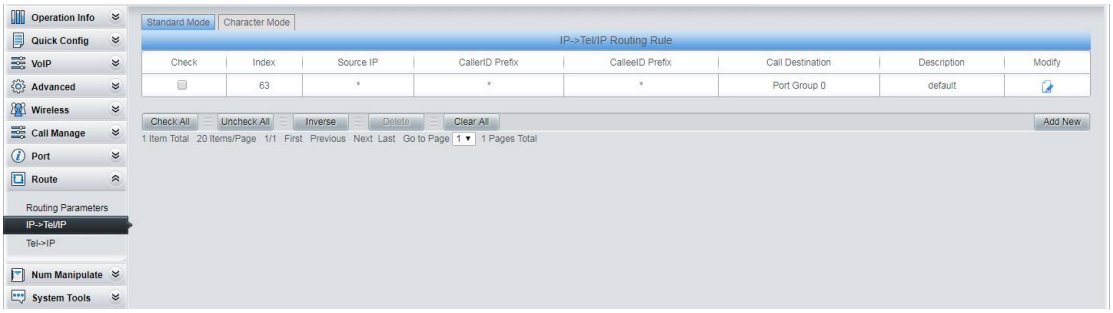

Step 1:

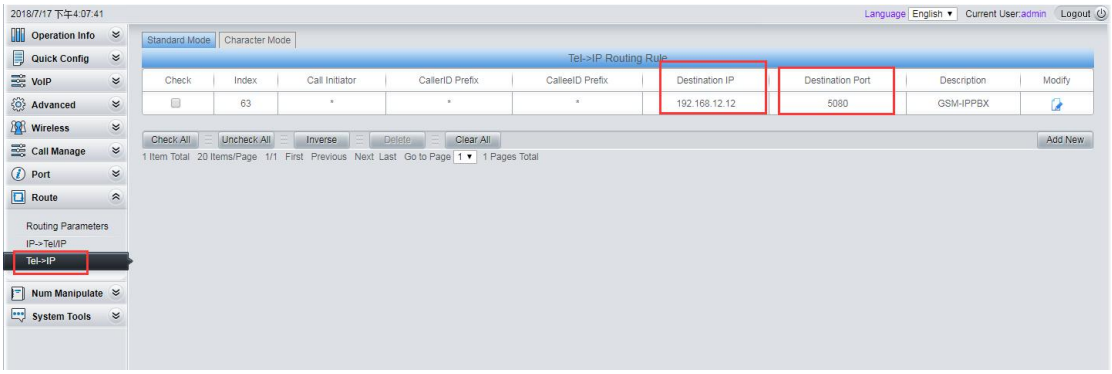



Step 2: Put the port into port group




Step 3: Set routes for IP-TEL and TEL-IP



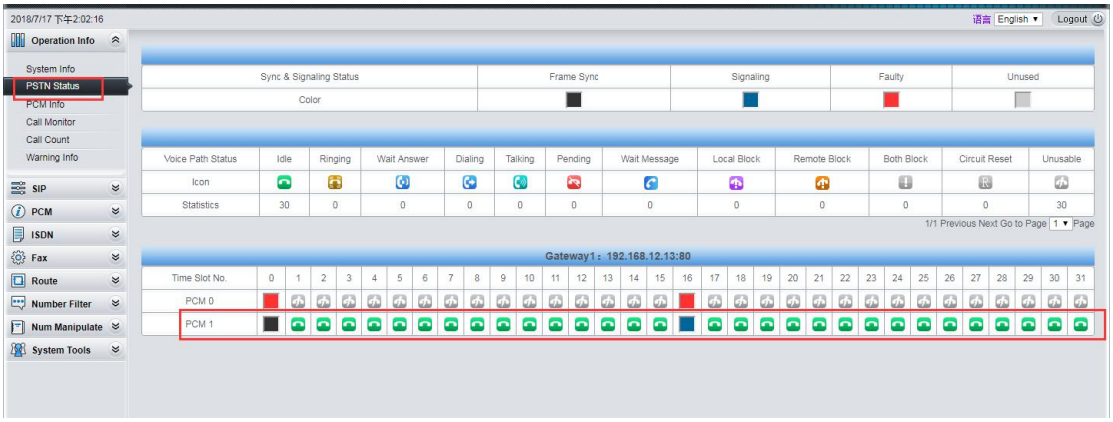



Destination port should be 5080 on IPPBX as peer to peer call.

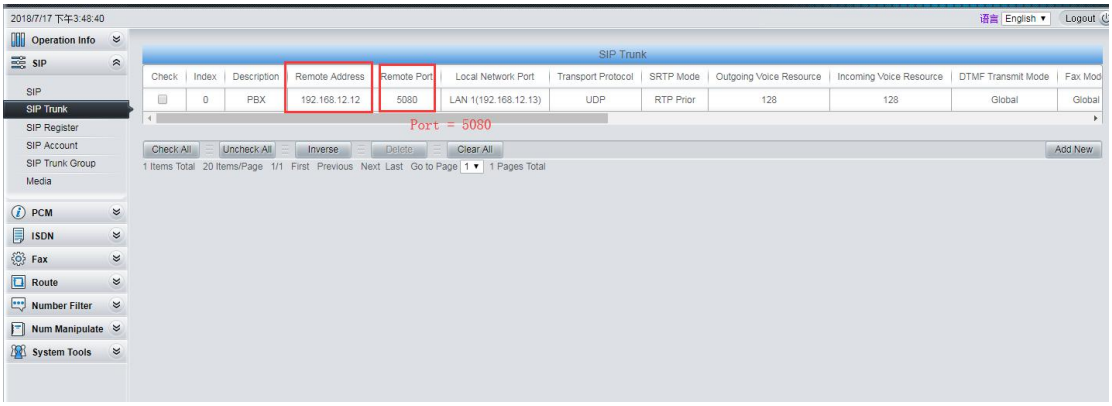
Call Test:


## 2.4.2. E1 gateways

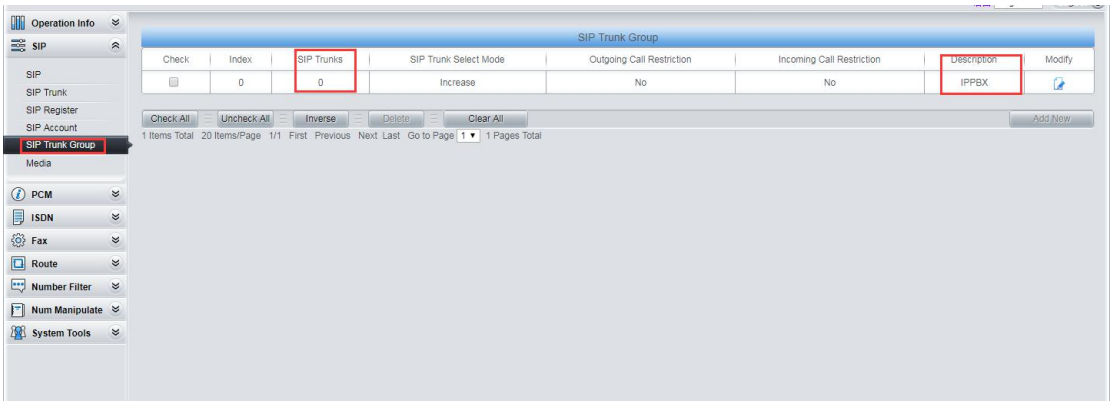
First make sure E1 gateway is well setup(PSTN staus ok, channels goes idle)



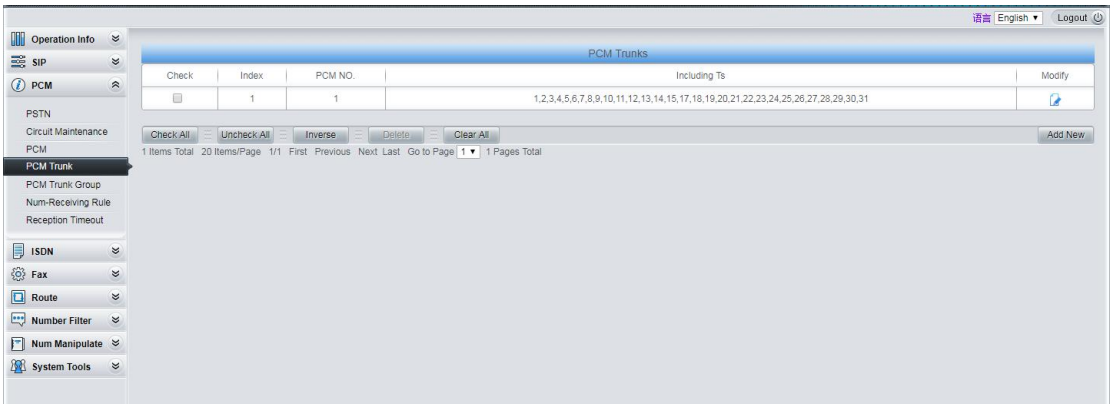

Step 1: Create SIP trunk



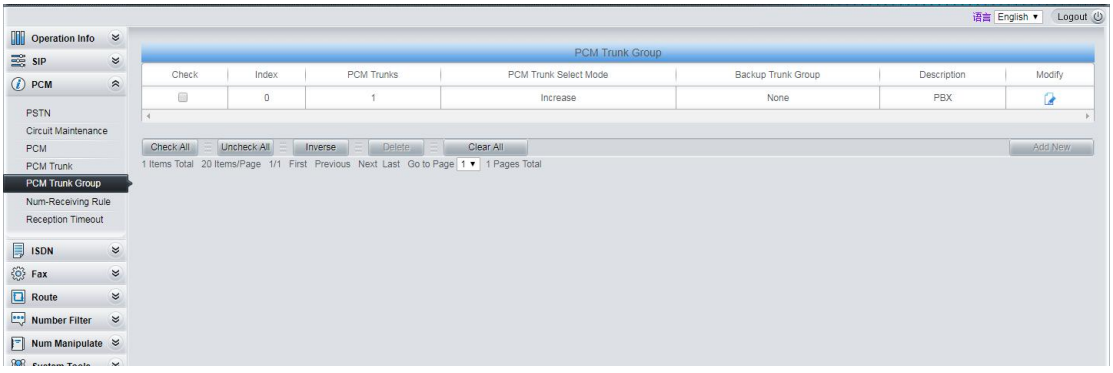

Step 2: Put sip trunk into sip trunk group



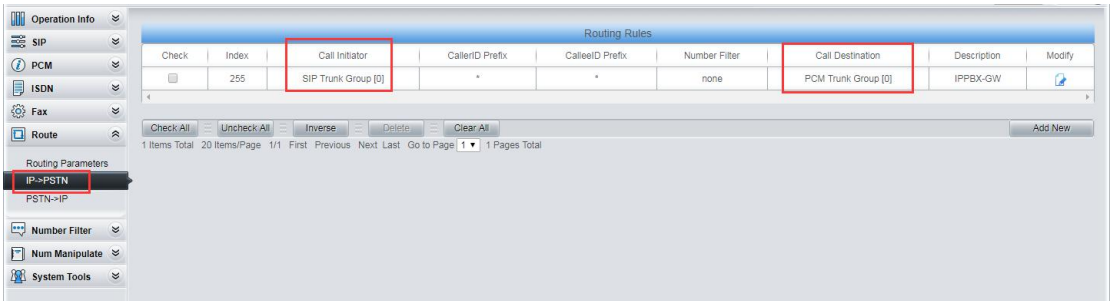

Step 3: Create PCM trunk:

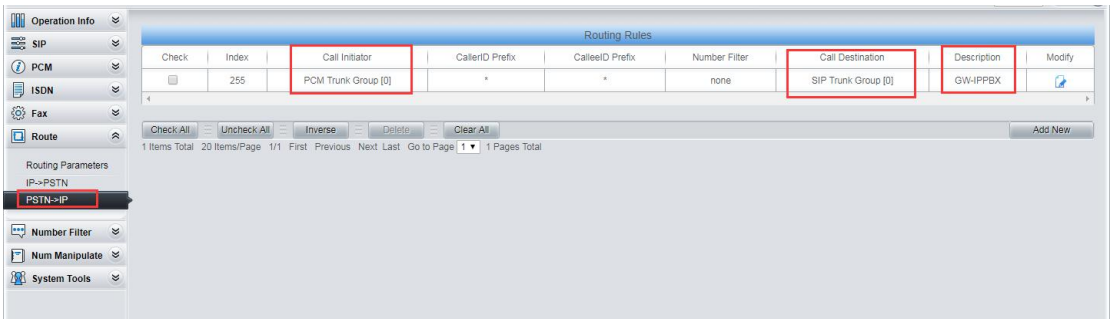



Step 4： Put sip trunk into sip trunk group



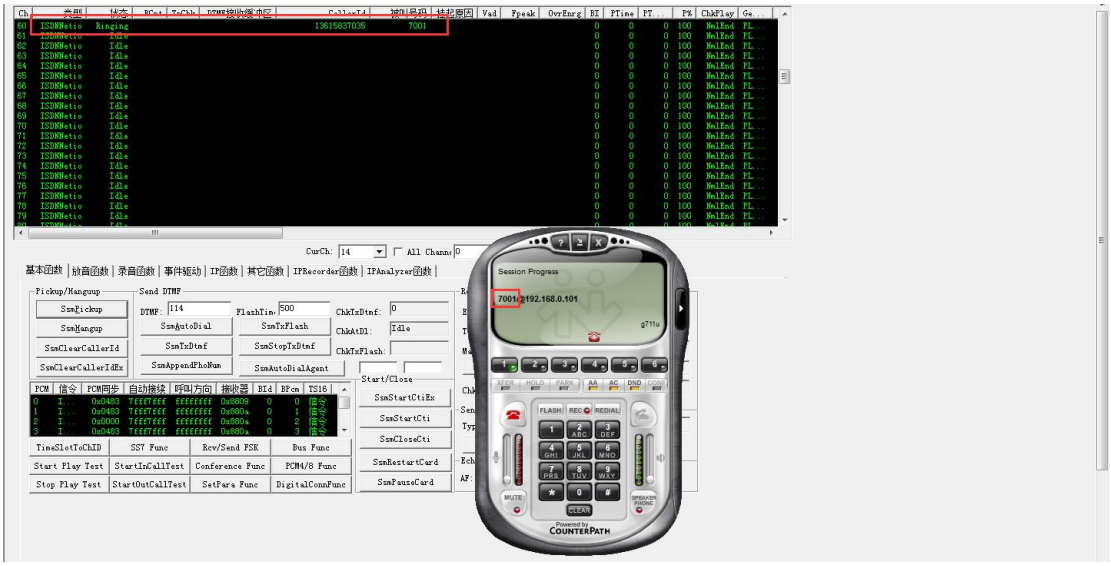

Step 5: Create IP-PSTN and PSTN-IP routes



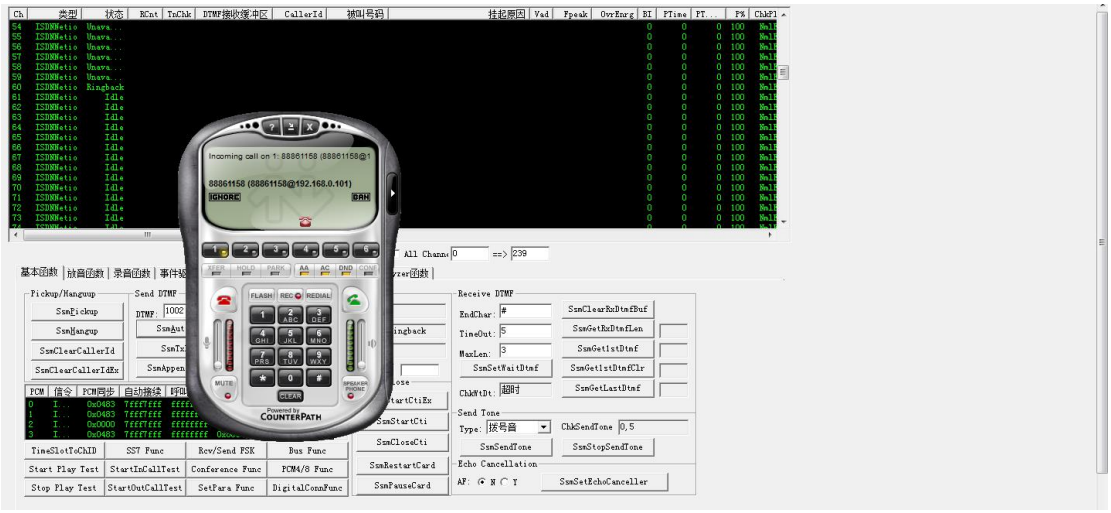




Call Test

IP-PSTN

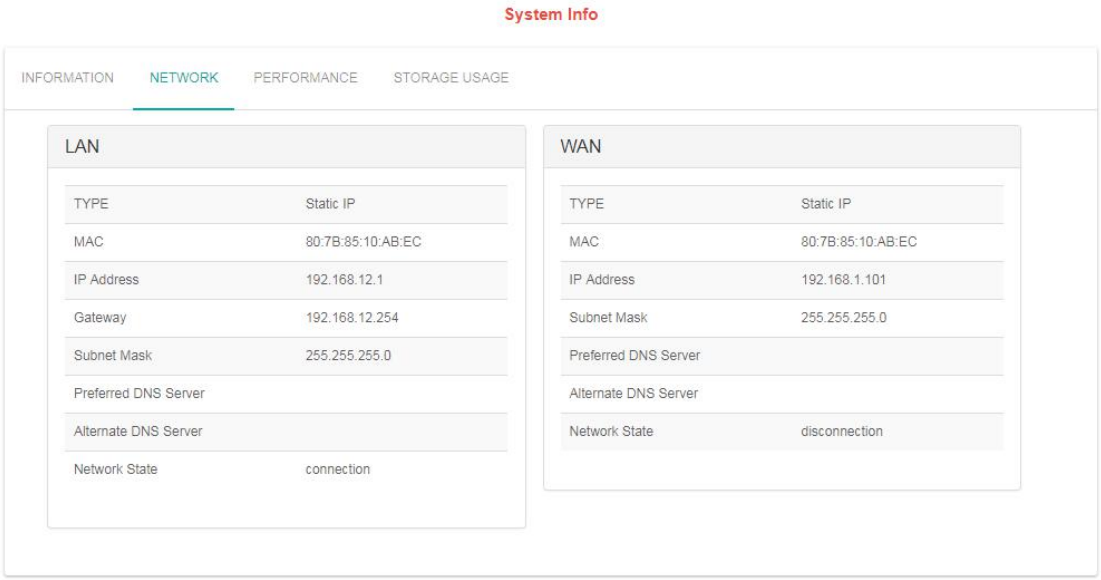


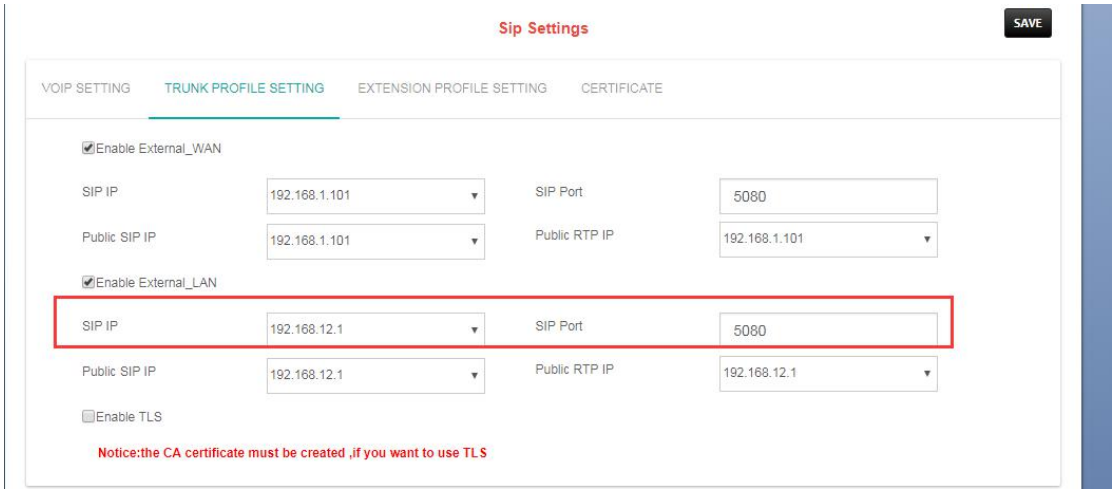

PSTN-IP



## 2.5. Connect to another UC200 or Other PBX(peer to peer)

UC200_1 LAN: 192.168.12.1 Subnet Mask:255.255.255.0 Gateway: 192.168.12.254
SIP trunk port :5080(default)
Extension range: 1000~1999

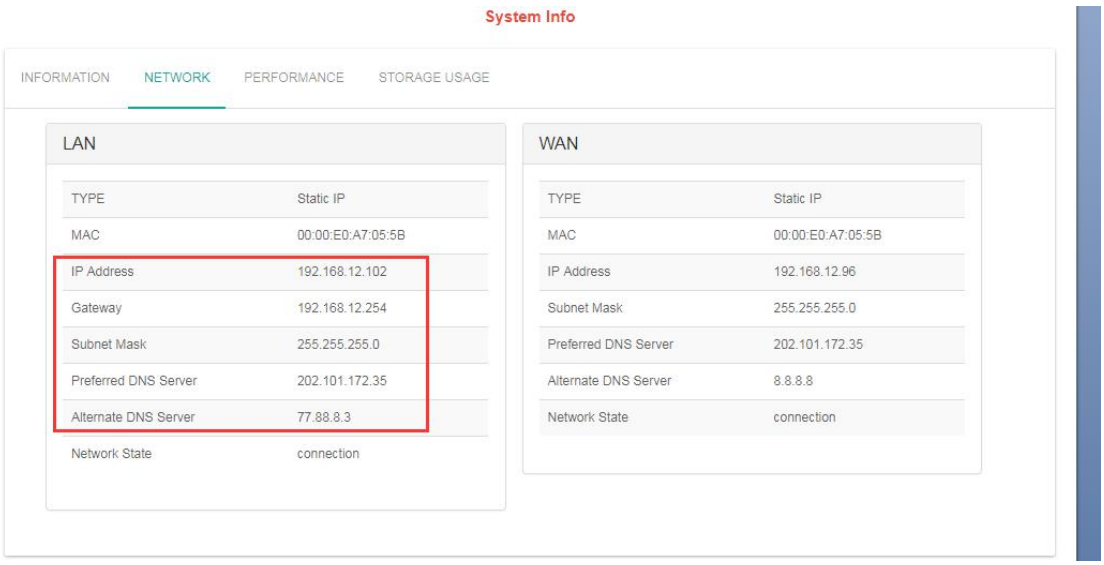


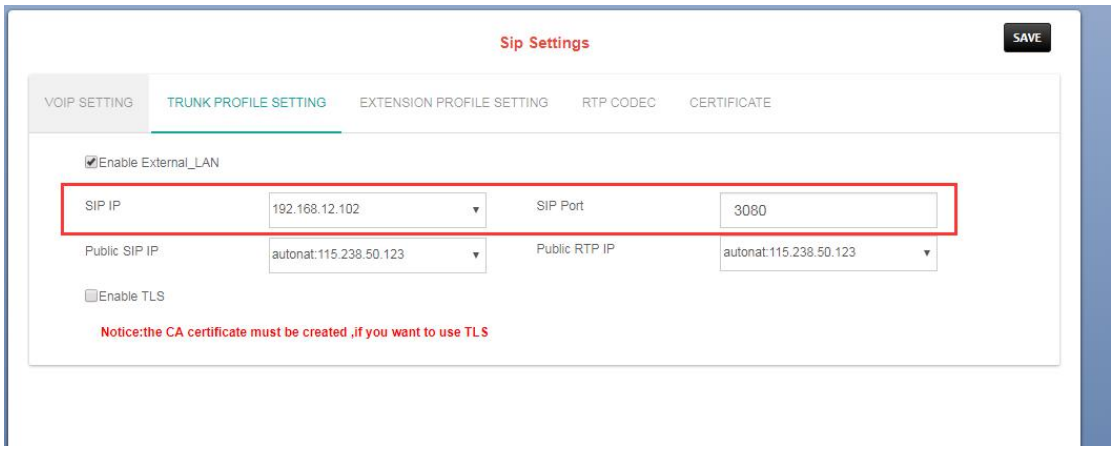



UC200_2 LAN:192.168.12.102 Subnet Mask:255.255.255.0 Gateway: 192.168.12.254

SIP trunk port: 3080

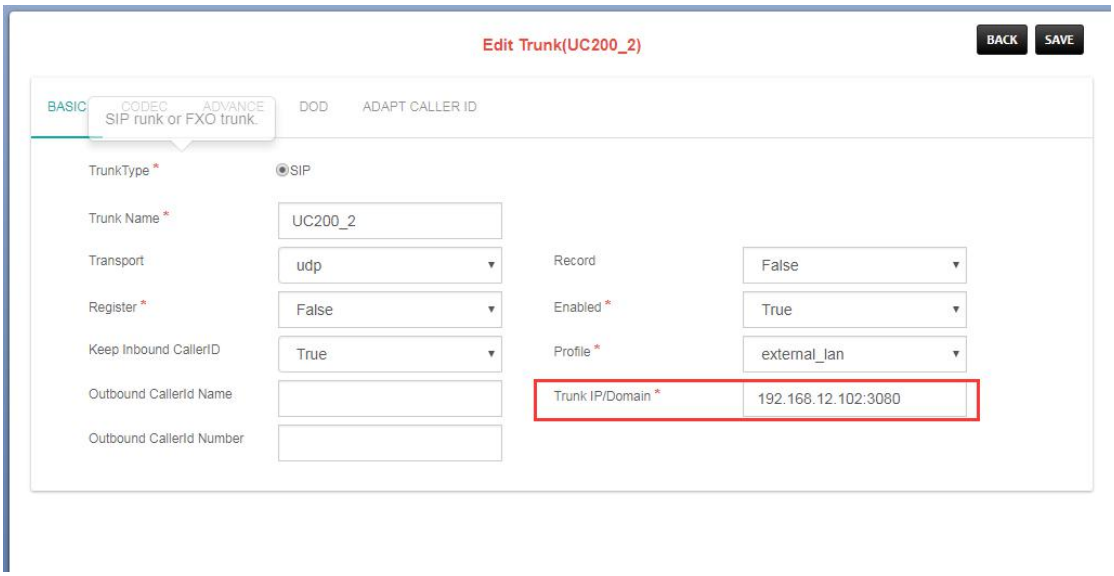
Extension range: 2000~2999



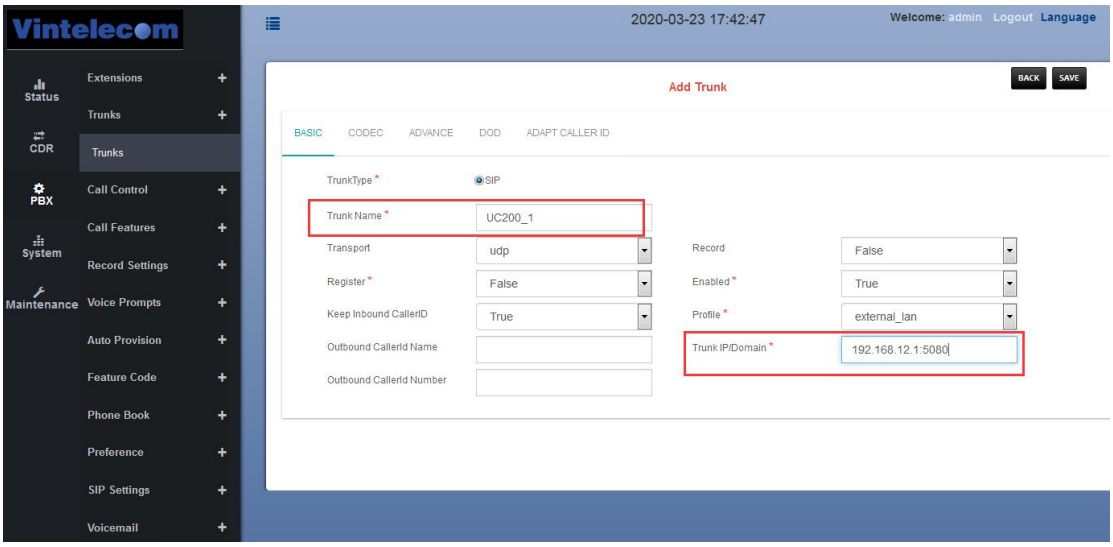




Create SIP trunk on each UC200 (PBX) for remote UC

UC200_1:

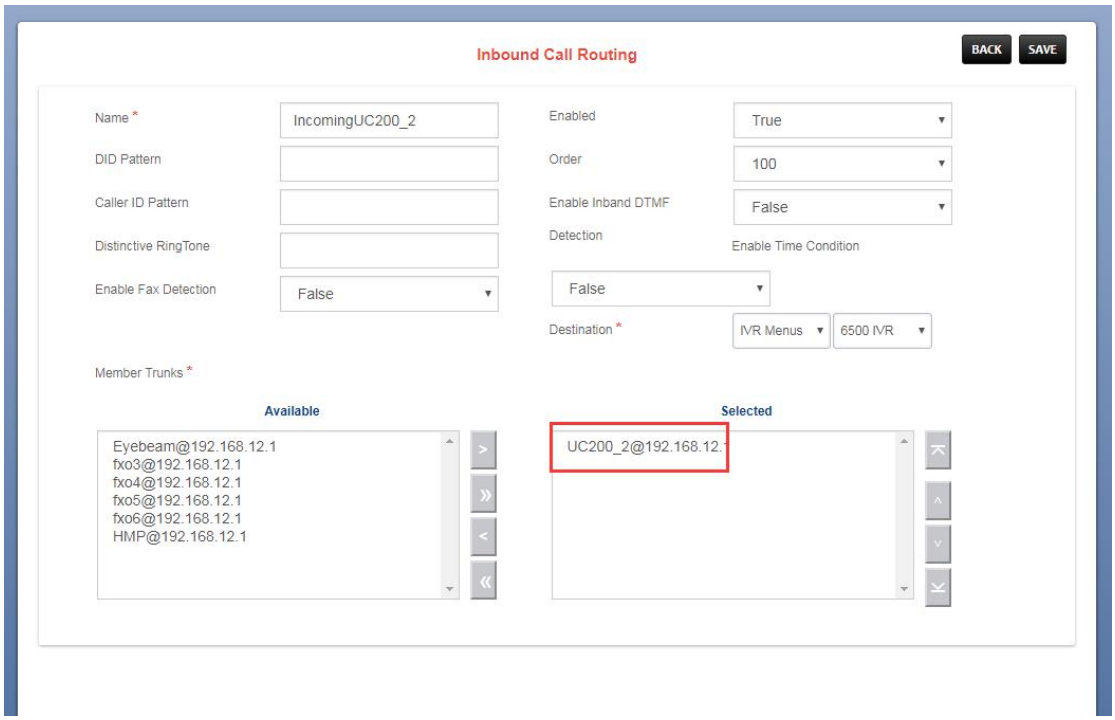



UC200_2:

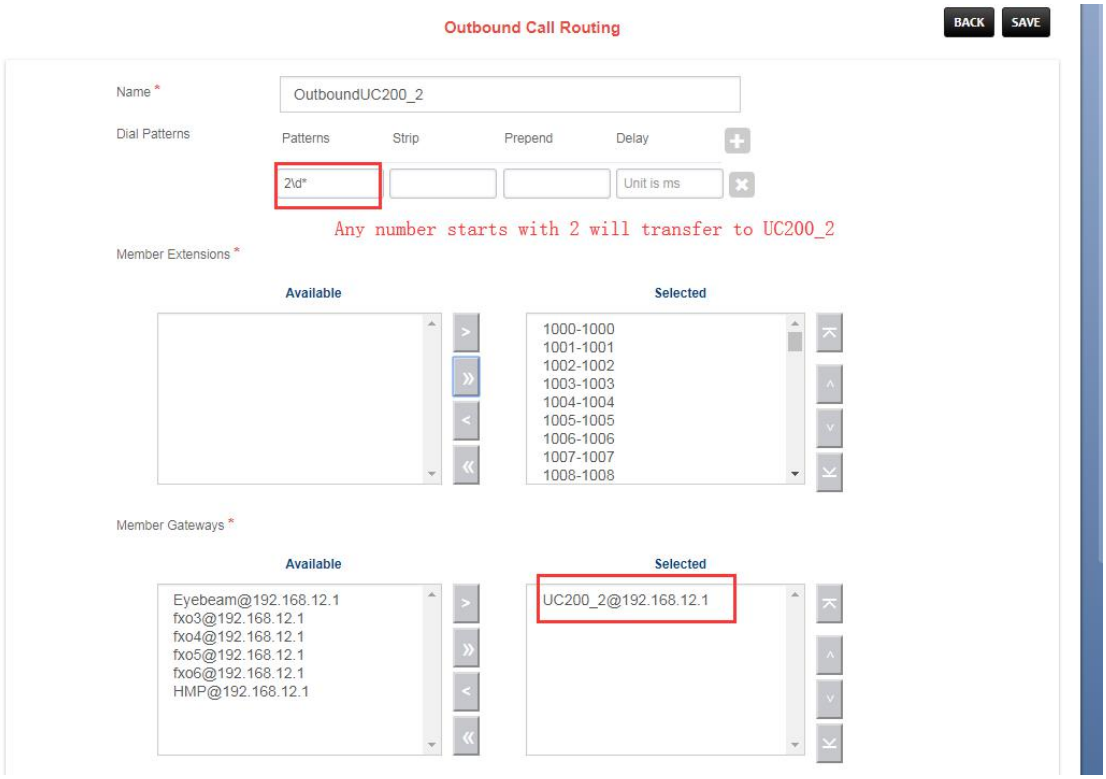



Create inbound and outbound route for both UC200

Inbound route on UC200_1:




Outbound route on UC200_1:




Use any extension in the outbound member extension to dial 2xxx will lead the call to UC200_2.

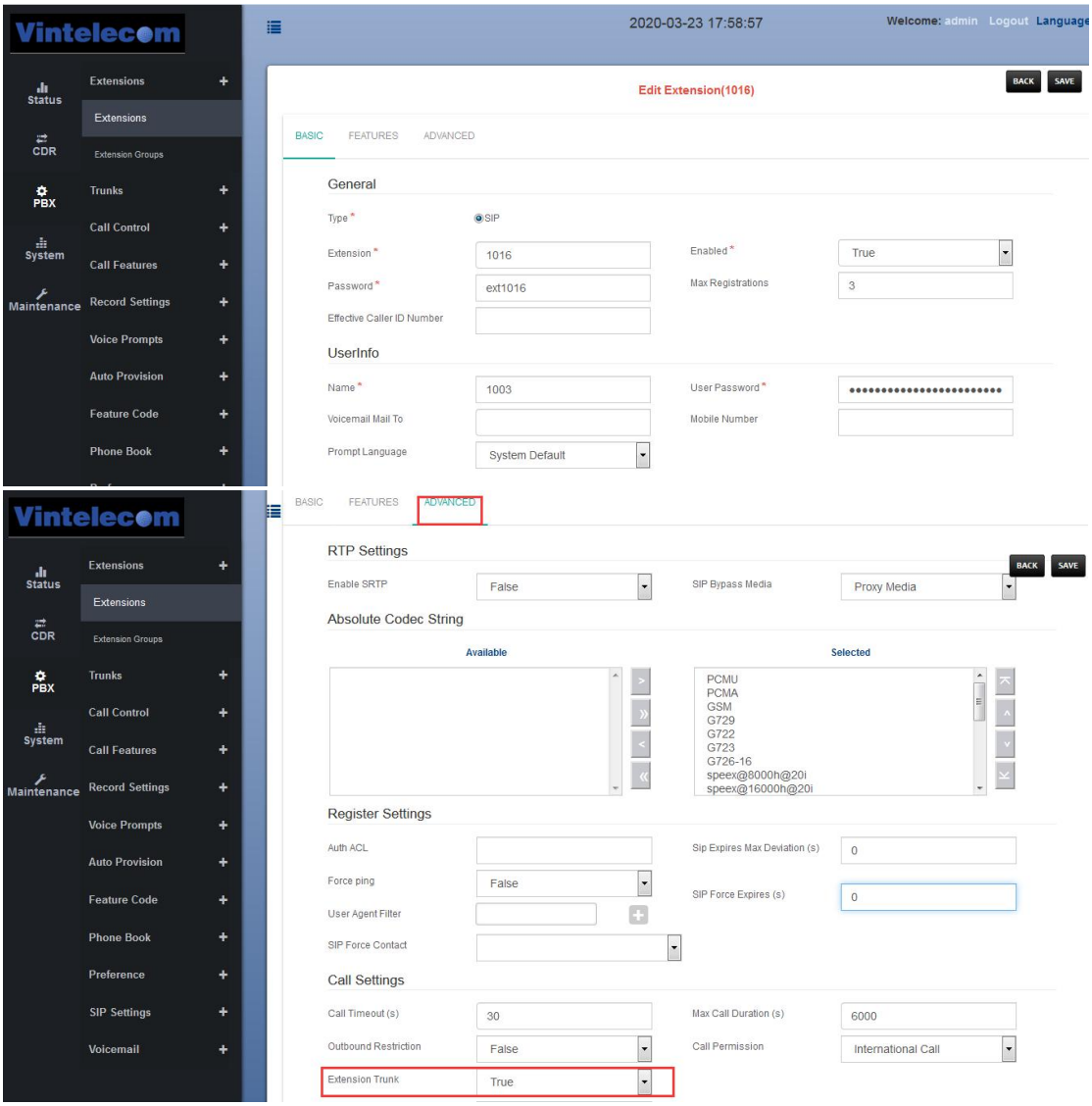
Same step on UC200_2 to create inbound and outbound for UC200_1.

# 3. Register Gateway to IPPBX as SIP trunk.

# (Extension Trunk)

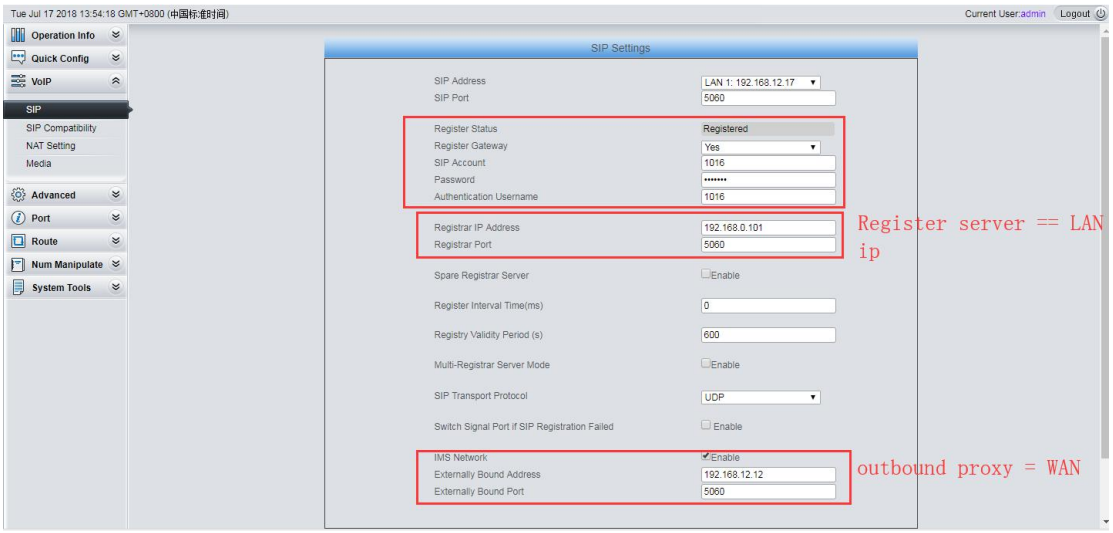
## 3.1. Create account

Follow steps in 1.1.1 Create Single Extension, but select Extension Trunk as True.



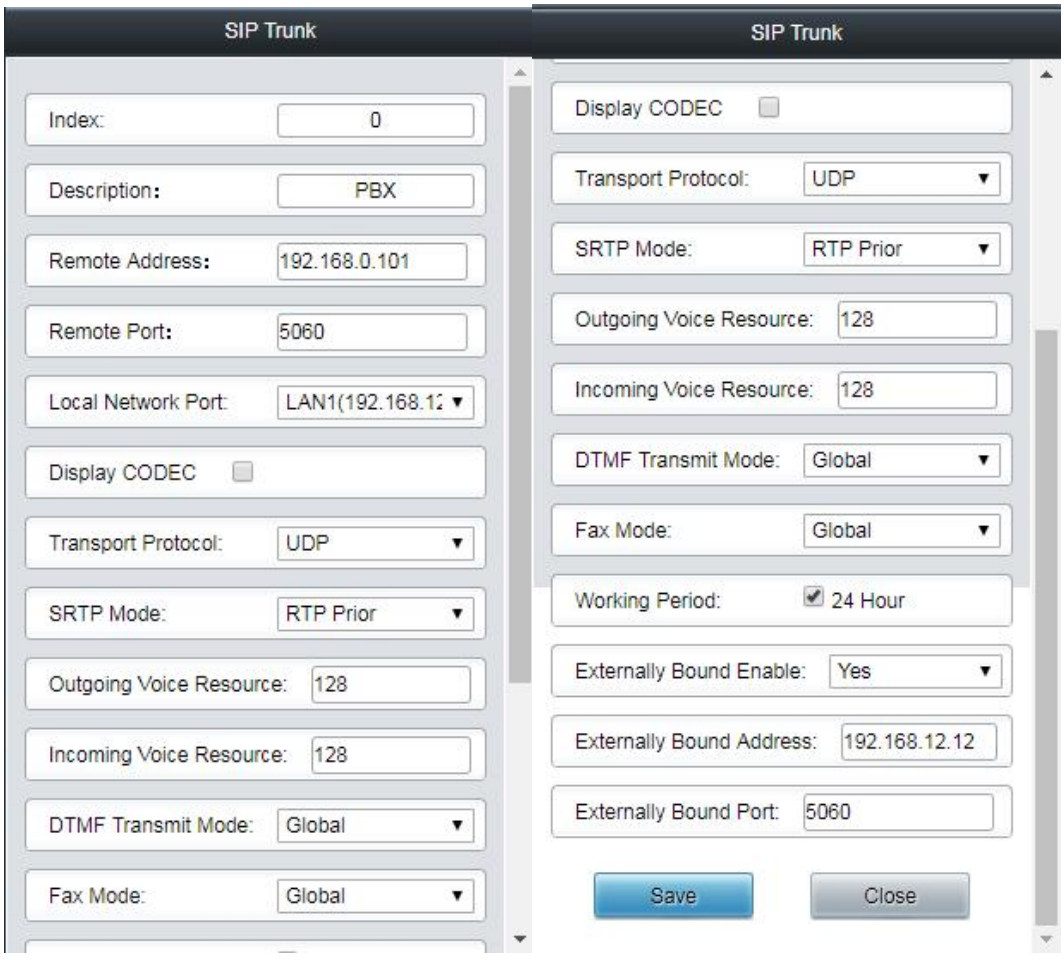

## 3.2. Register gsm gateway to UC200.

Analog, GSM gateways:

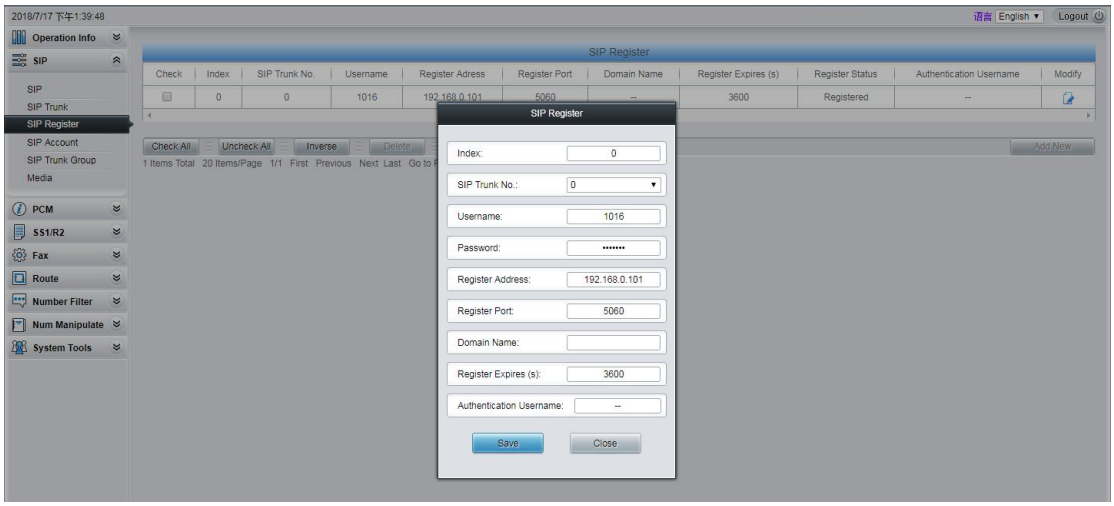



E1 gateways：

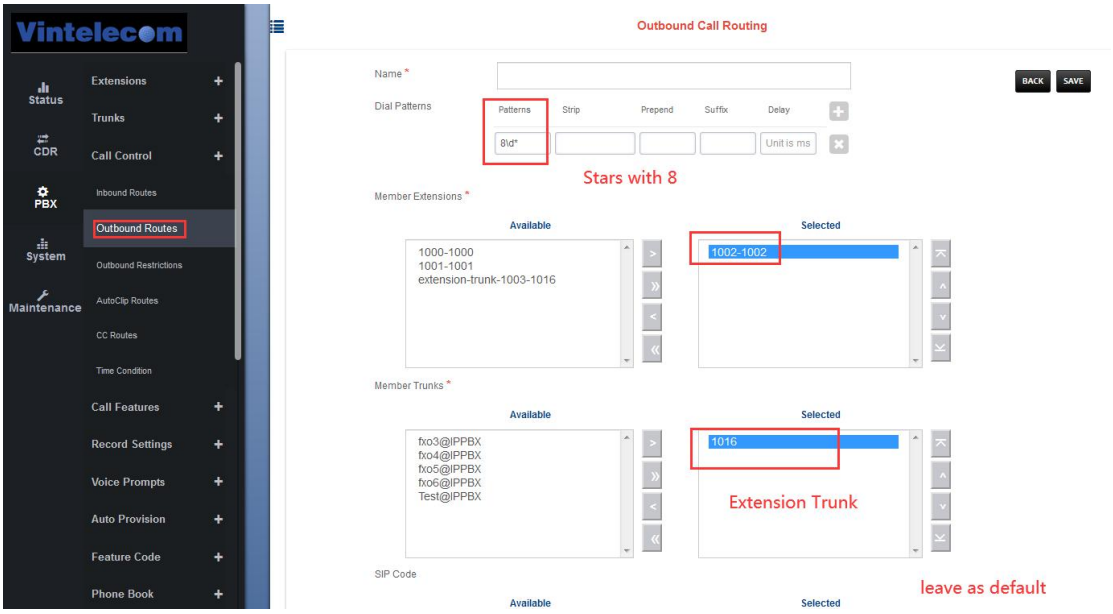
Create sip trunk:




Register sip trunk :




## 3.3. Create outbound route for extension trunk



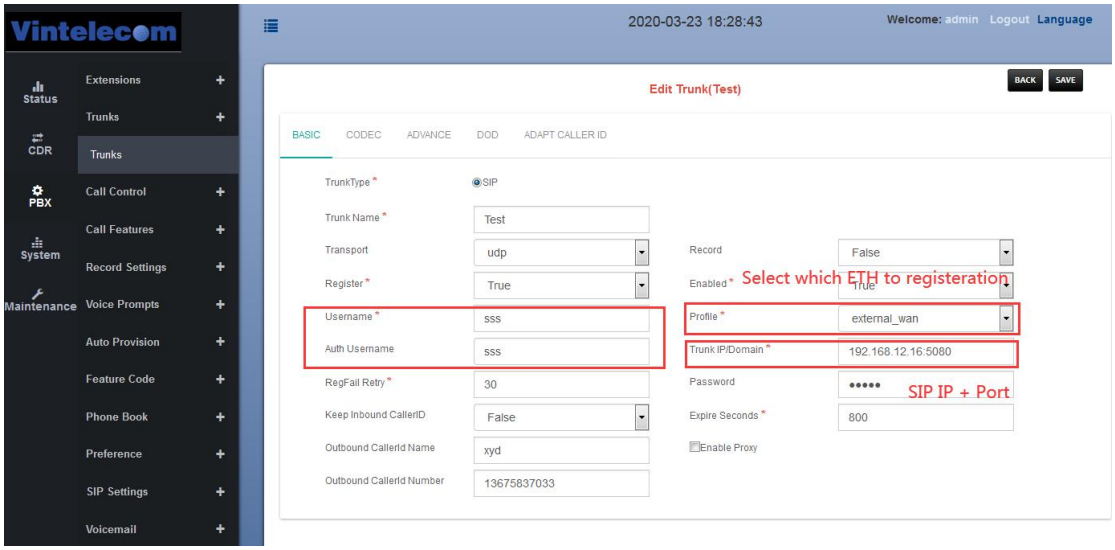

## 3.4. Inbound route for extension trunk

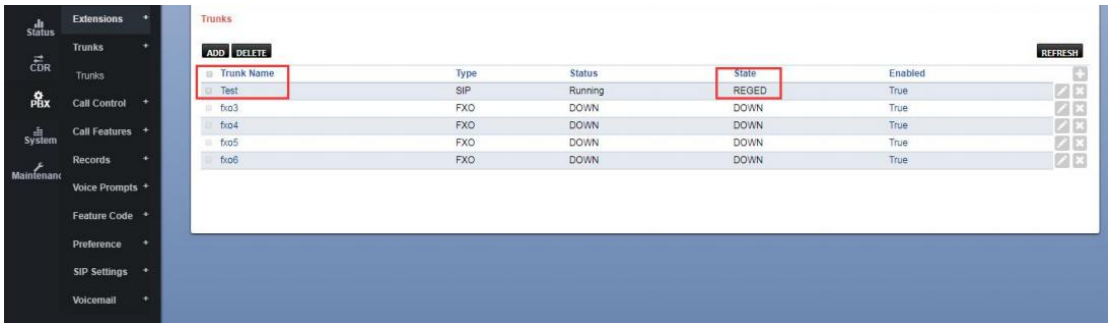
If the extension is selected as extension trunk, an auto inbound route will be inside IPPBX. The auto rule will be route by called numbers, for example, if 1002 is been called then 1002 extension will ringing.

The auto route will not be showing on Incoming route tables.

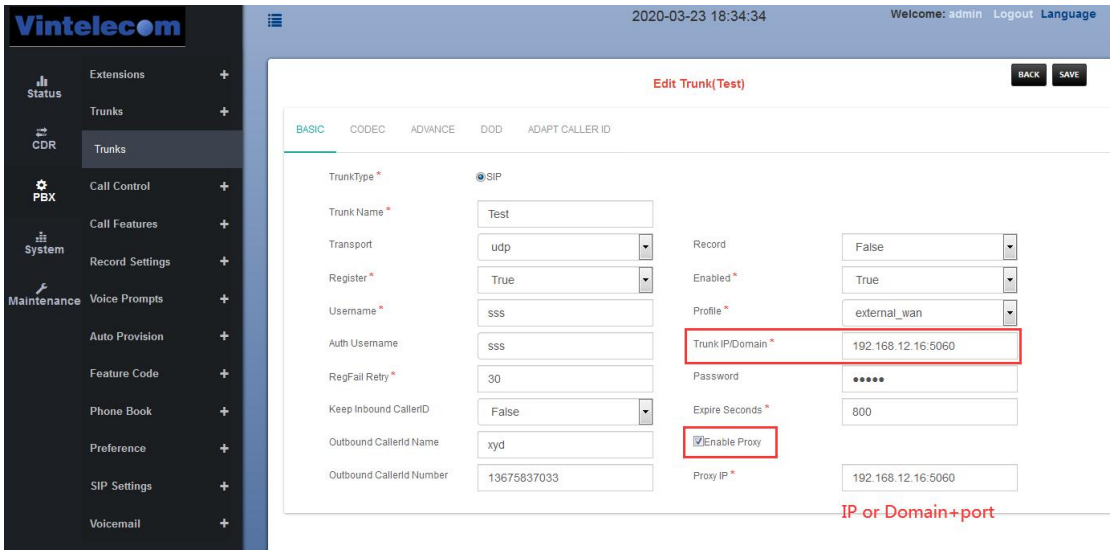
# 4. Connect UC to remote SIP server.

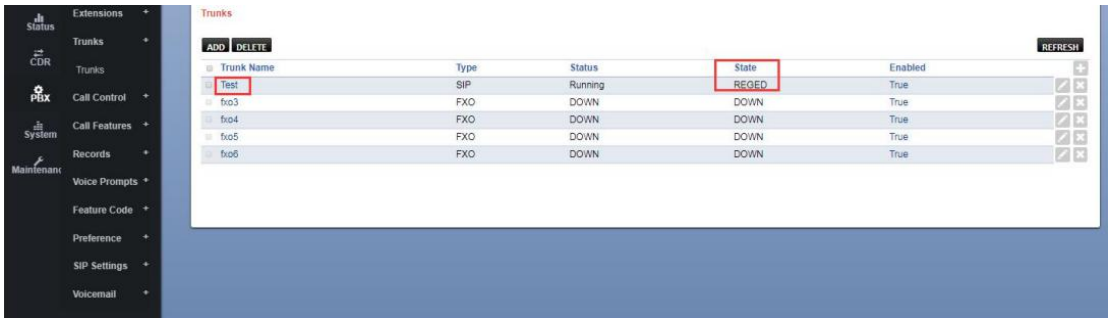
## 4.1 Without Outbound proxy



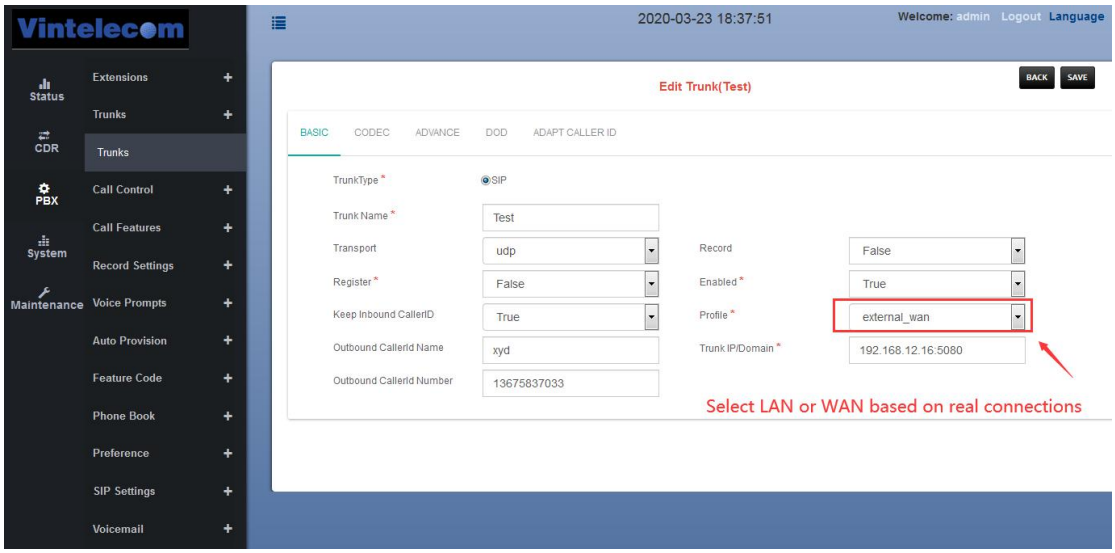

## 4.2 With outbound proxy



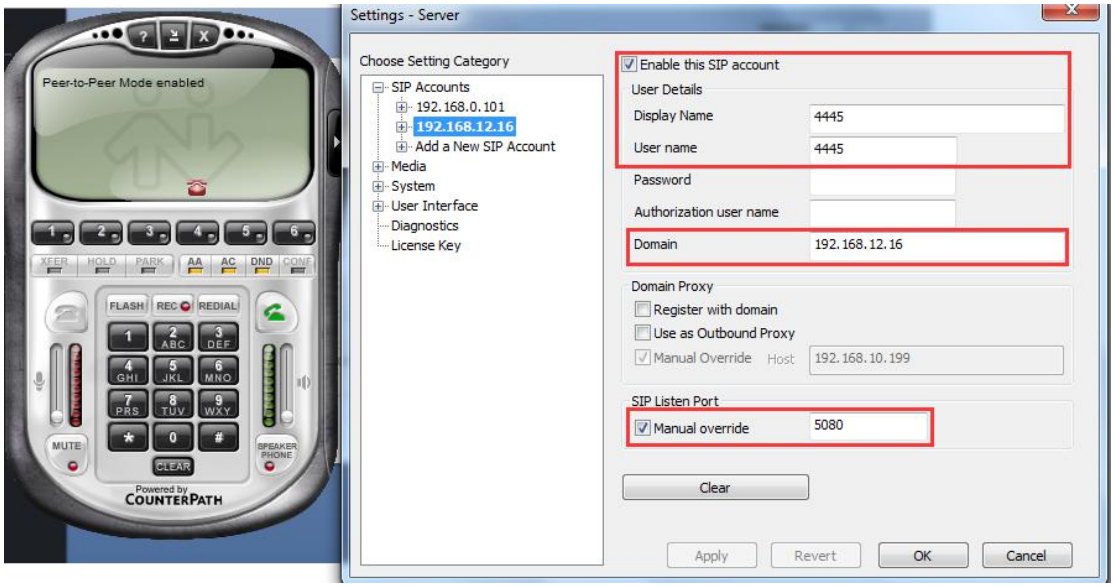



# 5. Call Functions

Create a peer to peer sip trunk name Test :




Use a soft phone on PC to set as in sip trunk IP/Domain:

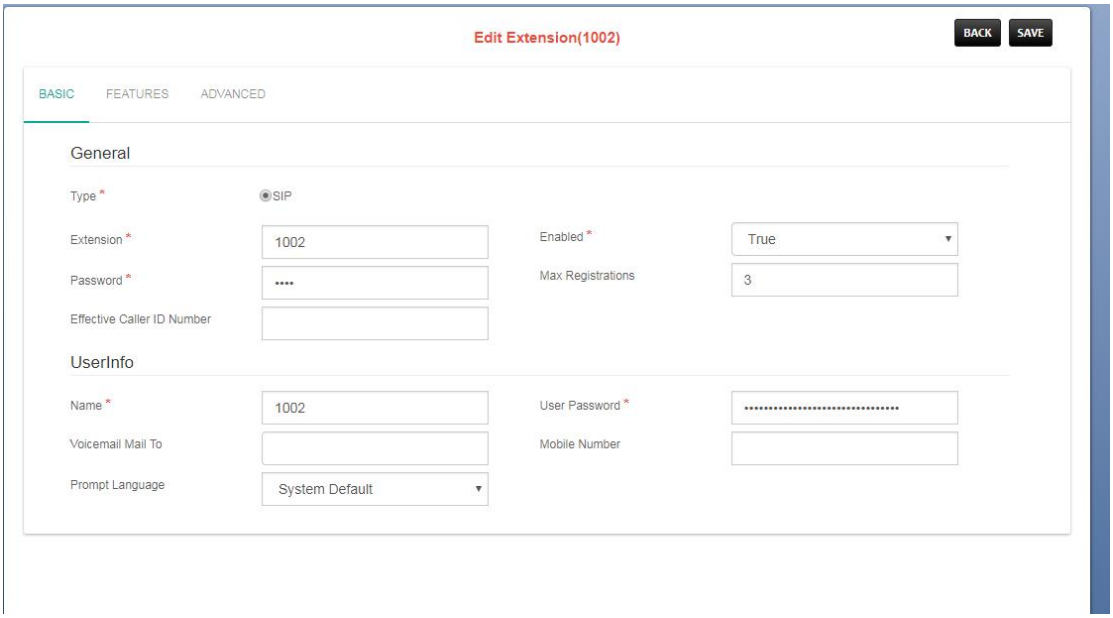


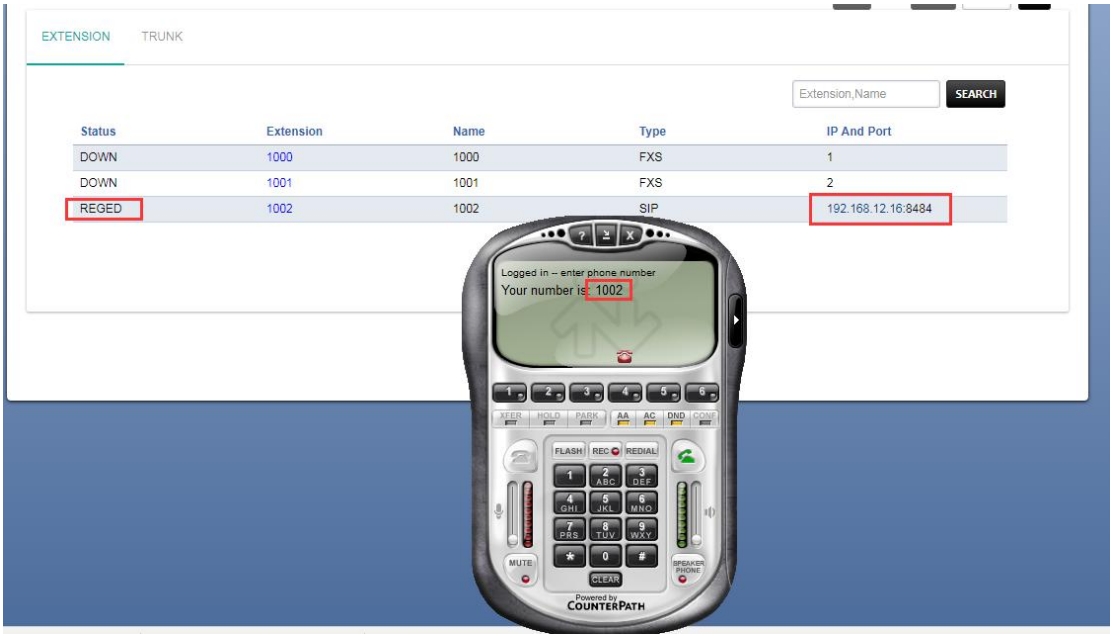

Topology:

IPPBX ------(peer to peer)----Eyebeam as a trunk

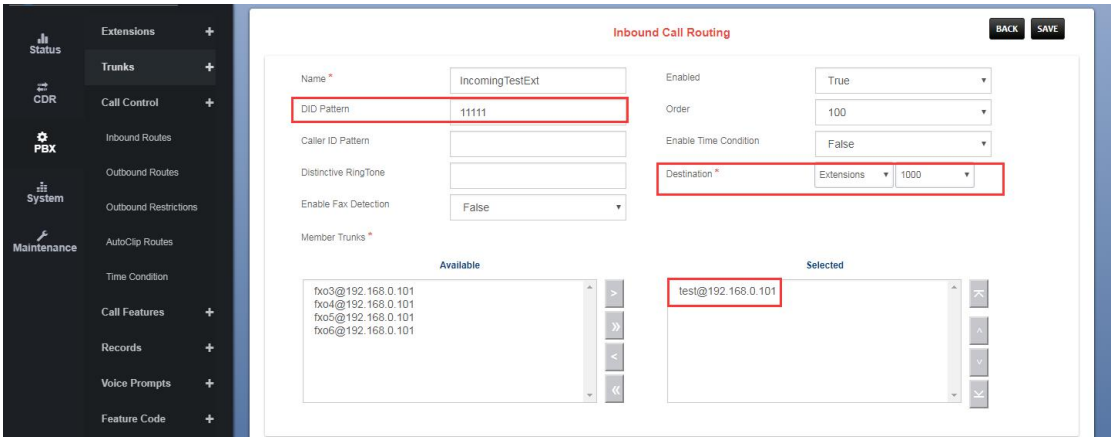
# 5.1. Incoming to Extension with fixed DID 11111

## 5.1.1. Make sure extension is created and registered.





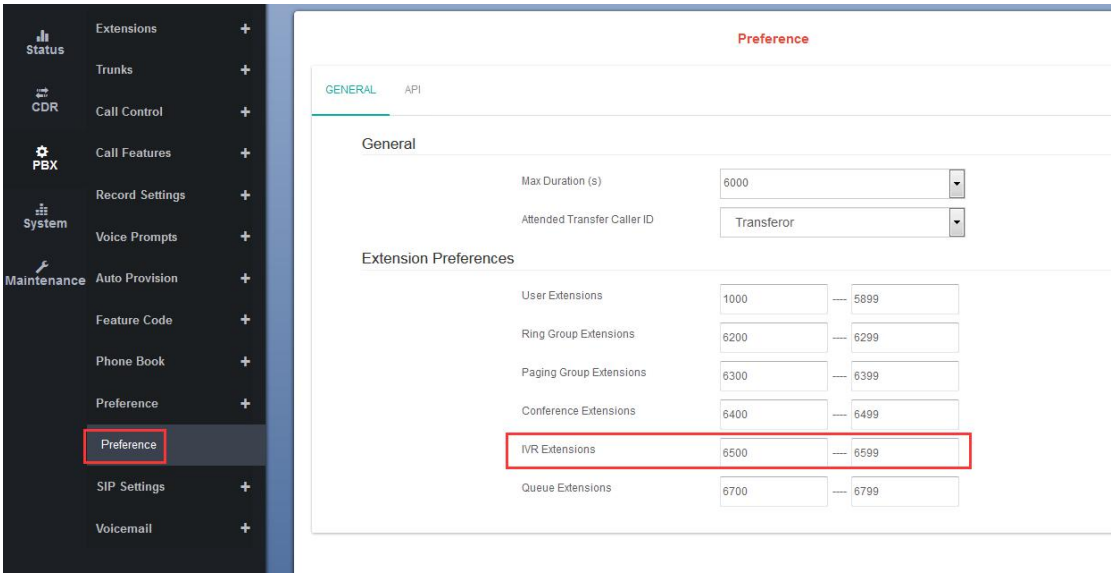

## 5.1.2. Create incoming route with fixed DID



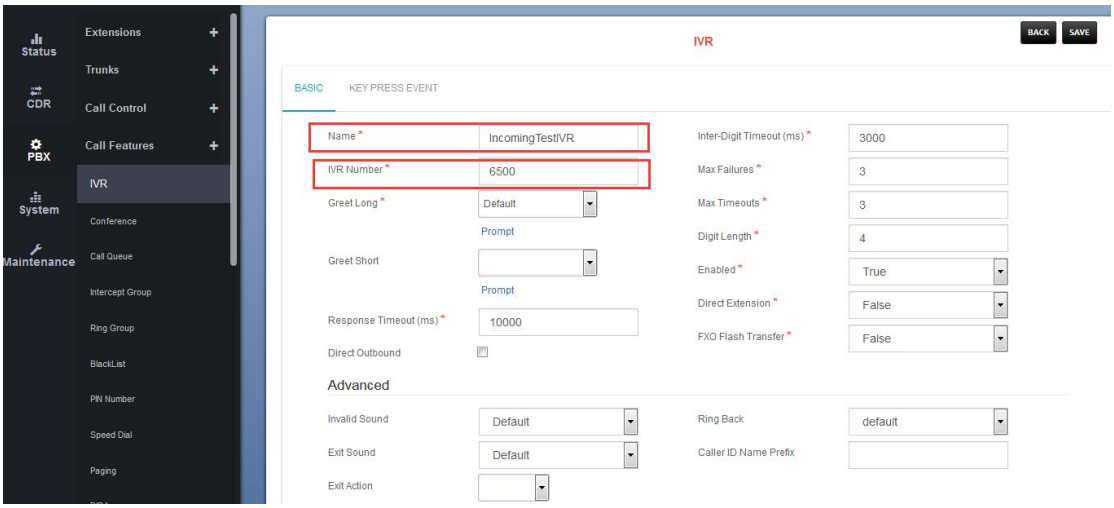

Call Test:

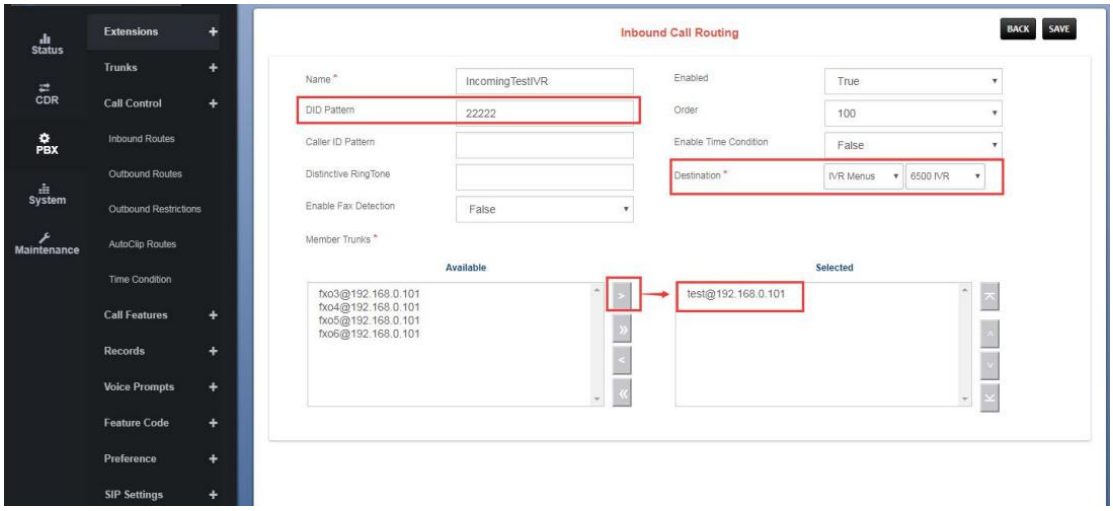

## 5.2. Incoming to IVR with fixed DID 22222

### 5.2.1. Check for IVR extension range



## 5.2.2. Create IVR profile



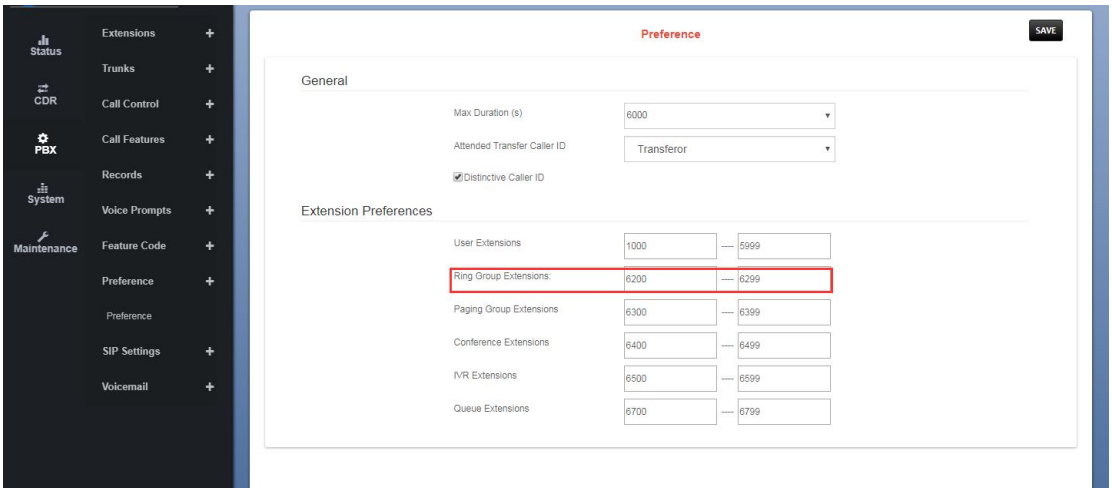




## 5.2.3. Create incoming route for IVR

Call Test:

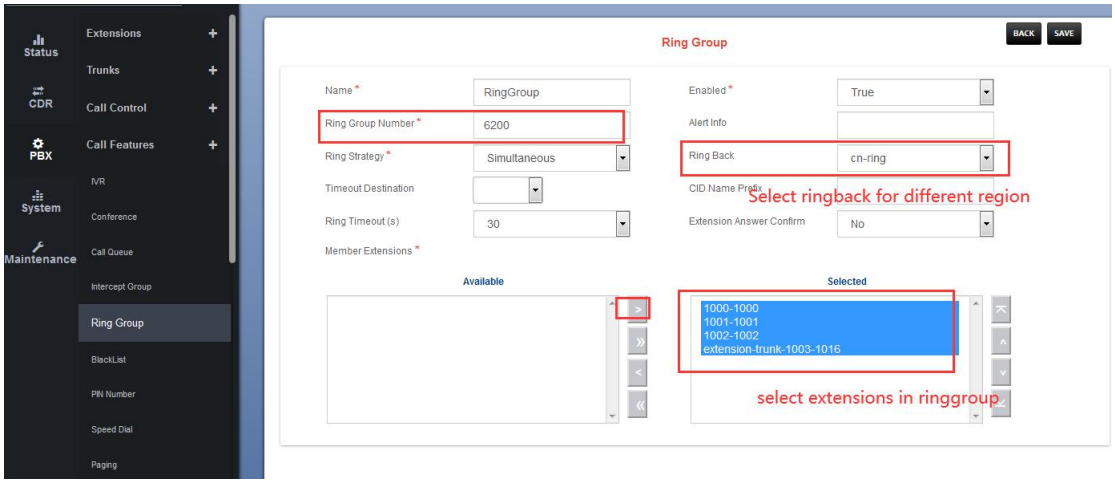

## 5.3. Incoming to Ring group with fixed DID 33333

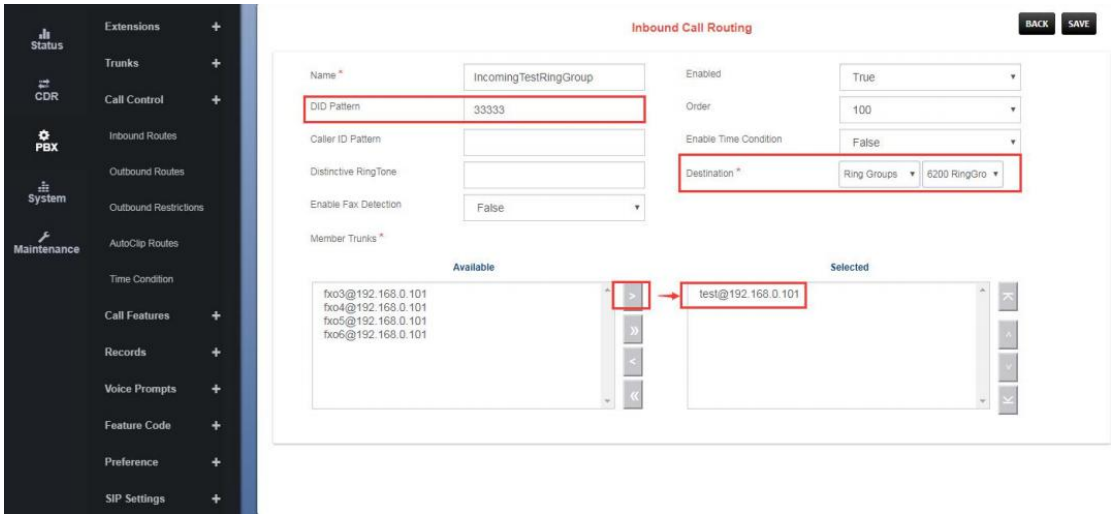
### 5.3.1. Check for Ring group number range in preference




### 5.3.2. Create Ring Group profile:

## 5.3.3. Create incoming route for Ring Group

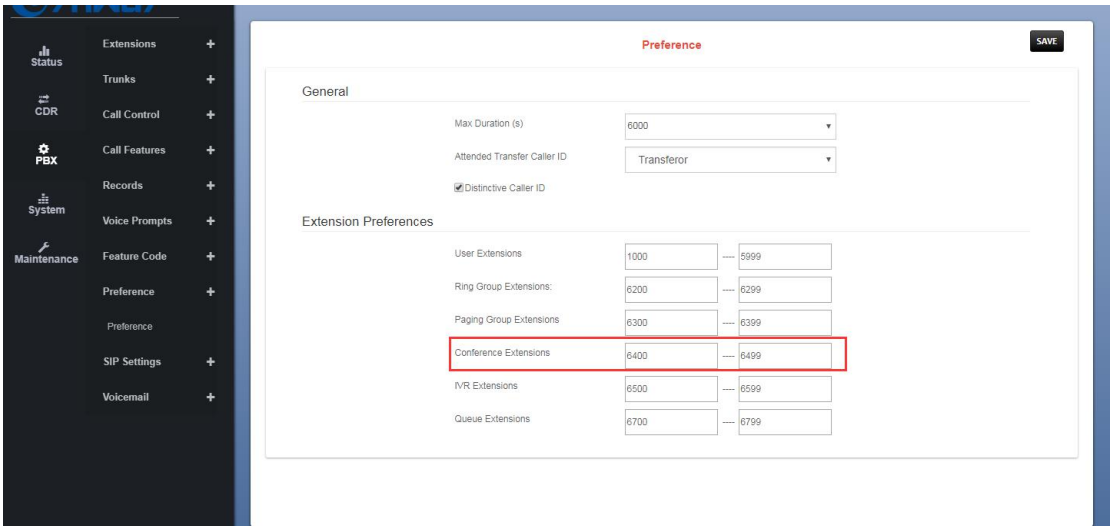


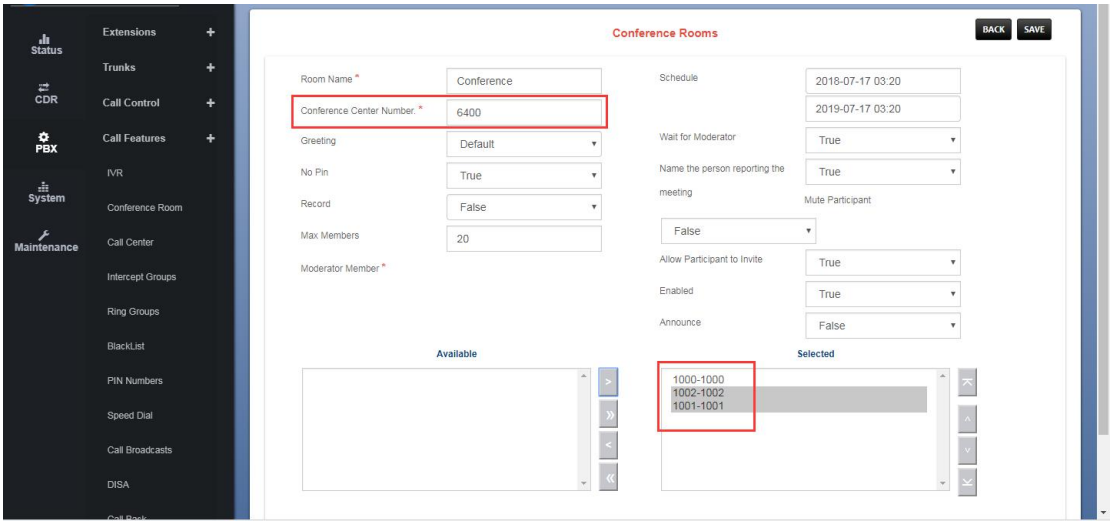

Call Test:


## 5.4. Incoming to Conference with fixed DID 44444

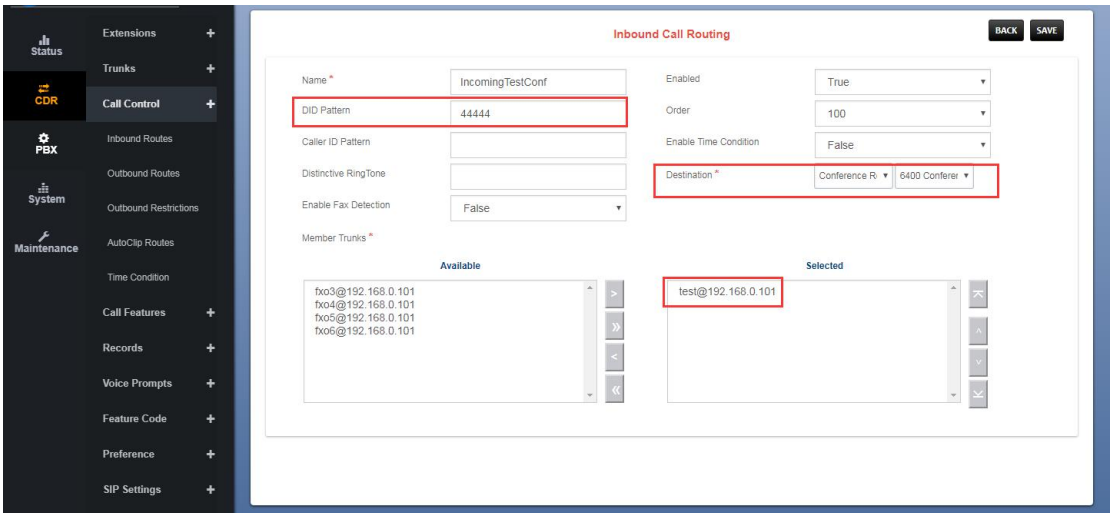
### 5.4.1. Check for Conference number range



## 5.4.2. Create conference profile



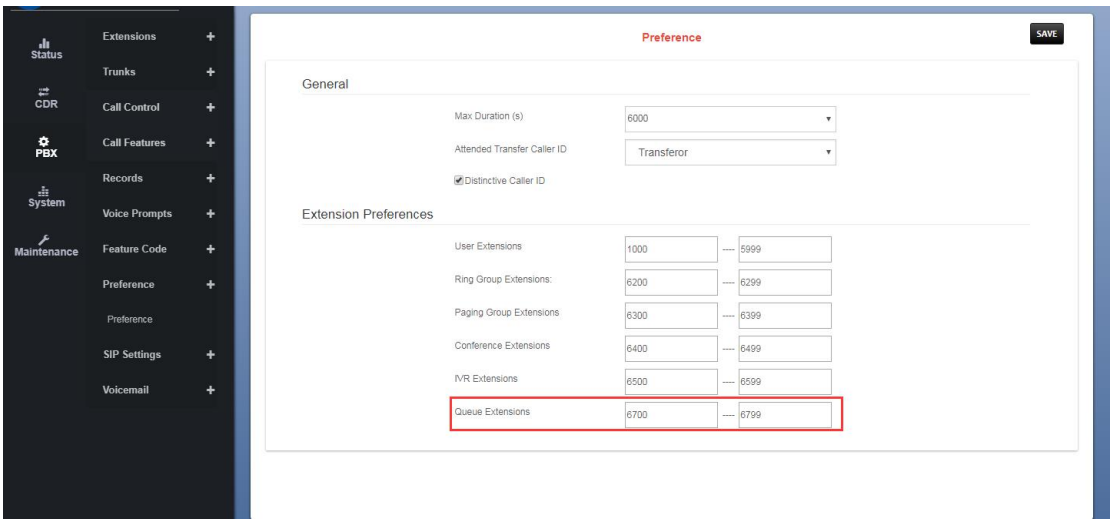

## 5.4.3. Create incoming route for conference



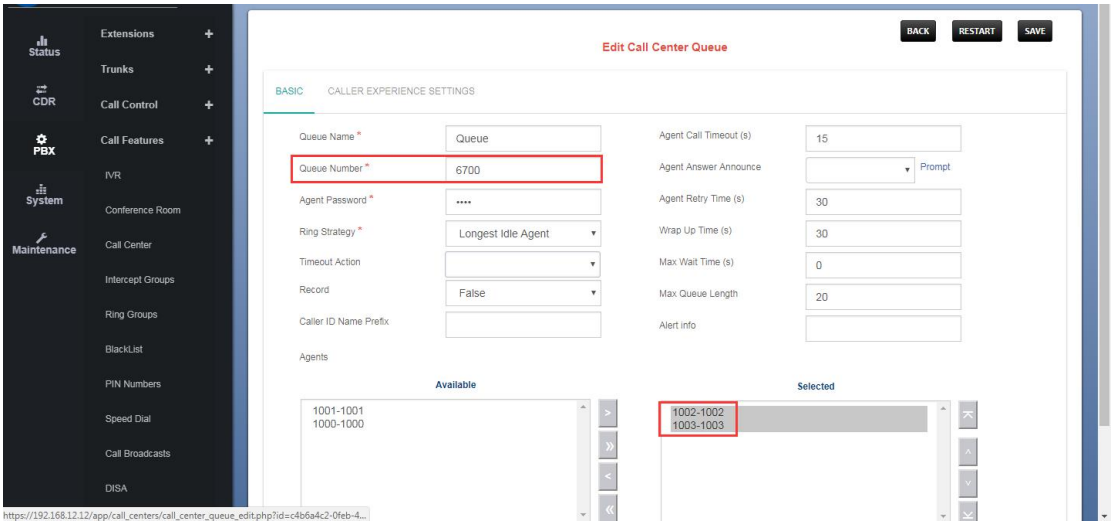

Call Test:


## 5.5. Call Center Queue with fixed DID 55555

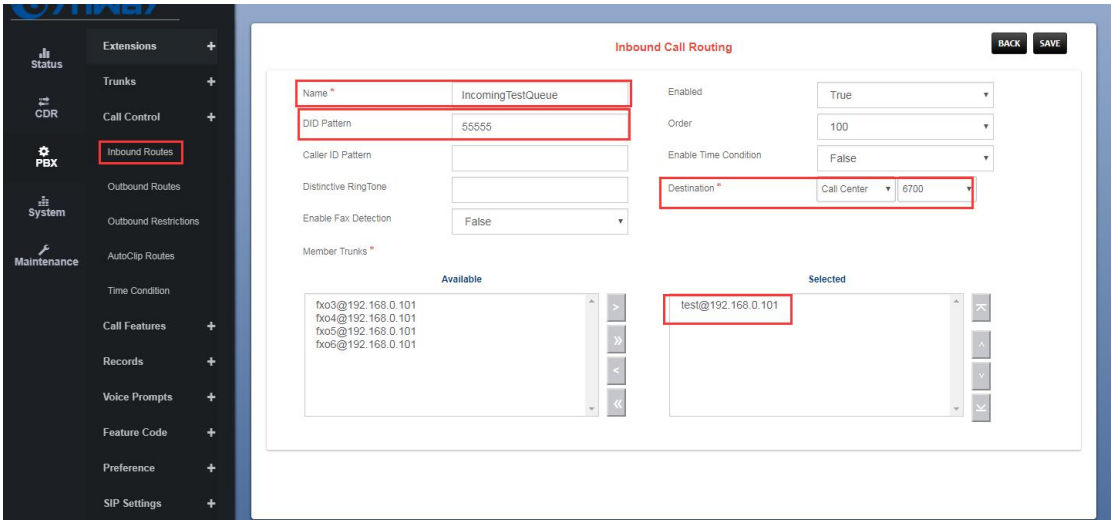
### 5.5.1. Check for call center queue number range in preference



## 5.5.2. Create call queue




## 5.5.3. Create incoming route for Call Queue

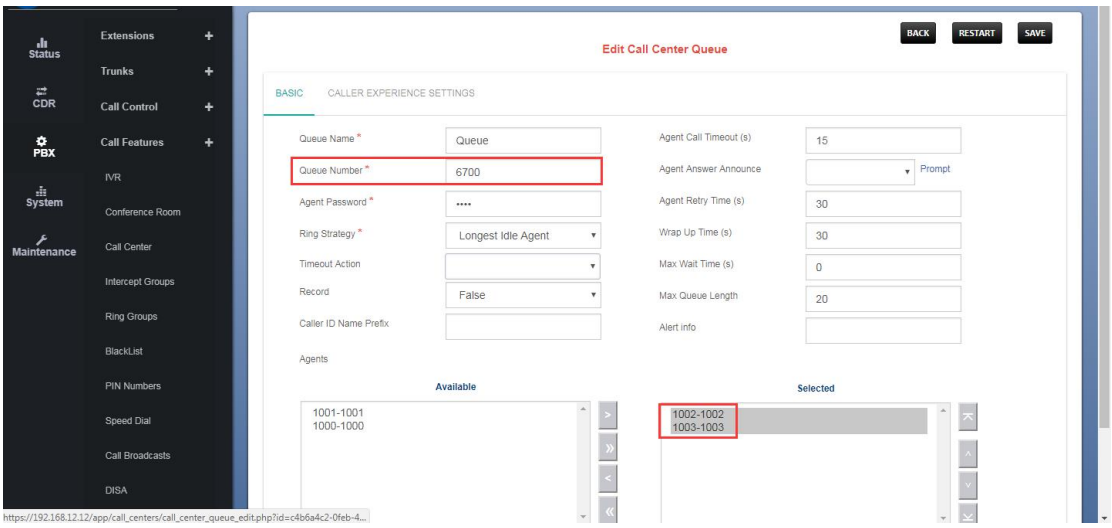



Call Test:

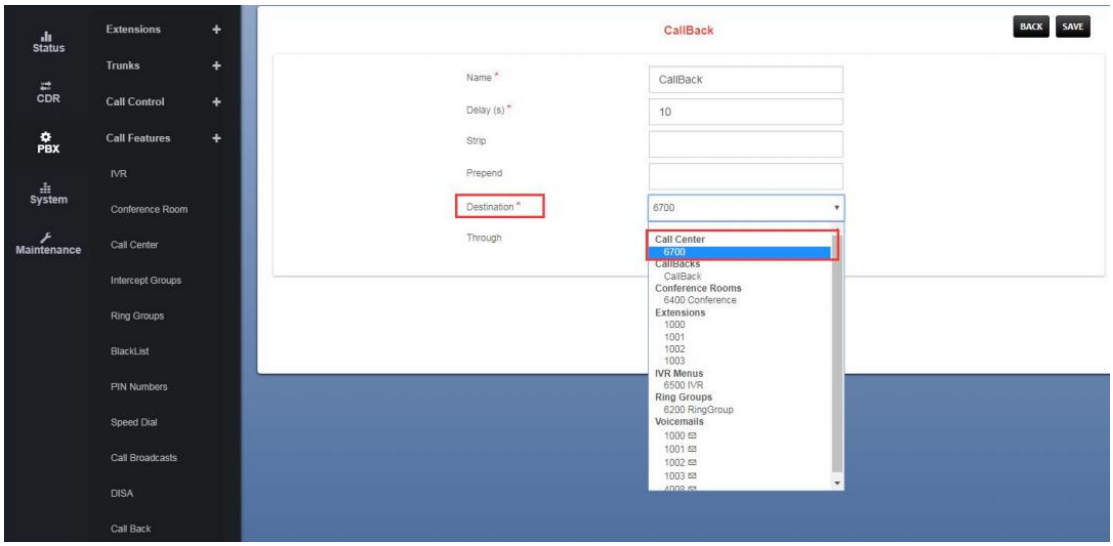

## 5.6. Call Back

Sample case : Call back with call Queue with fixed DID 66666

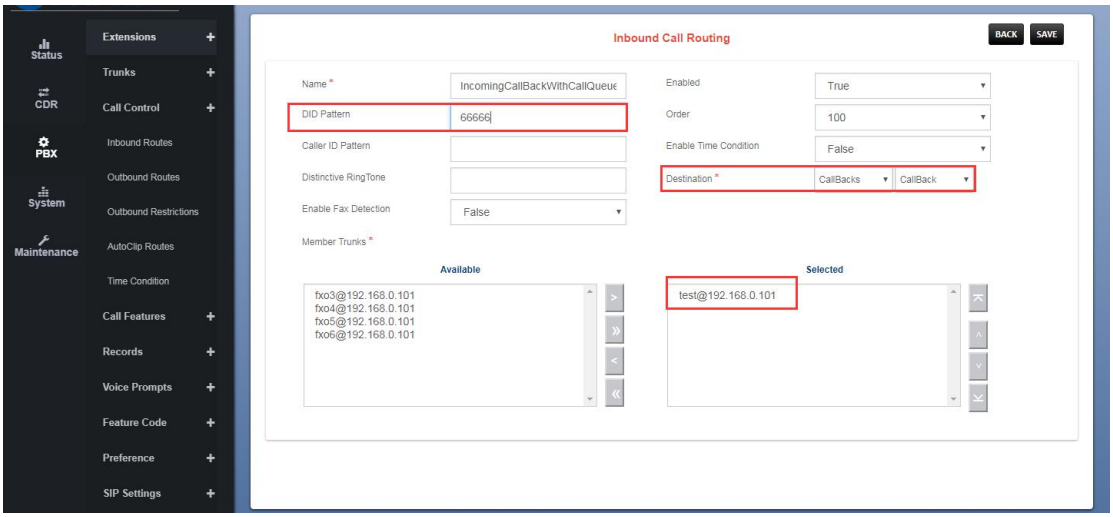
### 5.6.1. Follow 5.5 create a call center queue



## 5.6.2. Create call back profile, and lead call to call center.




## 5.6.3. Create incoming route for call back




Call Test:


*In this case, when there's incoming call, the call will be lead to callback profile. So the call will be replied 486 to remote sip trunk(hangup), and after 10 seconds IPPBX will callback to the caller while the call will be directed to call queue inside IPPBX at the same time. After remote caller is pickup, extensions in call queue will ring.*