

UC200 IPPBX Test Scenarios

By XIAOYINDI

Content

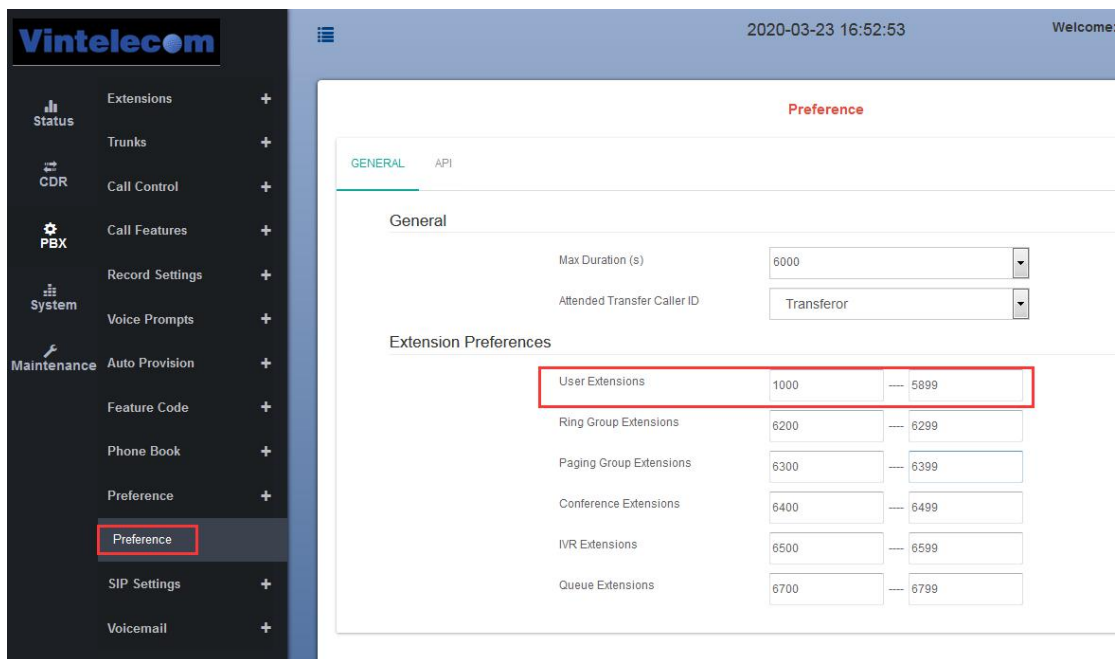
UC200 IPPBX Test Scenarios.....	1
1. Register Extensions.....	4
1.1. Create Extension.....	4
1.1.1. Create single extension.....	4
1.1.2. Mass/bulk create extensions.....	5
1.2. Registration.....	6
2. Peer to Peer SIP trunks.....	8
2.1. Create sip trunk.....	8
2.2. Inbound Route.....	8
2.3. Outbound Route.....	9
2.4. Connect to gsm/4g gateways.....	9
2.4.1. Analog, GSM gateway.....	9
2.4.2. E1 gateways.....	11
2.5. Connect to another UC200 or Other PBX(peer to peer).....	14
3. Register GSM/4g Gateway to IPPBX as SIP trunk.(Extension Trunk).....	18
3.1. Create account.....	18
3.2. Register GSM gateway to UC.....	18
3.3. Create outbound route for extension trunk.....	20
3.4. Inbound route for extension trunk.....	20
4. Connect UC to remote SIP server.....	21
4.1 Without Outbound proxy.....	21
4.2 With outbound proxy.....	21
5. Call Functions.....	22
5.1. Incoming to Extension with fixed DID 11111.....	23
5.1.1. Make sure extension is created and registered.....	23
5.1.2. Create incoming route with fixed DID.....	24
5.2. Incoming to IVR with fixed DID 22222.....	25
5.2.1. Check for IVR extension range.....	25
5.2.2. Create IVR profile.....	26
5.2.3. Create incoming route for IVR.....	26
5.3. Incoming to Ring group with fixed DID 33333.....	27
5.3.1. Check for Ring group number range in preference.....	27
5.3.2. Create Ring Group profile:.....	27
5.3.3. Create incoming route for Ring Group.....	28
5.4. Incoming to Conference with fixed DID 44444.....	29
5.4.1. Check for Conference number range.....	29
5.4.2. Create conference profile.....	30
5.4.3. Create incoming route for conference.....	30
5.5. Call Center Queue with fixed DID 55555.....	31
5.5.1. Check for call center queue number range in preference.....	31
5.5.2. Create call queue.....	32
5.5.3. Create incoming route for Call Queue.....	32
5.6. Call Back.....	33

5.6.1. Follow 5.5 create a call center queue.....	33
5.6.2. Create call back profile, and lead call to call center.....	34
5.6.3. Create incoming route for call back.....	34

1. Register Extensions.

1.1. Create Extension

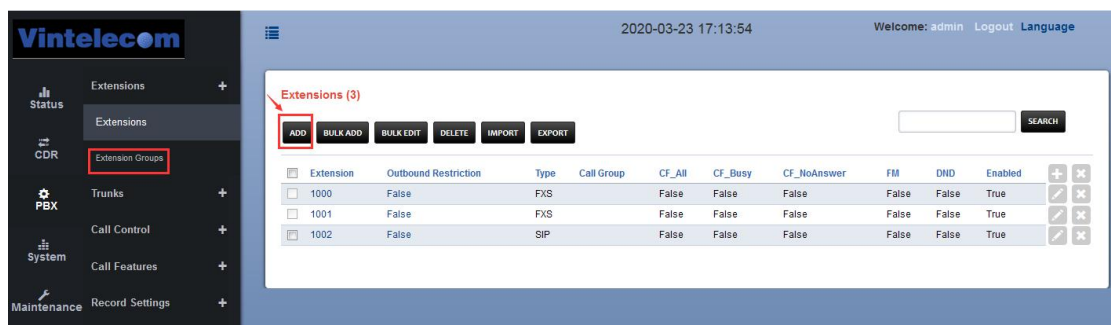
Default Extension Range:



The screenshot shows the Vintelecom web interface. On the left is a dark sidebar with a menu. The 'Preference' option is highlighted with a red box. The main content area is titled 'Preference' and has a 'GENERAL' tab selected. Under the 'Extension Preferences' section, there is a table of extension ranges. The 'User Extensions' row is highlighted with a red box.

Extension Preference	Start	End
User Extensions	1000	5899
Ring Group Extensions	6200	6299
Paging Group Extensions	6300	6399
Conference Extensions	6400	6499
IVR Extensions	6500	6599
Queue Extensions	6700	6799

1.1.1. Create single extension



The screenshot shows the Vintelecom web interface displaying a list of extensions. The 'Extensions Groups' option in the sidebar is highlighted with a red box. The main content area shows a table of extensions with columns for Extension, Outbound Restriction, Type, Call Group, CF_All, CF_Busy, CF_NoAnswer, FM, DND, Enabled, and actions. The 'ADD' button is highlighted with a red box.

Extension	Outbound Restriction	Type	Call Group	CF_All	CF_Busy	CF_NoAnswer	FM	DND	Enabled	+	x
1000	False	FXS		False	False	False	False	False	True	✓	✗
1001	False	FXS		False	False	False	False	False	True	✓	✗
1002	False	SIP		False	False	False	False	False	True	✓	✗

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.

Vintelecom 2020-03-23 17:07:53 Welcome: admin Logout Language

Bulk Add Extension [BACK] [SAVE]

BASIC FEATURES ADVANCED

General

Type * SIP

Start Extension * Enabled *

Create Number * Max Registrations **Create 5 extension in sequence**

Registration Password *

Fixed Password *

Effective Caller ID Number

Userinfo

Voicemail Mail To Mobile Number

Prompt Language

Select CODEC

Bulk Add Extension [BACK] [SAVE]

BASIC FEATURES **ADVANCED**

RTP Settings

Enable SRTP SIP Bypass Media

Absolute Codec String

Available Selected

PCMU
PCMA
GSM
G729
G722
G723
G726
speex
AMR

Register Settings

Auth ACL

Force ping

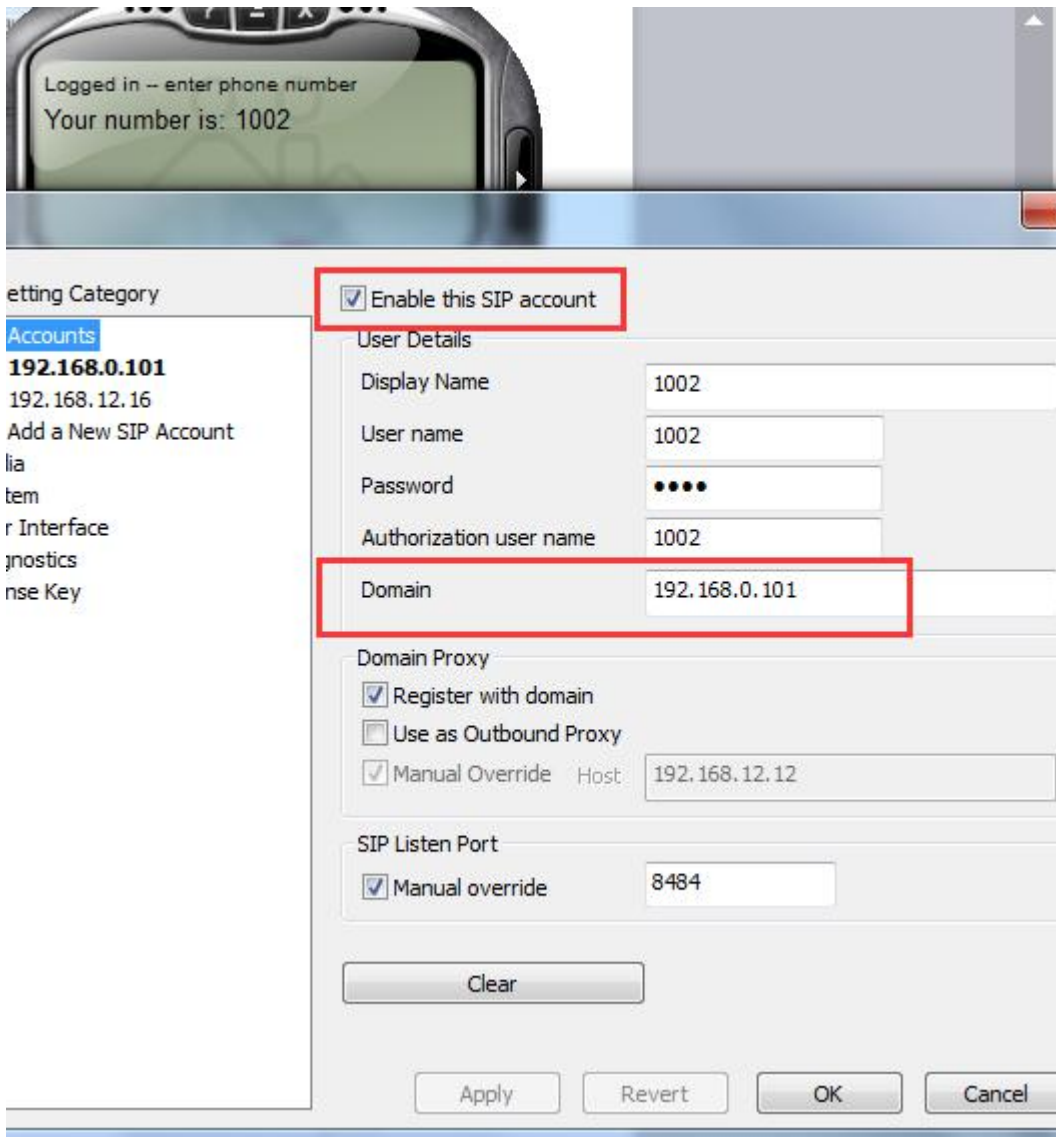
User Agent Filter +

Sip Expires Max Deviation (s)

SIP Force Expires (s)

1.2. Registration

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



2. Peer to Peer SIP trunks

2.1. Create sip trunk

The screenshot shows the 'Add Trunk' configuration page in the Vintelecom interface. The page is titled 'Add Trunk' and has tabs for 'BASIC', 'CODEC', 'ADVANCE', 'DOD', and 'ADAPT CALLER ID'. The 'BASIC' tab is active. The configuration includes the following fields:

- TrunkType: SIP
- Trunk Name: Test
- Transport: udp
- Register: False (highlighted with a red box)
- Record: False
- Enabled: True
- Profile: external_lan
- Keep Inbound CallerID: True
- Trunk IP/Domain: 192.168.12.13:5060 (highlighted with a red box and labeled 'Remote device IP+domain')
- Outbound CallerID Name: xyd
- Outbound CallerID Number: 13675837033

Buttons for 'BACK' and 'SAVE' are located at the top right of the configuration area.

2.2. Inbound Route

Incoming call with DID 12345 will be lead to extension 1002

The screenshot shows the 'Inbound Call Routing' configuration page in the Vintelecom interface. The page is titled 'Inbound Call Routing' and has buttons for 'BACK' and 'SAVE'. The configuration includes the following fields:

- Name: Incoming Test
- Enabled: True
- DID Pattern: 12345 (highlighted with a red box)
- Priority: 100
- Caller ID Pattern: (empty)
- Enable Inband DTMF Detect...: False
- Distinctive RingTone: (empty)
- Enable Time Condition: False
- Enable Fax Detection: False
- Destination: Extensions 1002 (highlighted with a red box)
- Send RingTone: Remote
- Member Trunks: (empty)

At the bottom, there are two lists of available trunks:

- Available: fx03@IPPBX, fx04@IPPBX, fx05@IPPBX, fx06@IPPBX
- Selected: Test@IPPBX

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:

Port	Type	State	Voice Type	Direction	CallerID	CalleeID	Cell Phone No.	Connection	Signal	SIP Reg Status
1	GSM	Unusable	---	---	---	---	---	Disconnect		Unregistered
2	GSM	Idle	---	---	---	---	---	Connect		Unregistered
3	GSM	Unusable	---	---	---	---	---	Disconnect		Unregistered
4	GSM	Unusable	---	---	---	---	---	Disconnect		Unregistered
5	GSM	Unusable	---	---	---	---	---	Disconnect		Unregistered
6	GSM	Unusable	---	---	---	---	---	Disconnect		Unregistered
7	GSM	Unusable	---	---	---	---	---	Disconnect		Unregistered
8	GSM	Unusable	---	---	---	---	---	Disconnect		Unregistered

Analog:

Channel State									
Channel	Type	Number	Voltage(V)	State	Direction	CallerID	CalleeID	Reg Status	Polarity Reversal Count
1	FXS	8001	0		---	---	---	Unregistered	---
2	FXS	8002	0		---	---	---	Unregistered	---
3	FXS	8003	0		---	---	---	Unregistered	---
4	FXS	8004	0		---	---	---	Unregistered	---
5	FXS	8005	0		---	---	---	Unregistered	---
6	FXS	8006	0		---	---	---	Unregistered	---
7	FXS	8007	0		---	---	---	Unregistered	---
8	FXS	8008	0		---	---	---	Unregistered	---
9	FXO	8009	48		---	---	---	Unregistered	---
10	FXO	8010	0		---	---	---	Unregistered	---
11	FXO	8011	0		---	---	---	Unregistered	---
12	FXO	8012	0		---	---	---	Unregistered	---
13	FXO	8013	0		---	---	---	Unregistered	---
14	FXO	8014	0		---	---	---	Unregistered	---
15	FXO	8015	0		---	---	---	Unregistered	---
16	FXO	8016	0		---	---	---	Unregistered	---

Connected

Step 1:

Port: 2

Register Port: No

SIP Account: 8002

Connection Method: Static Binding

Bound Number: 12345

Echo Canceller: Enable

Forbid Outgoing Call: Enable

Forbid Ingoing Call: Enable

Caller ID Detection: Enable

Buttons: Modify, Reset, Cancel

Available port

Bind DID set on inbound route

Step 2: Put the port into port group

Index: 0

Description: default

Register Port Group: NO

Authentication Mode: Do Not Register

Port Select Mode: Cyclic Increase

Port: Port 1 Port 2 Port 3 Port 4 Port 5 Port 6 Port 7 Port 8

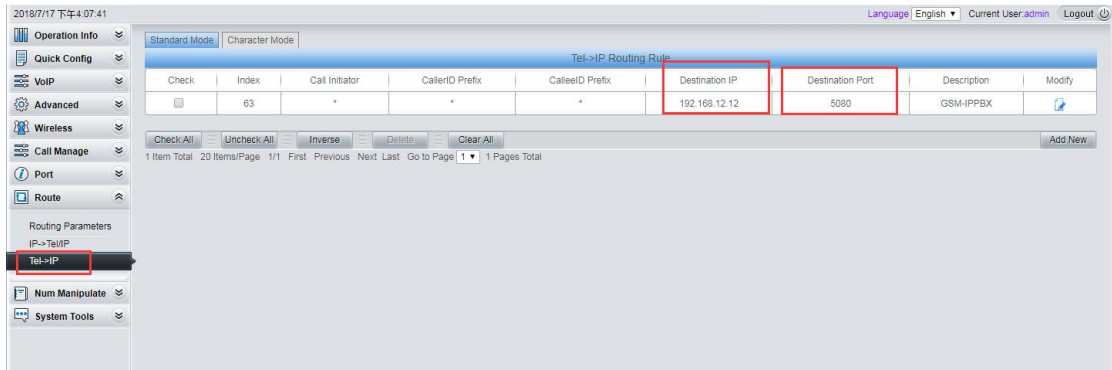
Buttons: Save, Cancel

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

IP->Tel/IP Routing Rule							
Check	Index	Source IP	CallerID Prefix	CalleeID Prefix	Call Destination	Description	Modify
<input type="checkbox"/>	83	*	*	*	Port Group 0	default	

1 Item Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total

Buttons: Check All, Uncheck All, Inverse, Delete, Clear All, Add New



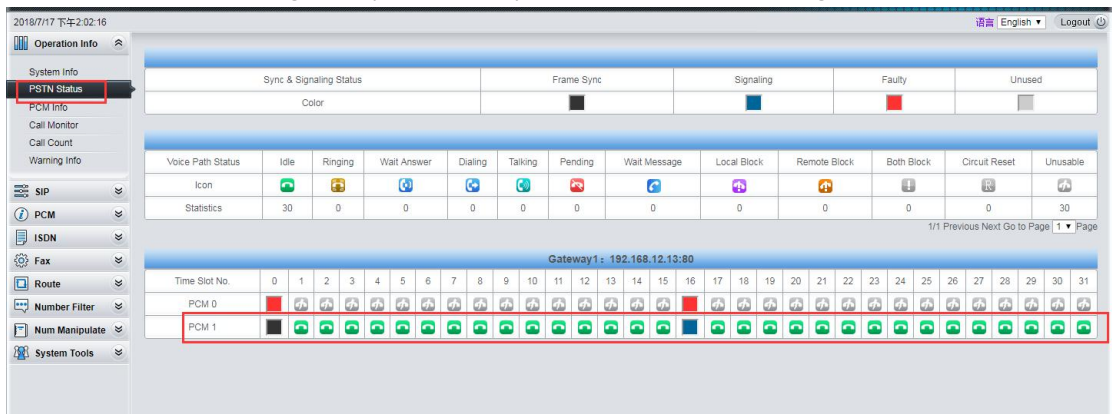
Destination port should be 5080 on IPPBX as peer to peer call.

Call Test:

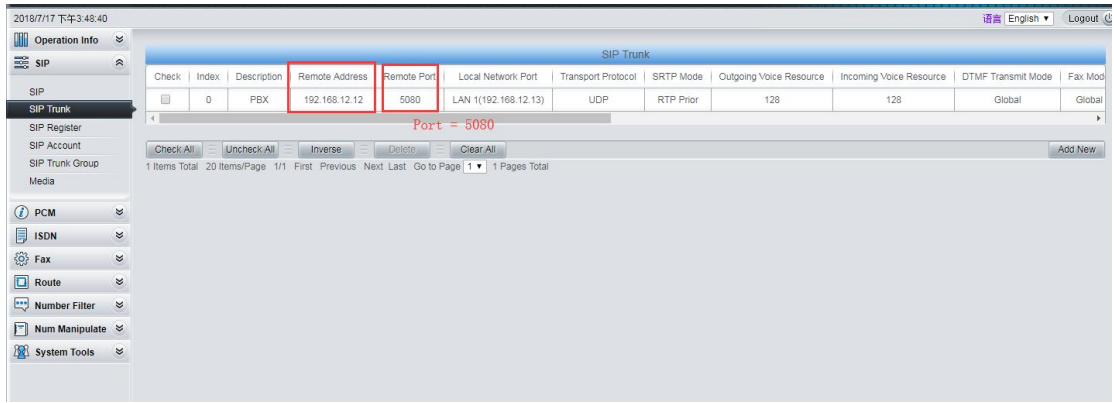


2.4.2. E1 gateways

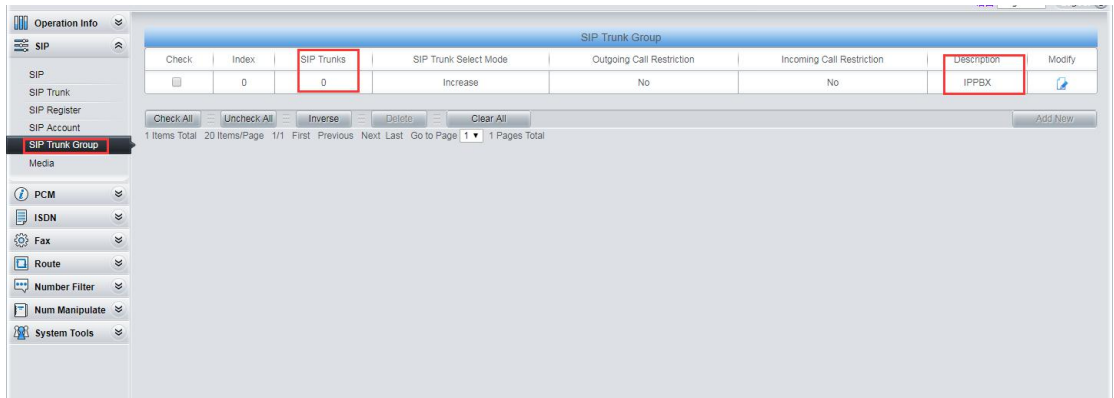
First make sure E1 gateway is well setup(PSTN status ok, channels goes idle)



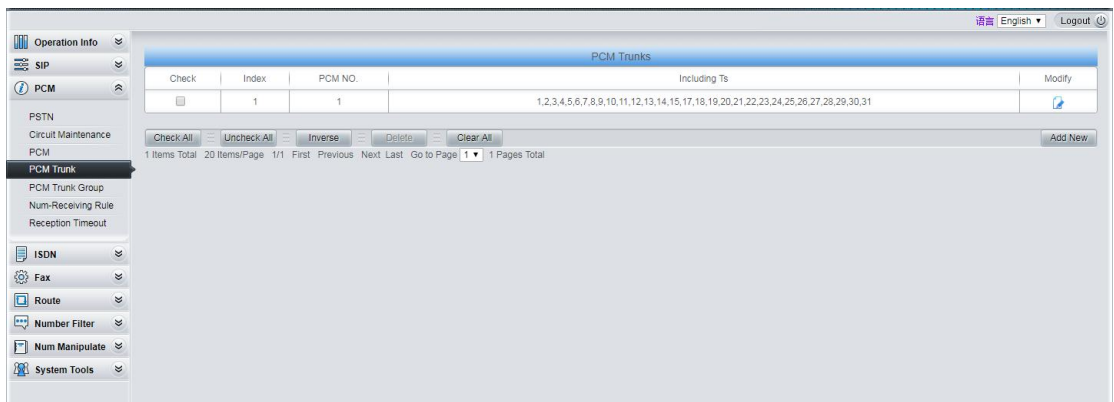
Step 1: Create SIP trunk



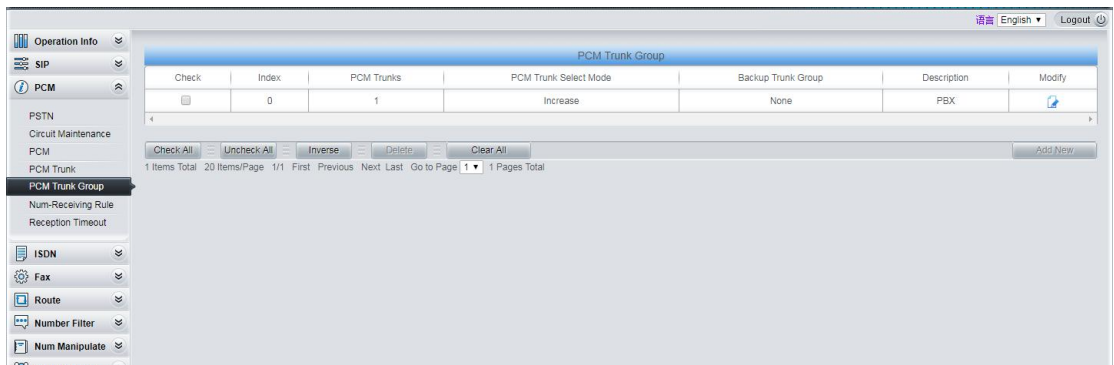
Step 2: Put sip trunk into sip trunk group



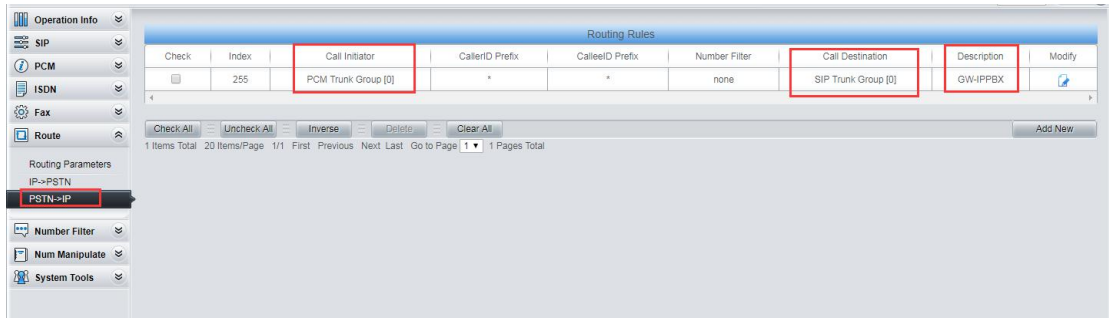
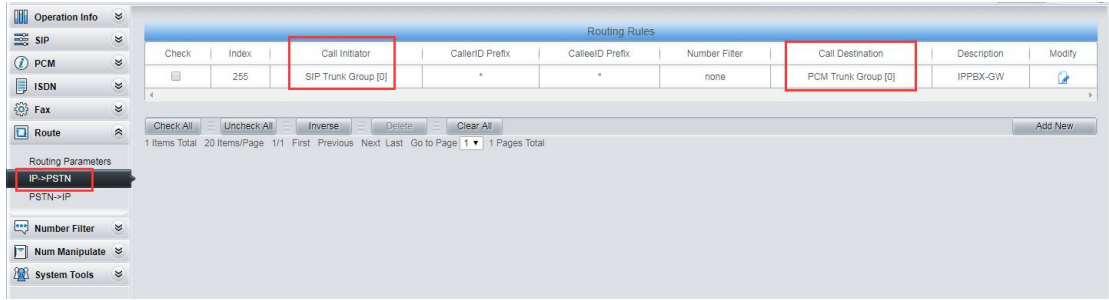
Step 3: Create PCM trunk:



Step 4: Put sip trunk into sip trunk group



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



Call Test

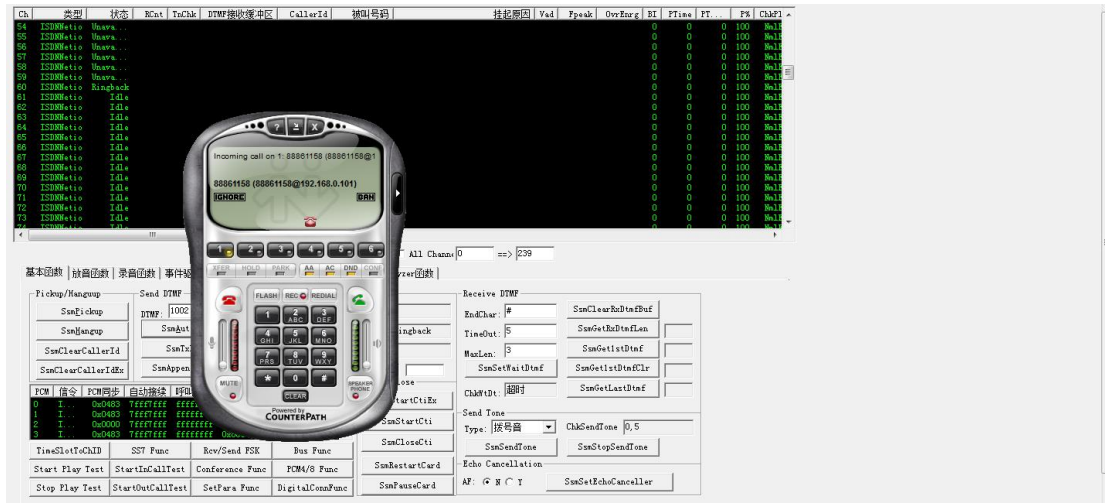
IP-PSTN

Call Log Data:

Ch	类型	状态	PC	Loc	Conn	Call ID	CallerID	被叫号码	呼叫原因	Val	Peak	OverErg	BT	PTime	PT	Pk	ChkPly	Co			
60	ISDN	Ringin					13615837035	7001								0	0	0	100	WaitEd	PL
61	ISDN	Ringin														0	0	0	100	WaitEd	PL
62	ISDN	IdLe														0	0	0	100	WaitEd	PL
63	ISDN	IdLe														0	0	0	100	WaitEd	PL
64	ISDN	IdLe														0	0	0	100	WaitEd	PL
65	ISDN	IdLe														0	0	0	100	WaitEd	PL
66	ISDN	IdLe														0	0	0	100	WaitEd	PL
67	ISDN	IdLe														0	0	0	100	WaitEd	PL
68	ISDN	IdLe														0	0	0	100	WaitEd	PL
69	ISDN	IdLe														0	0	0	100	WaitEd	PL
70	ISDN	IdLe														0	0	0	100	WaitEd	PL
71	ISDN	IdLe														0	0	0	100	WaitEd	PL
72	ISDN	IdLe														0	0	0	100	WaitEd	PL
73	ISDN	IdLe														0	0	0	100	WaitEd	PL
74	ISDN	IdLe														0	0	0	100	WaitEd	PL
75	ISDN	IdLe														0	0	0	100	WaitEd	PL
76	ISDN	IdLe														0	0	0	100	WaitEd	PL
77	ISDN	IdLe														0	0	0	100	WaitEd	PL
78	ISDN	IdLe														0	0	0	100	WaitEd	PL
79	ISDN	IdLe														0	0	0	100	WaitEd	PL
80	ISDN	IdLe														0	0	0	100	WaitEd	PL

Mobile Phone Screen: Session Progress, 7001 2192.168.0.101

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

System Info

INFORMATION	NETWORK	PERFORMANCE	STORAGE USAGE
LAN			
TYPE	Static IP		
MAC	80:7B:85:10:AB:EC		
IP Address	192.168.12.1		
Gateway	192.168.12.254		
Subnet Mask	255.255.255.0		
Preferred DNS Server			
Alternate DNS Server			
Network State	connection		
WAN			
TYPE	Static IP		
MAC	80:7B:85:10:AB:EC		
IP Address	192.168.1.101		
Subnet Mask	255.255.255.0		
Preferred DNS Server			
Alternate DNS Server			
Network State	disconnection		

Sip Settings

SAVE

VOIP SETTING **TRUNK PROFILE SETTING** EXTENSION PROFILE SETTING CERTIFICATE

Enable External_WAN

SIP IP: 192.168.1.101 SIP Port: 5080

Public SIP IP: 192.168.1.101 Public RTP IP: 192.168.1.101

Enable External_LAN

SIP IP: 192.168.12.1 SIP Port: 5080

Public SIP IP: 192.168.12.1 Public RTP IP: 192.168.12.1

Enable TLS

Notice:the CA certificate must be created ,if you want to use TLS

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

System Info

INFORMATION **NETWORK** PERFORMANCE STORAGE USAGE

LAN

TYPE	Static IP
MAC	00:00:E0:A7:05:5B
IP Address	192.168.12.102
Gateway	192.168.12.254
Subnet Mask	255.255.255.0
Preferred DNS Server	202.101.172.35
Alternate DNS Server	77.88.8.3
Network State	connection

WAN

TYPE	Static IP
MAC	00:00:E0:A7:05:5B
IP Address	192.168.12.96
Subnet Mask	255.255.255.0
Preferred DNS Server	202.101.172.35
Alternate DNS Server	8.8.8.8
Network State	connection

Sip Settings

SAVE

VOIP SETTING **TRUNK PROFILE SETTING** EXTENSION PROFILE SETTING RTP CODEC CERTIFICATE

Enable External_LAN

SIP IP: 192.168.12.102 SIP Port: 3080

Public SIP IP: autonat:115.238.50.123 Public RTP IP: autonat:115.238.50.123

Enable TLS

Notice:the CA certificate must be created ,if you want to use TLS

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

UC200_1:

Edit Trunk(UC200_2) BACK SAVE

BASIC CODEC ADVANCE
SIP trunk or FXO trunk

DOD ADAPT CALLER ID

TrunkType * SIP

Trunk Name *

Transport Record

Register * Enabled *

Keep Inbound CallerID Profile *

Outbound CallerId Name

Outbound CallerId Number

Trunk IP/Domain *

UC200_2:

2020-03-23 17:42:47 Welcome: admin Logout Language

Add Trunk BACK SAVE

BASIC CODEC ADVANCE DOD ADAPT CALLER ID

TrunkType * SIP

Trunk Name *

Transport Record

Register * Enabled *

Keep Inbound CallerID Profile *

Outbound CallerId Name

Outbound CallerId Number

Trunk IP/Domain *

Create inbound and outbound route for both UC200

Inbound route on UC200_1:

Inbound Call Routing

BACK SAVE

Name *	IncomingUC200_2	Enabled	True
DID Pattern		Order	100
Caller ID Pattern		Enable Inband DTMF	False
Distinctive RingTone		Detection	False
Enable Fax Detection	False	Enable Time Condition	
Member Trunks *		Destination *	IVR Menus 6500 IVR

Available

- Eyebeam@192.168.12.1
- fxo3@192.168.12.1
- fxo4@192.168.12.1
- fxo5@192.168.12.1
- fxo6@192.168.12.1
- HMP@192.168.12.1

Selected

- UC200_2@192.168.12.1

Outbound route on UC200_1:

Outbound Call Routing

BACK SAVE

Name *	OutboundUC200_2		
Dial Patterns	Patterns	Strip	Prepend
	2x*		Unit is ms
	Any number starts with 2 will transfer to UC200_2		
Member Extensions *			
	Available	Selected	
		<ul style="list-style-type: none"> 1000-1000 1001-1001 1002-1002 1003-1003 1004-1004 1005-1005 1006-1006 1007-1007 1008-1008 	
Member Gateways *			
	Available	Selected	
	<ul style="list-style-type: none"> Eyebeam@192.168.12.1 fxo3@192.168.12.1 fxo4@192.168.12.1 fxo5@192.168.12.1 fxo6@192.168.12.1 HMP@192.168.12.1 	<ul style="list-style-type: none"> UC200_2@192.168.12.1 	

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

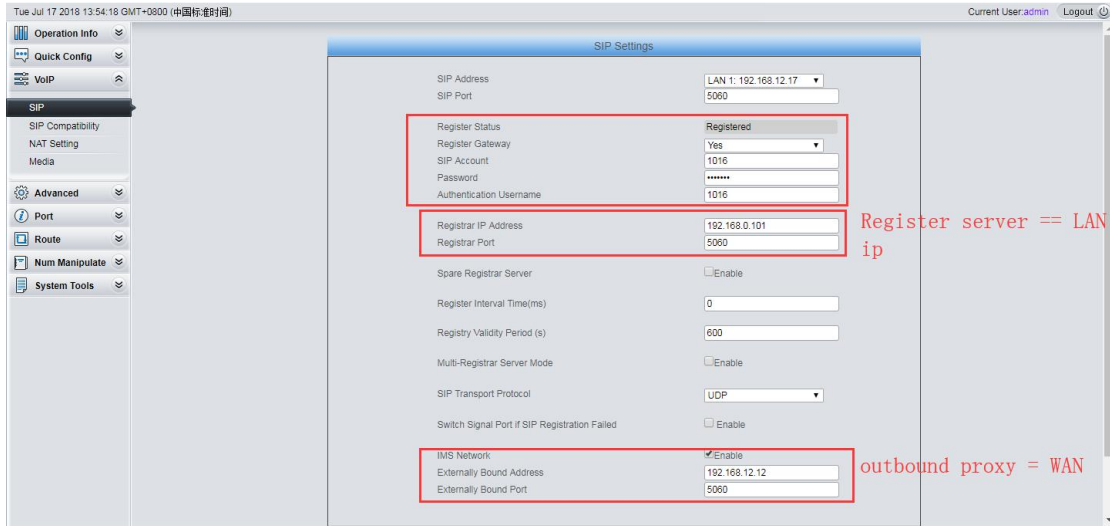
Follow steps in 1.1.1 Create Single Extension, but select Extension Trunk as True.

The screenshot shows the Vintelecom web interface for editing extension 1016. The interface is divided into several sections:

- General:** Type is SIP, Extension is 1016, Password is ext1016, Max Registrations is 3, and Enabled is True.
- User Info:** Name is 1003, User Password is masked, and Prompt Language is System Default.
- ADVANCED (Selected):**
 - RTP Settings:** Enable SRTP is False, SIP Bypass Media is Proxy Media.
 - Absolute Codec String:** A list of codecs is shown, with PCMU, PCMA, GSM, G729, G722, G723, and G726-16 selected.
 - Register Settings:** Auth ACL, Force ping (False), User Agent Filter, and SIP Force Contact are visible.
 - Call Settings:** Call Timeout (30s), Max Call Duration (6000s), Outbound Restriction (False), and Call Permission (International Call) are visible.
 - Extension Trunk:** This checkbox is checked and highlighted with a red box.

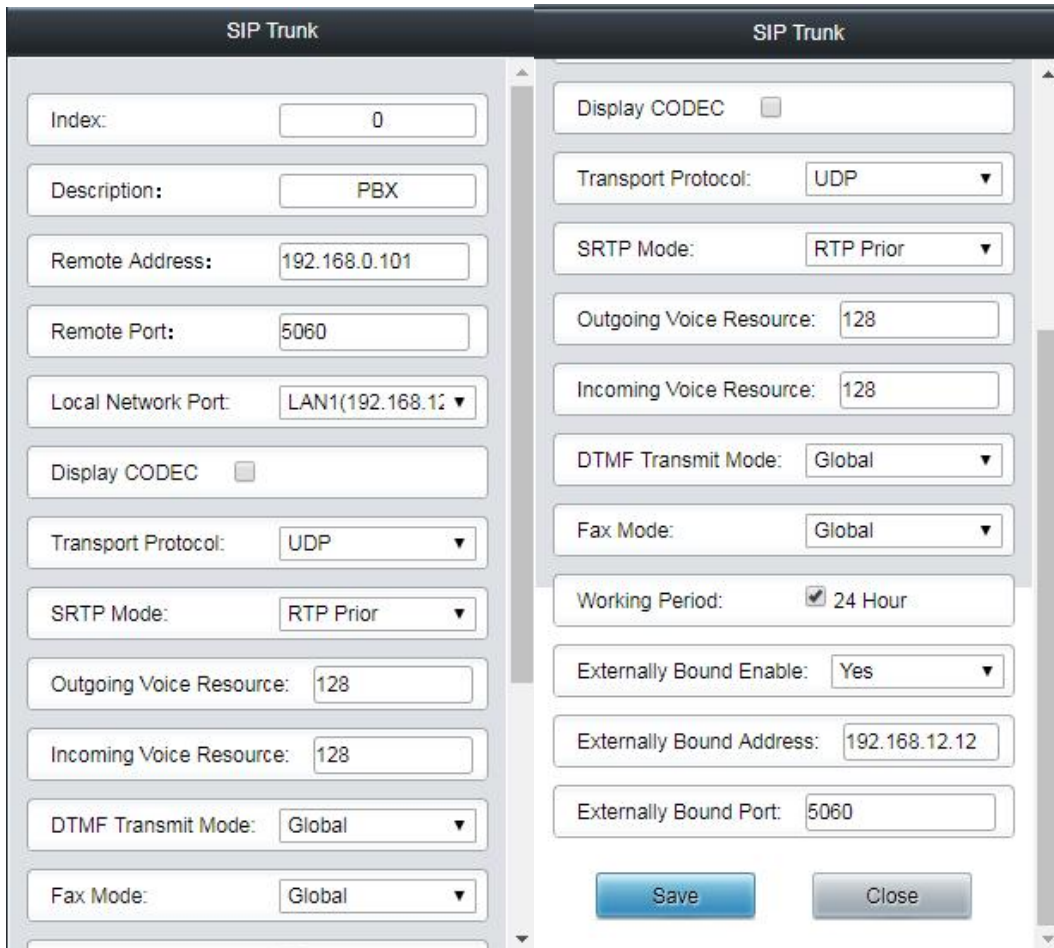
3.2. Register gsm gateway to UC200.

Analog, GSM gateways:

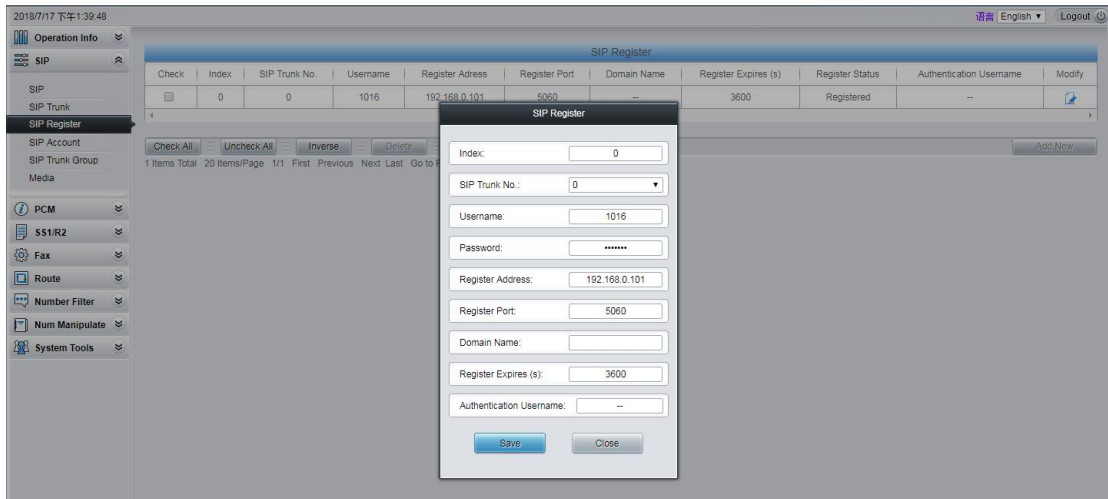


E1 gateways:

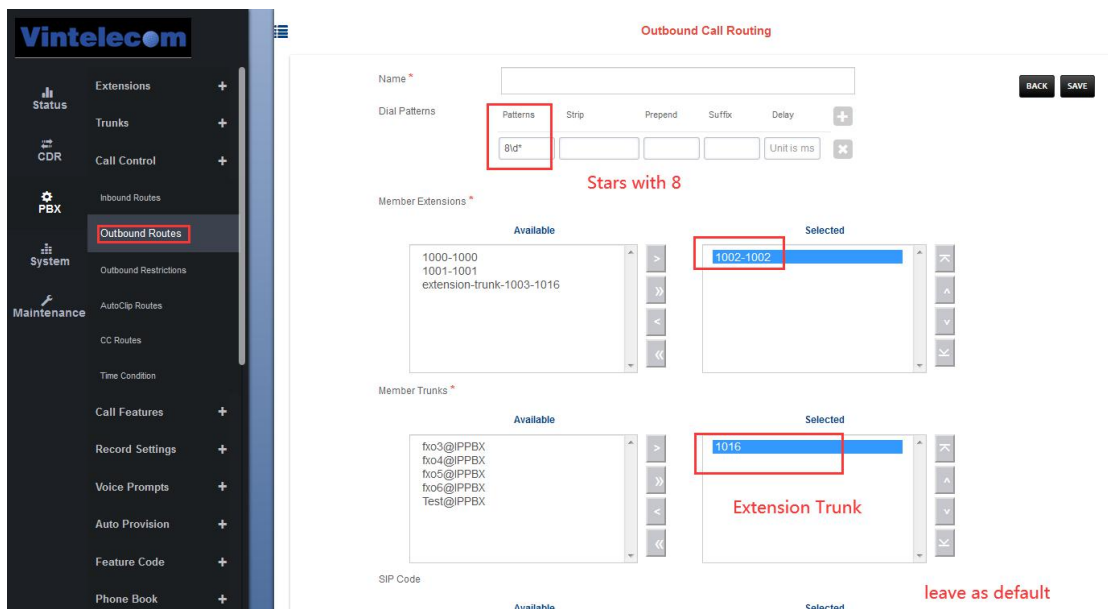
Create sip trunk:



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

Edit Trunk(Test)

TrunkType * SIP

Trunk Name *

Transport

Register *

Username *

Auth Username

RegFail Retry *

Keep Inbound CallerID

Outbound CallerID Name

Outbound CallerID Number

Record

Enabled *

Profile *

Trunk IP/Domain *

Password

Expire Seconds *

Enable Proxy

Trunks

Trunk Name	Type	Status	State	Enabled
Test	SIP	Running	REGED	True
fro3	FXO	DOWN	DOWN	True
fro4	FXO	DOWN	DOWN	True
fro5	FXO	DOWN	DOWN	True
fro6	FXO	DOWN	DOWN	True

4.2 With outbound proxy

Edit Trunk(Test)

TrunkType * SIP

Trunk Name *

Transport

Register *

Username *

Auth Username

RegFail Retry *

Keep Inbound CallerID

Outbound CallerID Name

Outbound CallerID Number

Record

Enabled *

Profile *

Trunk IP/Domain *

Password

Expire Seconds *

Enable Proxy

Proxy IP *

IP or Domain+port

Trunk Name	Type	Status	State	Enabled
Test	SIP	Running	REGED	True
fxo3	FXO	DOWN	DOWN	True
fxo4	FXO	DOWN	DOWN	True
fxo5	FXO	DOWN	DOWN	True
fxo6	FXO	DOWN	DOWN	True

5. Call Functions

Create a peer to peer sip trunk name Test :

Edit Trunk(Test) [BACK] [SAVE]

Tabs: BASIC | CODEC | ADVANCE | DOD | ADAPT CALLER ID

TrunkType * SIP

Trunk Name *

Transport Record

Register * Enabled *

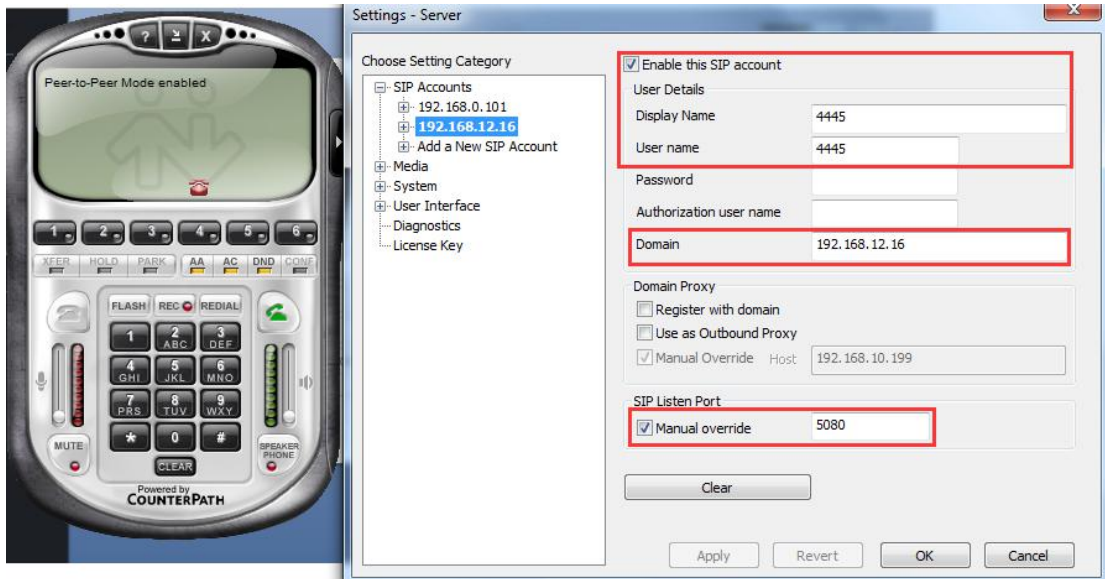
Keep Inbound CallerID Profile *

Outbound CallerID Name Trunk IP/Domain *

Outbound CallerID Number

Select LAN or WAN based on real connections

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.

Edit Extension(1002)
BACK
SAVE

BASIC
FEATURES
ADVANCED

General

Type * SIP

Extension * Enabled *

Password * Max Registrations

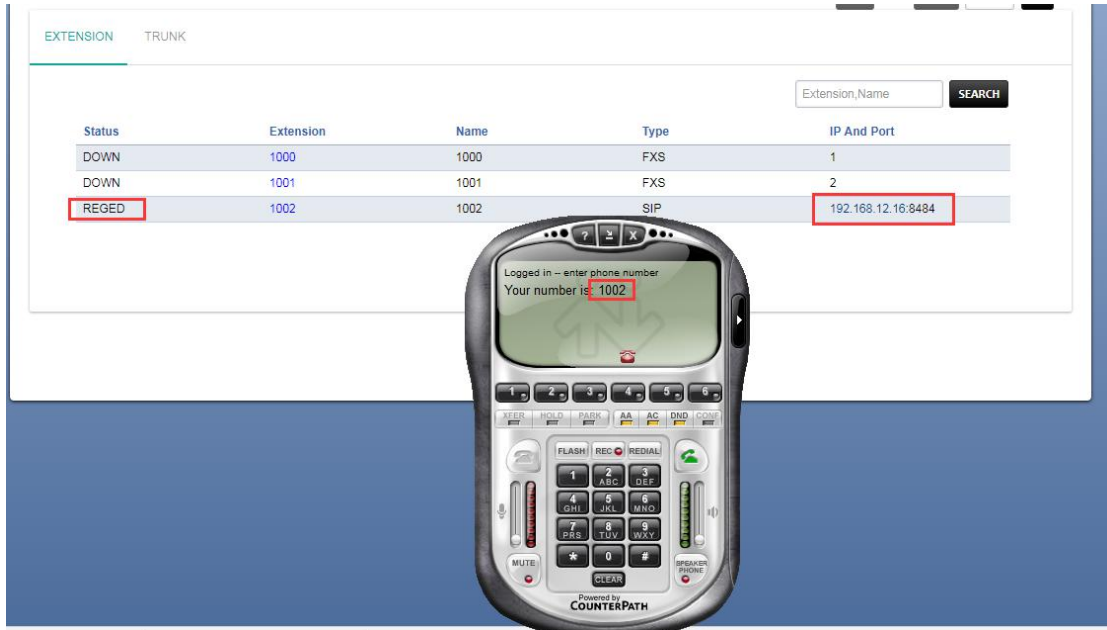
Effective Caller ID Number

UserInfo

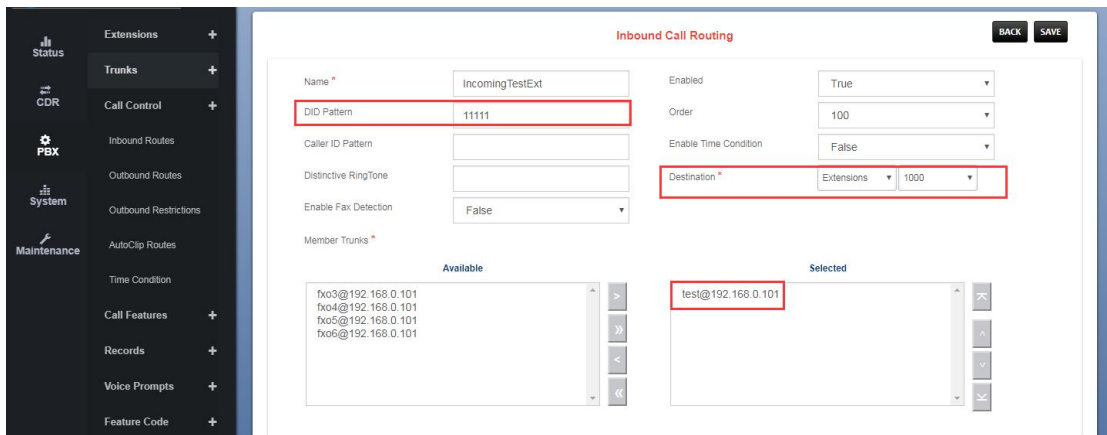
Name * User Password *

Vicemail Mail To Mobile Number

Prompt Language



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

The screenshot shows a web-based configuration interface for a PBX system. On the left is a dark sidebar with a menu of options: Status, Extensions, Trunks, CDR, Call Control, Call Features, Record Settings, System, Voice Prompts, Maintenance, Auto Provision, Feature Code, Phone Book, Preference, SIP Settings, and Voicemail. The 'Preference' option is highlighted with a red box. The main content area is titled 'Preference' and has two tabs: 'GENERAL' (selected) and 'API'. Under the 'GENERAL' tab, there are two sections: 'General' and 'Extension Preferences'. The 'General' section has two dropdown menus: 'Max Duration (s)' set to 6000 and 'Attended Transfer Caller ID' set to 'Transferor'. The 'Extension Preferences' section contains several rows of input fields for extension ranges. The 'IVR Extensions' row is highlighted with a red box and shows a range from 6500 to 6599. Other rows include User Extensions (1000-5899), Ring Group Extensions (6200-6299), Paging Group Extensions (6300-6399), Conference Extensions (6400-6499), and Queue Extensions (6700-6799).

Category	Start	End
User Extensions	1000	5899
Ring Group Extensions	6200	6299
Paging Group Extensions	6300	6399
Conference Extensions	6400	6499
IVR Extensions	6500	6599
Queue Extensions	6700	6799

5.2.2. Create IVR profile

IVR [BACK] [SAVE]

BASIC KEY PRESS EVENT

Name * IncomingTestIVR

IVR Number * 6500

Inter-Digit Timeout (ms) * 3000

Max Failures * 3

Max Timeouts * 3

Digit Length * 4

Enabled * True

Direct Extension * False

FXO Flash Transfer * False

Greet Long * Default

Prompt

Greet Short

Prompt

Response Timeout (ms) * 10000

Direct Outbound

Advanced

Invalid Sound Default

Exit Sound Default

Exit Action

Ring Back default

Caller ID Name Prefix

Inbound Call Routing [BACK] [SAVE]

Name * IncomingTestIVR

Enabled True

Order 100

Enable Time Condition False

Destination * IVR Menu 6500 IVR

DID Pattern 22222

Caller ID Pattern

Distinctive Ring Tone

Enable Fax Detection False

Member Trunks *

Available

fxo3@192.168.0.101

fxo4@192.168.0.101

fxo5@192.168.0.101

fxo6@192.168.0.101

Selected

test@192.168.0.101

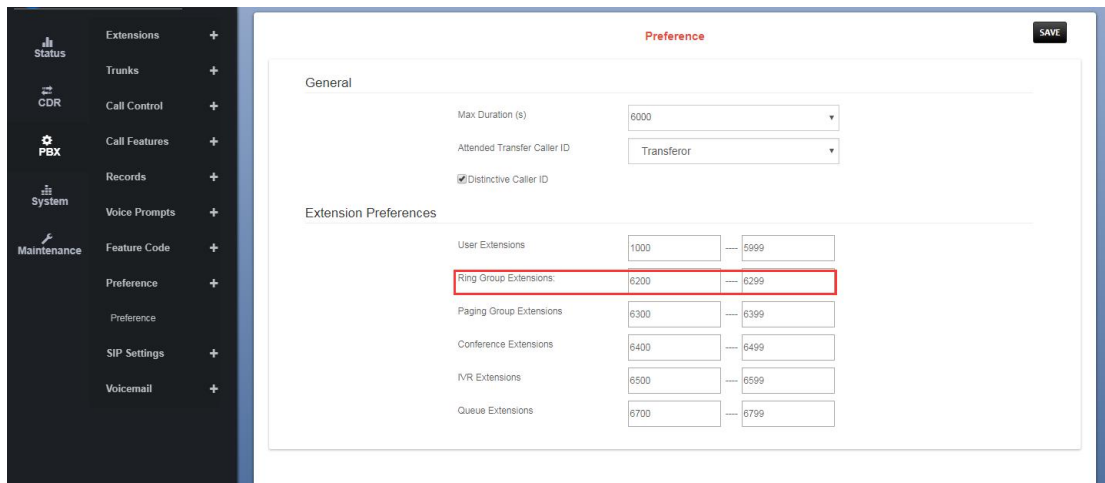
5.2.3. Create incoming route for IVR

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

Ring Group [BACK] [SAVE]

Name * RingGroup

Enabled * True

Ring Group Number * 6200

Alert Info

Ring Strategy * Simultaneous

Ring Back cn-ring

Timeout Destination

CID Name Prefix

Ring Timeout (s) 30

Extension Answer Confirm No

Member Extensions *

Available

Selected

1000-1000
1001-1001
1002-1002
extension-trunk-1003-1016

select ringback for different region

select extensions in ringgroup

Inbound Call Routing [BACK] [SAVE]

Name * IncomingTestRingGroup

Enabled True

DID Pattern 33333

Order 100

Caller ID Pattern

Distinctive RingTone

Enable Time Condition False

Destination * Ring Groups 6200 RingGro

Enable Fax Detection False

Member Trunks *

Available

Selected

test@192.168.0.101

fxo3@192.168.0.101
fxo4@192.168.0.101
fxo5@192.168.0.101
fxo6@192.168.0.101

Call Test:



5.4. Incoming to Conference with fixed DID 44444

5.4.1. Check for Conference number range

The screenshot shows a configuration page for a PBX system. The left sidebar contains navigation options: Status, CDR, PBX, System, and Maintenance. The main content area is titled 'Preference' and has a 'SAVE' button in the top right corner. Under the 'General' section, there are settings for 'Max Duration (s)' (6000), 'Attended Transfer Caller ID' (Transferor), and a checked 'Distinctive Caller ID' option. The 'Extension Preferences' section contains several rows of input fields for different extension ranges:

Extension Type	Start	End
User Extensions	1000	5999
Ring Group Extensions	6200	6299
Paging Group Extensions	6300	6399
Conference Extensions	6400	6499
IVR Extensions	6500	6599
Queue Extensions	6700	6799

5.4.2. Create conference profile

Conference Rooms [BACK] [SAVE]

Room Name *

Conference Center Number *

Greeting

No Pin

Record

Max Members

Moderator Member *

Schedule

Wait for Moderator

Name the person reporting the meeting

Mute Participant

Allow Participant to Invite

Enabled

Announce

Available

Selected

- 1000-1000
- 1002-1002
- 1001-1001

5.4.3. Create incoming route for conference

Inbound Call Routing [BACK] [SAVE]

Name *

DID Pattern

Caller ID Pattern

Distinctive Ring Tone

Enable Fax Detection

Member Trunks *

Enabled

Order

Enable Time Condition

Destination *

Available

Selected

- fxo3@192.168.0.101
- fxo4@192.168.0.101
- fxo5@192.168.0.101
- fxo6@192.168.0.101

- test@192.168.0.101

Call Test:



5.5. Call Center Queue with fixed DID 55555

5.5.1. Check for call center queue number range in preference

The screenshot shows a web interface for configuring a PBX system. The left sidebar contains a navigation menu with the following items: Status, Extensions, Trunks, CDR, Call Control, PEK, Call Features, Records, System, Voice Prompts, Maintenance, Feature Code, Preference, SIP Settings, and Voicemail. The main content area is titled 'Preference' and has a 'SAVE' button in the top right corner. It is divided into two sections: 'General' and 'Extension Preferences'. The 'General' section includes 'Max Duration (s)' set to 6000 and 'Attended Transfer Caller ID' set to 'Transferor'. The 'Extension Preferences' section lists various extension ranges: User Extensions (1000-5999), Ring Group Extensions (6200-6299), Paging Group Extensions (6300-6399), Conference Extensions (6400-6499), IVR Extensions (6500-6599), and Queue Extensions (6700-6799). The 'Queue Extensions' field is highlighted with a red box.

Section	Field	Value
General	Max Duration (s)	6000
	Attended Transfer Caller ID	Transferor
Extension Preferences	User Extensions	1000 - 5999
	Ring Group Extensions	6200 - 6299
	Paging Group Extensions	6300 - 6399
	Conference Extensions	6400 - 6499
	IVR Extensions	6500 - 6599
	Queue Extensions	6700 - 6799

5.5.2. Create call queue

Edit Call Center Queue [BACK] [RESTART] [SAVE]

BASIC CALLER EXPERIENCE SETTINGS

Queue Name * Queue

Queue Number * 6700

Agent Password *

Ring Strategy * Longest Idle Agent

Timeout Action

Record False

Caller ID Name Prefix

Agents

Available: 1001-1001, 1000-1000

Selected: 1002-1002, 1003-1003

Agent Call Timeout (s) 15

Agent Answer Announce Prompt

Agent Retry Time (s) 30

Wrap Up Time (s) 30

Max Wait Time (s) 0

Max Queue Length 20

Alert info

https://192.168.12.12/app/call_centers/call_center_queue_edit.php?id=c4b6a4c2-0feb-4...

5.5.3. Create incoming route for Call Queue

Inbound Call Routing [BACK] [SAVE]

Name * IncomingTestQueue

DID Pattern 55555

Caller ID Pattern

Distinctive RingTone

Enable Fax Detection False

Member Trunks *

Available: fxo3@192.168.0.101, fxo4@192.168.0.101, fxo5@192.168.0.101, fxo6@192.168.0.101

Selected: test@192.168.0.101

Enabled True

Order 100

Enable Time Condition False

Destination * Call Center 6700

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

The screenshot shows a web interface for editing a call center queue. The page title is "Edit Call Center Queue". On the left is a navigation menu with options like "Status", "Extensions", "Trunks", "CDR", "Call Control", "Call Features", "IVR", "Conference Room", "Call Center", "Intercept Groups", "Ring Groups", "BlackList", "PIN Numbers", "Speed Dial", "Call Broadcasts", and "DISA". The main content area is titled "BASIC CALLER EXPERIENCE SETTINGS". It contains several form fields: "Queue Name" (set to "Queue"), "Queue Number" (set to "6700", highlighted with a red box), "Agent Password", "Ring Strategy" (set to "Longest Idle Agent"), "Timeout Action", "Record" (set to "False"), and "Caller ID Name Prefix". On the right, there are fields for "Agent Call Timeout (s)" (15), "Agent Answer Announce" (set to "Prompt"), "Agent Retry Time (s)" (30), "Wrap Up Time (s)" (30), "Max Wait Time (s)" (0), "Max Queue Length" (20), and "Alert info". At the bottom, there are two lists: "Available" (containing "1001-1001" and "1000-1000") and "Selected" (containing "1002-1002" and "1003-1003", highlighted with a red box). Buttons for "BACK", "RESTART", and "SAVE" are at the top right. A URL is visible at the bottom left: "https://192.168.12.12/app/call_centers/call_center_queue_edit.php?id=c4b6e42-0feb-4-...".

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

The screenshot shows the 'CallBack' configuration page. The left sidebar contains navigation options: Status, Trunks, Call Control, Call Features, IVR, Conference Room, Call Center, Intercept Groups, Ring Groups, BlockList, PIN Numbers, Speed Dial, Call Broadcasts, DISA, and Call Back. The main form has the following fields:

- Name: CallBack
- Delay (s): 10
- Strip: (empty)
- Prepend: (empty)
- Destination: 6700 (dropdown menu is open, showing 'Call Center' selected)
- Through: (empty)

Buttons: BACK, SAVE

5.6.3. Create incoming route for call back

The screenshot shows the 'Inbound Call Routing' configuration page. The left sidebar contains navigation options: Status, Trunks, Call Control, Inbound Routes, Outbound Routes, Outbound Restrictions, AutoClip Routes, Time Condition, Call Features, Records, Voice Prompts, Feature Code, Preference, and SIP Settings. The main form has the following fields:

- Name: IncomingCallBackWithCallQueue
- Enabled: True
- DID Pattern: 6666
- Order: 100
- Enable Time Condition: False
- Destination: CallBack
- Member Trunks: (empty)

Available Member Trunks:

- fxo3@192.168.0.101
- fxo4@192.168.0.101
- fxo5@192.168.0.101
- fxo6@192.168.0.101

Selected Member Trunk:

- test@192.168.0.101

Buttons: BACK, SAVE

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.