

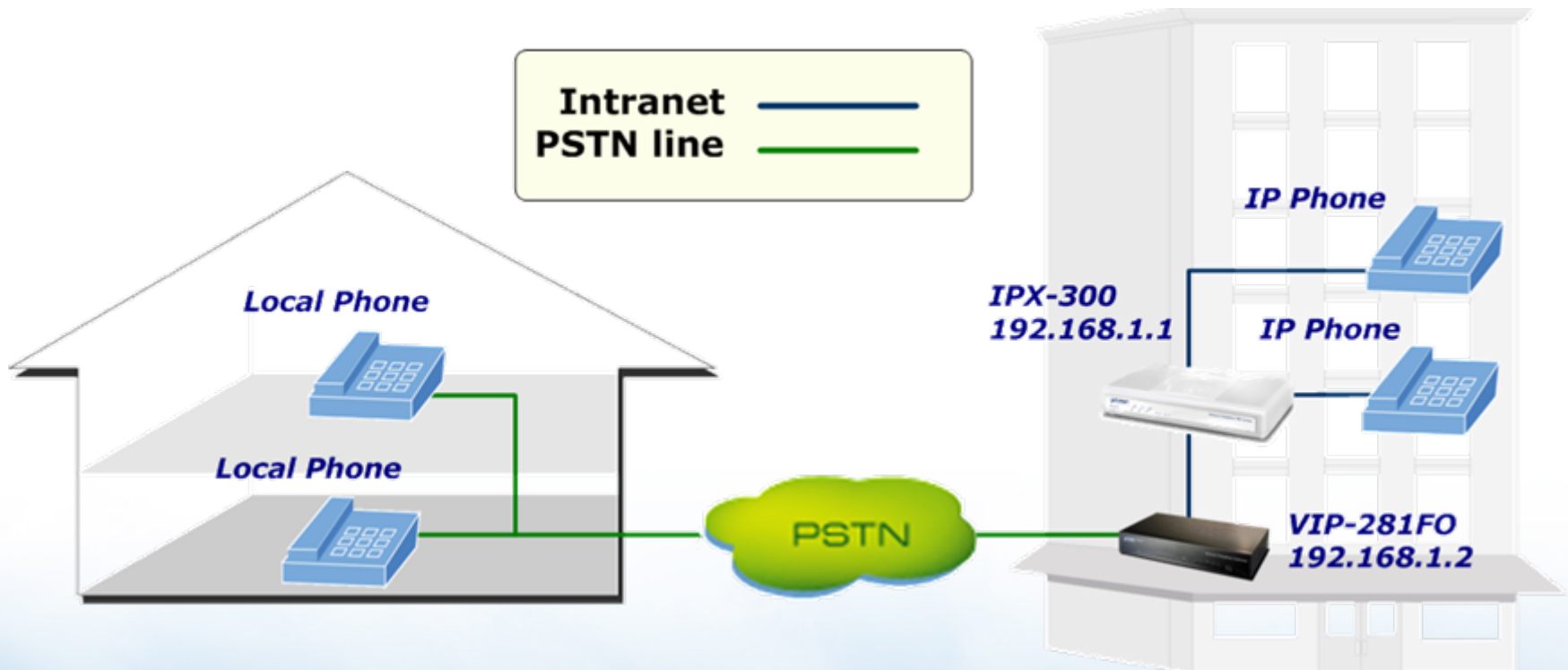
Internet Telephony PBX System

**IPX-300 Series
PBX with FXO GW Configuration**



Case 4: IP PBX-Trunk-Gateway one-stage to dial Configuration

We're using the IPX-300 and VIP-281FO to perform the calling party. Please refer to the following descriptions and insert proper parameters configurations into the each device for establish the voice communications.



Select the [UIP PBX Setup->Gateway Trunk] of IPX-300 to create the Gateway Trunk for the FXO gateway.

- >> **IP PBX Setup**
- SIP Basic Setting
- SIP Extension Setup
- Trunk Management
 - SIP Trunk
 - Gateway Trunk**
 - Trunk Group
 - Dialing Rules

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IP : 192.168.0.48 (Gateway IP Address)
Port : 5060 (Gateway SIP Port)
Afterward click the Insert to add that

IP PBX Setup

• Gateway Trunk Setting

Add Gateway trunk **Gateway trunk Max is 10**

IP	Port	Action
<input type="text"/>	<input type="text"/>	<input type="button" value="Insert"/> <input type="button" value="Change"/>
192.168.0.48	5060	<input type="button" value="Edit"/> <input type="button" value="Delete"/>

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>> IP PBX Setup

- SIP Basic Setting
- SIP Extension Setup
- Trunk Management
 - SIP Trunk
 - Gateway Trunk
 - Trunk Group**
 - Dialing Rules

Select the [IPX PBX Setup->Trunk Group] of IPX-300 to add the Trunk Group

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• Trunk Group

Group Name:

Number:

Trunk Group:

All Trunks:

<<< >>>

Submit

Click the **Add** button add a new Trunk Group
 Group Name : **PSTN**
 Group Number : **99**
 Trunk Group : **192.168.0.48:5060**
 Click the **Submit** button after confirmed whole configuration

Group Name List **Trunk Group Max is 10**

Group Name	Group Number	Number	Action
PSTN	99	192.168.0.48:5060	Edit Delete

- >> **IP PBX Setup**
- SIP Basic Setting
- SIP Extension Setup
- Trunk Management
- Attendant Management
 - Attendant Number**
 - Attendant Message
 - Attendant Time

Select the [Attendant Number] of **IPX-300** to assign a AA extension number for Gateway hotline setting.

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IP PBX Setup

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- **Attendant Number**

Attendant Number 1	<input type="text" value="300"/>
Attendant Number 2	<input type="text"/>

Assign a AA extension number :
Attendant Number 1 : **300**

- >> Wizard
- >> IP PBX Setup
- >> Infomation
- >> Network Setup
- >> Management
- >> **Save & Logout**

Select the [**Save & Logout**] of **IPX-300** to create a SIP trunk which connect with Gateway

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Save & Logout

- Save configuration

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Save

- Save configuration & Logout

Save & Logout

- Save configuration & Reboot

Save & Reboot

Click on the **Save & Reboot** button to save previous modification and take effect, the system will rebooting, please wait for 50 sec

Save & Logout

System is rebooting now ,please wait for 50 sec .

- VoIP Setup
- VoIP Basic**
- Dialing Plan
- Advance Setting
- Hot Line Setting
- Port Status

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Select the [Advance Setup ->VoIP Basic] of gateway to setting port number configuration.

VoIP Basic Configuration:
 Number : 555
 Check Reg
 Account: 555
 Password: 555
 Use Public Account: Enable
 Check Hunting Member
 Port 1 and Port 2

This function is when the port 1 be communicating, and it another SIP calls from network will forward this call to port 2 automatically.

VoIP Basic Configuration

VoIP Protocol Setting SIP

Port Number / Password Setting(MAX 20 digit) :

No.	Number	Reg	Account	Password	Register Status	Reason
1	555	<input checked="" type="checkbox"/>	555	...	Success	OK
2		<input type="checkbox"/>				

Use Public Account (PORT 1) Enable Disable

SIP Hunting Table :

No.	Hunting Member
1	<input checked="" type="checkbox"/> Port 1 <input checked="" type="checkbox"/> Port 2
2	<input checked="" type="checkbox"/> Port 1 <input checked="" type="checkbox"/> Port 2

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SIP Proxy Setting :	
Domain/Realm	192.168.1.1
SIP Proxy Server	192.168.1.1/5060
Register Interval (seconds)	100
SIP Authentication	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Outbound Proxy Server	192.168.1.1/5060

SIP Proxy Setting:
Domain/Realm:
192.168.1.1
SIP Proxy Server:
192.168.1.1/5060
SIP Authentication :
Enable
Outbound Proxy Server:
192.168.1.1/5060

About the IP address
please refer to previous
topology.

NAT Pass Setting:

NAT Pass Method	<input type="radio"/> STUN <input checked="" type="radio"/> Symmetric RTP
STUN Server IP Address	64.69.76.21
STUN Server port	3478

Local Setting:

Local SIP Port	5060
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Apply

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Click the **Apply** button to complete the VoIP Basic configuration.

VoIP Setup

- VoIP Basic
- Dialing Plan**
- Advance Setting
- Hot Line Setting
- Port Status

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Select the [Dialing Plan] of gateway to add the Dialing Plan Number.

Incoming Dial Plan Settings:
 Outgoing no: **1X / 2X**
 Length of Number: **3 ~ 3 / 8 ~ 8**
 Delete Length: **0**
 Prefix no.: **Null**
 Destination Telephone Port : **1,2,3 / 4**

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Incoming Dial Plan: (maximum 50 entries, maximum length of prefix digits is 16 digit, maximum length of number is 20 digit):

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Operation
1	1x	3 ~ 3	0	None	1,2,3	
2	2x	8 ~ 8	0	None	4	
		<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	ADD
DELETE Inbound Dial Plan		From <input type="text"/> To <input type="text"/>				

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- VoIP Setup
- VoIP Basic
- Dialing Plan
- Advance Setting
- Hot Line Setting**
- Port Status

Select the [**Hot Line Setting**] of gateway to add the Hot Line Number.

Hot Line Number Setting (Hotline Setting)

Hotline Delay	<input checked="" type="radio"/> Disable <input type="radio"/> Enable
Hotline Delay Time(Max. 20 sec)	<input type="text" value="3"/> sec
Port 1 number	<input type="text" value="300"/>
Port 2 number	<input type="text" value="300"/>

Apply

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Hot Line Number Setting :
Port 1 Number : **300**
Port 2 Number : **300**

[Main Menu](#) [Reboot](#) [Save Configuration](#) [Logout](#)

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Click the **Save Configuration** hyper-link to save the whole parameters and then click the **Reboot** to restart system.

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Click on the **Apply** button to complete the modification.

For example:

1.From IP Phone make a call to FXO (One-stage dial method)

Step 1. Pickup the handset and hearing a dial-tone, then to input a number **9922199518#**.

Step 2. You will be hear to the ring-back tone, waiting for remote to pickup the call.

2.From IP Phone make a call to FXO. (Two-stage dial method)

Step 1. Pickup the handset and hearing a dial-tone, then input a number **99#**.

Step 2. You will be hear a ring-back tone afterward, and then get a dial-tone form PSTN line.

Step 3. To input a local phone number, such as **22999158**

2.From local phone (PSTN) make a call into IPPBX through FXO.

Step 1. Pickup the handset of local phone and hearing a dial-tone, input number **858075**

Step 2. You will be hear a ring-back tone afterward, and then entered the IVR of IPPBX

Step 3. To input a extension number of SIP client, then the SIP client should be ring up.

NOTE:

The mentioned parameters and network settings in this sample be carefully modified to meet the real-world applications.

ACTIVATING IP POWER

