

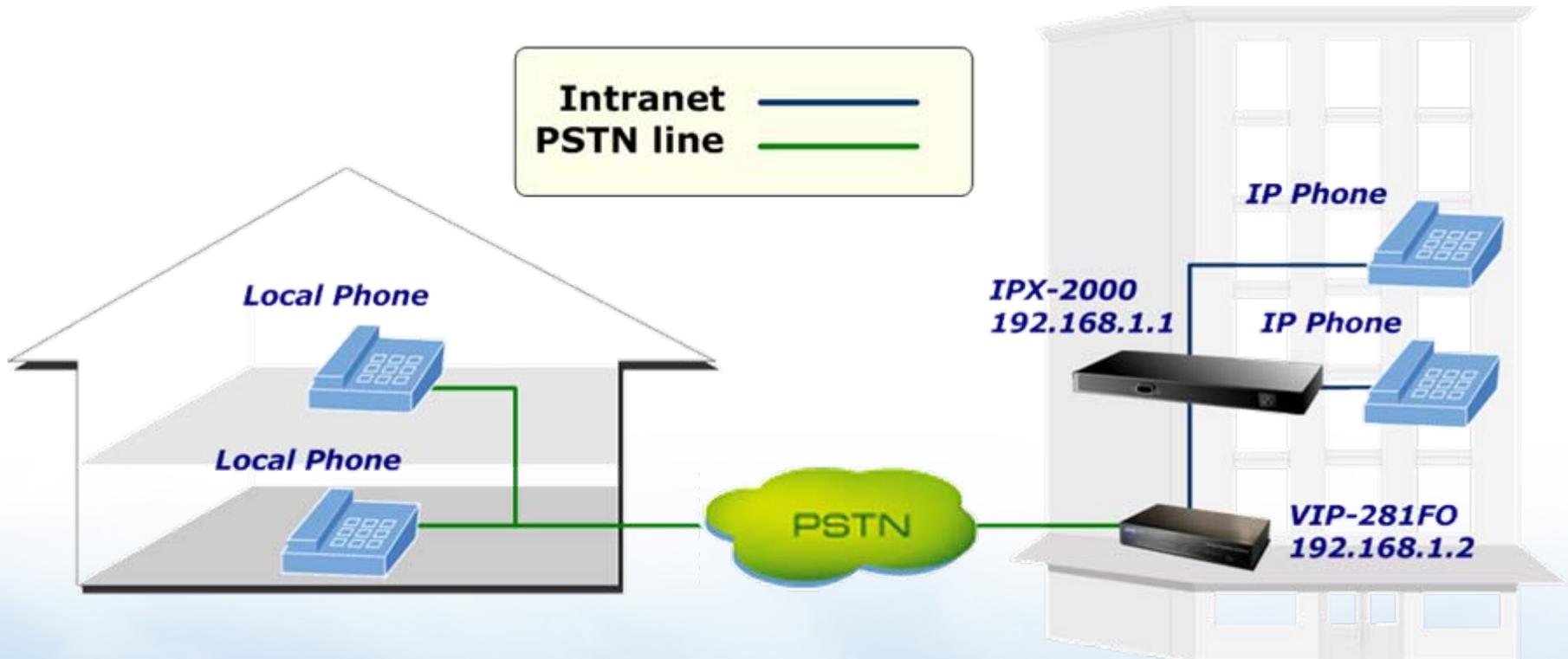
Internet Telephony PBX System

**IPX-2000/1800 Series
PBX with FXO GW Configuration II**

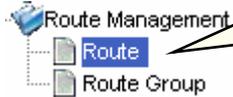


Case 4: IP PBX-Trunk-Gateway one-stage to dial Configuration

We're using the IPX-2000 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.



1



Select the [Route->Route] of IP PBX to create the Routes for the gateway.

:: ROUTE MANAGEMENT

Route ID	Description	Destination Number Pattern	Number of Stripped Digits	Prefix
<input type="text"/>	<input type="text"/>	<input type="text"/>	0 <input type="button" value="v"/>	<input type="text"/>
<input type="button" value="ADD"/>				
<input type="button" value="DEL"/>				
Route ID	Description	Destination Number Pattern	Number of Stripped Digits	Prefix
<input type="checkbox"/> IPX_TO_GW	<input type="text"/>	9.	0 <input type="button" value="v"/>	<input type="text"/>
<input type="checkbox"/> IPX_TO_GW_A	<input type="text"/>	8	1 <input type="button" value="v"/>	555

Create a Route ID : **PBX_TO_GW**
 Destination number pattern : **9.**
 Number of stripped digits: **0**
 Create a Route ID : **PBX_TO_GW_A**
 Destination number pattern : **8**
 Number of stripped digits: **1**
 Prefix: **555**



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Select the [Route->Routegroup] of IP PBX to add the previous Routes to a Routegroup

:: ROUTEGROUP MANAGEMENT

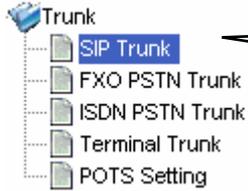
ROUTE GROUP ADD

Group ID	RG_PBX_TO_GW					
Description	<input type="text"/> SET					
Associated Routes	<table border="1"><tr><td>IPX_TO_GW</td><td rowspan="2">ADD DEL</td><td>IPX_TO_GW</td></tr><tr><td>IPX_TO_GW_A</td><td>IPX_TO_GW_A</td></tr></table>	IPX_TO_GW	ADD DEL	IPX_TO_GW	IPX_TO_GW_A	IPX_TO_GW_A
IPX_TO_GW	ADD DEL	IPX_TO_GW				
IPX_TO_GW_A		IPX_TO_GW_A				
BACK						

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Create a new Routegroup **RG_PBX_TO_GW**
Then to press the **ADD** button to get the both **IPX_TO_GW** and **IPX_TO_GW_A** added to this Routegroup.

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Select the [Trunk->SIP trunk] of IP PBX to create a SIP trunk which connect with Gateway

:: SIP TRUNK MANAGEMENT

Trunks Add New

Trunk Identifier	555
Description	
<input checked="" type="checkbox"/> Dynamic Peer	
Auth. Name	555
Auth. Password	•••
<input checked="" type="checkbox"/> Registration Required	
Outbound Routegroup	RG_PBX_TO_GW
<input type="checkbox"/> DID by Privilege	
DID of Extension	
DID Prefix	
DID Stripping	
Language	English
IVR List	
Usergroup of Privilege	UG_DEF
<input type="checkbox"/> Advanced Settings	

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Trunk identifier: 555 (this number is for Gateway to hot line to auto attendant)
 Check Dynamic peer
 Auth. Name: 555
 Auth. password: 555
 Outbound routegroup: RG_PBX_TO_GW

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Select the [User->Usergroup] of IP PBX to associate the SIP trunk 555 to corresponding usergroup

:: USER GROUP MANAGEMENT

Group ID	UG_DEF								
Description	<input type="text"/> <input type="button" value="SET"/>								
Associated Trunks	<table border="1"> <tr><td>555</td></tr> <tr><td>888</td></tr> <tr><td>pstnl</td></tr> </table>	555	888	pstnl	<table border="1"> <tr> <th>Group ID</th> <th>Weight</th> </tr> <tr> <td>0</td> <td><input type="text"/></td> </tr> </table>	Group ID	Weight	0	<input type="text"/>
	555								
888									
pstnl									
Group ID	Weight								
0	<input type="text"/>								
	<input type="button" value="ADD"/>	<table border="1"> <tr><td>888,0,0</td></tr> <tr><td>555,0,0</td></tr> <tr><td>pstnl,0,0</td></tr> </table>	888,0,0	555,0,0	pstnl,0,0				
888,0,0									
555,0,0									
pstnl,0,0									
	<input type="button" value="DEL"/>								
Reachable User Groups	<table border="1"> <tr><td>UG_DEF</td></tr> </table> <input type="button" value="ADD"/> <input type="button" value="DEL"/>	UG_DEF	<table border="1"> <tr><td>UG_DEF</td></tr> </table>	UG_DEF					
UG_DEF									
UG_DEF									
Associated PBX Features	<table border="1"> <tr><td>mm</td></tr> <tr><td>parkedcalls</td></tr> <tr><td>vm</td></tr> </table> <input type="button" value="ADD"/> <input type="button" value="DEL"/>	mm	parkedcalls	vm	<table border="1"> <tr><td>mm</td></tr> <tr><td>parkedcalls</td></tr> <tr><td>vm</td></tr> </table>	mm	parkedcalls	vm	
mm									
parkedcalls									
vm									
mm									
parkedcalls									
vm									
Member List	<table border="1"> <tr><td>User:admin</td></tr> <tr><td>Callers_from_SIP_Trunk:888</td></tr> <tr><td>Callers_from_SIP_Trunk:555</td></tr> <tr><td>Callers_from_PSTN_Trunk:pstnl</td></tr> </table>		User:admin	Callers_from_SIP_Trunk:888	Callers_from_SIP_Trunk:555	Callers_from_PSTN_Trunk:pstnl			
User:admin									
Callers_from_SIP_Trunk:888									
Callers_from_SIP_Trunk:555									
Callers_from_PSTN_Trunk:pstnl									
<input type="button" value="BACK"/>									

8

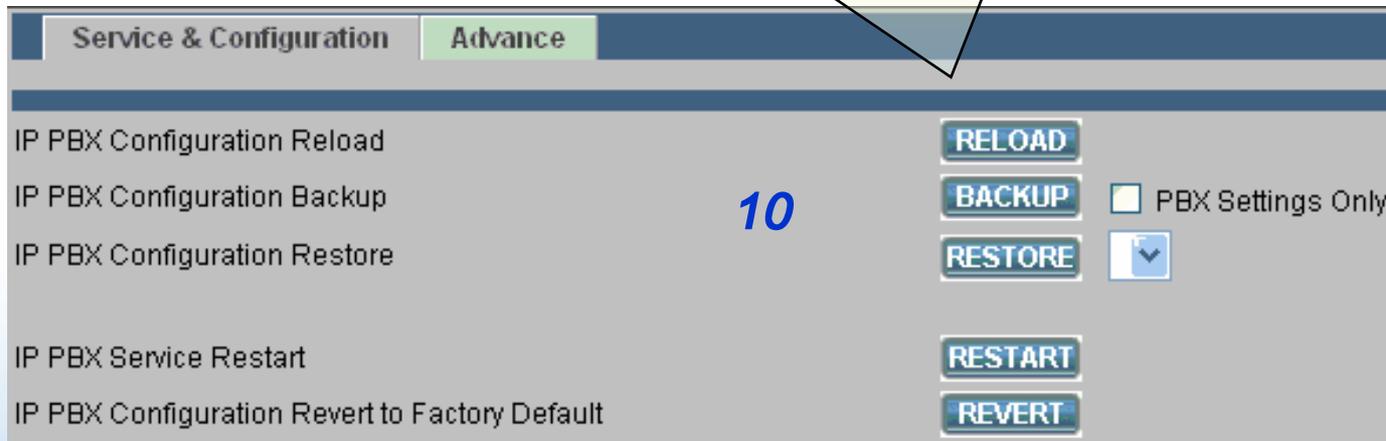
Associate the SIP trunk 555 to the usergroup UG_DEF



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Select the [Service->IP PBX service] of IP PBX to reload the IP PBX configuration.

Click on the **RELOAD** button to reload just modification to take effect, and after complete, please connect to gateway to finish rest of procedures.



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- 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.

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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

SIP Proxy Setting :

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

NAT Pass Setting:

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

Local Setting:

Local SIP Port	<input type="text" value="5060"/>
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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.

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Click the 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.

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Outgoing Dial Plan Settings:

Outgoing no: **0555**

Length of Number: **4~4**

Delete Length: **1**

Destination IP/DNS: **192.168.1.1**

Destination Port : **5060**

This Dialing Plan is for gateway to hot line **0555** then striped **0** dialed **555** call be transferred to the IVR of IP PBX.

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

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Destination Port	Operation
1	0555	4 ~ 4	1	None	192.168.1.1	5060	
	<input type="text"/>	<input type="text"/> ~ <input type="text"/>	ADD				

Outbound Dial Plan
 From To

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

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Operation
1	9	1 ~ 10	1	None	1,2	
	<input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text" value="17"/>	<input type="text"/>	ADD

DELETE Inbound Dial Plan From To

Incoming Dial Plan Settings:

Incoming no: **9**

Length of Number: **1~10**

(Please according to your environment to modified this value)

Delete Length: **1**

Destination telephone port : **1,2**

This Dialing Plan is when has an incoming call form client of IPPBX and it number is 922199518 into the gateway, that would be drop the prefix **9** automatically, then forward the number 22199518 to the **port 1 or 2**.

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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="0555"/>
Port 2 number	<input type="text" value="0555"/>

Apply

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Hot Line Number Setting :
Port 1 Number : **0555**
Port 2 Number : **0555**

[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 IPPBX make a call to FXO (One-stage dial method)

Step 1. Pickup the handset and hearing a dial-tone, then to input a number **922199518#**.

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

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

Step 1. Pickup the handset and hearing a dial-tone, then input a number **8#**.

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

