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# 1. Overview

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This document presents a complete guide for integrating VoiceGear Skype gateways with Pbxnsip phone systems running version 3 of Pbxnsip software. The document assumes reader familiarity with both VoiceGear gateways and Pbxnsip PBX. For more details on VoiceGear gateway configuration, please refer to the VoiceGear SkyBridge or VoiceGear Connect user guides available at [www.industrydynamics.ca](http://www.industrydynamics.ca).

## 1.1 Setting up Pbxnsip PBX

To get started with your Pbxnsip PBX please follow steps outlined below.

1. Make sure the system running Pbxnsip software has an assigned static IP or a DHCP assigned IP that does not expire
2. Log into the Pbxnsip management interface by starting your default browser and navigating to <http://0.0.0.0> where 0.0.0.0 is the IP or a fully qualified domain name of the system Pbxnsip has been installed on
3. Follow instructions outlined in section 2.1 to configure Pbxnsip to communicate with your VoiceGear gateway

## 1.2 Setting up VoiceGear Gateway

To get started with VoiceGear, please follow steps outlined below.

1. Connect the supplied power brick/cord and network cable to your new VoiceGear gateway
2. Connect monitor and keyboard to the gateway and use username: root, password: vgcroot123 credentials to access the administration console. Using the console, configure networking settings and set a static IP for the gateway
3. Open the VoiceGear web configuration interface in a web browser using the static IP you have assigned and TCP port 8080. For example, if the static IP assigned is 192.168.2.194, point your browser to <http://192.168.2.194:8080>
4. Once the web configuration interface has been loaded, please login with the following credentials: username: admin, password: admin
5. Once logged in, register at least one valid Skype account with the gateway via the “Skype-> Accounts->Add New” page
6. Please refer to section 2.2 for more details on setting up a SIP trunk to enable VoiceGear gateway to communicate with Pbxnsip PBX

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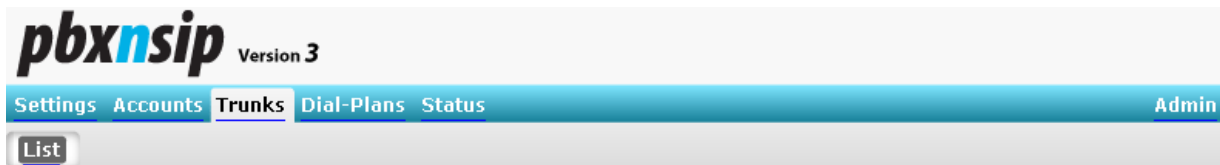
## 2. VoiceGear-Pbxnsip SIP Integration

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The main purpose of this section is to outline both VoiceGear gateway and Pbxnsip PBX configuration to enable integration of both systems via SIP protocol.



## 2.1 Pbxnsip Configuration

1. Navigate to the Pbxnsip “Trunks” screen by selecting the appropriate domain to manage and clicking on “Trunks” from the top menu



### Current Trunks

This list shows the currently available trunks in this domain.  
Please be careful clicking the delete button, because the trunk data will be lost and dial plan references might break.

Name	Status	3 Edit	Delete	Action
vgc (2)	200 OK (No refresh interval)			<a href="#">REGISTER</a>

**New Trunk:**

Name:  **2**

Type:  **2**

**2**

2. Create a new trunk for the VoiceGear gateway by entering “vg” as name and “SIP Registration” as type and clicking “Create” in the “New Trunk” box at the bottom of the screen
3. Once the “vg” trunk has been created, click on the “Edit” icon to the right of its name in the list of “Current Trunks”
4. In the “Edit Trunk” screen that comes up, make sure “Direction” is set to “Inbound and outbound”
5. Enter “pbx” for “Display Name”, “Account”, “Username”, “Password”, and “Password (repeat)” fields. Please note that value “pbx” has been arbitrarily picked for this guide and can be set differently in a production environment
6. Enter the IP address of the VoiceGear gateway as defined in step 2 of section 1.2 for both “Domain” and “Outbound Proxy” fields
7. Enter the number of concurrent calls supported by the VoiceGear gateway as the value for “CO Lines” field
8. Make sure only G.711A and G.711U codecs are set in the “Override Codec Preference” field
9. Make sure all other fields are left with default values and click on “Save” to save your changes.

List Edit

## Edit Trunk vg

Name:	<input type="text" value="vg"/>	
Type:	SIP Registration	
Direction:	Inbound and outbound	4
<hr/>		
Display Name:	<input type="text" value="pbx"/>	5
Account:	<input type="text" value="pbx"/>	5
Domain:	<input type="text" value="192.168.2.188"/>	6
Username:	<input type="text" value="pbx"/>	5
Password:	<input type="password" value="....."/>	5
Password (repeat):	<input type="password" value="....."/>	5
Outbound Proxy:	<input type="text" value="192.168.2.188"/>	6
CO Lines:	<input type="text" value="2"/>	7
Permissions to monitor this account:	<input type="text"/>	
Override Codec Preference:	<div style="display: flex; justify-content: space-between;"> <div style="width: 45%;"> <p>8</p> <p>G.711U G.711A</p> </div> <div style="width: 45%;"> <p>G.726 GSM 6.10 G.722 G.729A</p> </div> </div> <div style="display: flex; justify-content: center; margin-top: 5px;"> <input type="button" value="Up"/> <input type="button" value="Down"/> <input type="button" value="Remove"/> <input type="button" value="Add"/> </div>	
Proposed Duration (s):	<input type="text" value="3600"/>	
Keepalive Time:	<input type="text"/>	
Send email on status change:	<input type="radio"/> yes <input checked="" type="radio"/> no	
Strict RTP Routing:	<input type="radio"/> yes <input checked="" type="radio"/> no	
Avoid RFC4122 (UUID):	<input type="radio"/> yes <input checked="" type="radio"/> no	


10. Bring up the dial plan settings screen by clicking on "Dial-Plans" from the top menu

11. Click on the "Edit" icon to the right of "Standard Dialplan" in the "Current Dial Plans" list to bring up the "Edit Dial Plan" screen



## Current Dial Plans

This list shows the currently available dial plans on this system. Please be careful clicking the delete button, because the dial plan will be deleted permanently.

Name	Edit	Delete
Standard Dialplan	<b>11</b> 	<b>X</b>

### New Dial Plan:

Name:

- Under the "Trunk" heading, select "vg" from the drop-down list
- Under the "Pattern" heading, enter "6\*" as the value. This will tell the PBX to route all calls prefixed with 6 to the VoiceGear gateway
- Click on "Save" to save your changes

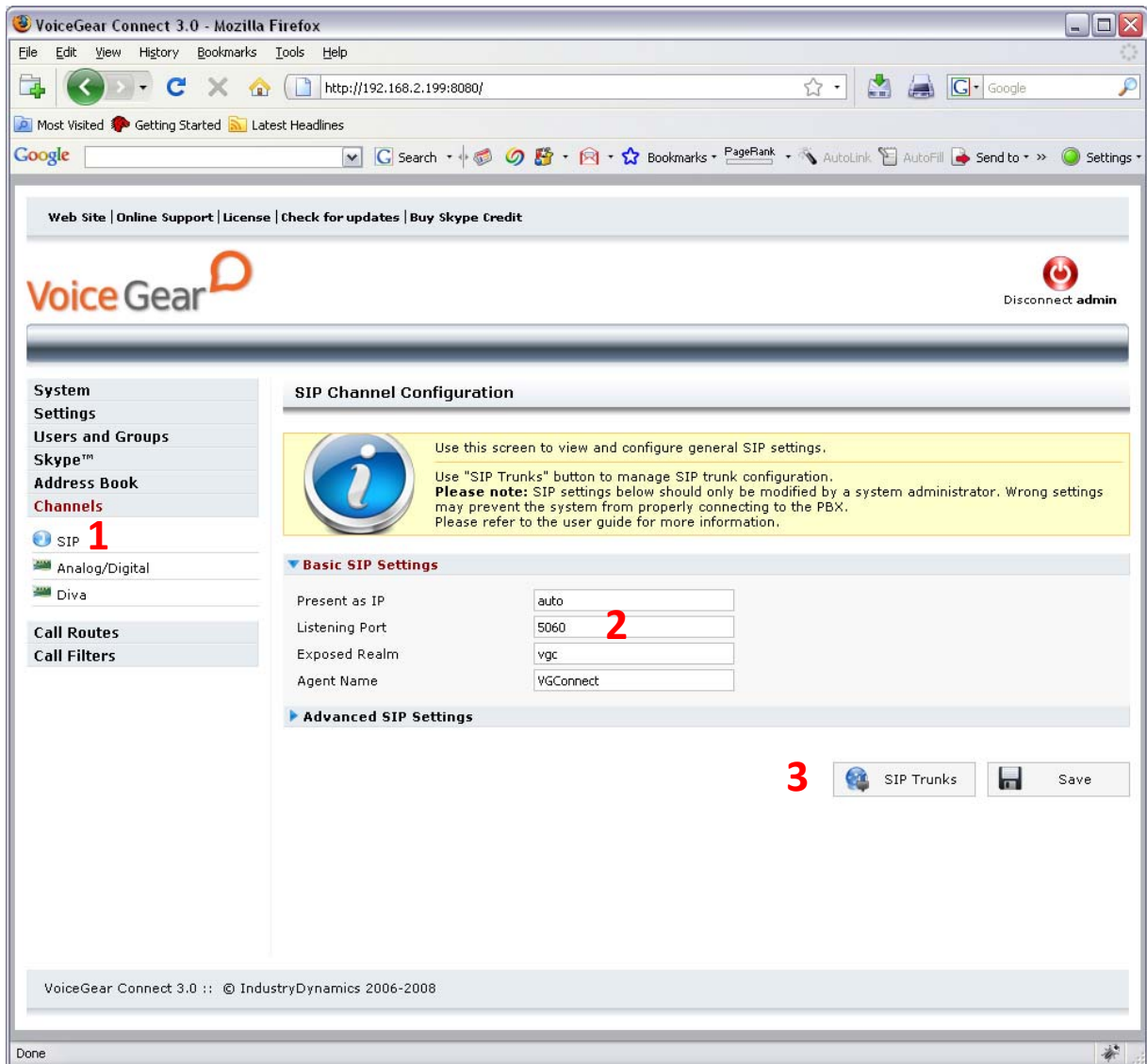
## Edit Dial Plan Standard Dialplan

Quick Usage: Use simple patterns for matching the input (for example, "9\*" or "911"), and just leave the replacement empty. Please see the online help for more information on how to use the advanced features of the dial plan.

Pref	Trunk	Pattern	Replacement
<input type="text" value="100"/>	Unassigned <input type="button" value="v"/>	<input type="text"/>	<input type="text"/>
<input type="text" value="100"/>	<b>12</b> vg <input type="button" value="v"/>	<b>13</b> 6*	<input type="text"/>
<b>14</b> <input type="button" value="Save"/>			

## 2.2 VoiceGear Gateway Configuration

1. Navigate to the “Channels->SIP” screen via the main menu of the VoiceGear web configuration interface



2. Make sure “listening port” is set to 5060
3. Click on the “SIP trunks” button to start the configuration of a SIP trunk that will communicate with Pbxnsip PBX
4. Under the SIP trunks page that opens, click on the “Add new” button to start adding a new trunk

## SIP Channel: pbxnsip



Use this screen to configure SIP trunk settings.

Please consult your PBX administrator or manufacturer for detailed configuration parameters.

**Basic Settings**

Description	<input type="text" value="pbxnsip"/>	5
Authentication Mode	Based on IP (Trusted) <input type="button" value="v"/>	6
Link Mode	PBX <input type="button" value="v"/>	7
Username	<input type="text" value="pbx"/>	8
Password	<input type="password"/>	
Auth Username	<input type="text"/>	
Remote Side IP	<input type="text" value="192.168.2.142"/>	9
Remote Side Port	<input type="text" value="5060"/>	10
DTMF Mode	RFC2833 <input type="button" value="v"/>	
Connection Enabled	Yes <input type="button" value="v"/>	

**Advanced Settings**

<input type="button" value="←"/> Back	<input type="button" value="Clear"/>	<input type="button" value="✖"/> Delete	<input type="button" value="Save"/>
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11

5. Under “Basic settings”, set the Description to “pbxnsip” to help future identification of this SIP trunk
6. Select the “Authentication mode” to be “Based on IP (trusted)”
7. Select the link mode to be “PBX”, this will tell VoiceGear gateway that it is communicating with a PBX as opposed to an individual user
8. Set the “Username” to be the username of a SIP trunk configured in step 5 of section 2.1
9. Set the “Remote Side IP” to the static IP of the Pbxnsip system
10. Set the “Remote Side Port” to 5060
11. Click on the “Save” button to save your changes. Once clicked, an “Apply settings” dialog will come up. Please click on “Apply now” button to apply your settings

Modifications need to be applied, please click on the "Apply Now" button.

Apply Now

SIP Channel: pbxnsip



Use this screen to configure SIP trunk settings.

Please consult your PBX administrator or manufacturer for detailed configuration parameters.



Successfully saved

Basic Settings

Description	<input type="text" value="pbxnsip"/>
Authentication Mode	Based on IP (Trusted) <input type="button" value="v"/>
Link Mode	PBX <input type="button" value="v"/>
Username	<input type="text" value="pbx"/>
Password	<input type="password"/>
Auth Username	<input type="text"/>
Remote Side IP	<input type="text" value="192.168.2.142"/>
Remote Side Port	<input type="text" value="5060"/>
DTMF Mode	RFC2833 <input type="button" value="v"/>
Connection Enabled	Yes <input type="button" value="v"/>

Advanced Settings

Back
 
 Delete

12. Navigate to the "Channels -> Call Routes -> Inbound Routes" screen via the main menu of the VoiceGear web configuration interface and click on "Incoming Calls Any/Any" route
13. In the subsequent screen, remove the existing destination by clicking on the "X" button to the right of it and add a new destination by selecting SIP PBX trunk identified with "pbxnsip". Enter the DID value to be the id or the alias of a desired destination set up on pbxnsip. This value can point not only to auto attendants but also to individual extensions
14. Click "Add" followed by "Save" to apply your changes
15. To try a test call from any extension connected to the Pbxnsip PBX, please dial "6123". This will route you to the Skype call testing center
16. If Skype account registered with the gateway has SkypeOut credits, you can try dialing a landline though Skype by entering "64168481850" from any extension. This will access IndustryDynamics head office