VoiceGear/Trixbox CE Integration Guide Ver. 0.2

Last Updated: May 27, 2009

1. Overview

This document presents a complete guide for integrating VoiceGear Skype gateways with Trixbox CE phone systems running version 2.6.x of Trixbox CE distribution software with Asterisk 1.4.x. The document assumes reader familiarity with both VoiceGear and Trixbox CE. For more details on VoiceGear configuration, please refer to the VoiceGear user guide available on www.industrydynamics.ca.

1.1 Setting up Trixbox CE PBX

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

1. Make sure the system running Trixbox CE software has an assigned static IP or a DHCP assigned IP that does not expire, the system is connected to the same LAN as VoiceGear gateway, and there is no firewall between the two systems
2. Switch Trixbox CE to Admin mode and login with administrator credentials
3. Set up a new SIP trunk and an outbound call route to enable the Trixbox CE system to talk to VoiceGear gateway. For more information, please refer to section 2.1

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 10.3.1.1, point your browser to http://10.3.1.1:8080

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 a Trixbox CE PBX

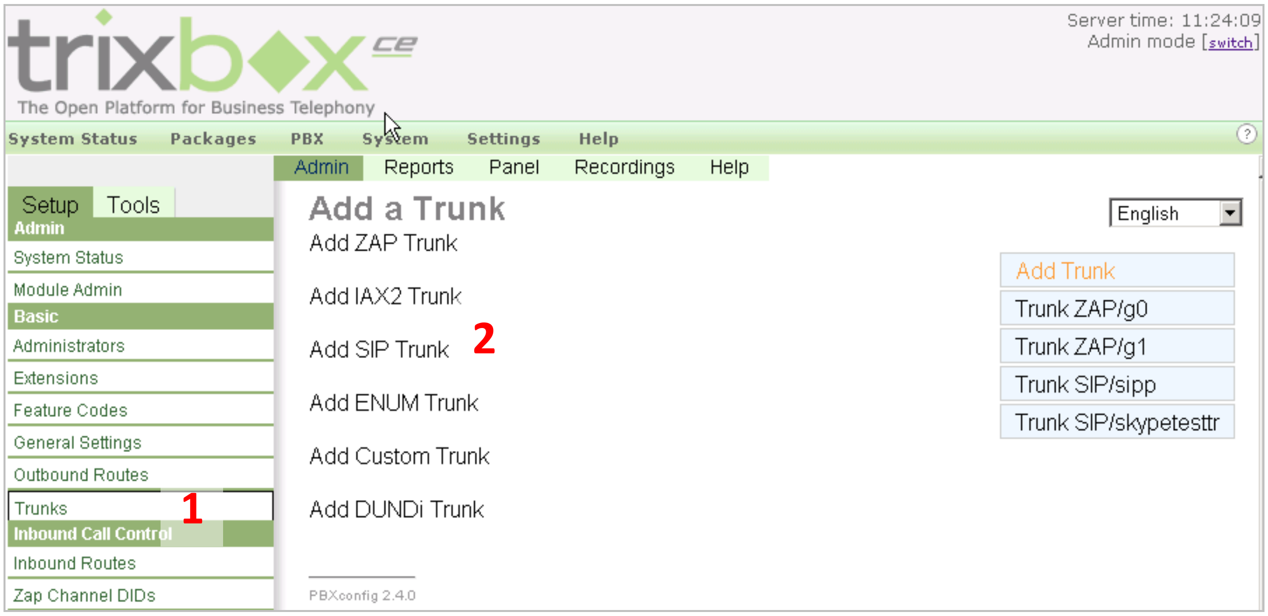
2. VoiceGear-Trixbox CE SIP Integration

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

2.1 Trixbox CE Configuration

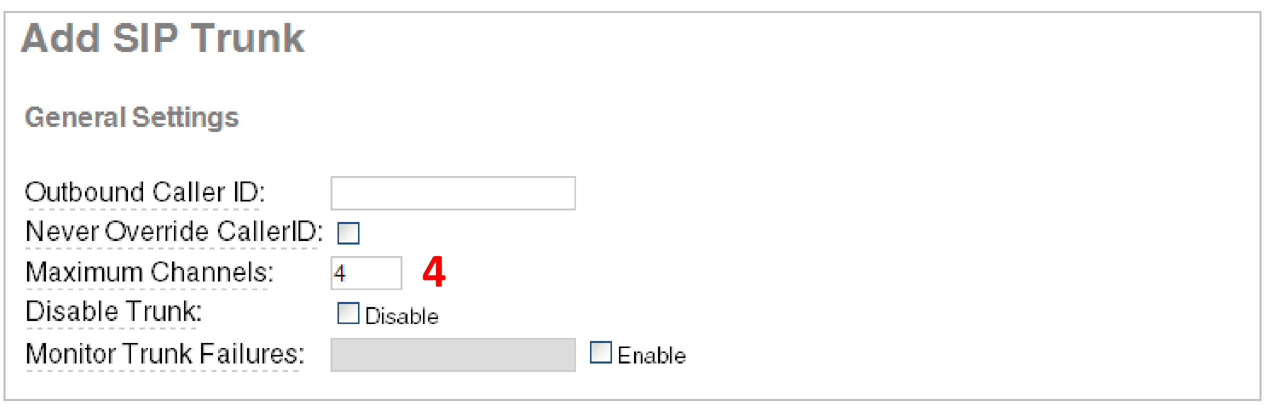
1. Navigate to the "PBX -> PBX Settings" screen via the top menu of the Trixbox CE Web user interface. Then click on "Basic -> Trunks" menu option.

2. Click on "Add SIP Trunk" link to create a new SIP trunk to be used for Skype calls via VoiceGear Skype gateway.

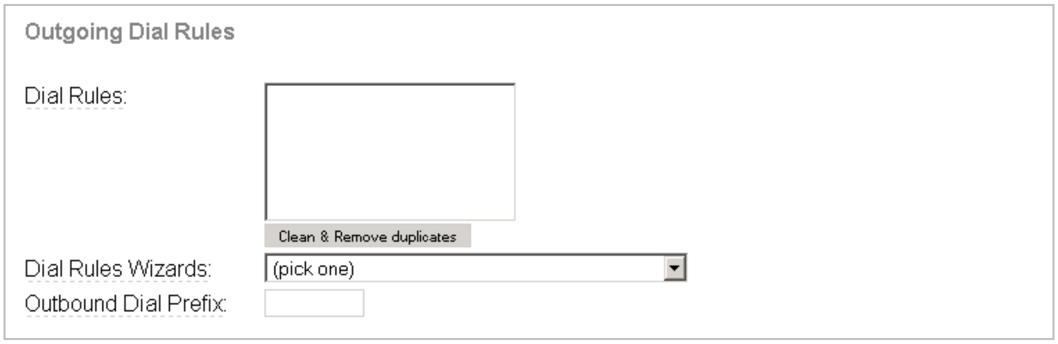


3. In the "Add SIP Trunk" screen, there are 5 main sections: General Settings, Outgoing Dial Rules, Outgoing Settings, Incoming Settings, and Registration.

4. In the "General Settings" section specify the maximum number of outgoing channels (simultaneous calls) to use with this trunk, or leave blank to specify no maximum.

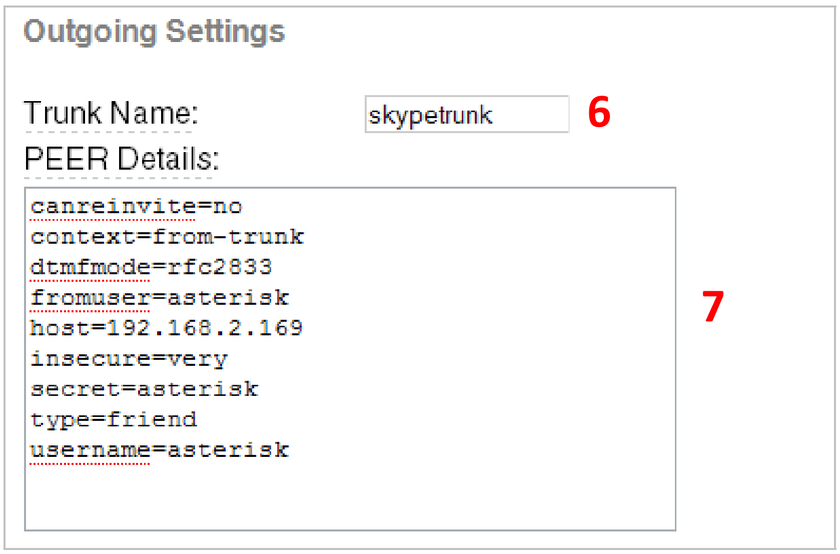


5. There are no changes required in the “Outgoing Dial Rules” section



6. Specify “Trunk Name” in the “Outgoing Settings” section

7. Type in the following information into the “PEER Details” input box:



Make sure to specify the static IP assigned to your VoiceGear gateway in the “host” property. Make a note of the “username” property specified here; you will need to enter the same user name in the VoiceGear configuration in section 2.2 below.

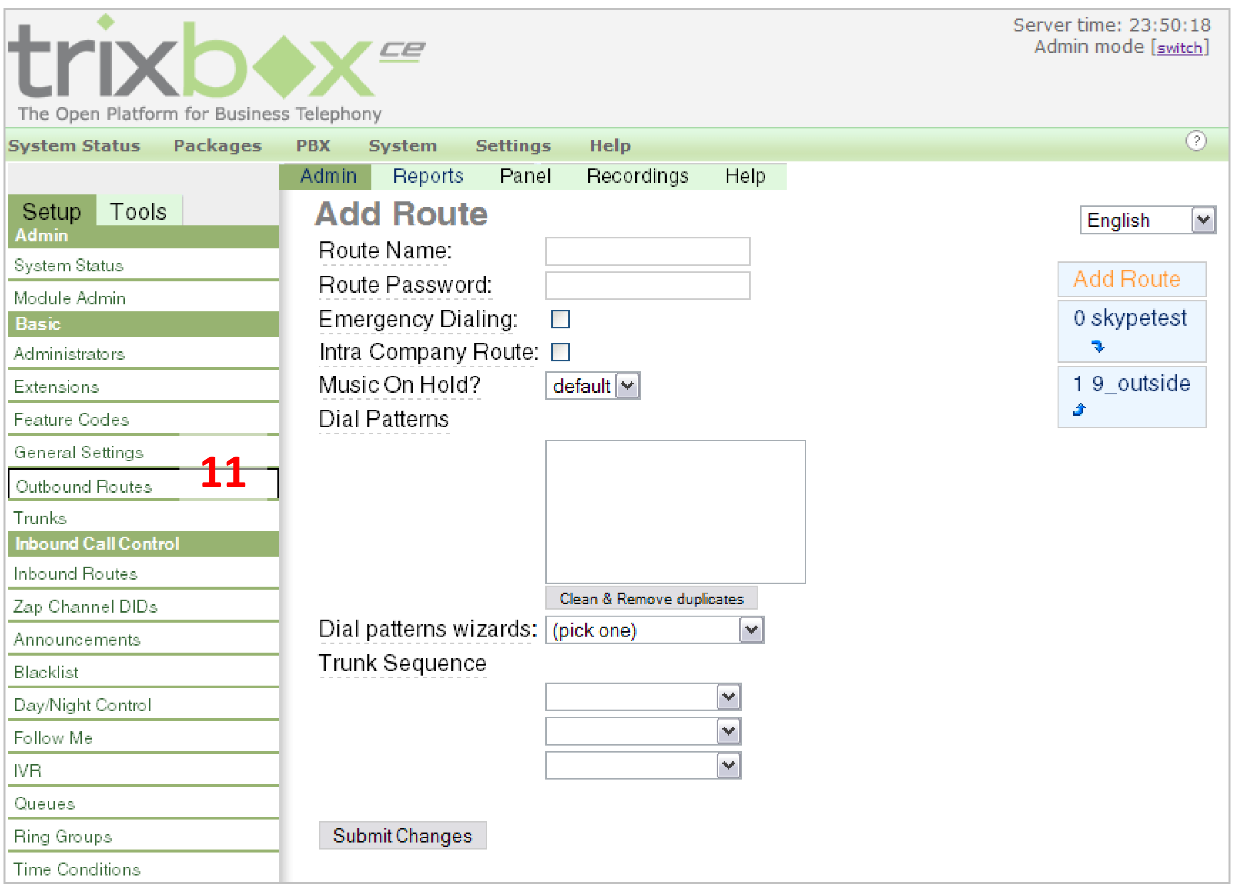
8. Specify “USER Context” in the “Incoming Settings” section as show below:

9. Specify “Register String” in the “Registration” section as show below:

Register string is used to register Trixbox CE system with VoiceGear gateway as a Skype provider. Make sure to specify the register string in the following format: *username:password@voicegear_ip_address*

10. Click “Submit Changes” button. This will create a new trunk that will be used for all Skype calls via VoiceGear gateway.

11. In the main menu on the left side, click on “Basic -> Outbound Routes” menu option to create a new outbound call route to be used for Skype calls via VoiceGear Skype gateway.

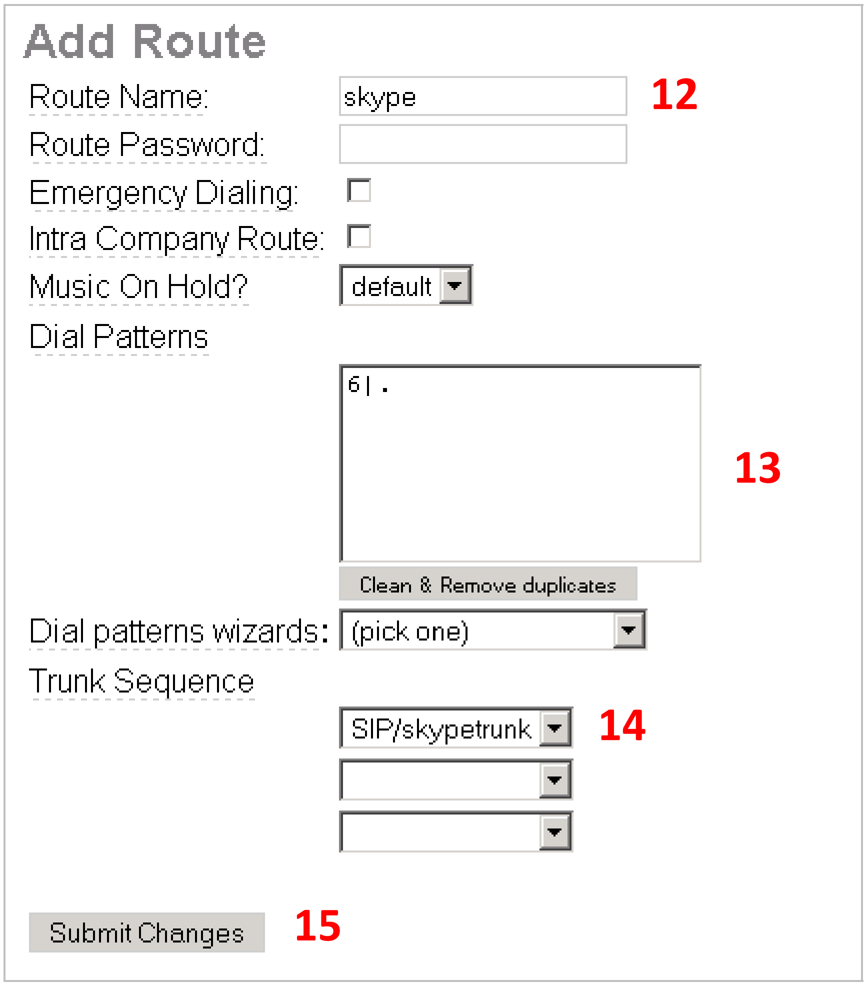


12. Specify “Route Name” in the “Add Route” screen

13. Specify “Dial Pattern” to be used for selecting the trunk connected to VoiceGear gateway (called “Skype trunk” below). In the example below, the digit 6 is used as a prefix to send outgoing calls via the Skype

trunk. Hence, if you dial 64168481850, the system would recognize the digit 6 as a prefix for Skype trunk, and will place the call to 4168481850 via Skype.

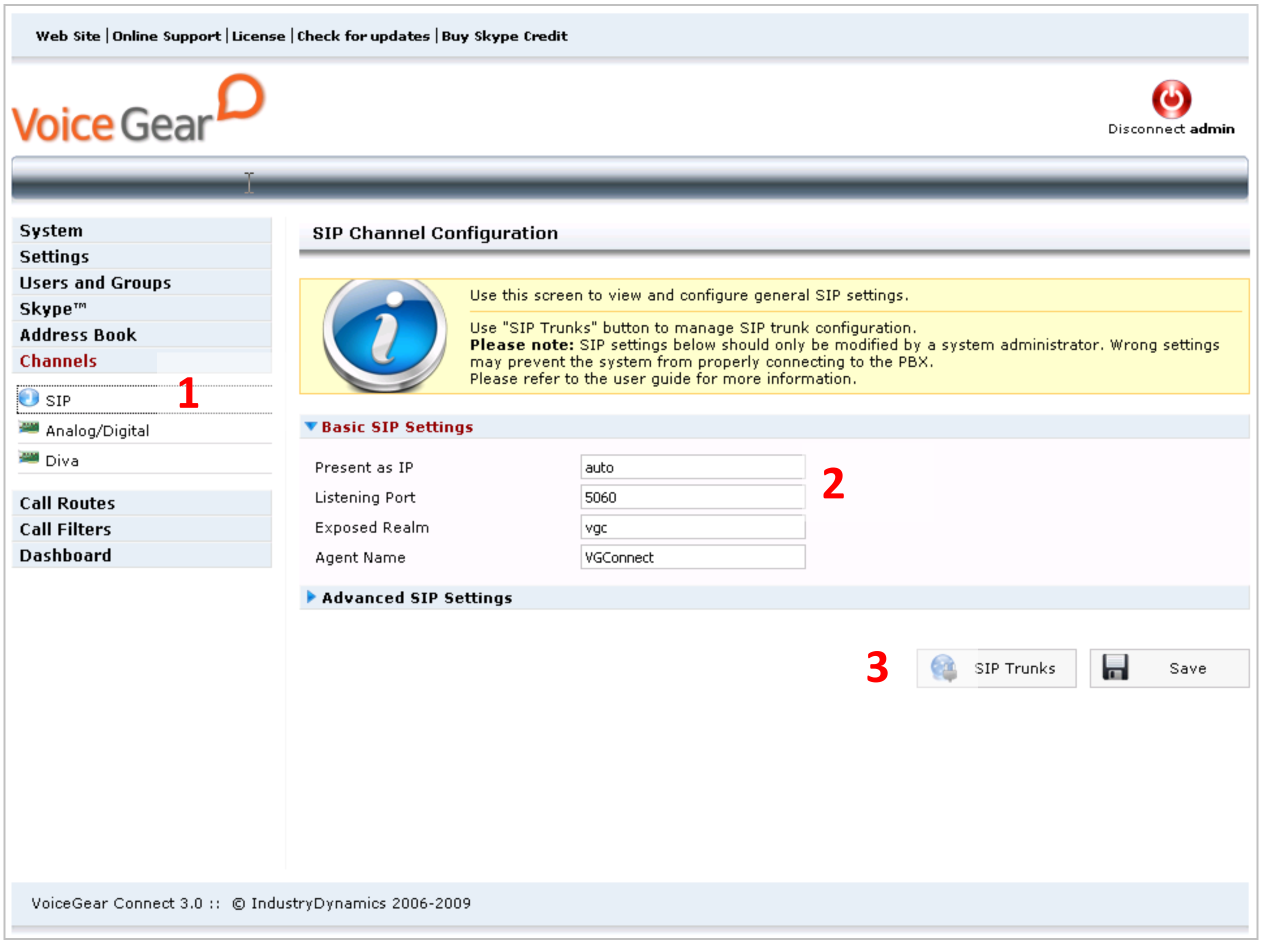
14. Select a predefined Skype trunk for the "Trunk Sequence" (you created a new SIP trunk for Skype in steps 1-10 above). This trunk will be used to place the call when the above Dial Pattern is matched.



15. Click "Submit Changes" button. This will create a new outgoing call route that will be used for outgoing Skype calls via VoiceGear gateway.

2.2 VoiceGear Gateway Configuration

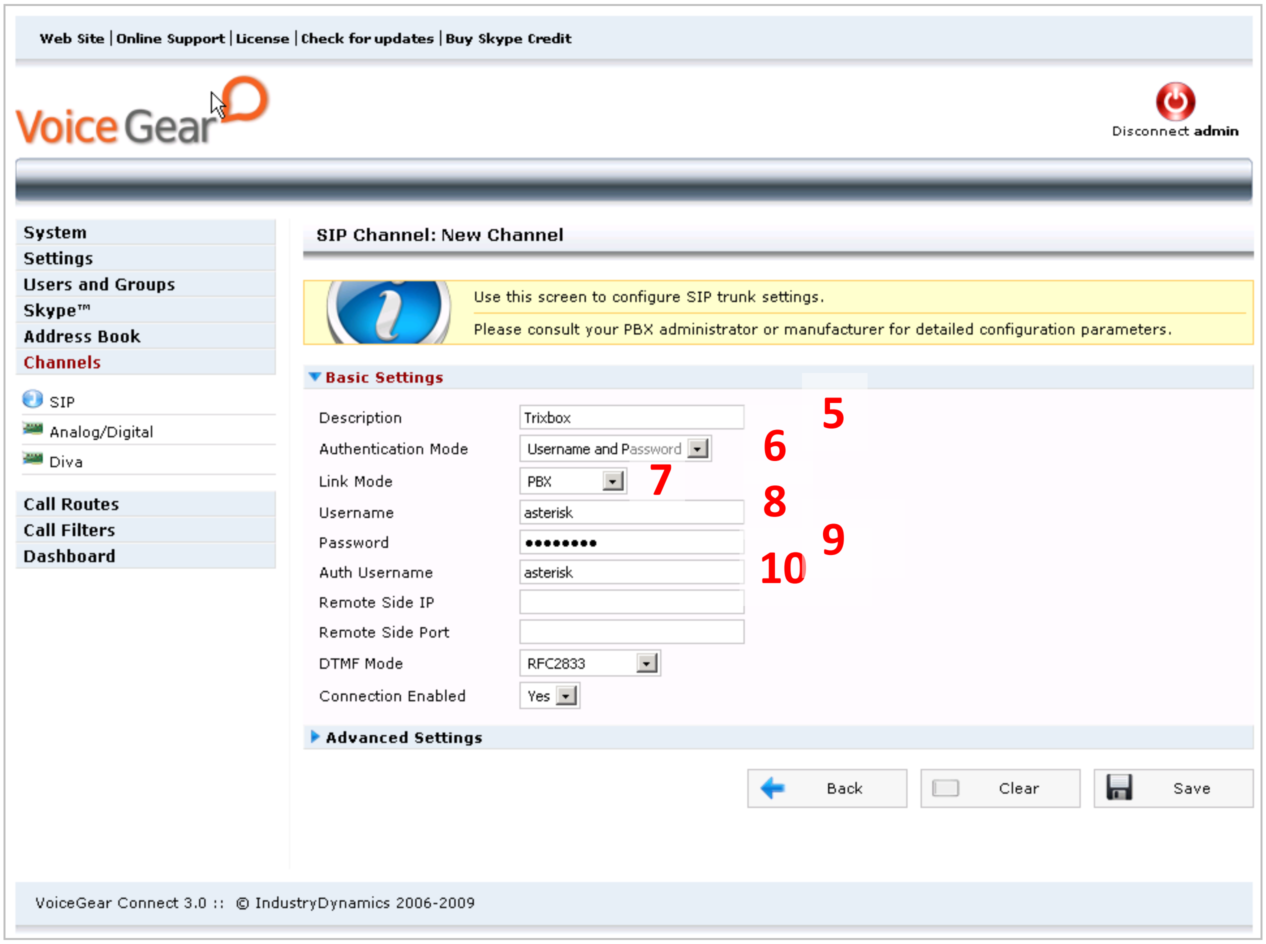
1. Navigate to the “Channels->SIP” screen via the main menu of the VoiceGear web configuration interface



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 Trixbox CE PBX

4. Under the SIP trunks page that opens, click on the “Add new” button to start adding a new trunk



5. Under "Basic settings", set the "Description" to '*Trixbox'* to help future identification of this SIP trunk

6. Select the "Authentication mode" to be "Username and Password"

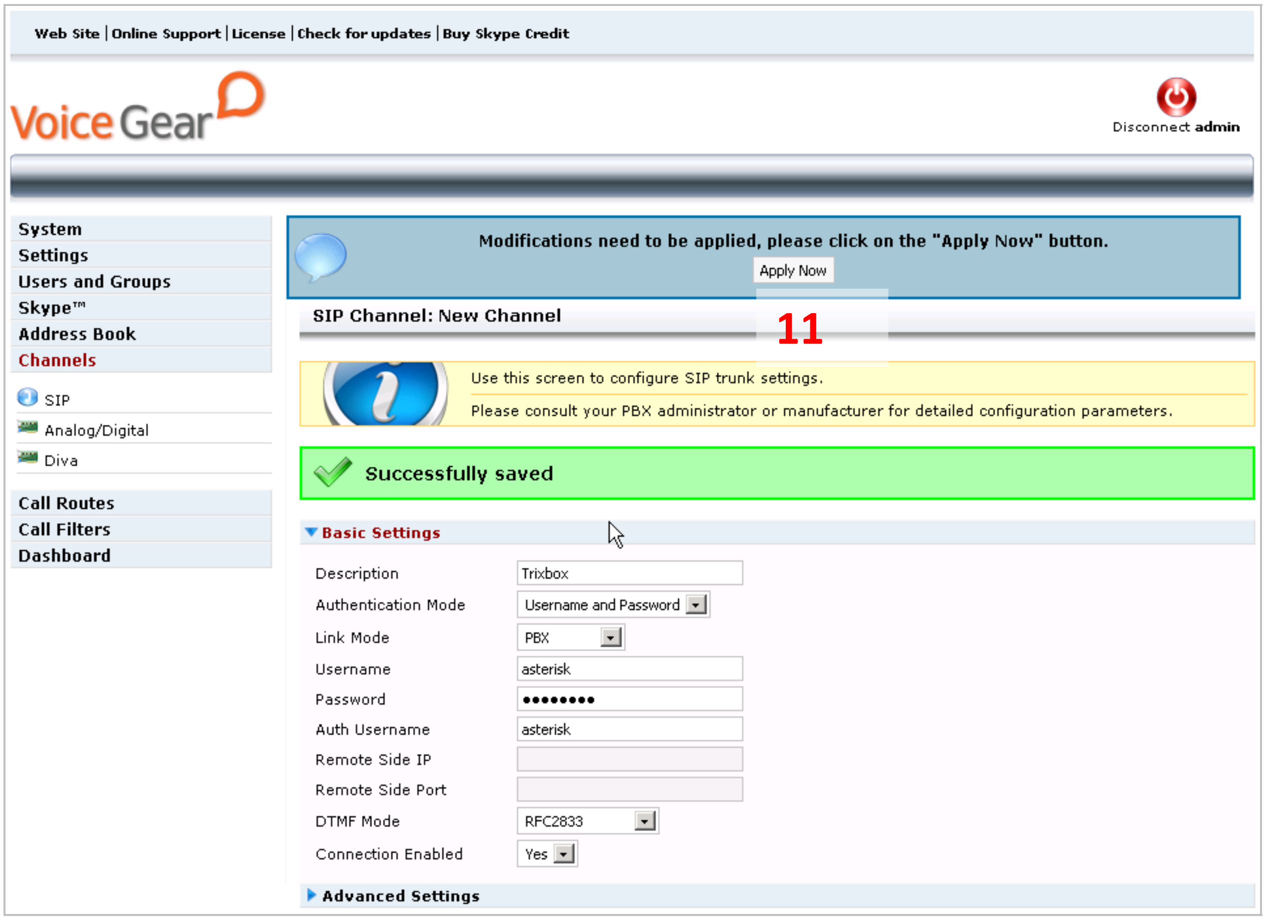
7. Select the link mode to be "PBX", this will tell VoiceGear gateway that it is communicating with a PBX as opposed to an individual user

8. Set the "Username" to be '*asterisk*'. Please note that this must match the "username" property set in step 7 above (in the Trixbox trunk configuration section)

9. Set the "Password" to be '*asterisk'*. Please note that this must match the "secret" property set in step 7 above (in the Trixbox trunk configuration section)

10. Set the "Auth Username" to '*asterisk*'. Please note that this must match the "fromuser" property set in step 7 above (in the Trixbox trunk configuration section)

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



12. Navigate to the “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 second part of the screen (under “2. Route Them to the First Available Destination in the Following List”), remove the default destination by clicking on the red “X” button to the right of it.

14. Add a new destination by selecting a SIP PBX trunk identified as “SIP PBX asterisk” from the “Select Destination Trunk” drop-down box, then type your extension off the Trixbox system into “Use this Number as Destination” text field. For example, if your office extension is “110”, then type in ‘*110*’ into this field. You can also specify ‘*s*’ as asterisk default extension, in which case your Trixbox will send the call to the default route, like your IVR or auto-attendant.

15. Click “Add” followed by “Save” to apply your changes

16. To try a test call from any extension connected to the Trixbox CE PBX, please dial “6123”. This will route you to the Skype call testing service

17. 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 dial into IndustryDynamics head office.