

Tech Bulletin 2011-006

IPitomy – NexVortex SIP Provider Configuration

Description

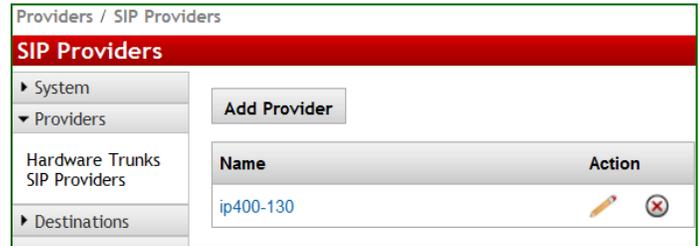
This guide is intended to streamline the installation of NexVortex SIP trunks in the IPitomy IP PBX.

Procedure – Add Provider

1. Navigate to the IPitomy IP PBX web interface as shown (usually 192.168.1.249/ippbx). (Your network may be different.) Under Providers select SIP Providers. The current Providers are listed—if this is the first, none will be listed here.

Add Provider

2. Select **Add Provider**
3. The screen at the right opens.
4. Input a name for this provider... "NexVortex".
5. Match all of the fields as they are listed.
6. "HOST" is "px1.nexvortex.com" unless otherwise instructed by NexVortex.
7. Register is usually "No" when NexVortex has been given a Static IP for this site. If the site is a Dynamically assigned IP Address site you must obtain an Authentication string from NexVortex and input this into the Custom parameter.
8. The Outbound Proxy must be: **px1.nexvortex.com**
9. The Username and Secret must be those provided to you by NexVortex.
10. Input the Call Limit...based on the subscription. (Ours was "2")
11. Select a "Default Destination" from those available if so desired. If one is selected the



SIP Provider	
Name:	NexVortex
User Type:	peer
DTMF Mode:	rfc2833
RFC2833 Compensate:	No
Host:	px1.nexvortex.com
Port:	<input checked="" type="radio"/> Default <input type="radio"/> Custom
Register:	<input type="radio"/> Yes <input checked="" type="radio"/> No <input type="radio"/> Custom
Authentication:	<input type="radio"/> Yes <input checked="" type="radio"/> No <input type="radio"/> Custom
Auth User:	<input checked="" type="radio"/> Default <input type="radio"/> Custom
From User:	<input checked="" type="radio"/> Default <input type="radio"/> Custom
From Domain:	<input type="radio"/> Default <input checked="" type="radio"/> Custom nexvortex.com
Realm:	<input checked="" type="radio"/> Default <input type="radio"/> Custom
Outbound Proxy:	<input type="radio"/> Disabled <input checked="" type="radio"/> Enabled px1.nexvortex.com
Username:	your_NexVortex_usernam
Secret:	your_NexVortex_password
Inbound Caller ID:	
Outbound Caller ID Name:	
Outbound Caller ID Number:	9413063700
Call Limit:	2
Qualify:	30000
Default Destination:	Menus Menu: Main Menu
Dial Prefix:	
RTP Keep-alive:	
Generate Ringing on outbound calls:	<input type="checkbox"/>

destination of non-DID incoming calls on this carrier will route to the destination in Call Routing—Incoming.

- Check “Allow Outbound Caller to Transfer” ONLY if you wish for calls being placed over these trunks to be allowed to control the PBX.

TYPICALLY this is NOT checked!

- Allow Call Recording is also optional.

- Choose from the available CODECs and click **Add** to “Add” them.

NexVortex’s native CODEC is G.711u. However they also support G.729 as a “fall-back”.

- Select each CODEC to be added and use the Up and Down buttons to position them to select the most desired first (top). Notice that the G.711u CODEC is on the top of our list... it will be the first CODEC protocol selected. G.729 is the fallback CODEC.

- If there are DID (Direct Inward Dial) numbers to be assigned. Add these one at a time in the Phone Numbers field at the bottom.

- Enter the number and then press the **Add** button.

- Once added select that number and assign a destination using the drop-down.

Notice that the test DID’s we were assigned are listed in our table... we assigned the “9413063701” number to ring at extension 2254.

Note: It is not necessary to define the destination of the prime number (lead number) as this will follow the Default Destination OR Call Routing—Incoming destination if none is assigned here.

- Don’t forget to click **Save Changes**

if not saved, all the information on this page must be entered again.

Allow Outbound Caller to transfer:	<input checked="" type="checkbox"/>	
Allow Call Recording:	<input checked="" type="checkbox"/>	
Ext CID Override:	<input type="checkbox"/>	
Restrict CID Override:	<input type="checkbox"/>	
Can Reinvite:	<input type="radio"/> Yes <input checked="" type="radio"/> No <input type="radio"/> N/A	
Send Remote Party ID	<input type="radio"/> Yes <input checked="" type="radio"/> No <input type="radio"/> N/A	
Trust Remote Party ID:	<input type="radio"/> Yes <input checked="" type="radio"/> No <input type="radio"/> N/A	
Insecure:	<input type="text" value="Very"/>	
Allow Codecs:	<div style="display: flex; justify-content: space-between;"> <div style="border: 1px solid gray; padding: 5px; width: 45%;"> <p style="text-align: center; margin: 0;">Disabled</p> <ul style="list-style-type: none"> G.711 (alaw) G.723.1 G.726 G.722 GSM </div> <div style="border: 1px solid gray; padding: 5px; width: 45%;"> <p style="text-align: center; margin: 0;">Enabled</p> <ul style="list-style-type: none"> G.711 (ulaw) G.729 </div> </div> <div style="margin-top: 10px; text-align: center;"> <input type="button" value="Add"/> <input type="button" value="Delete"/> </div>	
Phone Numbers		
This section contains phone numbers, (sometimes called DIDs) associated with this provider.		
<input type="text" value=""/>	<input type="text" value="9413063700"/> <input style="background-color: #e0e0e0;" type="text" value="9413063701"/>	<input type="button" value="Remove"/>
<input type="button" value="Add"/>		
Destination: <input type="text" value="Extension: 2254"/>		<input type="button" value="Set"/>
<input type="button" value="Save Changes"/>		

Note: “Insecure” is a Protocol matching parameter and has nothing to do with this carriers security. (Set it to “Very”)

Procedure—SIP (Global)

1. Navigate to the PBX Setup/SIP Setup page.

Note:

This is where “Global” settings are established. These settings are referenced whenever they are not specifically set in the SIP Provider definition—some are unique to this page and hence general to all SIP Providers.

2. The only item that we set in this area of system programming was the RTP Timeout. (In Advanced)
We set this to 120 seconds as a precaution to disconnect inactive calls with no voice traffic during a 2-minute period.
3. Don't forget to click

Save Changes

if not saved, the information on this page must be entered again.

The screenshot displays the 'PBX Setup / SIP Setup' interface. A red header bar reads 'SIP Setup'. On the left is a navigation menu with options: System, Providers, Destinations, Call Routing, PBX Setup (expanded), General, Database, Voicemail, SIP, Prompts, Music On Hold, Feature Codes, Services, and Reporting. The main content area is titled 'SIP Networking Settings' and includes a section for 'Local Networks & Subnet M...'. A table of settings is visible, with 'RTP Timeout' set to '120'. A yellow highlight box is placed over the '120' value with the text 'RTP Timeout: 120'. Other settings include 'Relax timer handling' (Yes/No), 'RTP Keep-alive', 'RTP Timeout on Hold', and 'Trust Remote Party ID' (Yes/No/N/A). An 'External IP:' field is also present. At the bottom, there is an 'Advanced' link.

Procedure—Call Routing-Outgoing

1. Navigate to the Call Routing/Outgoing page.
Note:
This is where user dialing strings are associated to trunks for use with what was dialed.
2. This may be an existing Outbound Route or new and specific for the trunk being added.
3. In this example the digits to be routed are those expected when calling long distance numbers beginning with "1".
...The dialed number will begin with "1" and then any digits... up to 11 digits maximum
In this digits-dialed-pattern Exact Length is set to "Yes".
4. Refer to the IPitomy 1100+ Manual for details on routing dialed digits.

<http://www.ipitomy.com/webrelease/IPitomy/IP1100+/IPitomy%20IP1100+%20Manual.pdf>

5. Notice that the added SIP Trunk (Provider) is now available for selection in the drop-down list.

6. Select your added Provider (ours is NexVortex) and click 

7. You must also click



before other changes to this trunk can be applied to the routing characteristics.

8. Notice that we placed this trunk at the top of the list for this dialing string. That means it will be selected first for calls placed with this digit string. (The trunks in "Boston" will be selected second.)

9. Here again reference the IPitomy 1100+ Manual for details on programming parameters.

10. When you're done, don't forget to click



...if not saved, the information on this page must be entered again.

Call Routing / Outgoing / Edit Outgoing Route

Edit Outgoing Route

System	Edit Outbound Route		
Providers	Route Name	NexVortex 11 Digit	
Destinations	Route Type	NexVortex 11 Digit	
Call Routing	Start Pattern	1N	
Incoming	Digits	11	
Outgoing	Exact Length	Yes	
Class Of Service	Subroute Digits	3	
PBX Setup	Subroute Offset	1	
Reporting	Trunks:	NexVortex-Boston	Strip Digits 0
		Up	Prefix Digits
		Dn	
		Add	NexVortex
		Delete	
	Disable Ext CID Override	no	
	Force Use PSTN CID	no	
	Override Default CID(name)	no	
	Override CID Name		
	Override Default CID(number)	no	
	Override CID Number		
			

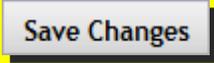
Note:

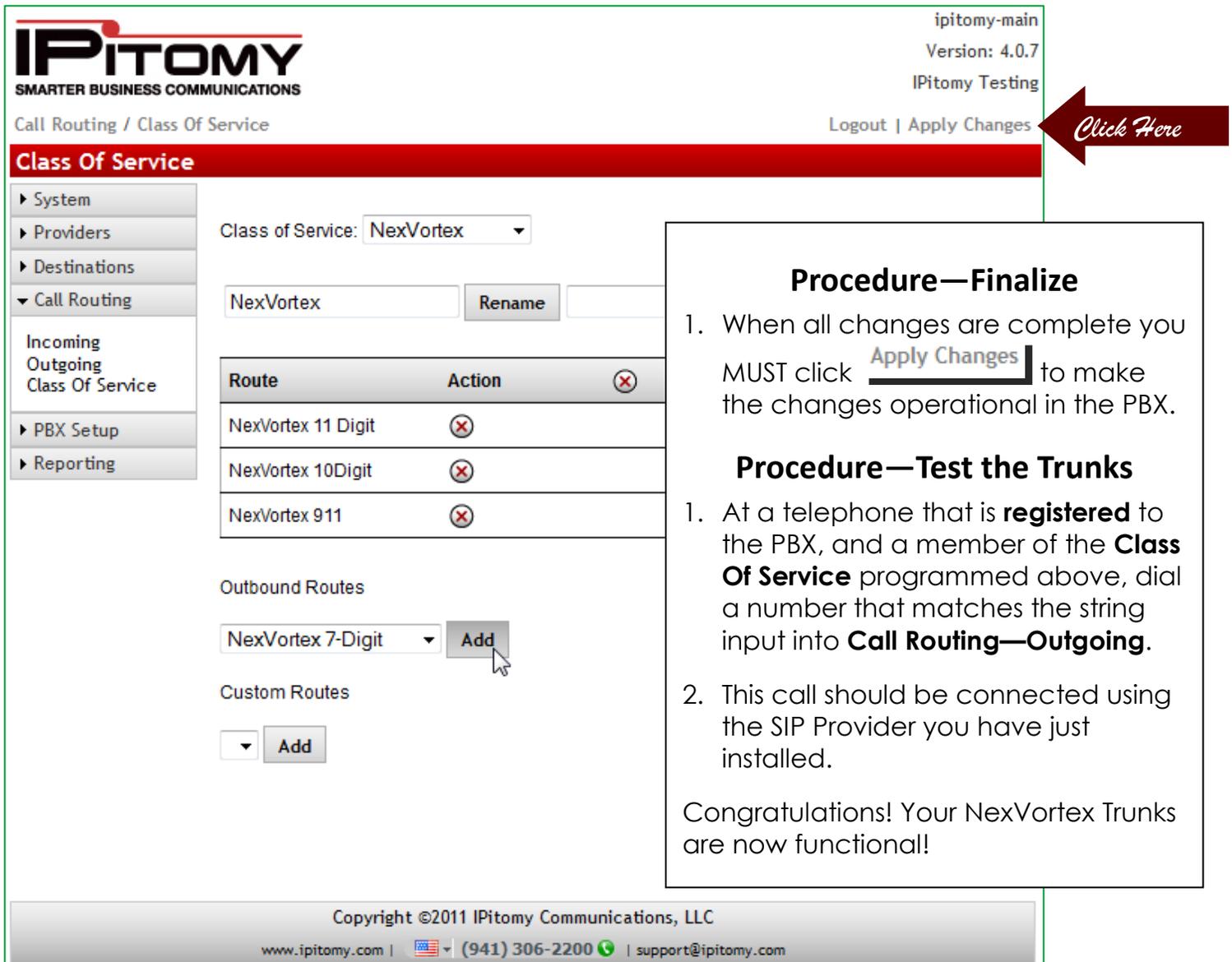
"When adding your "911" Route, also add a "311" Route with the same properties. NexVortex has a great feature to test the 911 operation without actually signaling emergency personnel. This is done by dialing 311. You must assure that the programming matches each of these Route assignments.

Procedure—Class Of Service

1. Navigate to the Call Routing/Class Of Service page.
Note:
This is where trunks are assigned as those allowed to be used by the various classes of service.
2. ONLY if a NEW Outgoing Route was created is this step necessary. ...generally there are only a few Classes Of Service. In the picture below the test system COS "NexVortex" is shown. Notice that the newly created Outbound Route "NexVortex 11 Digit" and others also added are listed here.
3. Selected any/all those from the drop-down list that should be utilized by this COS.
4. Click  to add this route to this COS.

Note:

This page does not have a  button.



ipitomy-main
Version: 4.0.7
IPitomy Testing
Logout | Apply Changes *Click Here*

Class Of Service

Class of Service: NexVortex

Route	Action
NexVortex 11 Digit	
NexVortex 10Digit	
NexVortex 911	

Outbound Routes
NexVortex 7-Digit 

Custom Routes


Procedure—Finalize

1. When all changes are complete you MUST click  to make the changes operational in the PBX.

Procedure—Test the Trunks

1. At a telephone that is **registered** to the PBX, and a member of the **Class Of Service** programmed above, dial a number that matches the string input into **Call Routing—Outgoing**.
2. This call should be connected using the SIP Provider you have just installed.

Congratulations! Your NexVortex Trunks are now functional!

Copyright ©2011 IPitomy Communications, LLC
www.ipitomy.com | (941) 306-2200 | support@ipitomy.com