

## Tech Bulletin 2011-002

### IPitomy—Windstream

## Description

This guide is intended to streamline the installation of Windstream 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.)
2. Under Providers select SIP Providers. The current Providers are listed—if this is the first, none will be listed here.
3. Select Add Provider
4. The screen at the right opens.
5. Input a name for this provider... being descriptive is helpful... ours is "Windstream".
6. Match all of the fields as they are listed here.
7. User Name and Secret (password) are necessary only if instructed to use them by Windstream.
8. Input the Caller ID data desired. The CID data here represents the content of our testing.
9. Input the maximum calls that can be placed on this provider. (Ours is Call Limit "2".)
10. Select a "Default Destination" from those available if so desired. If none is selected here the destination of any incoming call on this carrier that is NOT defined as a DID will route to the destination in Call Routing—Incoming.
11. Check "Allow Outbound Caller to Transfer" ONLY if you wish for calls being placed over these trunks to be allowed to control the PBX
12. Continue on through the remaining fields (shown on next page).

Name	Action
ip400-130	

**SIP Provider**

Name:

User Type:

DTMF Mode:

RFC2833 Compensate:

Host:

Port:  Default  Custom

Register:  Yes  No  Custom

Authentication:  Yes  No  Custom

Auth User:  Default  Custom

From User:  Default  Custom

From Domain:  Default  Custom

Realm:  Default  Custom

Outbound Proxy:  Disabled  Enabled

Username:

Secret:

Inbound Caller ID:

Outbound Caller ID Name:

Outbound Caller ID Number:

Call Limit:

Qualify:

Default Destination:   ...

Dial Prefix:

Area Code:

Generate Ringing on outbound calls:

Allow Outbound Caller to transfer:

**TYPICALLY this is NOT checked!**

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13. Allow Call Recording is also optional.
14. Choose from the available CODECs and click to "Add" them.
15. If there is a preferred order, select each added CODEC and use the Up and Down buttons to position them hierarchically to select the most desired first (top).
16. 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.

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

- Enter the number and



then press the button.

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

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.

17. Don't forget to click



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

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 type="radio"/> No <input checked="" type="radio"/> N/A																		
Trust Remote Party ID:	<input type="radio"/> Yes <input type="radio"/> No <input checked="" type="radio"/> N/A																		
Insecure:	Very <input type="button" value="v"/>																		
Allow Codecs:	<table style="width: 100%; border: none;"> <tr> <td style="text-align: center;">Disabled</td> <td style="text-align: center;">Enabled</td> <td></td> </tr> <tr> <td style="border: 1px solid gray; padding: 2px;">G.723.1</td> <td style="border: 1px solid gray; padding: 2px;">G.711 (ulaw)</td> <td rowspan="4" style="text-align: center; vertical-align: middle;"> <input type="button" value="Up"/>  <input type="button" value="Down"/> </td> </tr> <tr> <td style="border: 1px solid gray; padding: 2px;">G.726</td> <td style="border: 1px solid gray; padding: 2px;">G.711 (alaw)</td> </tr> <tr> <td style="border: 1px solid gray; padding: 2px;">iLBC</td> <td style="border: 1px solid gray; padding: 2px;">GSM</td> </tr> <tr> <td style="border: 1px solid gray; padding: 2px;">Speex</td> <td></td> </tr> <tr> <td style="border: 1px solid gray; padding: 2px;">LPC10</td> <td></td> <td></td> </tr> <tr> <td style="text-align: center;"><input type="button" value="Add"/></td> <td style="text-align: center;"><input type="button" value="Delete"/></td> <td></td> </tr> </table>	Disabled	Enabled		G.723.1	G.711 (ulaw)	<input type="button" value="Up"/> <input type="button" value="Down"/>	G.726	G.711 (alaw)	iLBC	GSM	Speex		LPC10			<input type="button" value="Add"/>	<input type="button" value="Delete"/>	
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iLBC	GSM																		
Speex																			
LPC10																			
<input type="button" value="Add"/>	<input type="button" value="Delete"/>																		
<b>Phone Numbers</b>																			
This section contains phone numbers, (sometimes called DIDs) associated with this provider.																			
	<input type="button" value="Remove"/>																		
<input type="button" value="Add"/>																			
Destination:	None <input type="button" value="Set"/>																		
<input type="button" value="Save Changes"/>																			

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### IPitomy—Windstream

#### 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 needed to change in this area of system programming was the RTP Timeout. (In Advanced)

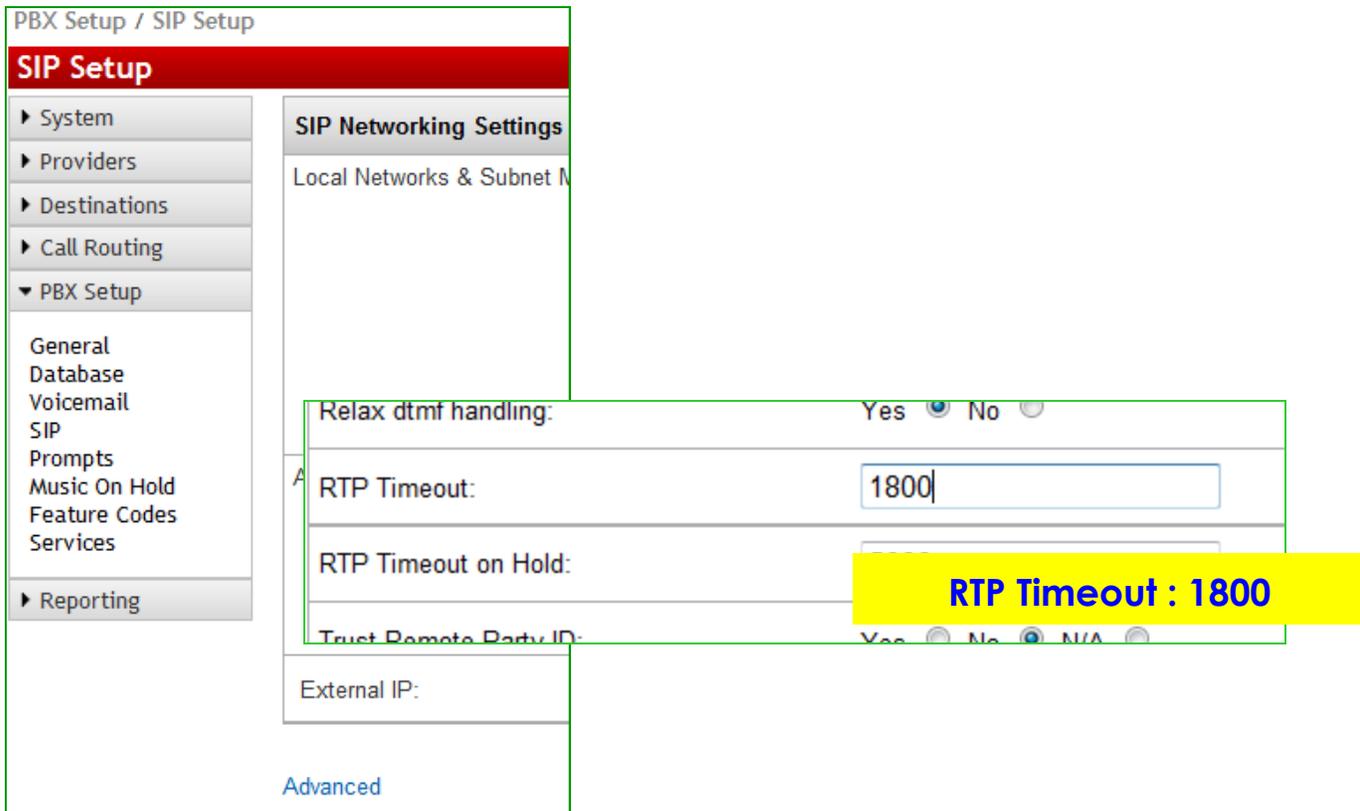
We discovered that Windstream very efficiently suppresses silence and hence the IPitomy PBX may conclude that the connection has gone idle. If this happens the PBX disconnects the call.

Windstream has requested that this timer be set to 1800s to match their Safeguard parameter which periodically (30minutes) checks for an active connection. If an active call is not confirmed the connection session is ended.

3. Don't forget to click



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



PBX Setup / SIP Setup

### SIP Setup

- ▶ System
- ▶ Providers
- ▶ Destinations
- ▶ Call Routing
- ▼ PBX Setup
  - General
  - Database
  - Voicemail
  - SIP
  - Prompts
  - Music On Hold
  - Feature Codes
  - Services
- ▶ Reporting

#### SIP Networking Settings

Local Networks & Subnet M

Relax dtmf handling: Yes  No

RTP Timeout:

RTP Timeout on Hold:

Trust Remote Party ID: Yes  No  N/A

External IP:

Advanced

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### IPitomy—Windstream

## Procedure—Call Routing-Outgoing

- Navigate to the Call Routing/Outgoing page.  
Note:  
This is where user dialing strings are associated to trunks for use with what was dialed.
- This may be an existing Outbound Route or something specific for the trunk being added.
- In this example the digits to be routed are those expected when calling international numbers.  
-The dialed number will begin with "011" and then any digits... Exact Length is set to "No".
- Refer to the IPitomy 1100+ Manual for details on routing dialed digits.
- Notice that the SIP Trunk is now available for selection in the drop-down list. (At the bottom)

- We selected Windstream and clicked **Add**
- You must also click **Save Changes** before other changes to this trunk can be applied to the routing characteristics.
- 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 "Houston" will be selected second.)
- Here again reference the IPitomy 1100+ Manual for details on programming parameters.
- When you're done, don't forget to click **Save Changes**

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

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### IPitomy—Windstream

#### 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 System Default COS is shown. Notice that the newly created Outbound Route "Windstream" is listed here.

3. Click  to add this route to this COS.

#### Note:

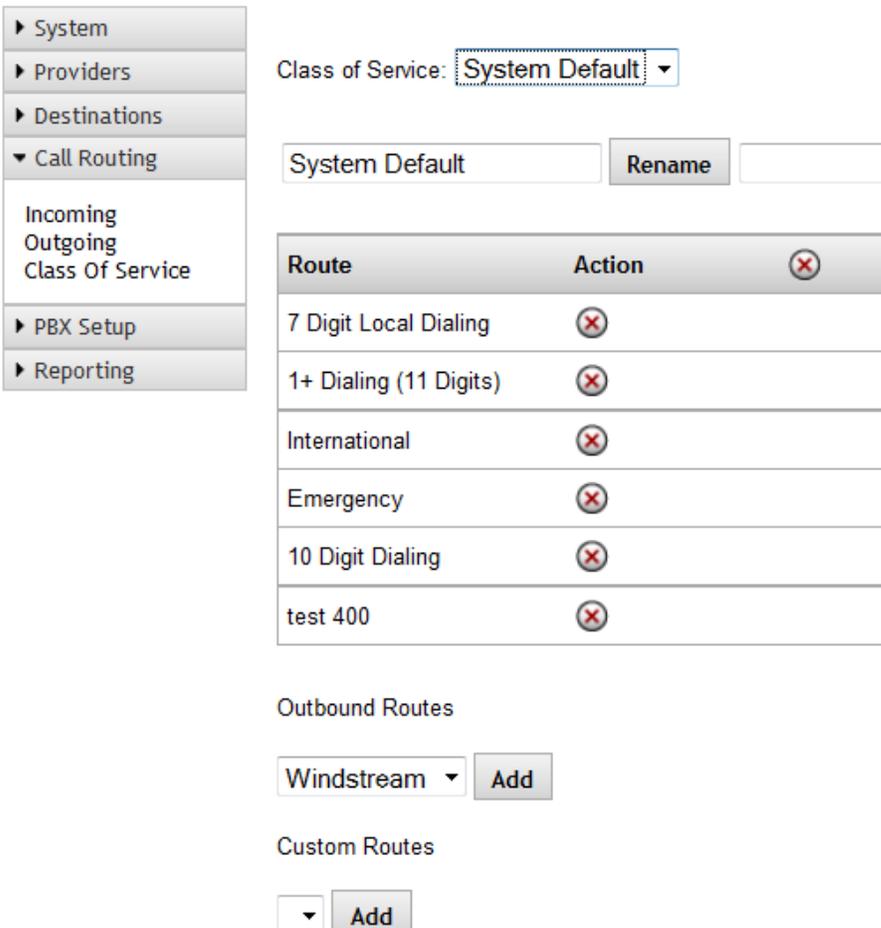
This page does not have a  button.

Call Routing / Class Of Service

Logout | Apply Changes

 Click Here

#### Class Of Service



Route	Action
7 Digit Local Dialing	
1+ Dialing (11 Digits)	
International	
Emergency	
10 Digit Dialing	
test 400	

#### 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 Windstream Trunks are now functional!