

Tech Bulletin 2011-003

IPitomy – MegaPath SIP Provider Configuration

Description

This guide is intended to streamline the installation of MegaPath (a.k.a. Speakeasy and Covad) 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... we used "MegaPath".
6. Match all of the fields as they are listed.
7. Your "HOST" will be different and provided by MegaPath.
8. Your "HOST" must also be input in Outbound Proxy.
9. The Register field MUST be custom. Enter it as shown substituting:
 - a. "IPitomy" for the *Username* they gave you.
 - b. The *HostipAddressDomain* MUST match that of the HOST ...also provided by MegaPath.
10. User Name and Secret (password) are provided by MegaPath.
11. The Output Caller ID MUST match your MegaPath TN (provided by MegaPath).
12. Input the Call Limit...based on the subscription. (We tested "2")
13. Select a "Default Destination" from those available if so desired. If *None* is selected the destination

Providers / SIP Providers

SIP Providers

System

Providers **Add Provider**

Hardware Trunks

SIP Providers

Destinations

Name	Action
ip400-130	

SIP Provider

Name: MegaPath

User Type: peer

DTMF Mode: rfc2833

RFC2833 Compensate: No

Host: lab-1-siptrunk-a.voice.spe

Port: Default Custom

Register: Yes No Custom

register => IPitomy:fr33h2o2day@lab-1-siptrunk-a.voice.speakeasy.net/IPitomy

register => IPitomy:secret@HostipAddressDomain/IPitomy

Authentication: Yes No Custom

Auth User: Default Custom

From User: Default Custom ask164u

From Domain: Default Custom speakeasy.net

Realm: Default Custom

Outbound Proxy: Disabled Enabled lab-1-siptrunk-a.voice.spe

Username: IPitomy

Secret: fr33h2o2day

Inbound Caller ID:

Outbound Caller ID Name:

Outbound Caller ID Number: 3125334877

Call Limit: 2

Qualify: 0

Default Destination: Extensions Extension: 2254

Dial Prefix:

RTP Keep-alive:

Generate Ringing on outbound calls:

Allow Outbound Caller to transfer:

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

14. 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!

15. Allow Call Recording is also optional.

16. Choose from the available CODECs and click **Add** to "Add" them.

MegaPath's Native CODEC is G.711u.

17. Select each added CODEC and use the Up and Down buttons to position them to select the most desired first (top).

18. 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.

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.

19. Don't forget to click **Save Changes**

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

Allow Call Recording:	<input type="checkbox"/>						
Ext CID Override:	<input type="checkbox"/>						
Restrict CID Override:	<input type="checkbox"/>						
Can Reinvite:	<input checked="" type="radio"/> Yes <input type="radio"/> No <input type="radio"/> N/A						
Send Remote Party ID	<input checked="" type="radio"/> Yes <input type="radio"/> No <input 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;"> <thead> <tr> <th style="width: 50%; border: none;">Disabled</th> <th style="width: 50%; border: none;">Enabled</th> </tr> </thead> <tbody> <tr> <td style="border: none;"> <div style="border: 1px solid gray; padding: 2px;"> G.723.1 G.726 iLBC Speex LPC10 </div> </td> <td style="border: none;"> <div style="border: 1px solid gray; padding: 2px;"> G.711 (ulaw) G.711 (alaw) GSM </div> </td> </tr> <tr> <td style="border: none; text-align: center;"><input type="button" value="Add"/></td> <td style="border: none; text-align: center;"><input type="button" value="Delete"/></td> </tr> </tbody> </table>	Disabled	Enabled	<div style="border: 1px solid gray; padding: 2px;"> G.723.1 G.726 iLBC Speex LPC10 </div>	<div style="border: 1px solid gray; padding: 2px;"> G.711 (ulaw) G.711 (alaw) GSM </div>	<input type="button" value="Add"/>	<input type="button" value="Delete"/>
Disabled	Enabled						
<div style="border: 1px solid gray; padding: 2px;"> G.723.1 G.726 iLBC Speex LPC10 </div>	<div style="border: 1px solid gray; padding: 2px;"> G.711 (ulaw) G.711 (alaw) GSM </div>						
<input type="button" value="Add"/>	<input type="button" value="Delete"/>						

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

Phone Numbers
This section contains phone numbers, (sometimes called DIDs) associated with this provider.
<div style="border: 1px solid gray; width: 150px; height: 40px; margin-bottom: 5px;"></div> <div style="display: flex; align-items: center; gap: 10px;"> <input type="button" value="Add"/> <div style="border-left: 1px solid gray; border-right: 1px solid gray; height: 20px; width: 20px; margin: 0 5px;"></div> <input type="button" value="Remove"/> </div>
Destination: None <input type="button" value="Set"/>
<input type="button" value="Save Changes"/>

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 shows the 'PBX Setup / SIP Setup' interface. The 'SIP Setup' section is active, displaying 'SIP Networking Settings' and 'Local Networks & Subnet M'. A table of settings is visible, with 'RTP Timeout' set to '120'. A yellow highlight is placed over the '120' value with the text 'RTP Timeout: 120'. Other settings include 'Relax dtmf handling' (Yes/No), 'RTP Keep-alive', 'RTP Timeout on Hold', and 'Trust Remote Party ID' (Yes/No/N/A). The 'Advanced' tab is selected at the bottom.


Relax dtmf handling:	Yes <input type="radio"/> No <input type="radio"/>
RTP Keep-alive:	<input type="text"/>
RTP Timeout:	120
RTP Timeout on Hold:	<input type="text"/>
Trust Remote Party ID:	Yes <input type="radio"/> No <input type="radio"/> N/A <input checked="" type="radio"/>


External IP:

Advanced

Procedure—Call Routing-Outgoing

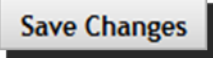
1. Navigate to the Call Routing/Outgoing page.
Note:
This is where user dialing strings are associated to trunks for use with what was dialed.
2. This may be an existing Outbound Route or new and specific for the trunk being added.
3. In this example the digits to be routed are those expected when calling international numbers.
...The dialed number will begin with "011" and then any digits... In this digits-dialed-pattern Exact Length is set to "No".
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. Selected your added Provider (ours is MegaPath) and clicked 

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 "Houston" 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

<ul style="list-style-type: none"> System Providers Destinations Call Routing Incoming Outgoing Class Of Service PBX Setup Reporting 	<p>Edit Outbound Route</p> <p>Route Name: <input type="text" value="MegaPath"/></p> <p>Route Type: <input type="text" value="MegaPath"/></p> <hr/> <p>Start Pattern: <input type="text" value="011X"/></p> <p>Digits: <input type="text" value="4"/></p> <p>Exact Length: <input type="text" value="No"/></p> <p>Subroute Digits: <input type="text" value="0"/></p> <p>Subroute Offset: <input type="text" value="0"/></p> <hr/> <p>Trunks: <input type="text" value="MegaPath - Houston"/></p> <p>Up <input type="button" value="Up"/></p> <p>Dn <input type="button" value="Dn"/></p> <p>Add <input type="button" value="Add"/></p> <p>Delete <input type="button" value="Delete"/></p> <p>Strip Digits: <input type="text" value="0"/></p> <p>Prefix Digits: <input type="text"/></p> <hr/> <p>Disable Ext CID Override: <input type="checkbox"/></p> <p>Force Use PSTN CID: <input type="checkbox"/></p> <p>Override Default CID(name): <input type="text" value="no"/></p> <p>Override CID Name: <input type="text"/></p> <p>Override Default CID(number): <input type="text" value="no"/></p> <p>Override CID Number: <input type="text"/></p> <p><input type="button" value="Save Changes"/></p>
---	---

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 "BW Demos" is shown. Notice that the newly created Outbound Route "MegaPath" is listed here and selected from the drop-down list.
3. Click to add this route to this COS.

Note:

This page does not have a button.

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 MegaPath Trunks are now functional!