

Tech Bulletin 2012-002



IPitomy – AccessLine SIP Provider Configuration

Description

This guide is intended to streamline the installation of AccessLine SIP trunks in the IPitomy IP PBX. In our combined testing we determined that analog FAX machines should be connected via an analog telephone line dedicated to the function or on a separate gateway configured for CODEC G.711. AccessLine makes use of the G.729 CODEC to optimize bandwidth usage and we found the voice quality very good during our testing. The configuration deploys both the G.720 and G.711 (as a backup CODEC).

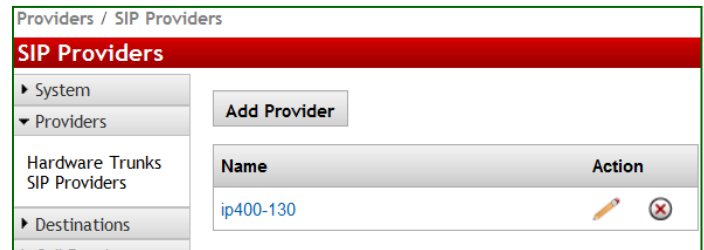
Refer to the IPitomy 1100+ Installation and Maintenance Manual on the IPitomy Wiki for details about adding SIP Providers. http://wiki.ipitomy.com/index.php/IP_PBX_Manual_Providers#Provisioning_a_SIP_Provider

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 Edit SIP Provider screen opens.



- Input a **Name** for this provider... we used "AccessLine".
- Match all of the fields as they are listed.
- "**HOST**" as provided by AccessLine is the **DNS NAME** in their configuration guide:

Provided by AccessLine	
SIP TRUNK ID	dgwsid9151
PASSWORD	N9Yn16JI
DNS NAME	interop-ex.sea.AccessLine.com
IP ADDRESS	64.28.122.44
PORT	6060
Primary CODEC	G.729
Secondary CODEC	G.711

In our test it was not necessary to use the provided IP Address. We used the DNS.

- Port** must be defined as "Custom" and set to **6060**.
- Register** must be Custom and filled with the details as shown. HOWEVER, it is easier to complete the other fields first and save your work as this field will auto-fill. (Come back to it.)
- Insert the **SIP Trunk ID** into **Username**.
- Insert the **PASSWORD** into **Secret**.
- Since G.729 is not a default CODEC, you must select it on the left-side "Disabled" list and click **Add** to "Add" it.
- Now select the G.729 CODEC on the right-side and click **Up** until it is at the top of the list.
- Now select all those CODEC's that are not to be used and click **Delete** to move them to the Disabled list. Only G.729 and G.711 should remain.
- Now click **Save Changes** at the bottom of the page.
- Go back to **Register** and select Custom. When you do this the field will be populate with the default content using your input configuration.
- At the end of the string insert ":6060" (colon6060).
- Now click **Save Changes** again.

Edit SIP Provider

- System
- Providers
- Hardware Trunks
- SIP Providers
- Destinations
- Call Routing
- PBX Setup
- Reporting
- Diagnostics

SIP Provider

Name: Accessline

User Type: peer

DTMF Mode: rfc2833

RFC2833 Compensate: No

Host: interop-ex.sea.accessline.com

Port: Default Custom 6060

Register: Yes No Custom

register => dgwsid9151:N9Yn16JI@interop-ex.sea.accessline.com:6060

Authentication: Yes No Custom auth = dgwsid9151:N9Yn16,

Auth User: Default Custom

From User: Default Custom

From Domain: Default Custom

Realm: Default Custom

Outbound Proxy: Disabled Enabled

Username: dgwsid9151

Secret: N9Yn16JI

Inbound Caller ID:

Outbound Caller ID Name: IPitomy Main

Outbound Caller ID Number: 2065172676

Call Limit: 2

Quality: 30000

Default Destination: None

Dial Prefix:

RTP Keep-alive:

Generate Ringing on outbound calls:

Allow Outbound Caller to transfer:

Allow Call Recording:

Ext CID Override:

Restrict CID Override:

Can Reinvite: Yes No N/A

Send Remote Party ID: Yes No N/A

Trust Remote Party ID: Yes No N/A

Insecure: Very

Allow Codecs:

Disabled	Enabled
G.711 (alaw)	G.729
G.723.1	G.711 (ulaw)
G.726	
G.722	
GSM	

Add Delete Up Down

Phone Numbers

This section contains phone numbers, (sometimes called DIDs) associated with this provider.

2065172793

Add Remove

Destination: None Set

18. 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!

19. Allow Call Recording is optional.

20. 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.

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

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

We assigned the "2065172793" number to ring at extension 5555.

Note: It is not necessary to define the destination of the prime number (lead number) as any number not defined in DID Phone Numbers will follow the Default Destination.


21. Don't forget to click

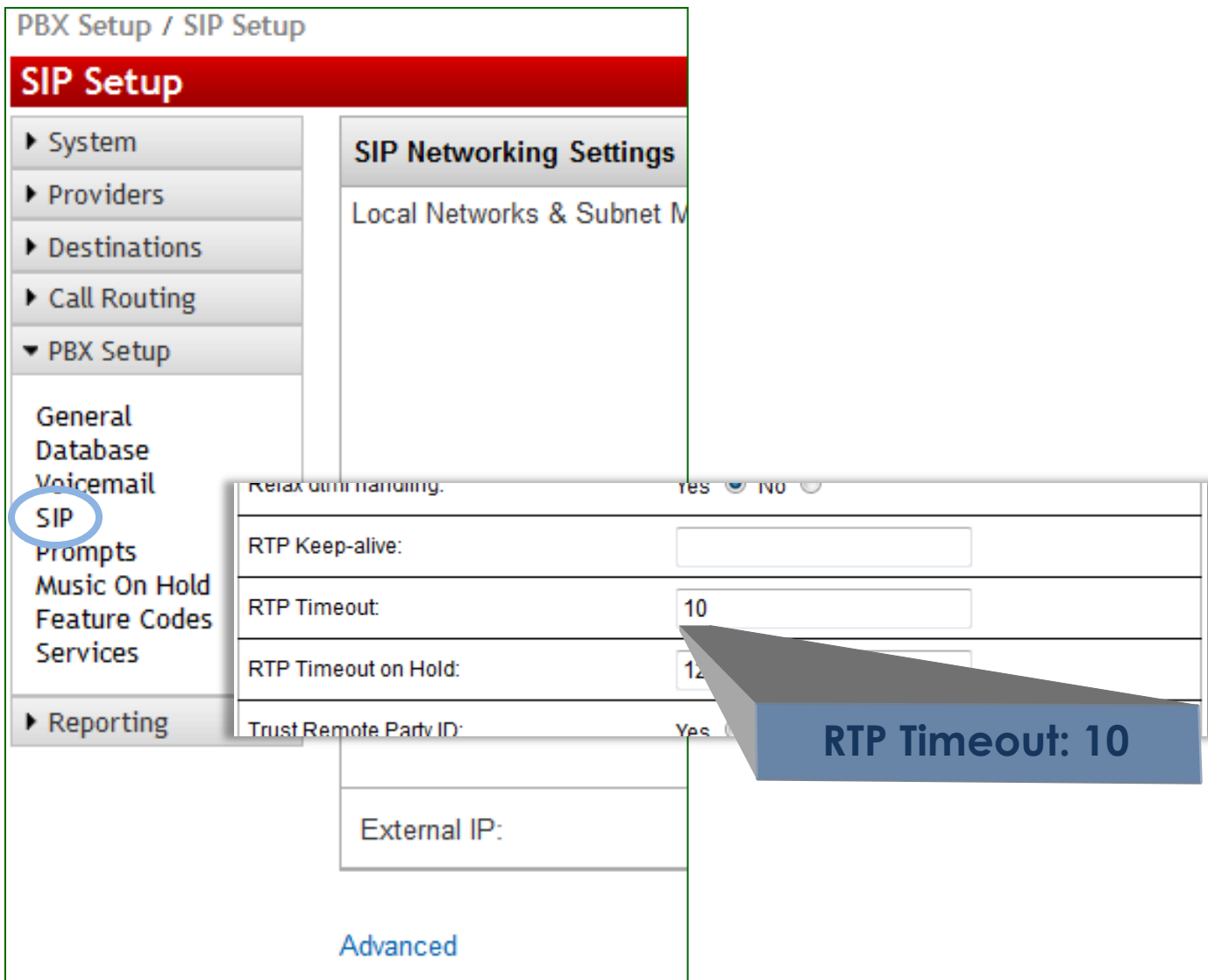
Save Changes

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

Allow Outbound Caller to transfer:	<input type="checkbox"/>	Note: "Insecure" is a Protocol matching parameter and has nothing to do with this carriers security. (Set it to "Very")									
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 checked="" type="radio"/> Yes <input type="radio"/> No <input type="radio"/> N/A										
Insecure:	<input type="text" value="Very"/>										
Allow Codecs: <table style="width: 100%; border: none;"> <tr> <td style="text-align: center; border: none;">Disabled</td> <td style="text-align: center; border: none;">Enabled</td> <td style="border: none;"></td> </tr> <tr> <td style="border: none;"> <input type="text" value="G.711 (alaw)"/> <input type="text" value="G.723.1"/> <input type="text" value="G.726"/> <input type="text" value="G.722"/> <input type="text" value="GSM"/> </td> <td style="border: none;"> <input type="text" value="G.729"/> <input type="text" value="G.711 (ulaw)"/> </td> <td style="border: none; text-align: center;"> <input type="button" value="Up"/> <input type="button" value="Down"/> </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> <td style="border: none;"></td> </tr> </table>			Disabled	Enabled		<input type="text" value="G.711 (alaw)"/> <input type="text" value="G.723.1"/> <input type="text" value="G.726"/> <input type="text" value="G.722"/> <input type="text" value="GSM"/>	<input type="text" value="G.729"/> <input type="text" value="G.711 (ulaw)"/>	<input type="button" value="Up"/> <input type="button" value="Down"/>	<input type="button" value="Add"/>	<input type="button" value="Delete"/>	
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<table style="width: 100%; border: none;"> <tr> <td style="width: 40%; border: 1px solid #ccc; height: 40px;"></td> <td style="width: 20%; border: 1px solid #ccc; padding: 2px;">2065172793</td> <td style="width: 40%; border: none;"></td> </tr> <tr> <td style="border: none;"><input type="button" value="Add"/></td> <td style="border: none;"></td> <td style="border: none; text-align: right;"><input type="button" value="Remove"/></td> </tr> </table>				2065172793		<input type="button" value="Add"/>		<input type="button" value="Remove"/>			
	2065172793										
<input type="button" value="Add"/>		<input type="button" value="Remove"/>									
Destination: <input type="text" value="Extension: 5555"/>		<input type="button" value="Set"/>									

Procedure—SIP (Global)

1. Navigate to 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. At the time of our testing the RTP Timeout was set to 10 seconds as a precaution to disconnect inactive calls with no voice traffic during a 10-second period. You may wish to extend this period to ignore longer periods of “silence”.
3. Here again, when changes are made you must click  to store and changes made on this page before continuing on to other programming. 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 0thn handling. Yes NO

RTP Keep-alive:

RTP Timeout: 10

RTP Timeout on Hold: 12

Trust Remote Party ID: Yes

External IP:

Advanced

RTP Timeout: 10

Procedure—Call Routing-Outgoing

1. Navigate to the Call Routing/Outgoing page. (This is where digits-dialed are associated to trunks and how the digits dialed will be handled on the selected trunk).
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 10-digit numbers (Area Code +). In this example Exact Length is set to "Yes". We have added Outgoing Routes for each of the calling scenarios:
 - 1+
 - 7-Digit
 - 10-Digit
 - Emergency
 - International

4. Refer to the IPitomy 1100+ Manual on the IPitomy WIKI for details on configuring Outgoing Routes. http://wiki.ipitomy.com/index.php/IP_PBX_Manual_Call_Routing#Outgoing_Call_Routing
5. Notice that the added SIP Trunk (Provider) is now available for selection in the drop-down list.

6. Select your added Provider (AccessLine) and clicked **Add**

7. You must also click **Save Changes** before other changes to this trunk can be applied to the routing characteristics.

If multiple trunks are available for use in the dialing pattern defined here (10-digit dialing), they may be added to the list.

When there are multiple trunks use the **Up** and

Dn to position the trunks as desired for first to last choices.

8. When you're done, click

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

Procedure—Class Of Service

1. Navigate to the Call Routing/Class Of Service page. (This is where trunks are assigned as those allowed to be used by extensions – by way of that extension's assigned class 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 "Paul Test" is shown. Notice that the newly created Outbound Route "AccessLine" is listed here and selected from the drop-down list.
3. Click **Add** to add this route to this COS. Notice that each of the other Routes that we defined for AccessLine have been programmed already when the screenshot was taken. (They are already listed for use in this Class Of Service.)

Note:

This page does not have a **Save Changes** button.

Procedure—Finalize

1. When all changes are complete you MUST click **Apply Changes** 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 AccessLine Trunks are now functional!!