PSUM 2.0 Configuration of Sip Trunks

2.5.1 The background of SIP Trunk

The SIP Trunks serve connecting PrivateServer to a local PBX or to a SIP Voice provider. The above statement implies that a Trunk can be used both for receiving and placing calls in a bidirectional way.

> By Secure Call design the SIP Trunks are available only for the **PrivateGSM Enterprise Edition**. **PrivateGSM Professional Edition** can't place calls on a Trunk.

The PrivateServer is designed to perform always as a client when configuring a SIP Trunk. This means that PrivateServer itself can not act as a Trunk Provider.

Even if a SIP Trunk can receive and place calls PrivateServer has to be configured to act in a bidirectional way. The Trunk setup should be done for the incoming calls first and after this function is enabled you can setup it for outgoing calls as well.

The incoming calls Trunks are configured in the **Inbound** entry that you access under **Main Menu -> PBX INTEGRATION**. Outgoing calls are configured into the **Outbound** entry that is under the Inbound one.

2.5.2 Inbound Configuration

Click on the "Inbound" link in menu on the left side to go to the "Sip Trunk List" page.

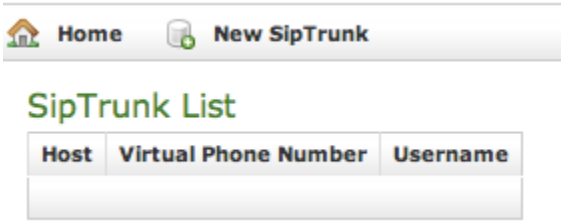

Figure 1. The "Sip Trunk List" page

Figure 1. The "Sip Trunk List" page illustrates this page, with an empty table and a "New Sip Trunk" icon above it. This is the default empty configuration.

2.5.2.1 Create an Inbound Trunk

Click on the **New Sip Trunk** to go to the Edit Sip Trunk page:

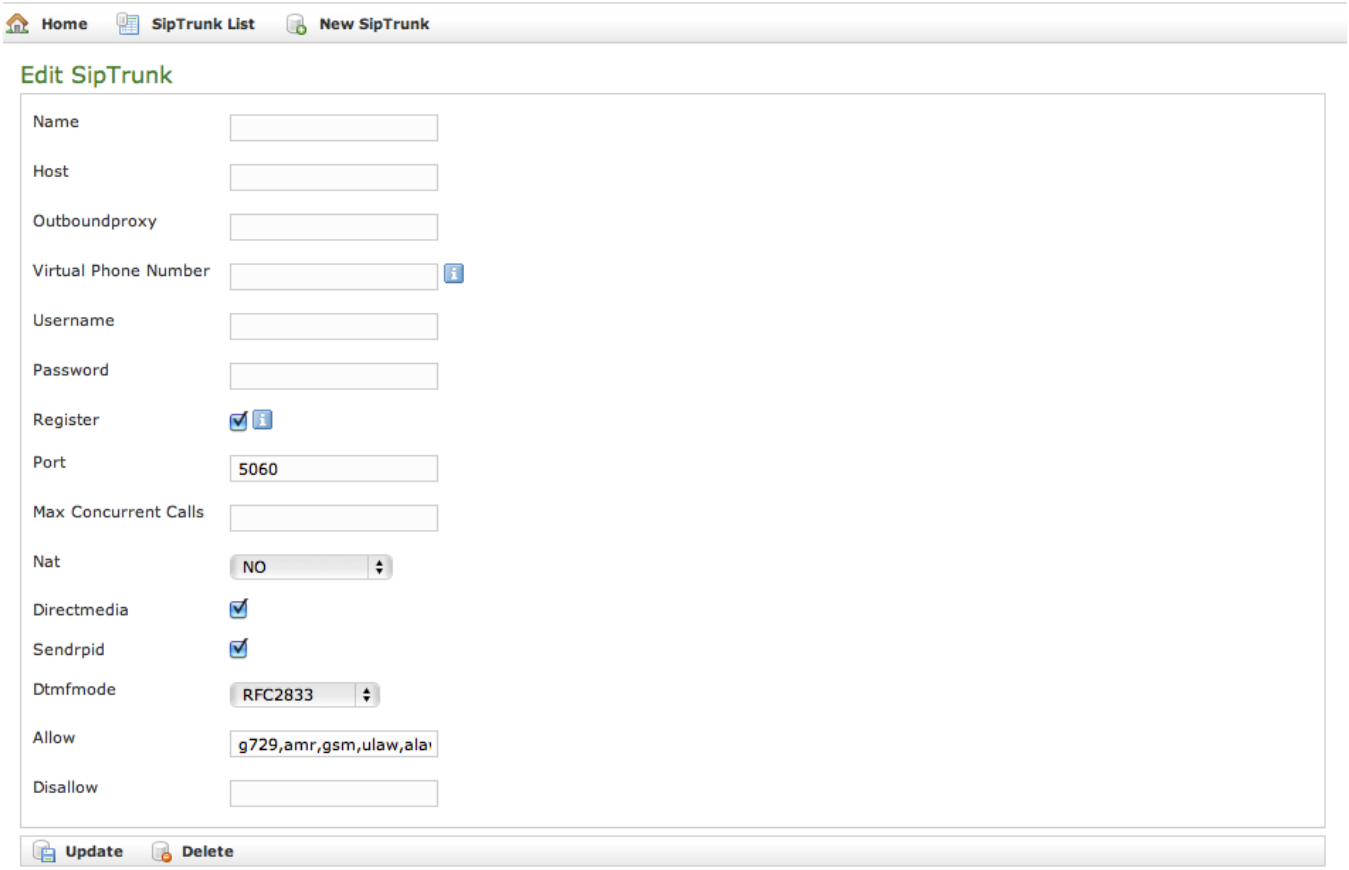

Figure 2. "Edit Sip Trunk" form

> There are two types of trunk that you can create, depending on the authentication type:
>
> 1. username and password: you must insert username and password
> 2. IP based: leave the username and password fields blank and fill in the HOST parameter.

The fields have to be configured as follows:

- **NAME:** a meaningful name for this trunk
- **HOST**: IP address of the SIP server provided by ITSP
- **OUTBOUND PROXY**: IP address of the outbound proxy server provided by ITSP
- **VIRTUAL PHONE NUMBER**: associated virtual number
- **USERNAME**: login username
- **PASSWORD**: password for the username entered above
- **PORT**: registration port of the service
- **REGISTER**: select if you want to have only outgoing calls or incoming (see below for deeper explanation)
- **CONCURRENT CALLS**: maximum number of calls on a single trunk
- **NAT**: enable/disable the Network Address Translation configuration
- **DIRECTMEDIA**: experimental sound management in P2P mode without routing the audio stream through the server
- **SENDRPID**: the Remote Party ID, used to interconnect with VoIP Provider for the management of privacy
- **DTMFMODE**: tone signaling technology used for this SIP interconnection
- **ALLOW**: voice codecs used in this trunk
- **DISALLOW**: voice codecs denied in this trunk

> The called number in inbound calls passing through the trunk should match the number of an internal account or a virtual phone number associated with an account. This way the correct SIP phone will ring.

The default values for **Port**, **Allow** and **Disallow** are usually work well, however feel free to change them to suite your needs better.

Mandatory fields are:

- Host

- Port
- Allow/Disallow

Register option

If the trunk is going to be also used for OUTBOUND calls (external calls) you need to be sure that the "Register" checkbox is checked. In this case the "Name" and "Secret" fields are mandatory as well. Otherwise you can uncheck the "Register" checkbox so the above fields will become optional.

Fill out the fields accordingly to the guidelines above and when you are done click on the "Update" icon at the bottom of the page.

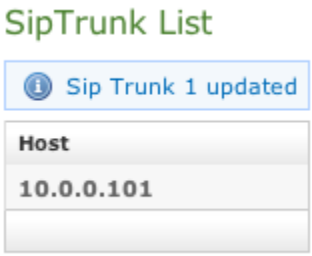

Figure 3. The new SipTrunk List Table

You can get back to the "Sip Trunk List" page where the Sip Trunk table is now filled based on your settings (Figure 3. The new SipTrunk List Table).

A notification will inform you about the progress of the operation. The number shown in the notification is an internal identifier in the database and thus can be ignored.
To check the entered data of the SIP Trunk click on the "Host" value into the table (for example in this case you should click on "10.0.0.101"). This will open again the **Edit Sip Trunk** page with the previously entered data.

2.5.2.2 Change Inbound Trunk settings

On the **Edit Sip Trunk** page you can change the values of the Inbound Trunk fields. When you are done commit the changes by clicking on the **Update** icon.
Click again on the "Host" value and verify the values of the fields are actually changed.

2.5.2.3 Delete an Inbound Trunk

To delete a trunk you need to select it in the trunk list. Click on the selected trunk and you'll be taken to the **Edit Sip Trunk** page. Click on the **Delete** icon at the bottom of the page to delete the Sip Trunk.

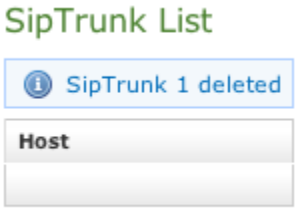

Figure 4. The Sip Trunk List after the Trunk deletion

A warning pop-up window will ask you for confirmation. After you click on the **Ok** button or press the Enter key on your keyboard you'll be taken back again to the "Sip Trunk List" page which will show an empty list and a notice as in Figure 4. The Sip Trunk List after the Trunk deletion.