

2. Configuring PrivateServer

The CUCMBE is ready to relate with PrivateServer using the SIP Trunk, but still it can't until we create the same configuration on the PrivateServer itself.





2.1 Prerequisites



If you came here then all the necessary prerequisites are matched.

2.2 PrivateServer configuration

You can refer to PrivateServer management documentation for creating a SIP Trunk. We are going to explain here which specific values need to be set up in order to create it properly.

Edit Sip Trunk

Name	<input type="text" value="CUCUMBE"/>
Host	<input type="text" value="192.168.11.11"/>
Outboundproxy	<input type="text" value="192.168.11.11"/>
Virtual Phone Number	<input type="text" value="2222"/> 
Username	<input type="text"/>
Password	<input type="password"/>
Register	<input type="checkbox"/> 
Port	<input type="text" value="5060"/>
Max Concurrent Calls	<input type="text" value="6"/>
Nat	<input type="text" value="NO"/> 
Directmedia	<input checked="" type="checkbox"/>
Sendrpid	<input checked="" type="checkbox"/>
Dtmfmode	<input type="text" value="RFC2833"/> 
Allow	<input type="text" value="ulaw"/>
Disallow	<input type="text" value="all"/>

 **Update**  **Delete**

In the fig. X you can watch a plain example of a working configuration. It's major differences from a standard one is that:

- *check out* the **Register** option
- **Dtmfmode** value must be *RFC2833*
- **Allow** codec is *ulaw* only
- **Disallow** codec must be *all*

The **Virtual phone number** is optional and can be left blank.

After you're done with the **Inbound** you have to check that the **Outbound** section has either a dialing rule set to let the trunk be used or the trunk set as the default one.

