

## 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.

