

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	CUCUMBE
Host	192.168.11.11
Outboundproxy	192.168.11.11
Virtual Phone Number	2222 
Username	
Password	
Register	☐ 
Port	5060
Max Concurrent Calls	6
Nat	NO 
Directmedia	☒
Sendrpid	☒
Dtmfmode	RFC2833 
Allow	ulaw
Disallow	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.

