***SiSky Enterprise Edition generate up to 16 Skype Trunks for Asterisk!***

**Method 1:** Config Asterisk for SiSky[EE] (For Example: **Asterisk without freePBX**)
→ http://www.yeastar.com/download/AsteriskConfig2.pdf

**Method 2:** Config Asterisk for SiSky[EE] (For Example: **Asterisk+freePBX or Trixbox**)

Following instructions are based on the example of 16 channels, SIP account from 501 to 516 for SiSky's SIP Endpoints.

**Config Asterisk for SiSky (For Example: Asterisk+freePBX or Trixbox)**

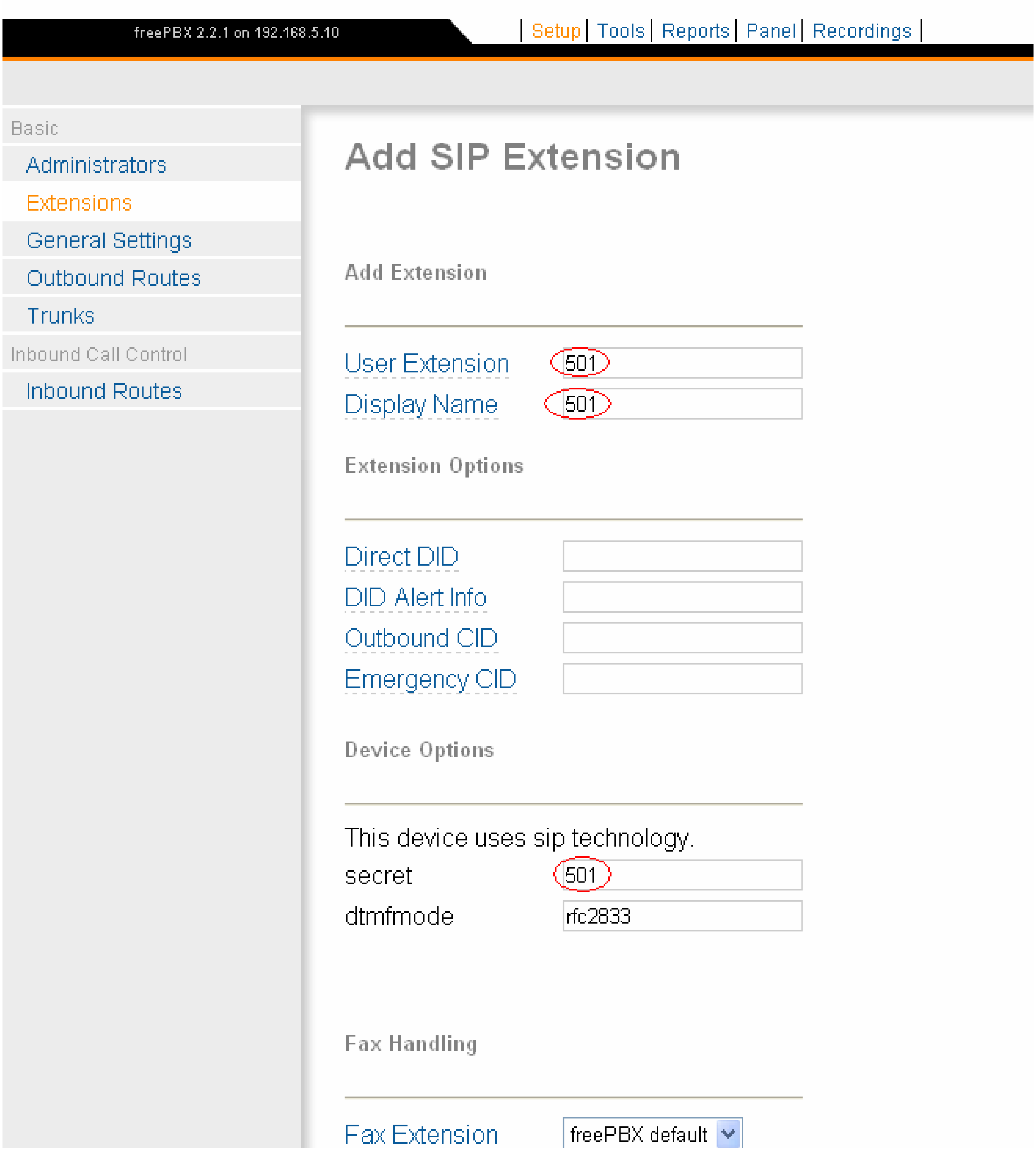
**1. In extensions_custom.conf, add below lines into the [from-internal-custom] context:**

```
exten => _500.,1,Dial(SIP/${EXTEN:0}@501)
exten => _501.,1,Dial(SIP/${EXTEN:0}@501)
exten => _502.,1,Dial(SIP/${EXTEN:0}@502)
exten => _503.,1,Dial(SIP/${EXTEN:0}@503)
exten => _504.,1,Dial(SIP/${EXTEN:0}@504)
exten => _505.,1,Dial(SIP/${EXTEN:0}@505)
exten => _506.,1,Dial(SIP/${EXTEN:0}@506)
exten => _507.,1,Dial(SIP/${EXTEN:0}@507)
exten => _508.,1,Dial(SIP/${EXTEN:0}@508)
exten => _509.,1,Dial(SIP/${EXTEN:0}@509)
exten => _510.,1,Dial(SIP/${EXTEN:0}@510)
exten => _511.,1,Dial(SIP/${EXTEN:0}@511)
exten => _512.,1,Dial(SIP/${EXTEN:0}@512)
exten => _513.,1,Dial(SIP/${EXTEN:0}@513)
exten => _514.,1,Dial(SIP/${EXTEN:0}@514)
exten => _515.,1,Dial(SIP/${EXTEN:0}@515)
exten => _516.,1,Dial(SIP/${EXTEN:0}@516)
```

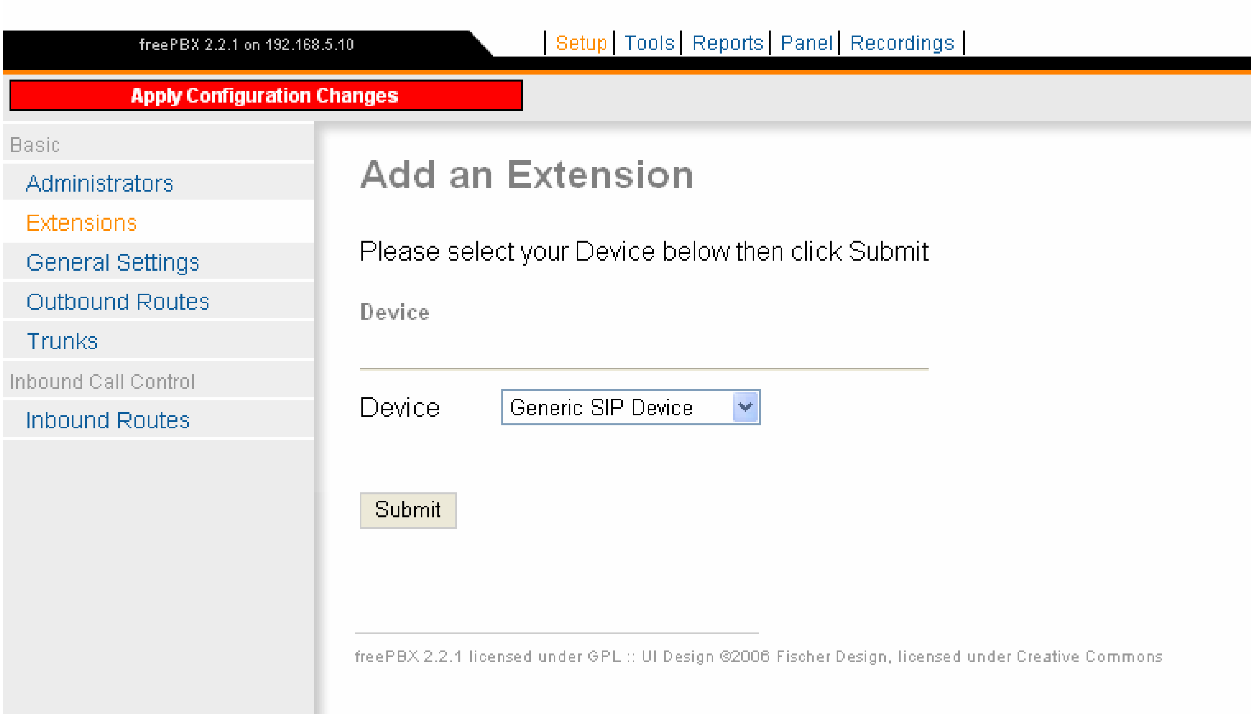
Notes:
exten => _500.,1,Dial (SIP/${EXTEN:0}@501)
---that means automatically finding idle Skype trunk from 1 to 16 to dial out.
exten => _501.,1,Dial(SIP/${EXTEN:0}@501)
---that means dial out through Skype trunk 1.

**2. Set up SIP Extension in freePBX.**

Set extension 501 as below:
"User Extension":501
"Display Name":501
"Secret":501
"dtmfmode":rfc2833




(3) Apply Configuration Changes or restart Asterisk.




Set other extensions as same steps as setting extension 501.

**Now you have successfully finished configuring Asterisk for SiSky Enterprise Edition. Next step you should Config SiSky's SIP Endpoints and Skype trunks.**