



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

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Basic

- [Administrators](#)
- [Extensions](#)**
- [General Settings](#)
- [Outbound Routes](#)
- [Trunks](#)

Inbound Call Control

- [Inbound Routes](#)

Add SIP Extension

Add Extension

User Extension

Display Name

Extension Options

Direct DID

DID Alert Info

Outbound CID

Emergency CID

Device Options

This device uses sip technology.

secret

dtmfmode

Fax Handling

Fax Extension

(3) Apply Configuration Changes or restart Asterisk.

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Apply Configuration Changes

Basic

- [Administrators](#)
- [Extensions](#)**
- [General Settings](#)
- [Outbound Routes](#)
- [Trunks](#)

Inbound Call Control

- [Inbound Routes](#)

Add an Extension

Please select your Device below then click Submit

Device

Device

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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.