

Quickstart Guide



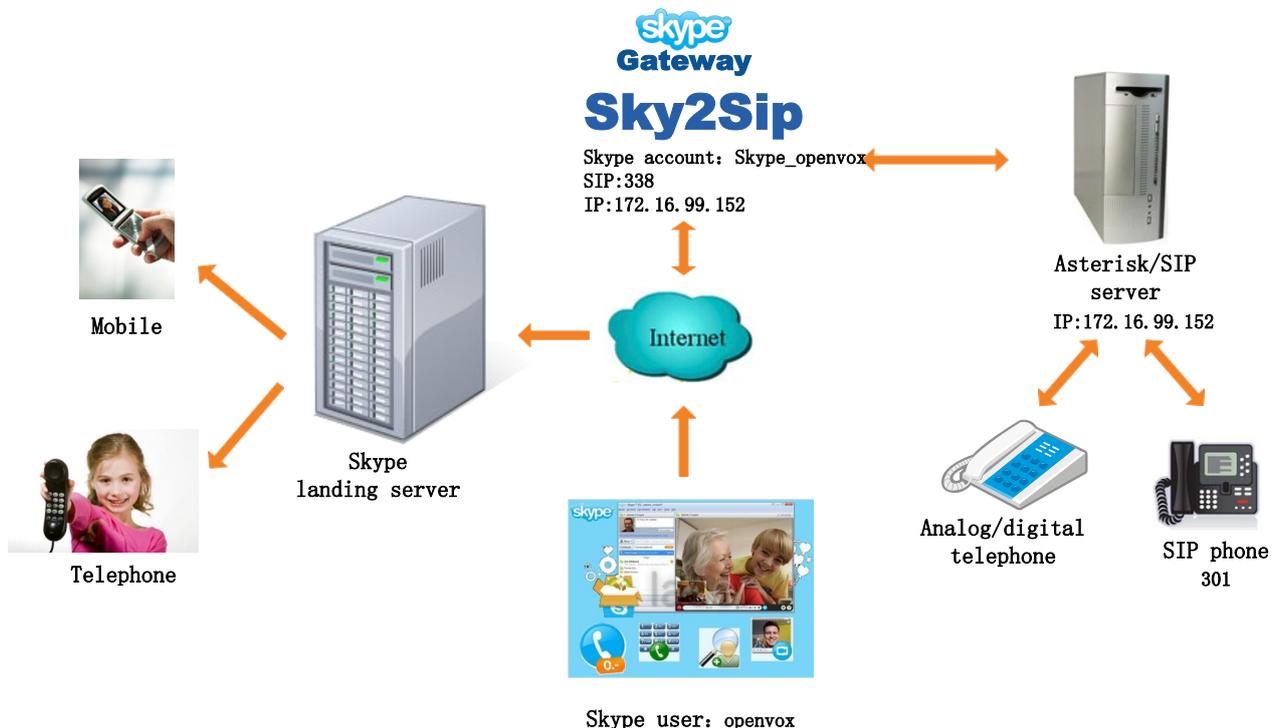
Ladies and gentlemen, let me guide you through the Skype gateway quickstart of

Sky2Sip

connect with

Asterisk®

First of all, thank you for choosing OpenVox Skype gateway Sky2Sip, we will make our best efforts for more creative products. Now please follow me to know how to install and set Sky2Sip which connect with Asterisk® server.



Like the above figure, the software package Sky2Sip can be installed in the Asterisk® which act as a SIP server or another server. The Sky2Sip server must be X86 platform, 32 bit or 64 bit are both compatible, and can maximum bear 32 concurrent calls.

Calling out flow

Local SIP phone 301 → Sky2Sip transfer SIP signaling to Skype → Skype landing server → Mobile/Telphone user; **(Direct dialing mode)**

Local SIP user 301 → SIP server → SIP extension 338 in Sky2Sip → Sky2Sip transfer SIP signaling to Skype → Skype landing server → Mobile/Telphone user; **(Secondary Dialing Mode)**

Calling in flow

Skype remote user → Sky2Sip transfer Skype to SIP → SIP server → Local SIP phone; **(Direct Dialing Mode)**

Skype remote user → Skype account gateway login → Sky2Sip transfer Skype to SIP → SIP server → SIP phone; **(Secondary Dialing Mode)**

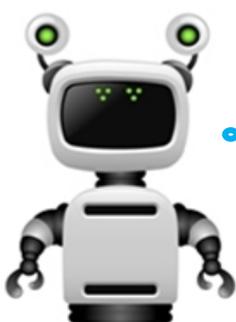
Installation

Run the following commands in your Linux CLI :

```
tar -zxvf SkypetoSipGw.tar.gz
chmod 777 install.sh
./install.sh
```

Certify

For example, if Sky2Sip is installed in the server whose IP is 172.16.99.152, please enter http://172.16.99.152:8888 in your IE browser to login web, default username/password is admin/admin, choose button to save your license. License keys also can be uploaded.



Step 3

SIP Settings

1. Select the right IP address for Skype communication from the drop-down list since there are a few IP detected. Assign a port for “Skype Gateway SIP Port” , 2000~65535 is available except those have be used such as

Step 1



Step 2



5060 for SIP port. Click will display detail help information.

SKYPE GATEWAY IP

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Skype Gateway IP Address:

Skype Gateway SIP Port:

2. Outbound Setting

- Direct Dial Mode

If you choose “Direct Dial Mode” for your “Outbound Dial Mode”, please press prefix+ (according to your dial rules) + country code + destination number when make calls. At the same time, “Prompt Tone Language” is not available. If Sky2Sip is installed in the SIP server, in another word, the Skype gateway server is the same server with SIP server, then “SIP server IP Address” is the same as “Skype Gateway IP Address”. The default setting for “SIP Server Port” is 5060, and there is no need to change it in general.

Registered SIP: This option enables Skype Gateway to register SIP accounts to SIP server.

User ID: The SIP ID that Skype gateway registered.

Password: SIP ID’s password that the gateway registered.

Display Name: Display name that gateway’s SIP ID.

Authorization Name: Name that SIP server authorizes SIP account.

Peer IP: The IP address(es) of SIP phone(s) allowed to make outbound call through the Skype gateway. Up to 32 IP addresses can be registered.

After finish enter username, password display name and peer IP, please register a SIP extension in your SIP server. If Asterisk has been installed in your SIP server, your can also write in /etc/asterisk/sip.conf like that :

```
[338]
type=friend
user=338
secret=338
host=dynamic
context=from-internal
allow=all
canreinvite=no

[from-skype]
type=friend
port=5600
Host=172.16.99.152
insecure=invite
context=from-pstn
```

OUTBOUND SETTING

Outbound Dial Mode: Direct Dial Mode

Prompt Tone Language: English

DTMF Mode: INFO

DTMF Generate: 100 ms

Audio Codec: U-law

SIP Server IP Address: 172.16.99.152

SIP Server Port: 5060

Registered SIP:

Registered Interval(s): 600

User ID: from-skype

Password:

Display Name: from-skype

Authorization Name:

Status: Unregistered

Index	Access IP	Delete
1	172.16.8.110	Delete
2	172.16.8.186	Delete

Add

The figure registered a SIP extension 338, and SIP extensions in 172.16.8.186, 172.16.8.110 these two SIP server are allowed to make and answer calls.

- Secondary Dial Mode

If you choose “Secondary Dial Mode” for your “Outbound Dial Mode”, Chinese and English Prompt Tone Language are available. When make calls, please dial SIP account registered in the SIP server firstly, then you will hear a piece of prompt tone, after that, dial destination country code and number. Other options is the same with direct dial mode.

3. Inbound Setting

- Direct Dial Mode

If you choose “Direct Dial Mode” for your “Inbound Dial Mode”, remote Skype user Dial Skype gateway’s account directly, and then the call will be get through to the assigned “Inbound Direct Dial number”. The following figure means when remote Skype user calls Skype gateway account, Sky2Sip will transfer the call to registered SIP extension 301.

INBOUND SETTING

Inbound Dial Mode: Direct Dial Mode

Inbound Direct Dial Num: 301

- Secondary Dial Mode

If you choose “Secondary Dial Mode” for your “Inbound Dial Mode”, the incoming call will connect to Sky2Sip’s account firstly, then the gateway plays a piece of prompt tone, after that, remote Skype user dials destination extension end with “#”.

Skype Settings

1. Skype Configure

Skype Account: The Skype account bound to Skype gateway, which will be logged in the Skype network.

Skype Password: The password of the account logged in the Skype network.

Skype Message: When gateway receives text from Skype user, the message will be replied to the remote sender automatically based on this gateway doesn’t support text conversation.



SKYPE CONFIG

Skype Account:	<input type="text" value="skype_openvox"/>
Skype Password:	<input type="password" value="••••••••"/>
Auto Replied Message:	<div style="background-color: #333; color: #eee; padding: 5px; font-family: monospace; font-size: 0.9em;"> This Skype version does not support text chat, please call directly. </div>

2. Outbound Route

This enables SIP extension calls out to Skype accounts. For example, the following Figure means when SIP phone dials 123, the call will be connect to the Skype account openvox if the Skype gateway’s account has added openvox as a contact.

OUTBOUND ROUTE

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Enable SIP To Skype:

Index	Call Num	SkypeID	Delete
1	123	openvox	<input type="button" value="Delete"/>

Dial Rules

Please add dial rules in your SIP server like that:

```
[from-pstn]
exten => _X.,1,Answer()
exten => _X.,n,Dial(sip/${EXTEN},,r)
exten => _X.,n,Hangup()

[from-internal]
exten => _9.,1,Dial(SIP/from-skype/${EXTEN:1})
exten => _9.,n,Hangup()
```

After all these steps, I would say “congratulations” that installation is completed.