

# Quickstart Guide



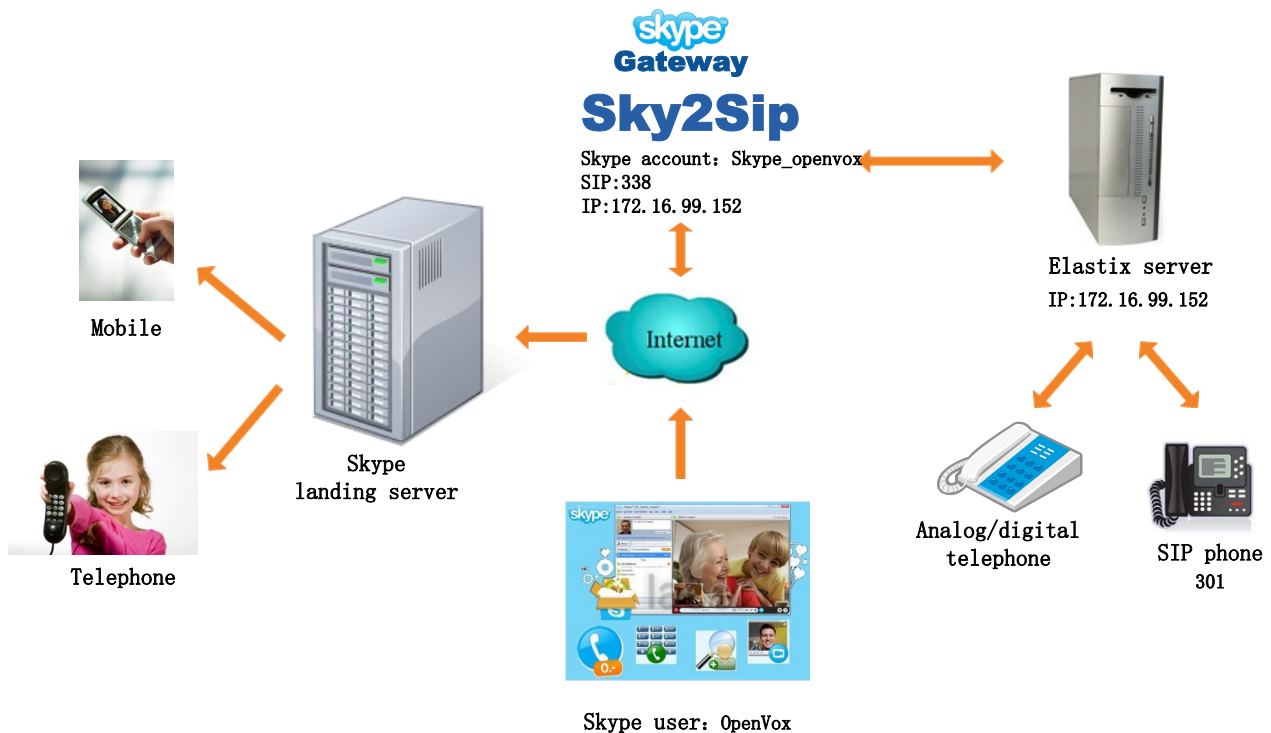
Ladies and gentlemen, let me guide you through the Skype gateway quickstart of

## Sky2Sip

Connect with

## Elastix®

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 connect with Elastix® server.



Like the above figure, the software package Sky2Sip can be installed in the SIP server or another server. The Sky2Sip server must be X86 platform until now, 32 bit and 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.



### 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 been used such as 5060 is for SIP port. Click ? will display detail help information.

SKYPE GATEWAY IP	
Skype Gateway IP Address:	172.16.99.152
Skype Gateway SIP Port:	5600

## 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.

OUTBOUND SETTING	
Outbound Dial Mode:	Direct Dial Mode
Prompt Tone Language:	中文
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:	<input checked="" type="checkbox"/>
Registered Interval(s):	600
User ID:	119
Password:	
Display Name:	119
Authorization Name:	

After Sky2Sip web configuration, please turn to Elastix web to Configure like the following to add a SIP trunk:

The screenshot shows the 'PBX Configuration' interface. On the left, a sidebar menu has 'Trunks' highlighted. The main area is titled 'Add a Trunk' and contains three options: 'Add SIP Trunk' (highlighted with a red box), 'Add DAHDI Trunk', and 'Add Zap Trunk (DAHDI compatibility mode)'. Below this is the 'General Settings' section for a trunk named 'skype-119'. The 'Trunk Name' field is highlighted with a red box. Other fields include 'Outbound Caller ID', 'CID Options' (set to 'Allow Any CID'), 'Maximum Channels', 'Disable Trunk' (checkbox), and 'Monitor Trunk Failures' (checkbox). The 'Dialed Number Manipulation Rules' section includes fields for '(prepend)', '+ prefix', and 'match pattern', along with buttons for '+ Add More Dial Pattern Fields' and 'Clear all Fields'. Below this is a 'Dial Rules Wizards' dropdown and an 'Outbound Dial Prefix' field. The 'Outgoing Settings' section shows 'Trunk Name' set to '119' and a 'PEER Details' text area containing: `host=172.16.8.119`, `username=119`, `port=5600`, `type=friend`, `insecure=invite`, and `context=from-pstn`. The 'PEER Details' area is also highlighted with a red box.

### Outbound routes setting in Elastix

The screenshot shows the 'Add Route' configuration page. The 'Route Name' field is highlighted with a red box and contains the value '9\_outside'. Other fields include 'Route CID', 'Route Password', 'Route Type' (with checkboxes for 'Emergency' and 'Intra-Company'), 'Music On Hold?' (set to 'default'), and 'Time Group' (set to '---Permanent Route---').

Route Position: Last after 9\_outside

Additional Settings

PIN Set: None

**Dial Patterns that will use this Route**

(prepend ) + 9 | [. / CallerId

(prepend ) + prefix | [match pattern / CallerId

+ Add More Dial Pattern Fields

Dial patterns wizards: (pick one)

**Trunk Sequence for Matched Routes**

0 skype-119

1

2

Submit Changes

Create a SIP extension in Elastix web:

Basic

**Extensions**

Feature Codes

General Settings

Outbound Routes

Trunks

Inbound Call Control

Inbound Routes

Zap Channel DIDs

Announcements

Blacklist

CallerID Lookup Sources

Day/Night Control

### Add an Extension

Please select your Device below then click Submit

Device

Device: Generic SIP Device

- Generic SIP Device
- Generic IAX2 Device
- Generic ZAP Device
- Generic DAHDI Device
- Other (Custom) Device
- None (virtual exten)

Submit

### Edit Extension

Display Name: 301

CID Num Alias

SIP Alias

**Device Options**

This device uses sip technology.

secret: 301

dtmfmode: rfc2833

canreinvite: no

context: from-internal

After that, please save your changes and reboot the gateway.

- 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 number with country code and end with “#”. In Sky2Sip web, please set parameters:

OUTBOUND SETTING	
Outbound Dial Mode:	Secondary Dial Mode
Prompt Tone Language:	中文
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:	<input checked="" type="checkbox"/>
Registered Interval(s):	600
User ID:	338
Password:	●●●
Display Name:	338
Authorization Name:	●●●

In your Elastix web, please set likt that:

Add Extension	
User Extension	338
Display Name	338

This device uses sip technology.	
secret	338
dtmfmode	rfc2833

### 3. Inbound Setting

- Direct Dial Mode

If inbound is direct dial mode, remote Skype user call Sky2Sip's account directly, the call will transfer to the assigned destination number. The following figure means when remote Skype user call Sky2Sip's Skype account, the call will be transferred to 301.

INBOUND SETTING	
Inbound Dial Mode:	Direct Dial Mode
Inbound Direct Dial Num:	301
Display Skype Account:	<input checked="" type="checkbox"/>

- Secondary Dial Mode

If inbound is secondary dial mode, Remote Skype user calls Sky2Sip's account, after hear a piece of prompt tone, then dials 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:	skype_openvox
Skype Password:	●●●●●●●●
Auto Replied Message:	This Skype version does not support text chat, please call directly.

## 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 ⌵ ?

Enable SIP To Skype:

Index	Call Num	SkypeID	Delete
1	123	openvox	Delete

Address: F/3, Building No.127, Jindi Industrial Zone,  
Shazui Road, Futian District, Shenzhen, Guangdong 518048, China  
Tel:+86-755-82535461, 82535095, 82535362, Fax:+86-755-83823074  
Business Contact: [sales@openvox.com.cn](mailto:sales@openvox.com.cn)  
Technical Support: [support@openvox.com.cn](mailto:support@openvox.com.cn)

Business Hours: 09:00-18:00(GMT+8) from Monday to Friday  
URL: [www.openvox.cn](http://www.openvox.cn)