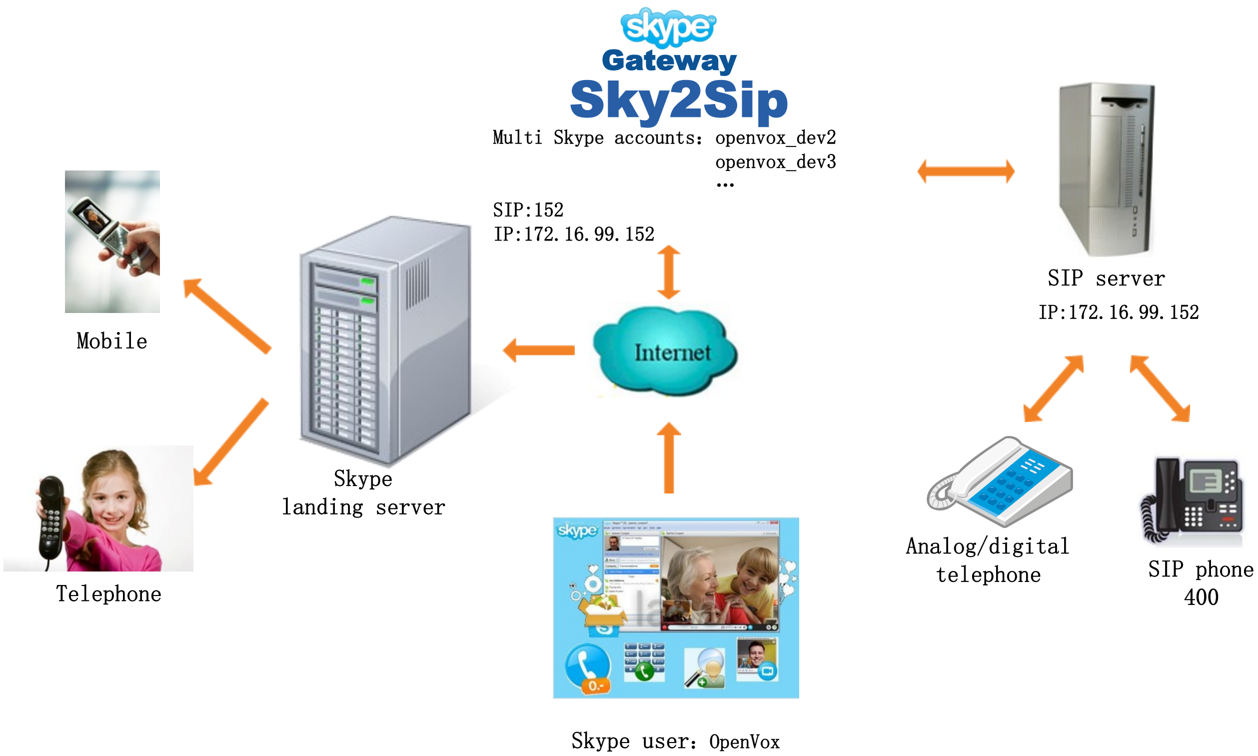
Quickstart Guide

Multi Sky2Sip

Connect with

Asterisk®

First of all, thank you for choosing OpenVox Skype gateway -Sky2Sip with multi accounts, we will make our best efforts for more creative products. Now please follow me to know how to install and set multi Sky2Sip 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 Linux X86 platform, 32 bit or 64 bit are both compatible, and can maximum bear 32 concurrent calls.

Dial-up Process

- **Calling out flow:**

1) Direct Dialing Mode

Local SIP phone 400 —> Sky2Sip transfer SIP signaling to Skype —> Skype landing server —> Mobile/Telphone user

Ex: 9 + country code + called number

2) Secondary Dialing Mode

Local SIP user 400 —> SIP extension 6001 in Sky2Sip —> Sky2Sip transfer SIP signaling to Skype —> Skype landing server —>Mobile/Telphone user

Ex: country code + called number

- **Calling in flow:**

1) Direct Dialing Mode

Skype remote user —> Sky2Sip transfer Skype to SIP —> SIP server —> Local SIP Phone

2) Secondary Dialing Mode

Skype remote user —> Skype account gateway login —> Sky2Sip transfer Skype to SIP—> SIP server —> SIP phone

I think you have known many things about Sky2sip through the above introduction. Now please follow me to set your Skype.

Skype configuration

Step One :

Installation

Run the following commands in your Linux CLI :(“**xxx**” means version number)

```
tar -zxvf SkypetoSipGwMulti+xxx.tar.gz
chmod 777 install.sh
./install.sh
```

If you want to unload your Sky2sip, run the following command in your Linux CLI:

```
./uninstall.sh
```

Please configured Trunks and dial out rules in the Asterisk command interface in /etc/asterisk/sip.conf as shown below.

```
[2311]
host=172.16.99.152
port=5600
type=friend
insecure=invite
context=from-pstn
fromuser=152

[152]
secret=152
dtmfmode=rfc2833
context=from-internal
host=dynamic
type=friend
nat=no
port=5060
allow=all

[6001]
secret=6001
dtmfmode=rfc2833
context=from-internal
host=dynamic
type=friend
nat=no
port=5060
allow=all
```

Then you can write your dial out rules in /etc/asterisk/extensions.conf like this:

```
[from-internal]
exten => _9.,1,Dial(SIP/2311/
${EXTEN:1})
exten => _9.,n,Hangup
exten => _3.,1,Dial(SIP/${EXTEN})
exten => _3.,n,Hangup
```

Step Two :

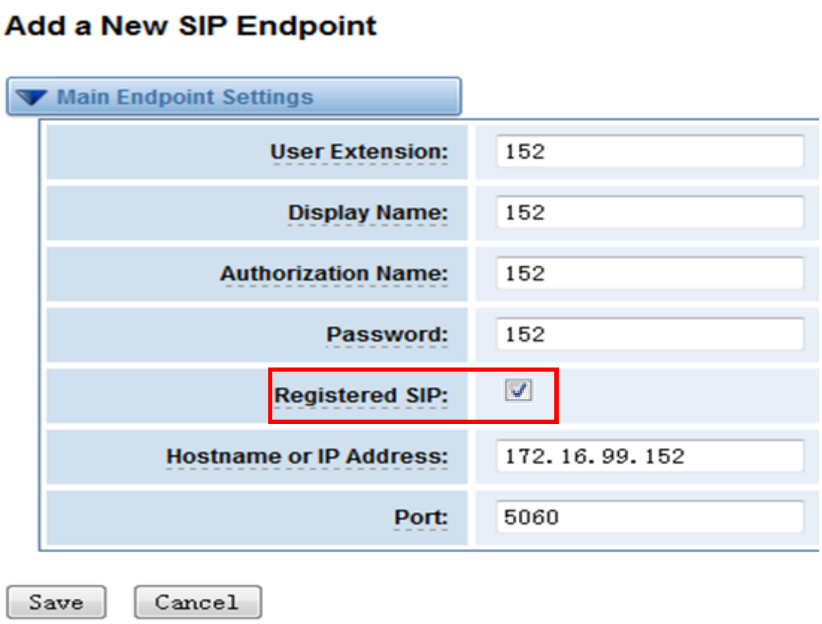
Log In

For example, if Sky2Sip is installed in the server Whose IP is **172.16.99.152**, please enter **http://172.16.99.152:9999** in your Firefox browser to login web, default username/ password is **admin/admin.**

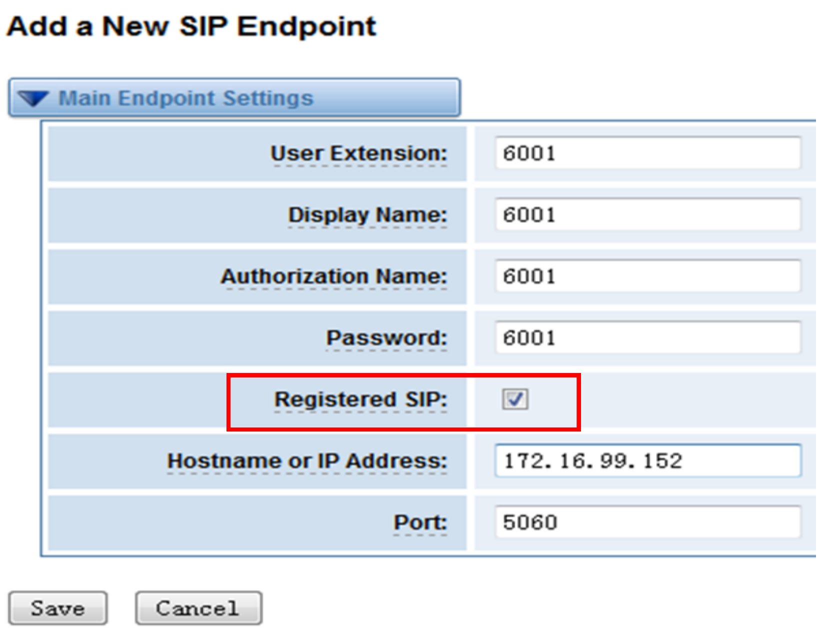
Step Three: Sip

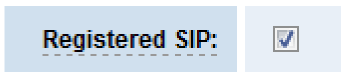
1.SIP Endpoints

Please press [Add New SIP Endpoint] then you will see following figure. We have added a sip trunk (for **Direct Dialing Mode**) named 152 in /etc/asterisk/sip.conf



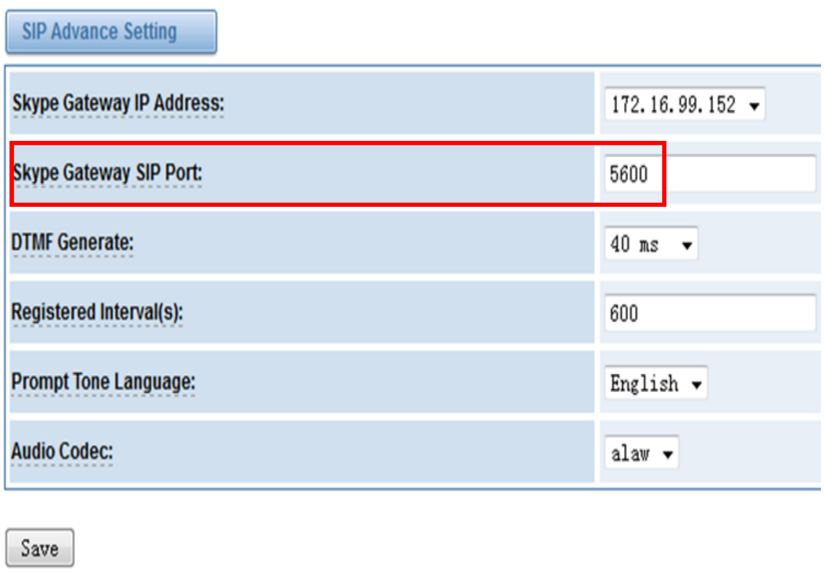
We have added an Extension 6001 in /etc/asterisk/extensions.conf for **Secondary Dialing Mode** . Then turn to Sky2sip webpage, go on installing like this:



Don't forget to chose 

2.Advanced Sip settings

You don't need to do any change except fill in the Skype Gateway SIP Port block with 5600.

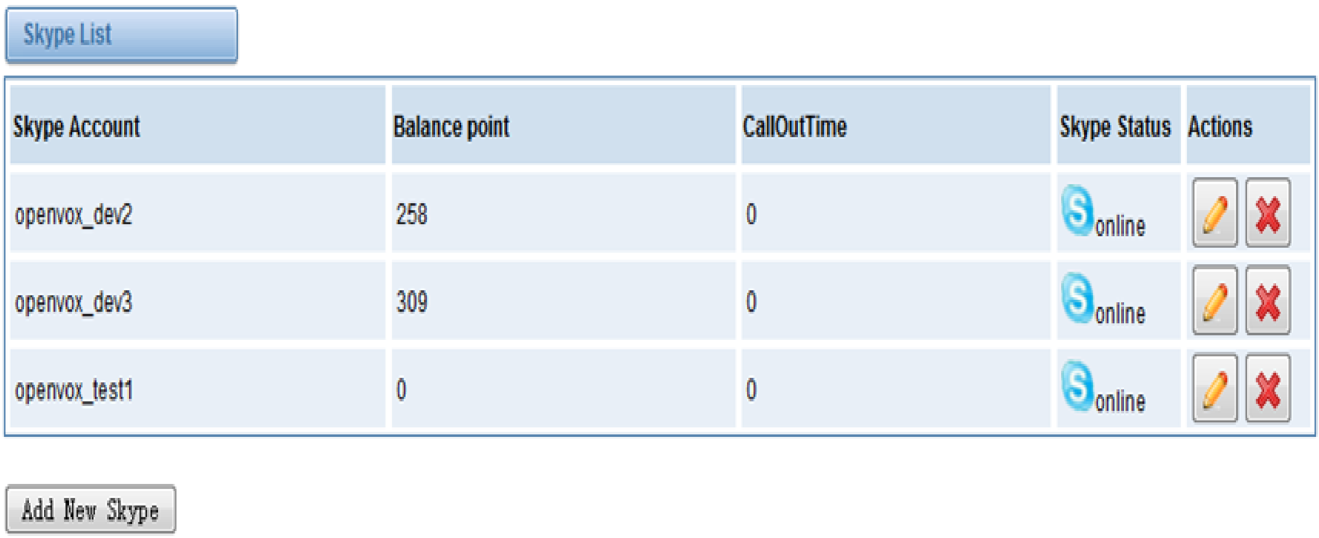


Step Four: Skype

1.Skype ID

You can add multi Skype accounts.

After saving, the Skype list like follows:

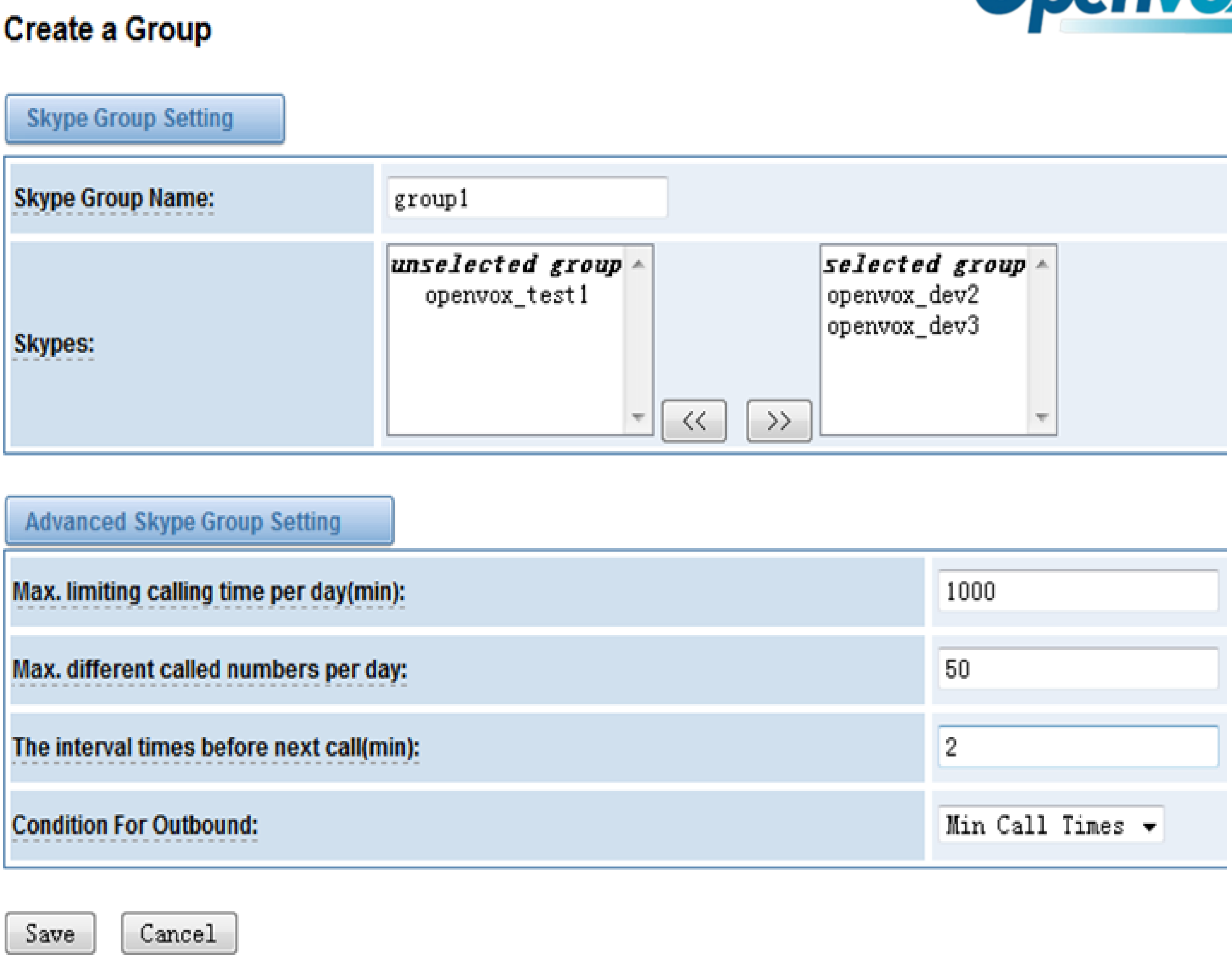

Skype List

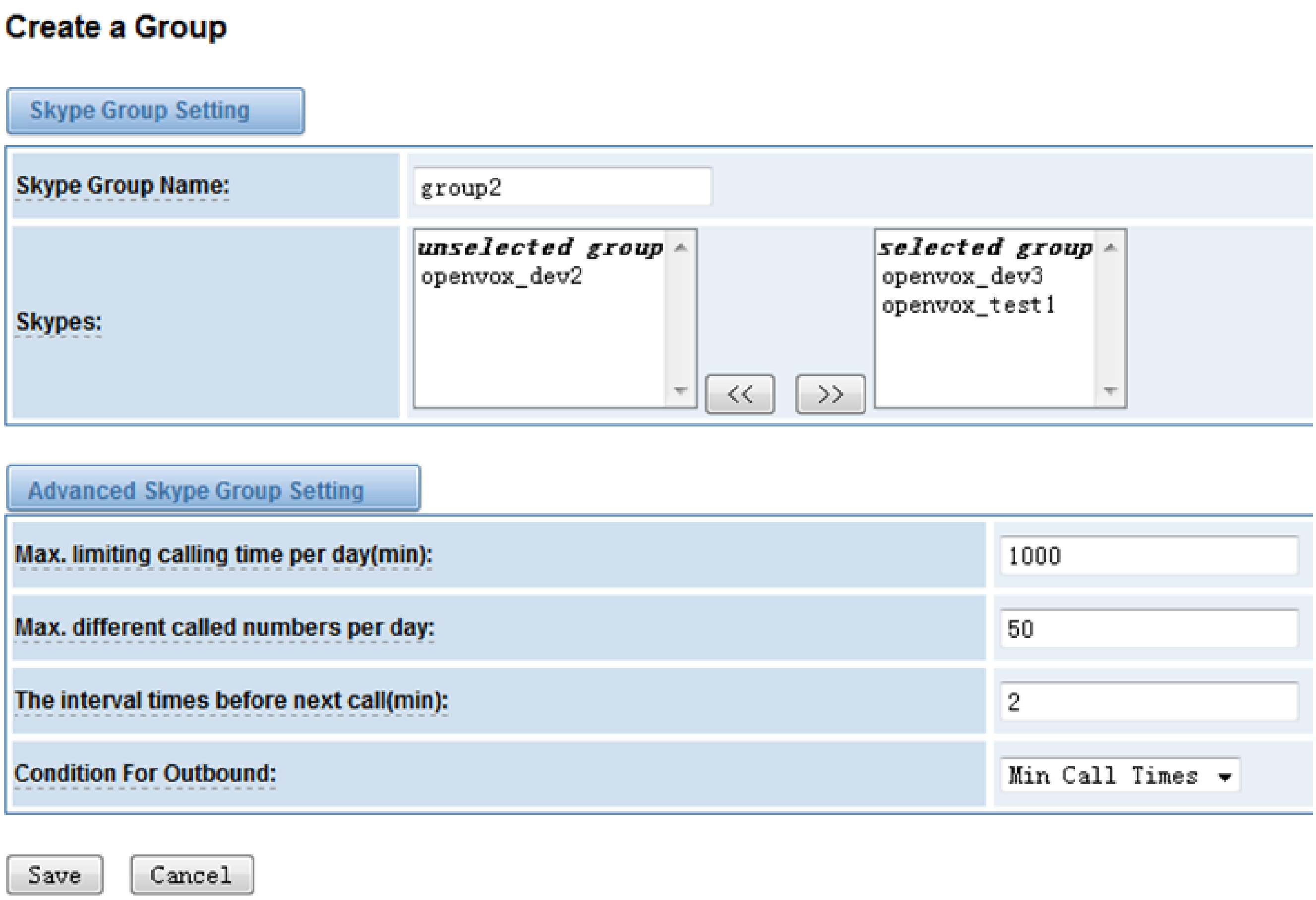
Skype Account	Balance point	CallOutTime	Skype Status	Actions
openvox_dev2	258	0	online	
openvox_dev3	309	0	online	
openvox_test1	0	0	online	

Add New Skype

2.Skype group settings

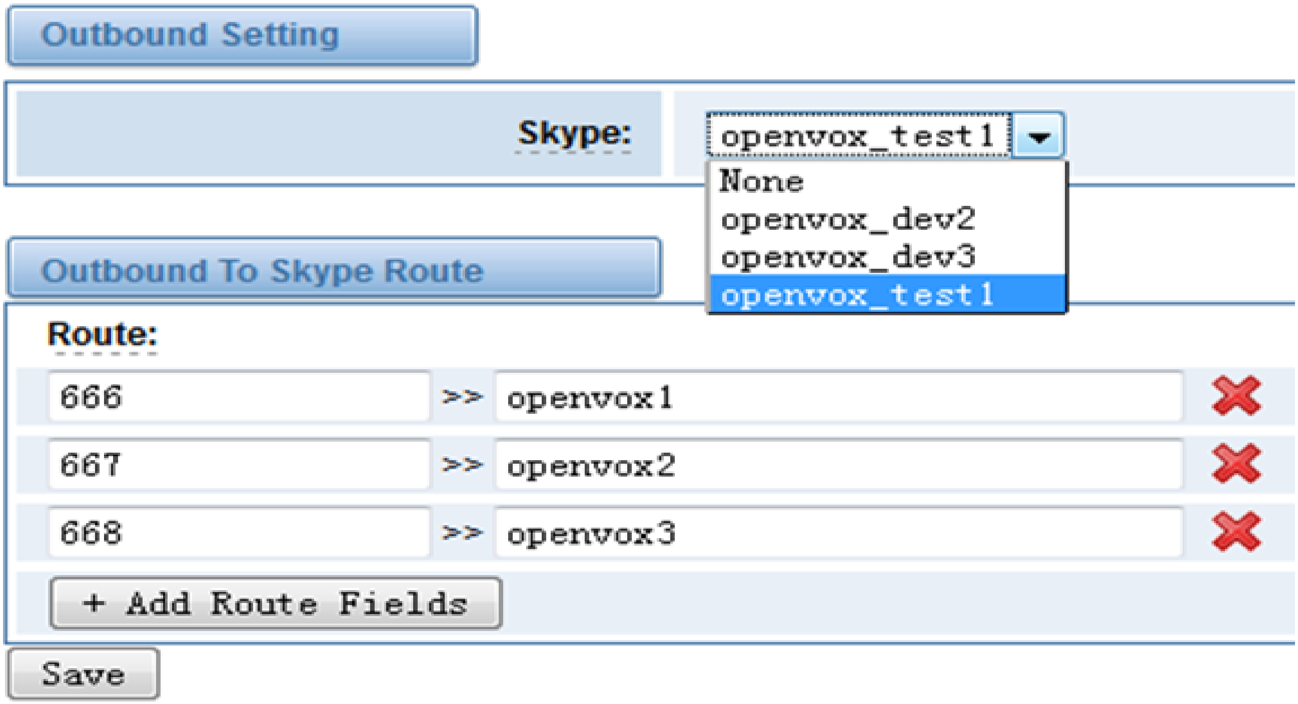
You can create one or more groups as shown below:


Create a Group
Skype Group Setting
Skype Group Name:
group1
Skypes:
unselected group
openvox_test1
selected group
openvox_dev2
openvox_dev3
Advanced Skype Group Setting
Max. limiting calling time per day(min):
1000
Max. different called numbers per day:
50
The interval times before next call(min):
2
Condition For Outbound:
Min Call Times
Save
Cancel


Create a Group
Skype Group Setting
Skype Group Name:
group2
Skypes:
unselected group
openvox_dev2
selected group
openvox_dev3
openvox_test1
Advanced Skype Group Setting
Max. limiting calling time per day(min):
1000
Max. different called numbers per day:
50
The interval times before next call(min):
2
Condition For Outbound:
Min Call Times
Save
Cancel

3.Advanced Skype Settings

The following figure means when sip openvox_test1 dials 666, the call will be connected to the Skype account openvox1, the same to 667 and 668.

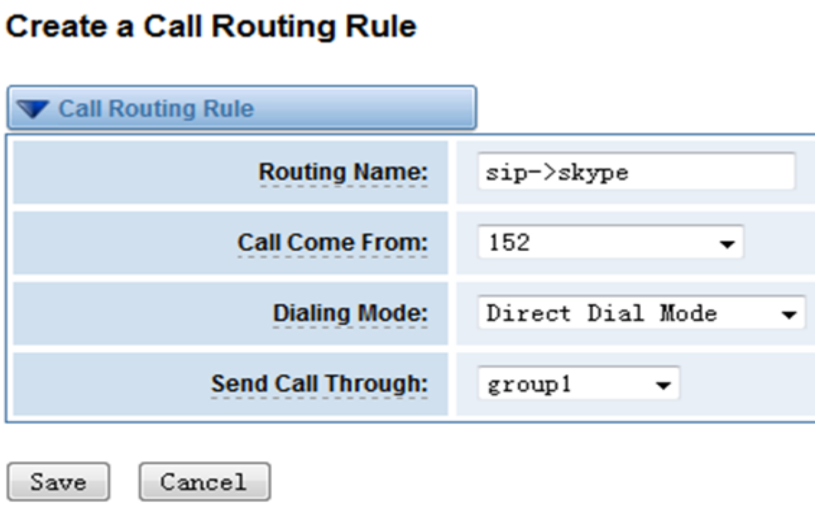


Step Five: Routing

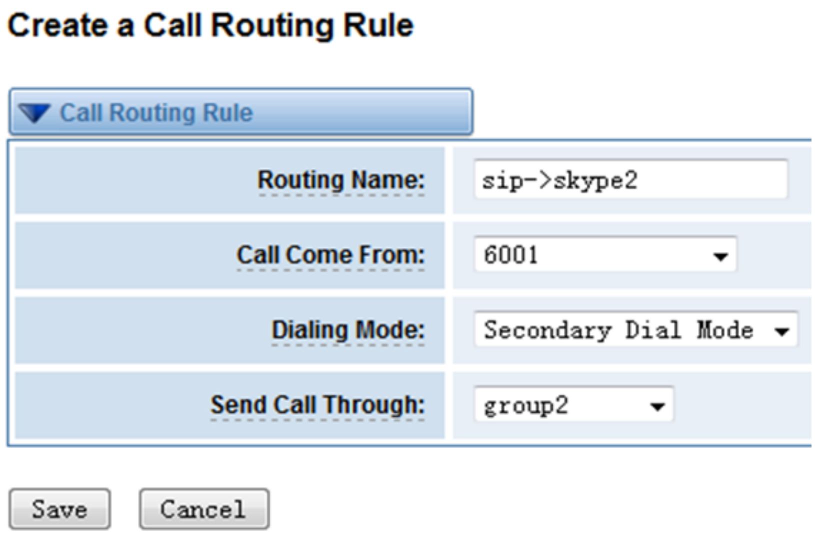
1.Call Routing Rules

1) Outbound

Create a call routing rule sip->skype for **Direct Dialing Mode.**

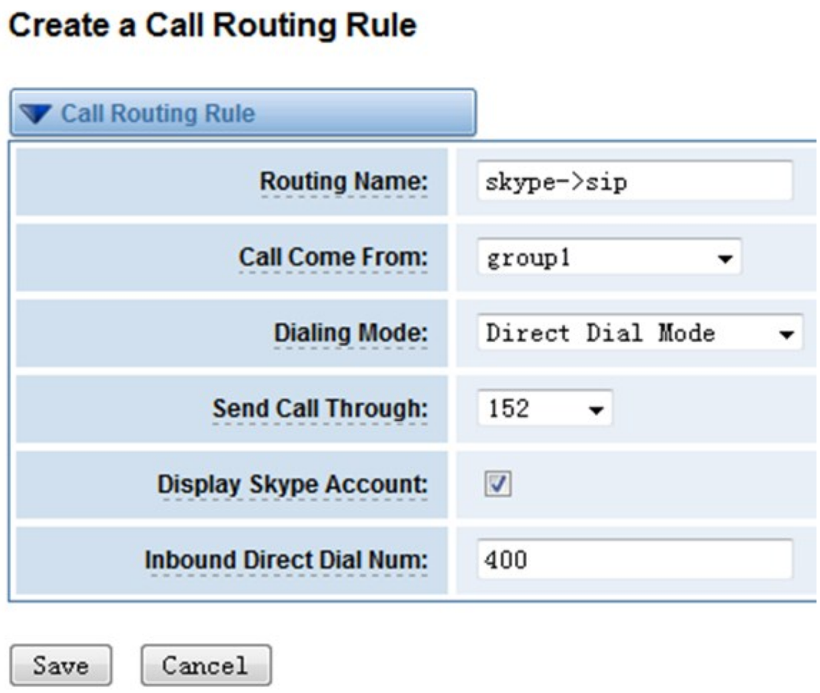


Create a call routing rule sip->skype2 for **Secondary Dialing Mode.**

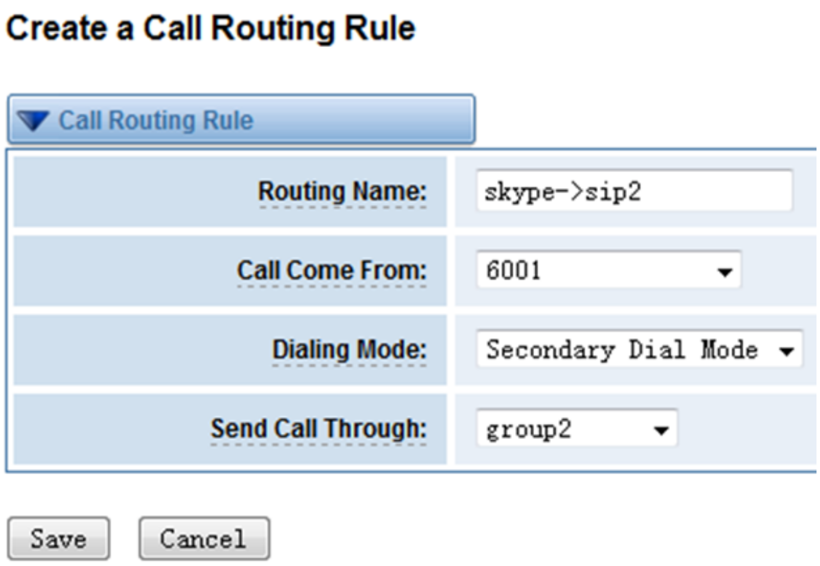


2) Inbound

For example, if you want to directly dial a Skype account in group1, Sip 400 rings, you can configure like this:



When you dial Skype account in group2 with Secondary Dialing Mode, you will be asked to input any extension numbers and end your input with “#” , after hearing warning tone. You can configure your routing as follows:



Step Six: Certify

You have two ways for choosing.

1) The first way: If you have many licenses, please upload licenses in form of files.

2) The second way: If you have very few licenses such as one or two, you can add licenses manually .

Display the results after save.

Authorization Code List

NO.	Authorization Code	Status	Start	End	Actions
1	pONvSLKpEjeqjcJ20DEI8aXkfYY+IfqMIfDdz4iuXqcJ2YOh1Dyb2RNagJUeSk9zTZCd+ol4LwS/EBy1xhqwog78NA0C50xxNKiiLnPVj8g99vFep5jyJEKbVkEWxGmRz/HTFI6dp0MOvlpQ/nuVulxip2PepLL9jSlkrqdV9S4	active	Indefinite Duration	Indefinite Duration	
2	VfCzfNLvifbSgpMLo+kFpUKr/ok++Y3kOdgzPvUzGPXeF4NUBBUZb+SZLz+lePEctyjAiF61p5B3L/wED+X2gPq7pi6+e8jbTUFwUZ+odQEI4CVhyUcEx/swyF7IS6mKbmNS0icidFaOkKZ7QPtytQQEUO3JrVEjWqFYDhkSsSo	active	Indefinite Duration	Indefinite Duration	
3	kfdcmh7AYBL3xUUWs37wY287lucxrQtlqaPLptuER2xVhRgJxL2ZB+FY7yR9ELAFXqCnq713Oix9Qoi+/iMTMzw2yzqQvxc1bdbdCUmKVZIoX83a6Abz2bb8uU6/OgKNZui+bd7hOLTI12vSC6E3y8V/DgsDibDuaZzqWLrP2ro	active	Indefinite Duration	Indefinite Duration	
4	zK40cB7vYcbTB8kQg3mIyoQ/86gNhmVSBY+k4mx6uWg/j4brJyzyGZ2CQHrHK05CqFX9AgREdqIb9yIlyx7zoVsyij0I1iSF2YdUJacYrkMK0VUZ+Soid7QAyvf+ORsjIFTspbunH5mkK9VbruF4uyHbS0X0XT5j9/tO01pwfyc	active	Indefinite Duration	Indefinite Duration	
5	ISO4WQNqHm9picFpMT705VX5qWuVt8R7XM9fmtDjKNmTbjQ1fC8COOXVLEhHnjzgqt4DmJVJN8arINj1z7Rrc6EBAD9KXKtsNkUYeaN3Mh6ztTZI7pRCXSV7XC6Jnpn/QYbh+xDYtJOv6P7n1X0BvK5rqDB7BN6tU7i+yW9IUwQ	active	Indefinite Duration	Indefinite Duration	

When you finish all the installation, you can check your main page state

Skype Status

Skype Status	Skype Account	Balance point	CallOutTime
online	openvox_dev2	258	0
online	openvox_dev3	309	0
online	openvox_test1	0	0

SIP Information

User Extension	Display Name	Authorization Name	Status	Credentials
152	152	152	unregisted	152@172.16.99.152
6001	6001	6001	registed	6001@172.16.99.152

Channels Status

Channel ID	Direction	Status	From	To	Cross
1	-	IDLE	-	-	-
2	-	IDLE	-	-	-
3	-	IDLE	-	-	-
4	-	IDLE	-	-	-
5	-	IDLE	-	-	-

Backup Configuration

Users can choose to backup configuration or not for your convenient. If you choose not to backup your configuration, When you unload your Sky2sip, your previous configuration will not be remained, and you must reinstall. Otherwise if you have backup your configuration, you just need to load.

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