SFLphone Documentation

Release 1.0

SFLphone Team

August 18, 2014

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SFLphone is a robust, standards-compliant enterprise softphone, for desktop and embedded systems. It is designed to handle several hundred calls a day. SFLphone is available under the GNU GPL license, version 3.

Please visit the official website for a complete list of features: http://sflphone.org.

CHAPTER 1

Contents

1.1 Getting started

1.1.1 Install SFLphone

First, you need to add the official SFLphone PPA (Personal Package Archive). This allows us to push new versions for older distributions.

Note: This step is not mandatory as Ubuntu provides SFLphone packages in its *universe* repository.

Solution 1: Software Center

- Open Software Center (in Unity, press the `Windows` key and type *software center*)

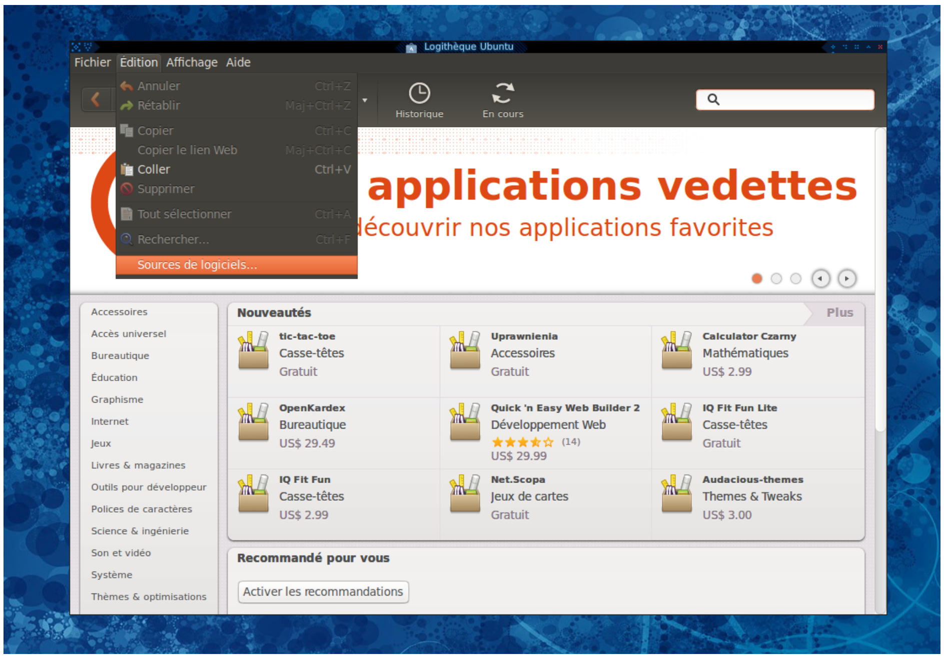


Figure 1.1: Ubuntu Software Center.

- In `Edit > Software` sources, select the *Other software* tab
- Click `add` and enter *ppa:savoirfairelinux*
- Click `close`
- Click the little arrow next to *All Software* and select sflphone
- Select **GNOME client for SFLphone** and click `Install`

Solution 2: Command-line

Add the repository to your software sources:

```
sudo add-apt-repository ppa:savoirfairelinux
```

Now, update the package list:

```
sudo apt-get update
```

You can now install the latest SFLphone version:

```
sudo apt-get install sflphone-client-gnome
```

Solution 3: Building from sources (not recommended)

Please refer to the instructions here to build SFLphone from source.

1.1.2 Configuring an existing account

The simplest way to configure SFLphone is to use the *First Run* wizard.

- In the Unity lens interface (top left dock icon or `Windows` key), type *sflphone*
- After a few seconds, a wizard window should appear on the screen with a welcome message
- Press `Continue` to start the wizard

Note: You can always return to the installation wizard in SFLphone by clicking the menu `Call > Configuration Assistant`

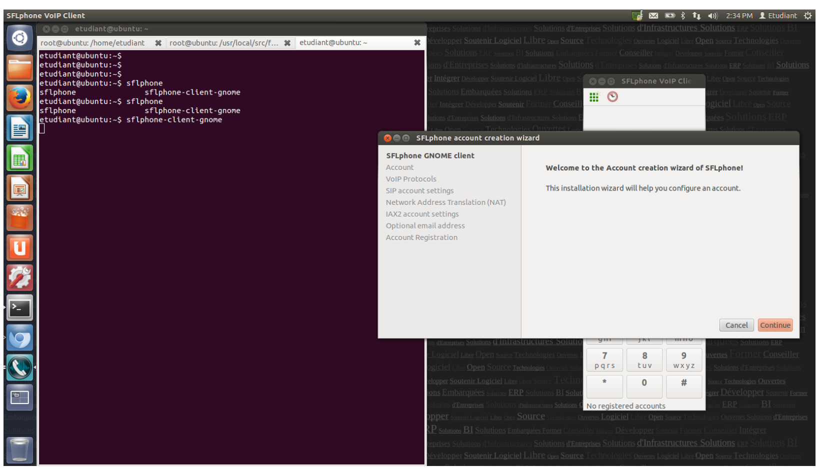


Figure 1.2: Account configuration wizard: first step.

Account

Select `Register an existing SIP or IAX2 account`

VoIP Protocols

You can select here the communication protocol to use to make calls (if unsure, use SIP).

Account settings

This step allows you to set up your account, by specifying the hostname, username, password, ...

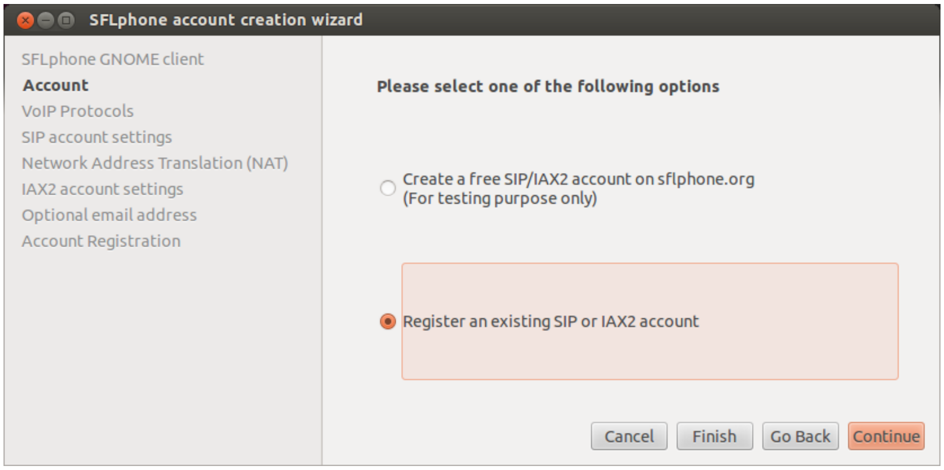


Figure 1.3: Account configuration wizard: Account window.

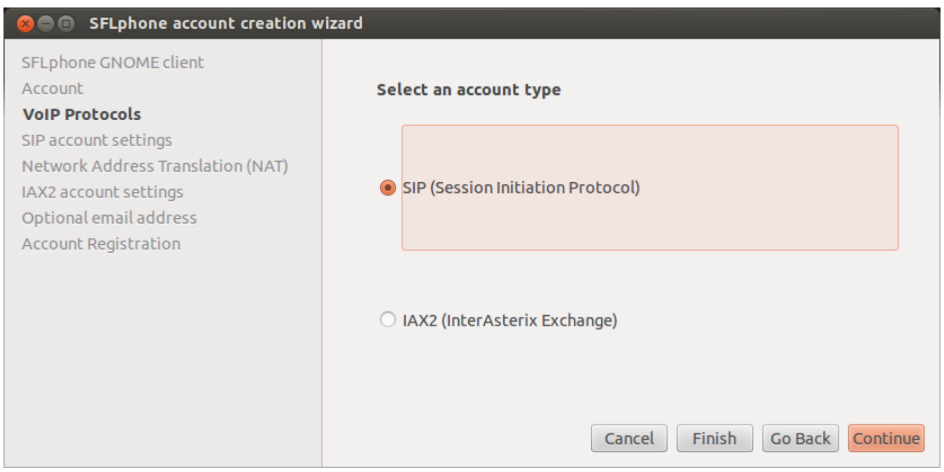


Figure 1.4: Account configuration wizard: VoIP Protocols window.

Figure 1.5: Account configuration wizard: Account settings.

Here are the details of each settings:

Fields	Description
Alias	A name you will remember (Example: Workplace)
Hostname	Usually the server address, (Example: sip.sflphone.org)
Username	Usually your phone extension (Example: 123), but may also b
Password	Your account password (Warning: will be stored in plain text)
Voicemail number (optional)	Another username you can call to play your voicemail. Not ev has one
Secure communications with ZRTP	Do not check (see security section)

Press `Continue` and `Apply` when you are ready to register your account.

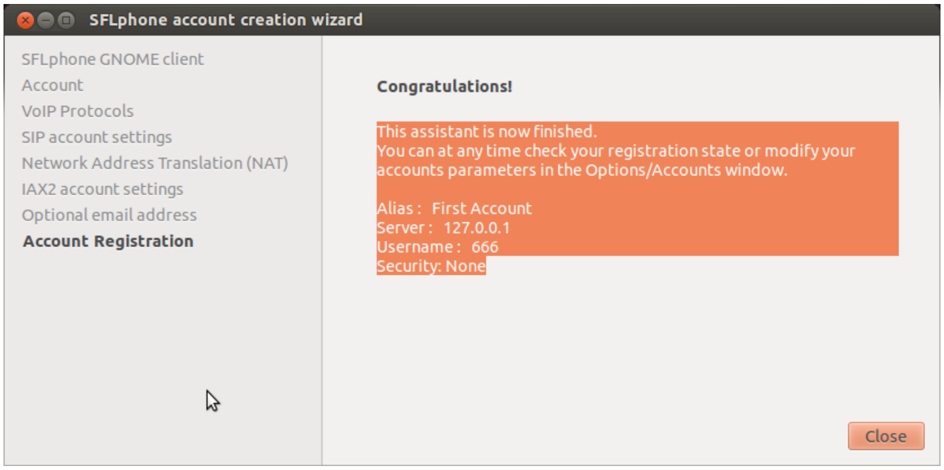


Figure 1.6: Account configuration wizard: Review

Account registration

The last panel displays an overview of your account settings. You may now click on `Close`, as the installation wizard is finishow now.

If you now select `Edit > Accounts`, your new account should be registered, and appear in green.

1.1.3 SIP security basics

The first thing to know about SIP security is that there is usually none at all. By default, **everything is transmitted unencrypted** and readable by any software that can grab traffic between you and your peer. If you are using proxies along the way, each of them may decide to ignore certain security options. The second important detail to retain is that SIP security alone (SIPS, SIP-TLS) does not encrypt your communications. SIPS only encrypts the handshake between both peers. To encrypt the media stream (aka, voice) itself, you also need to enable Secure RTP (aka, SRTP). Likewise, only having SRTP enabled may encrypt your audio but will not obscure information about the call itself (i.e. participants, IP addresses, etc.).

Most registrar and SIP servers have some level of support for security, however, all implementations are not created equal. As of Asterisk 1.8 (the default in many server operating systems), some security options are not as well supported as they should be. For a safer system, Freeswitch is the best free software option. Asterisk also needs to be recompiled to enable SRTP (see https://projects.savoirfairelinux.com/projects/sflphone/wiki/Security). This, in turn, will force you to manually update your SIP server everytime a security patch is available, introducing security vulnerabilities of its own.

Important: TODO show security options here

1.2 Setup a secure environment with Asterisk

1.2.1 Set up a basic Asterisk server

Important: Prerequisites: an Ubuntu server or virtual machine

Outcome: a basic SIP server with 2 accounts

Install Asterisk

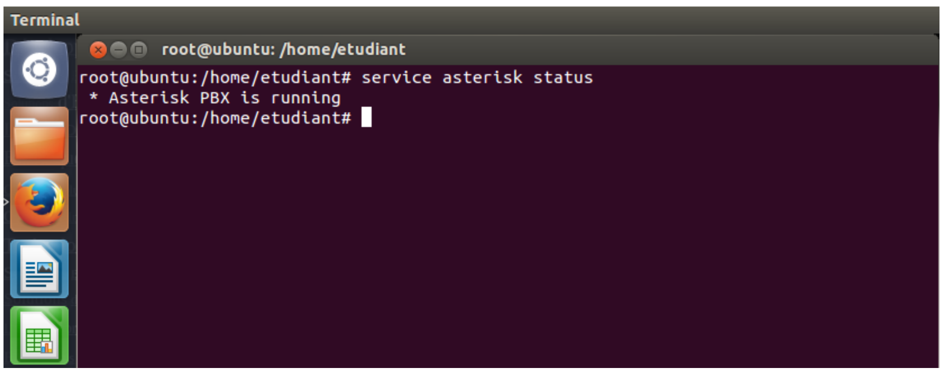
Install Asterisk with apt-get:

```
sudo su
apt-get install asterisk
```

It will ask you for a country code, you can check http://countrycode.org/ to get yours.

You can check if Asterisk is operational using:

```
service asterisk status
```



Basic configuration

Add SIP accounts

Now, using your favourite text editor, make a backup of `/etc/asterisk/sip.conf` and replace it with:

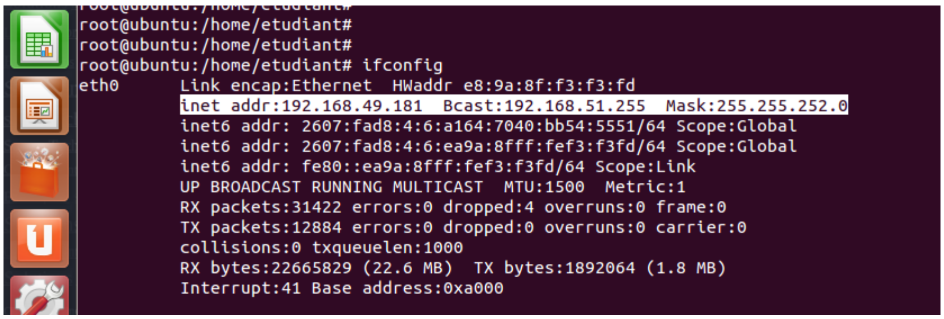
```
[general]
context=internal
allowguest=no
allowoverlap=no
bindport=5060
bindaddr=0.0.0.0
srvlookup=no
disallow=all
allow=ulaw
allow=g722
allow=alaw
allow=gsm
alwaysauthreject=yes
canreinvite=no
session-timers=refuse
localnet=192.168.1.0/255.255.255.0

[666]
```

```
callerid=anonymous
type=friend
host=dynamic
secret=123
context=internal

[777]
type=friend
host=dynamic
secret=456
context=internal
```

Be sure to update *localnet* to match your network settings. Run `ifconfig` command to check your public IP address:

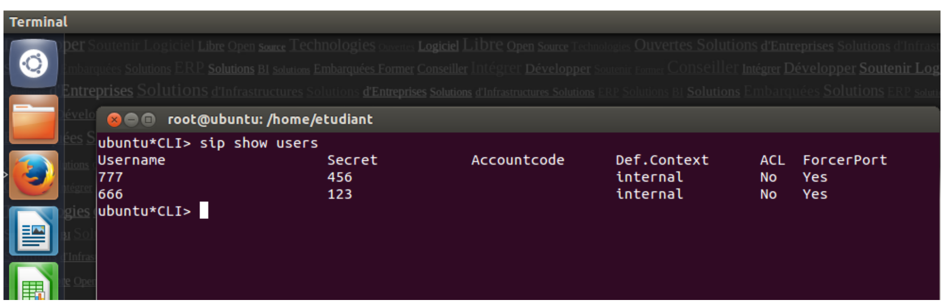


To apply the settings, execute `rasterisk` and type:

```
sip set debug on
sip reload
```

To confirm the new diaplan, run:

```
sip show users
```



Dialplan

Now, setup a new dialplan to be able to call other users. Edit `/etc/asterisk/extensions.conf` and add the following lines:

```
[users]
include => default
include => trunklocal
include => iaxtel700
include => trunktollfree
include => iaxprovider
exten => 777,n,Dial(SIP/777,777,Tt)
```

```
[internal]
exten => _XXX,1,Dial(SIP/${EXTEN})
```

1.2.2 Configuring Asterisk encryption

In this part, we will use TLS to **encrypt the data streams**.

```
# Operate as root to install openssl
sudo su -
apt-get install openssl

# Create the directory and get a certificate
mkdir /etc/certs
cd /etc/certs/
wget https://raw.github.com/rillian/asterisk-opus/master/contrib/scripts/ast_tls_cert
sh ast_tls_cert -C pbx.MyHostName -O "NSA proof(?) server" -d /etc/certs/

# Now fill out the form.
sh ast_tls_cert -m client -c /etc/certs/ca.crt -k /etc/certs/ca.key -C mynsauser.MyHostName -O "NSA p
chown asterisk:asterisk ./ -R
ls
```

Note: This *How-To* has assumed the same computer as server and client for simplicity's sake. In a real-world context, this will rarely be the case. So you will have to safely upload the certificates to the client computer. To do this, the `scp` command allows you to upload the key over an encrypted connection. In our case, you can simply copy them using `cp` as it doesn't change anything.

```
scp mynsauser.pem username@hostname:/tmp/
scp ca.crt username@hostname:/tmp/
```

Now, Asterisk need to be configured to use this certificate for encrypted calls. Backup your current `/etc/asterisk/sip.conf`, then open `sip.conf` and replace its contents with:

```
[general]
context=internal
allowguest=no
allowsubscribe=yes
allowoverlap=no
bindport=5060
bindaddr=0.0.0.0      ;192.168.48.213
tlsbindaddr=0.0.0.0   ;192.168.48.213
srvlookup=no
disallow=all
allow=ulaw
allow=g722
allow=alaw
allow=gsm
alwaysauthreject=yes
canreinvite=no
;nat=yes
session-timers=refuse
localnet=192.168.48.0/255.255.252.0
tlsenable=yes
tlscertfile=/etc/certs/asterisk.pem
tlscafile=/etc/certs/ca.crt
tlscipher=TLSv1
```

```
;tlsclientmethod=tlsv1
tlsdontverifyserver=no
tlsbindaddr=0.0.0.0

[777]
callerid=NSA
type=friend
secret=nsa
host=dynamic
transport=tls
port=5061
context=internal
dtmfmode=rfc2833
insecure = invite,port
nat = yes

[888]
callerid=NSA
type=friend
secret=nsa
host=dynamic
transport=tls
port=5061
context=internal
dtmfmode=rfc2833
insecure = invite,port
nat = yes

[999]
callerid=PLAY
type=friend
secret=play
host=dynamic
transport=tls
port=5061
context=local
dtmfmode=rfc2833
;insecure = invite,port
nat = yes
```

And in `extensions.conf`, in the *[local]* section, add:

```
exten => 999,1,Answer
exten => 999,2,Playback(tt-weasels)
exten => 999,3,Wait(10)
exten => 999,4,Hangup
```

Now, run the `rasterisk` command and type:

```
sip reload
dialplan reload
```

Asterisk should now be using TLS for message passing.

Warning: The stream itself is not yet encrypted, only the SIP messages are.

1.2.3 Configuring SFLphone with Asterisk

Once this is done, execute `sflphone-client-gnome`. A *configuration wizard* will launch if you started it for the first time.

Select the following values:

- Account: `Register an Existing SIP or IAX2 account`, then `Next`
- VoIP Protocols: `SIP`, then `Next`

As for the SIP account settings, input these values:

Settings	Values
Alias	My First Account
Hostname	<your asterisk server IP>
Username	666
Password	123

Note: You may run `ifconfig` to check your IP address.

Then `Next` and `Apply`.

If you go in `Edit > Account`, you should now see *MyFirstAccount* as **Registered** (in green). If you see a **Trying...** status and run **Asterisk** on the same computer as SFLphone, you have to change the default port in your account advanced settings (`Edit > Accounts > Select your account > Edit > Advanced`).

1.2.4 Configuring SFLphone security

Important: Prerequisites: You have both ca.crt and the .pem certificates

For Asterisk to **encrypt your stream**, select:

- In `Edit > Account > Your account > Security`, select `ZRTP` in `SRTP key exchanges` (Asterisk need to be compiled with SRTP support). Also select `Use TLS transports`.
- In the `Advanced` tab, set your ca.crt as authority and .pem as user certificate.
- Uncheck `Verify incoming certificate (as a server)`
- Click on `Apply`

You are now done!

Warning: Please note that this setup is still vulnerable to Man-in-the-middle attack, but not to packet sniffers.

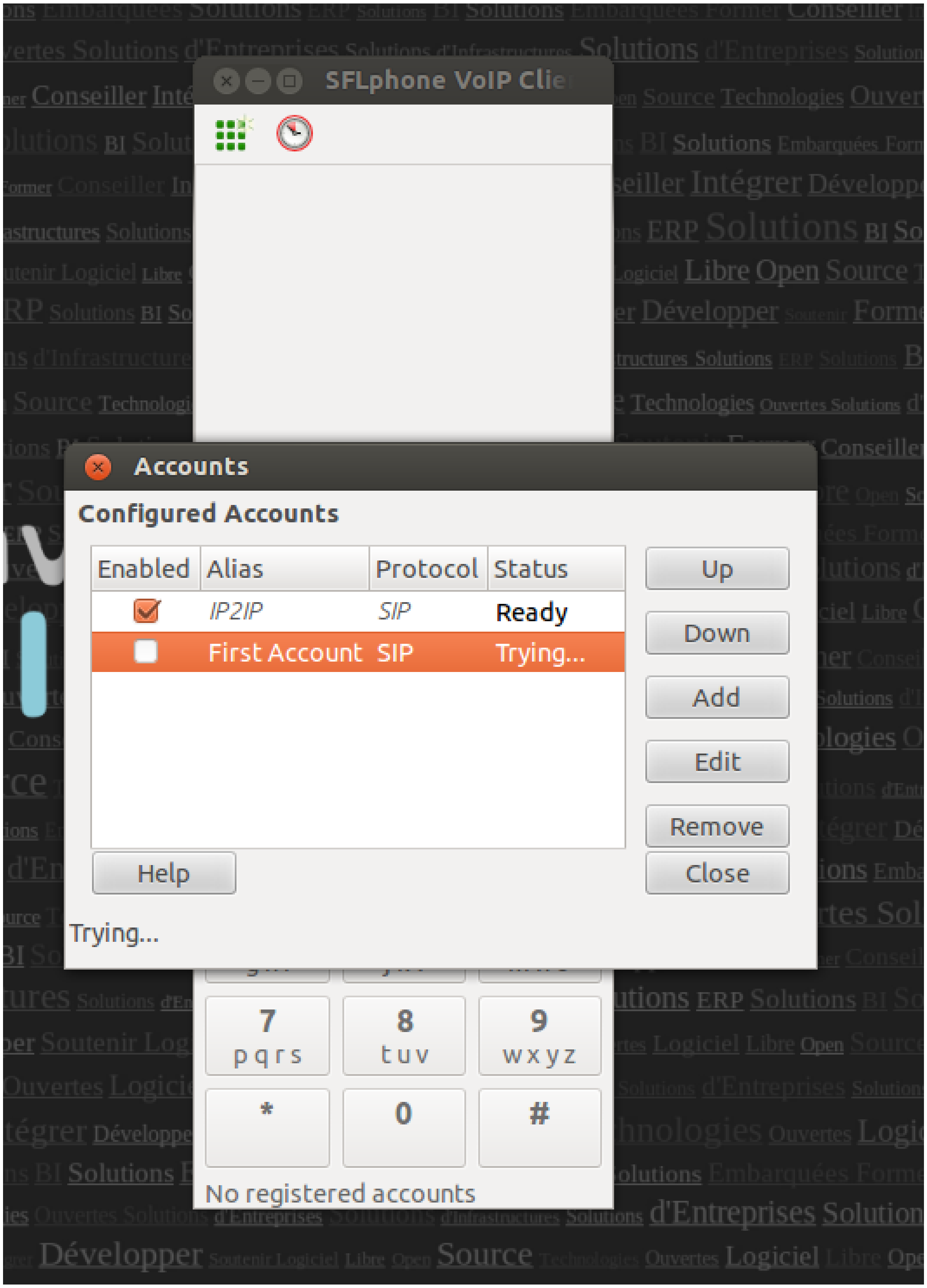


Figure 1.7: Account list window, showing the accounts registration status.

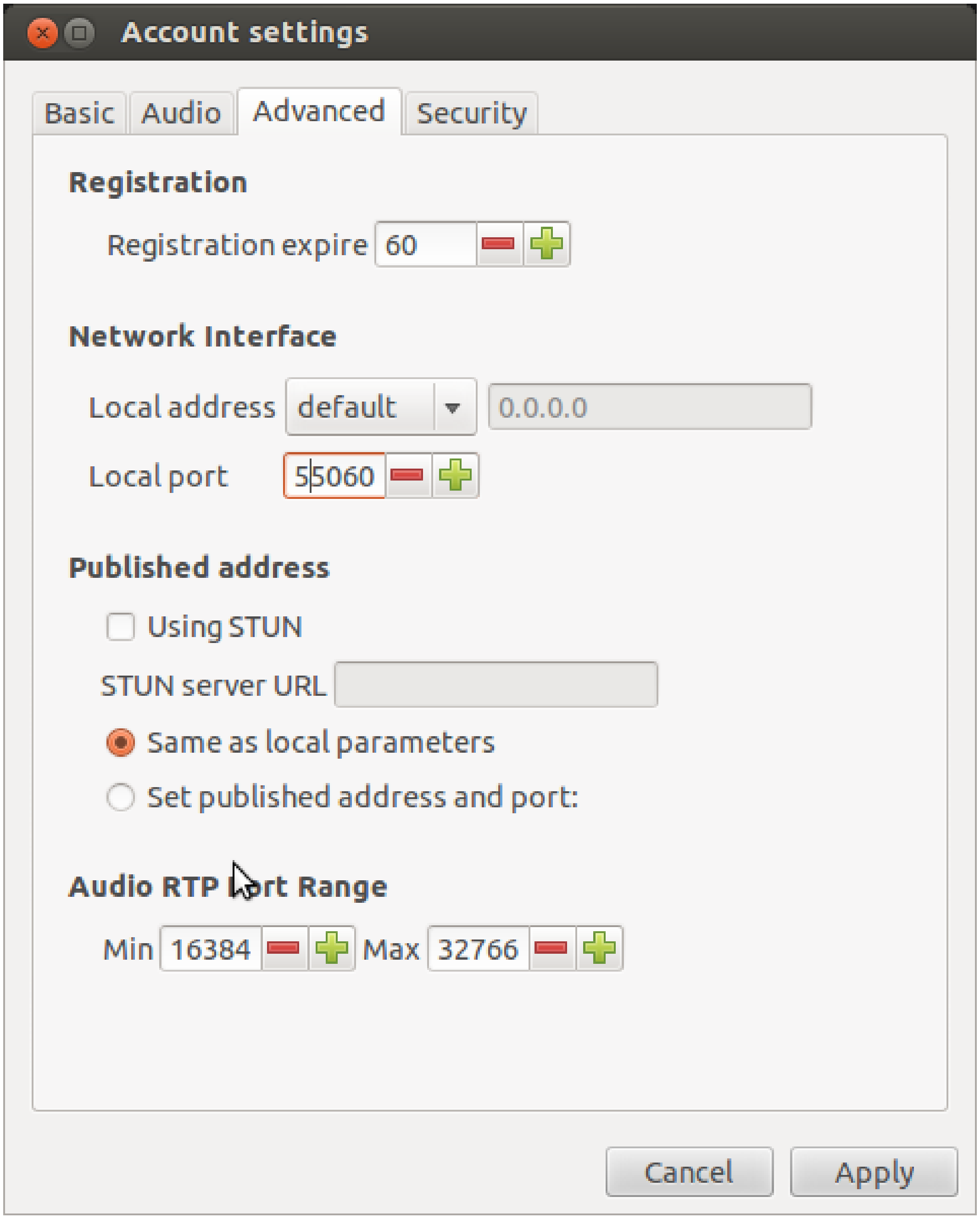
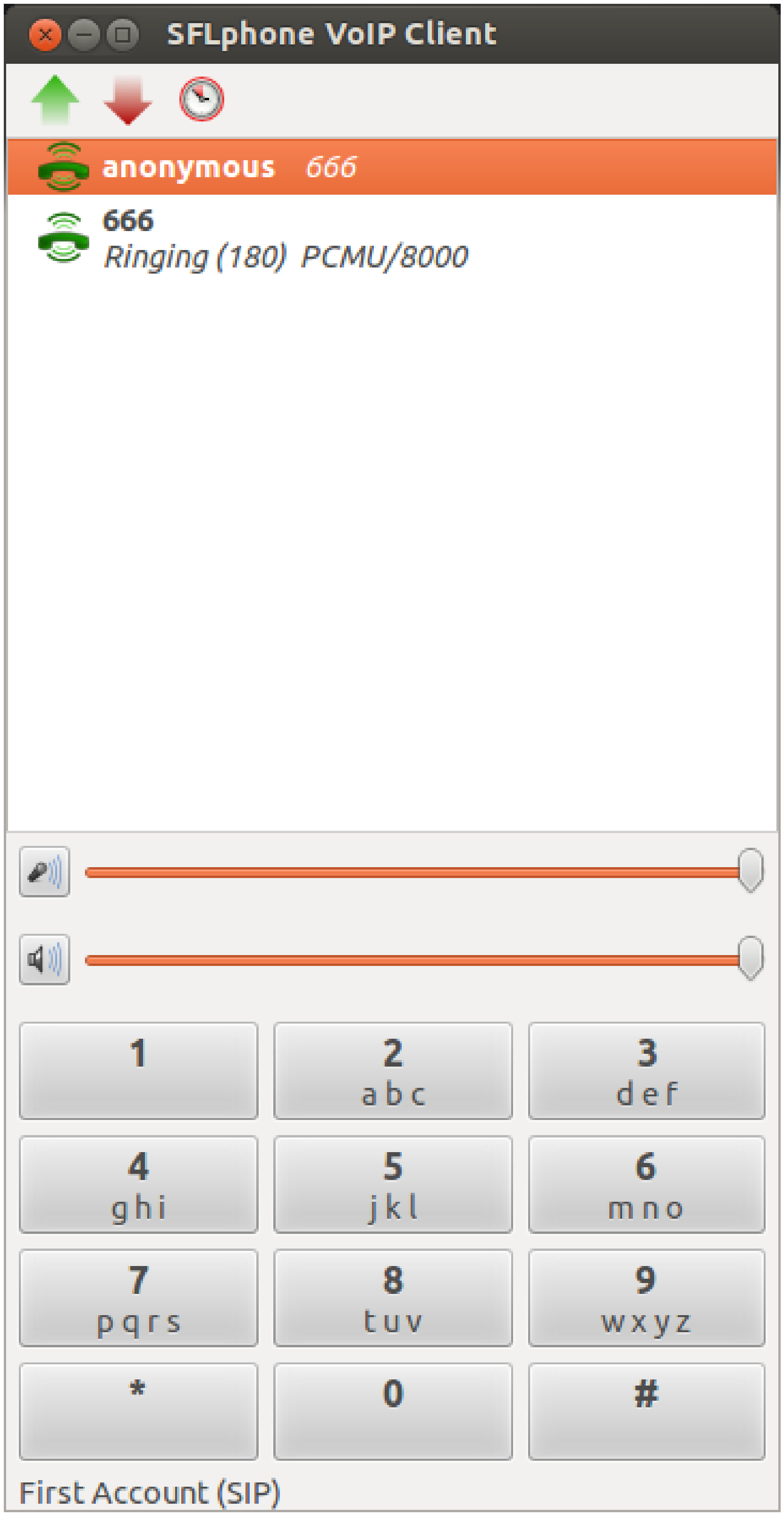


Figure 1.8: The account's advanced settings allows to configure the network interface, port range and registration expiration time.


SFLphone VoIP Client
anonymous 666
666
Ringing (180) PCMU/8000
1
2 abc
3 def
4 ghi
5 jkl
6 mno
7 pqrs
8 tuv
9 wxyz
*
0
#
First Account (SIP)

1.3 Setup a secure environment with Freeswitch

1.3.1 Installing Freeswitch

Please see the Ubuntu Quick Start Guide for an official guide to installing Freeswitch.

Otherwise, to install the latest FreeSwitch from git, run the following commands:

```
# Install all dependencies
sudo su -
apt-get install git-core build-essential autoconf automake libtool libncurses5 libncurses5-dev make l

# Download FreeSwitch via git
mkdir -p /usr/local/src
cd /usr/local/src
git clone git://git.freeswitch.org/freeswitch.git
cd freeswitch

# Build
./bootstrap.sh
./configure
make
# You may have to call "make" a few times before it works.

# Install in /usr/local
make all install cd-sounds-install cd-moh-install
```

Reference: http://wiki.freeswitch.org/wiki/Linux_Quick_Install_Guide to install FreeSwitch (install all optional packages).

Now, it is time to see if FreeSwitch is properly loaded. If you already have SFLphone running on the same computer, make sure to close it first.

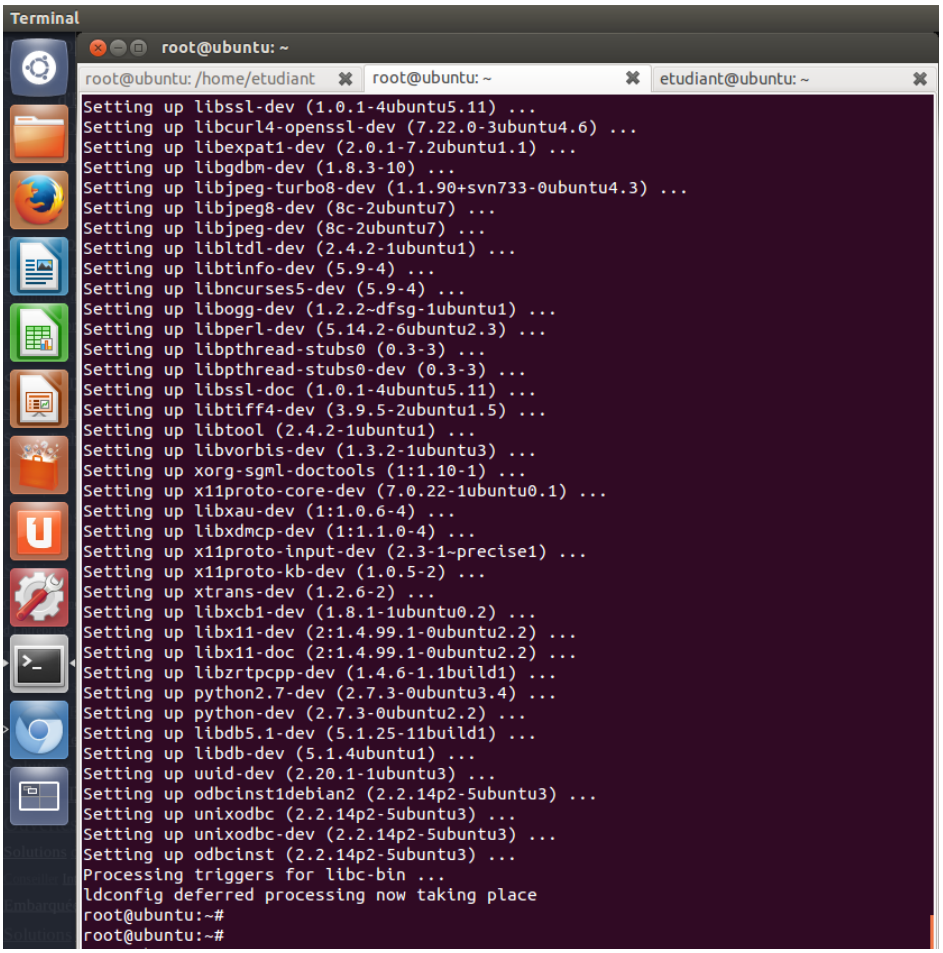
```
/usr/local/freeswitch/bin/freeswitch
```

After a few seconds, a shell will appear. To test if SIP is ready, enter:

```
sofia status profile internal
```

This should display something like:

```
Name internal
Domain Name N/A
Auto-NAT false
DBName sofia_reg_internal
Pres Hosts 192.168.48.185,192.168.48.185
Dialplan XML
Context public
Challenge Realm auto_from
RTP-IP 192.168.48.185
SIP-IP 192.168.48.185
URL sip:mod_sofia@192.168.48.185:5060
BIND-URL sip:mod_sofia@192.168.48.185:5060;transport=udp,tcp
HOLD-MUSIC local_stream://moh
OUTBOUND-PROXY N/A
CODECS IN G722,PCMU,PCMA,GSM
CODECS OUT G722,PCMU,PCMA,GSM
TEL-EVENT 101
DTMF-MODE rfc2833
CNG 13
SESSION-TO 0
NOMEDIA false
```


Terminal
root@ubuntu: ~
root@ubuntu: /home/etudiant
root@ubuntu: ~
etudiant@ubuntu: ~
Setting up libssl-dev (1.0.1-4ubuntu5.11) ...
Setting up libcurl4-openssl-dev (7.22.0-3ubuntu4.6) ...
Setting up libexpat1-dev (2.0.1-7.2ubuntu1.1) ...
Setting up libgdbm-dev (1.8.3-10) ...
Setting up libjpeg-turbo8-dev (1.1.90+svn733-0ubuntu4.3) ...
Setting up libjpeg8-dev (8c-2ubuntu7) ...
Setting up libjpeg-dev (8c-2ubuntu7) ...
Setting up libltdl-dev (2.4.2-1ubuntu1) ...
Setting up libtinfo-dev (5.9-4) ...
Setting up libncurses5-dev (5.9-4) ...
Setting up libogg-dev (1.2.2~dfsg-1ubuntu1) ...
Setting up libperl-dev (5.14.2-6ubuntu2.3) ...
Setting up libpthread-stubs0 (0.3-3) ...
Setting up libpthread-stubs0-dev (0.3-3) ...
Setting up libssl-doc (1.0.1-4ubuntu5.11) ...
Setting up libtiff4-dev (3.9.5-2ubuntu1.5) ...
Setting up libtool (2.4.2-1ubuntu1) ...
Setting up libvorbis-dev (1.3.2-1ubuntu3) ...
Setting up xorg-sgml-doctools (1:1.10-1) ...
Setting up x11proto-core-dev (7.0.22-1ubuntu0.1) ...
Setting up libxau-dev (1:1.0.6-4) ...
Setting up libxdmcp-dev (1:1.1.0-4) ...
Setting up x11proto-input-dev (2.3-1~precise1) ...
Setting up x11proto-kb-dev (1.0.5-2) ...
Setting up xtrans-dev (1.2.6-2) ...
Setting up libxcb1-dev (1.8.1-1ubuntu0.2) ...
Setting up libx11-dev (2:1.4.99.1-0ubuntu2.2) ...
Setting up libx11-doc (2:1.4.99.1-0ubuntu2.2) ...
Setting up libzrtpcpp-dev (1.4.6-1.1build1) ...
Setting up python2.7-dev (2.7.3-0ubuntu3.4) ...
Setting up python-dev (2.7.3-0ubuntu2.2) ...
Setting up libdb5.1-dev (5.1.25-11build1) ...
Setting up libdb-dev (5.1.4ubuntu1) ...
Setting up uuid-dev (2.20.1-1ubuntu3) ...
Setting up odbcinst1debian2 (2.2.14p2-5ubuntu3) ...
Setting up unixodbc (2.2.14p2-5ubuntu3) ...
Setting up unixodbc-dev (2.2.14p2-5ubuntu3) ...
Setting up odbcinst (2.2.14p2-5ubuntu3) ...
Processing triggers for libc-bin ...
ldconfig deferred processing now taking place
root@ubuntu:~#
root@ubuntu:~#

```
LATE-NEG true
PROXY-MEDIA false
ZRTP-PASSTHRU true
AGGRESSIVENAT false
CALLS-IN 0
FAILED-CALLS-IN 0
CALLS-OUT 0
FAILED-CALLS-OUT 0
REGISTRATIONS 1
```

1.3.2 Configuring Freeswitch security

To have an overview of SIP security, please read the section *SIP security basics*.

```
cd /usr/local/freeswitch/bin
sudo ./gentls_cert setup -cn 127.0.0.1 -alt DNS:localhost -org 127.0.0.1
sudo ./gentls_cert create_server -cn 127.0.0.1 -alt DNS:localhost -org 127.0.0.1
```

This will generate a self signed certificate in `/usr/local/freeswitch/conf/ssl/CA/cakey.pem`.

Important: Self-signed certificates are not entirely secure and void the chain of trust. If you want care about security, please generate a certificate signed by an authority.

Now, in `/usr/local/freeswitch/conf/vars.xml`, enable TLS:

Original:

```
<X-PRE-PROCESS cmd="set" data="sip_tls_version=tlsv1"/>

<!-- Internal SIP Profile -->
<X-PRE-PROCESS cmd="set" data="internal_auth_calls=true"/>
<X-PRE-PROCESS cmd="set" data="internal_sip_port=5060"/>
<X-PRE-PROCESS cmd="set" data="internal_tls_port=5061"/>
<X-PRE-PROCESS cmd="set" data="internal_ssl_enable=false"/>

<!-- External SIP Profile -->
<X-PRE-PROCESS cmd="set" data="external_auth_calls=false"/>
<X-PRE-PROCESS cmd="set" data="external_sip_port=5080"/>
<X-PRE-PROCESS cmd="set" data="external_tls_port=5081"/>
<X-PRE-PROCESS cmd="set" data="external_ssl_enable=false"/>
```

New:

```
<X-PRE-PROCESS cmd="set" data="sip_tls_version=tlsv1"/>

<!-- Internal SIP Profile -->
<X-PRE-PROCESS cmd="set" data="internal_auth_calls=true"/>
<X-PRE-PROCESS cmd="set" data="internal_sip_port=5060"/>
<X-PRE-PROCESS cmd="set" data="internal_tls_port=5061"/>
<X-PRE-PROCESS cmd="set" data="internal_ssl_enable=true"/>
<X-PRE-PROCESS cmd="set" data="internal_ssl_dir=$${base_dir}/conf/ssl"/>

<!-- External SIP Profile -->
<X-PRE-PROCESS cmd="set" data="external_auth_calls=false"/>
<X-PRE-PROCESS cmd="set" data="external_sip_port=5080"/>
<X-PRE-PROCESS cmd="set" data="external_tls_port=5081"/>
<X-PRE-PROCESS cmd="set" data="external_ssl_enable=true"/>
<X-PRE-PROCESS cmd="set" data="external_ssl_dir=$${base_dir}/conf/ssl"/>
```

Now, in the Freeswitch shell, execute:

```
reloadxml
```

Back in `/usr/local/freeswitch/bin`, it is now time to create certificate for users:

```
sudo ./gentls_cert create_client -cn 1002 -out 1002.pem
```

Be sure to *copy/scp* the following certificates to all relevants users.

Note: This *How-to* is always using the same computer as server and client for simplicity purpose. In a real context, this will rarely be the case. So you will have to safely upload the certificates to the client computer. To do this, the *scp* command allow you to create an encrypted upload of the key, thus, invulnerable to man in the middle attack. In this case, you can copy them using *cp* as it doesn't change anything.

```
scp /usr/local/freeswitch/conf/ssl/CA/cacert.pem username@hostname:/home/username/
scp /usr/local/freeswitch/conf/ssl/agent.pem username@hostname:/home/username/
```

If you use an alternate upload method, please double check if the target file owned by the same used as the sflphone process (usually your current username) and have **600** permissions. The *scp* lines will automatically do that for you.

Optional:

In this how-to, we run Freeswitch as root. This, of course, as Freeswitch is a network facing application, is a potential attack vector. If you change Freeswitch user, do not forget to use:

```
cd /usr/local/freeswitch/
find -iname conf/ssl/ | xargs chown myfreeswitchuser:myfreeswitchuser
```

Reference: http://wiki.freeswitch.org/wiki/SIP_TLS

1.3.3 Configuring SFLphone with Freeswitch

Freeswitch already provides a few usable accounts you can use right away. They are numbered from **1000** to **1015** and the **default password** is *1234*.

The configuration files for every accounts are stored in `/usr/local/freeswitch/conf/directory/default/`. You can edit accounts individually.

1.3.4 Configuring SFLphone security

Configuring a secure Freeswitch account is trivial.

First, make sure you uploaded the `cacert.pem` and `agent.pem` as described in the earlier steps. Once this is done, create your account using `Edit > Account > New` and in the `Security` tab, check `Use TLS transports` and select `ZRTP` in `SRTP key exchange`.

Now, in the `Edit` dialog, add both your `cacert.pem` and `agent.pem` and press `OK`.

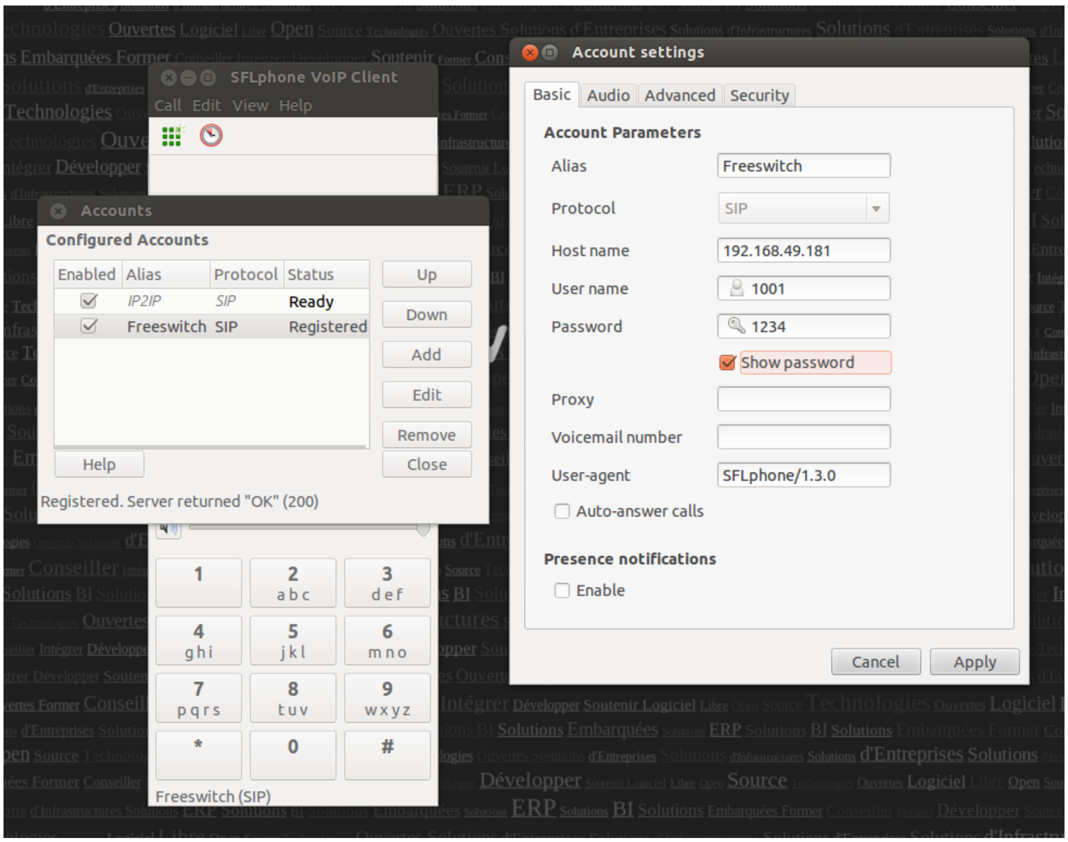


Figure 1.9: Account settings for a Freeswitch account.

CHAPTER 2

Indices and tables

- *genindex*
- *modindex*
- *search*