

VoIP with FreeSWITCH

TALK SOFT

FreeSWITCH is a powerful and versatile telephony platform that can scale from a softphone to a PBX and even to a carrier-class softswitch.

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Open source projects have lowered the barrier to entry into telephony for hobbyists and businesses alike. The popular Asterisk PBX tool, for instance, is a high-functioning and low-budget telephony alternative that has proven disruptive in the world of business telephone systems [1]. Perhaps the most disruptive software of all, however, comes in the form of the scalable softswitch library called FreeSWITCH.

The FreeSWITCH website [2] says that it can be used as a “simple switching engine, a PBX, a media gateway, or a media server to host IVR applications.”

True to that chameleon-like description, FreeSWITCH has appeared in production environments as a telco switch, in numerous business scenarios as a PBX, as a softphone for laptop users, and even as a voice routing engine for the OpenSim project.

Additionally, you can deploy FreeSWITCH as an application or use it as a library (libfreeswitch) that can be embedded in third-party software. This double identity is similar to the relation-

ship between cURL (application) and libcurl (library).

FreeSWITCH has been used in some exotic scenarios; however, those new to telephony applications are best served starting out with FreeSWITCH in its default configuration as a SOHO PBX. Although you might find the numerous FreeSWITCH configuration files and settings overwhelming, in a short time, you will appreciate the power and flexibility of this versatile telephony tool.

Installing FreeSWITCH

As of this writing, the current, stable version of FreeSWITCH is 1.0.4, which can be downloaded from the project website [3], or you can use the Subversion repository to install the latest version. Unlike many projects, the SVN trunk is extremely stable, so it might be convenient for you simply to use the latest trunk.

In any case, the FreeSWITCH developers have made it very easy to update to the latest trunk, even if you have installed from the latest stable version.

Some pre-compiled binaries are known to exist; however, I recommend compiling from source.

First, unpack and install the source. In this example, I will download into `/usr/src` and install from there:

```
tar zxvf freeswitch-1.0.4.tar.gz
cd freeswitch-1.0.4
./configure
make install
```

I recommend getting the sound files:

```
make cd-sounds-install
make cd-moh-install
```

FreeSWITCH comes with and can handle sound files at 8, 16, 32, and 48kHz sampling rates. Few, if any, telephony systems – open source or proprietary – can do the kinds of things FreeSWITCH can do with calls at multiple sampling rates.

The installation will take some time, so feel free to chain the commands together with the `&&` operator.

When the process is complete, you will have a basic FreeSWITCH installation. Because it is handy to have the two main FreeSWITCH executable files available in the *bin* path, I use symlinks:

```
ln -s /usr/local/freeswitch/bin/freeswitch /usr/local/bin/freeswitch
ln -s /usr/local/freeswitch/bin/fs_cli /usr/local/bin/fs_cli
```

To launch the application, enter *freeswitch*, and use the *fs_cli* utility to access the command line of any FreeSWITCH server to which you have access, even if it is on another server. At first, you'll see quite a bit of debug/log information, but don't worry about errors or warnings at this point.

Your New Softswitch

The first basic command is *version*, which displays the version of FreeSWITCH you are running, including the SVN revision. The *status* command shows information about your instance of FreeSWITCH, including uptime and the number of currently active sessions. Another basic command is the self-descriptive *help*.

To exit FreeSWITCH, type *fsctl shutdown* or use the shortcut command ... (three periods).

If you have a sound card or headset on your system then try this:

```
load mod_portaudio
```

After several debug lines, you'll see +OK when the module is loaded. Now list the PortAudio devices available on your system:

```
pa devlist

API CALL [pa(devlist)] output:
0;Built-in Microphone;2;0;i
1;Built-in Input;2;0;
2;Built-in Output;0;2;r,o
3;Logitech USB Headset;0;2;
4;Logitech USB Headset;1;0;
```

The *i* and *o* are next to the currently selected input and output devices. To change the input device, use the *pa indev* command with the device number,

```
pa indev #x
```

where *x* is the number of the input device. The output device can be changed with the *pa outdev* command. In my setup, I'll use device 4 as output and device 5 as input.

If you have the MOH (music on hold) files installed, this command will make a call to the MOH extension and send the MOH to the PortAudio output device:

```
pa call 9999
```

If you hear music, it was a successful call. If not, make sure the MOH files were installed properly. Hang up with the *pa hangup* command.

To make sure both input and output devices are working, do an echo test:

```
pa call 9996
```

Now when you speak into the input device, your voice should be echoed to the output device. The *pa hangup* command disconnects.

As you can see, it is possible to use FreeSWITCH and PortAudio to create a simple



Figure 2: A successfully registered X-Lite client.

softphone. In this way, you will be able to use any codec that your FreeSWITCH installation supports – and there are some very high quality codecs available – without installing a softphone client.

Connecting a SIP Phone

The most common way to use FreeSWITCH is with a SIP phone. A physical handset is often called a hard phone, whereas a software-based phone is appropriately called a softphone. Numerous types of hard phones are available from manufacturers. Softphones come in both proprietary and FLOSS varieties. A popular free – but not open source – softphone is X-Lite from CounterPath [4], which is available for Linux, Mac OS X, and Windows. Even if your FreeSWITCH server runs on Linux, your softphones can run on any operating system, as long as they are SIP compliant.

To begin, download X-Lite for your desktop or laptop operating system and perform a default install. (Installing a softphone on the same system that is running FreeSWITCH is not recommended; however, in many cases, it will work for the sake of testing.)

Session Initiation Protocol

SIP is a TCP/IP-based Application Layer protocol used for multimedia sessions over the Internet. The text-based protocol is similar to HTTP, with requests and responses, header fields, encoding rules, and status codes. In addition to VoIP, SIP is used in instant messaging, online gaming, and distribution of streaming multimedia.

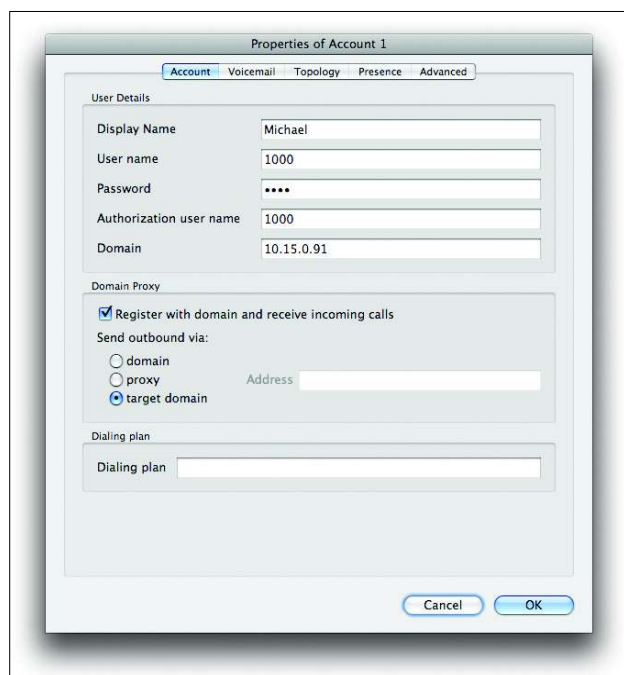


Figure 1: The X-Lite SIP account configuration screen.

Now launch X-Lite and open the SIP Account Settings menu by clicking the *Show Menu* button. (The button is the small down-arrow at the top of the phone, just left of center.)

Only one SIP account is available in X-Lite, so click on it then click the *Properties* button. As shown in Figure 1, fill in the fields and use 1234 as the password for extension 1000.

Next, clear out the *Dialing plan* field because it is not needed with FreeSWITCH. Now click *OK* then *Close*, and the phone will attempt to register. Figure 2 shows a successful registration.

To make a test call, dial 9999 and click the green button to send. In a few moments you should hear music on hold. Figure 3 shows a call in progress to extension 9999.

To shut down FreeSWITCH, use the *fsctl shutdown (...)* command.

Configuration Highlights

The default FreeSWITCH configuration has many features. Table 1 shows Dialplan highlights, and Table 2 shows some of the more important files and subdirectories in the FreeSWITCH configuration.

The default install will create a directory named */usr/local/freeswitch/conf*, which I'll refer to as *\$CONF*. All config files are XML and can include other XML files.

The top of this XML tree is *\$CONF/freeswitch.xml*, which includes various XML files that in turn include still more XML files. This arrangement allows you to make changes to the FreeSWITCH

configuration without sifting through a particularly large file.

FreeSWITCH as a Daemon

To start FreeSWITCH as a daemon, use the *-nc* (no console) command-line argument:

```
freeswitch -nc
```

FreeSWITCH will report that it is backgrounding and will give the process ID (PID). To stop FreeSWITCH, simply execute

freeswitch -stop at the Linux prompt. For now, leave FreeSWITCH running and use the *fs_cli* command to connect: You will be greeted with a brief welcome message and a CLI prompt and perhaps some debug messages on the screen. To turn down the debug level, type:

```
/log info
```

This will restrict log messages to the *INFO* level and lower and keep you from being inundated with debug information.

All commands to the *fs_cli* program begin with a slash, so type */help* to see a list of available commands. Anything else typed at the *fs_cli* prompt will be sent to the FreeSWITCH server, with one exception: The ellipsis (...) will not shut down FreeSWITCH from *fs_cli*; rather, it exits the *fs_cli* program. Instead of shutting down FreeSWITCH from within *fs_cli*, exit the program and then issue the *freeswitch -stop* command at the Linux prompt.

Just let FreeSWITCH continue to run in the background while you set up a new user and modify the Dialplan.

Adding a New SIP User

FreeSWITCH comes with 20 users pre-defined, but if you want to add more, just two steps modify the configuration:

- Add a user entry in *\$CONF/directory/default/*
- Modify the Dialplan to allow the new user to be called

To add a new user named "Katherine" with extension number 1500, change directory into *\$CONF/directory/default* and make a copy of *1000.xml*:

Table 1: Dialplan Highlights

Extension	Function
1000-1019	Default extensions
2000-2002	Sample dial groups
30xx	Narrowband conferences (8kHz)
31xx	Wideband conferences (16kHz)
32xx	Ultra-wideband conferences (32kHz)
33xx	CD-quality conferences (48kHz)
4000	Voicemail retrieval
5000	Sample IVR
5900	Call park
5901	Call park retrieval
9888	FreeSWITCH conference
9992	Information application
9996	Echo test
9999	Music on hold

```
cp 1000.xml 1500.xml
```

In an editor, open *1500.xml*, replace *1000* with *1500*, and change the *effective_caller_id_name* to *Katherine*. Your new file should look similar to Listing 1.

Now open *\$CONF/dialplan/default.xml* and locate the Dialplan entry (or "extension") named *Local_Extension*. In the line

```
<condition field=destination_number"
expression="^(10[01][0-9])$">
```

those familiar with regular expressions will recognize that *^(10[01][0-9])\$* means "match exactly four digits – of which the first two are *10*, the next one is *0* or *1*, and the last one is any digit from *0* to *9* – and store the matched value in the variable *\$1*." Put more simply, it matches anything from 1000-1019, inclusive.

To add the new user *Katherine* at extension 1500, use the pipe character, which is a logical OR:

```
<condition field=destination_number"
expression="^(10[01][0-9]|1500)$">
```

Now launch the *fs_cli* program and then type *reloadxml* or press F6. The new extension is now ready for use.

The best way to test it is to set up a softphone (or a hard phone, if you have one) as 1500 with FreeSWITCH. In your X-Lite client, open the SIP account settings and change the display name to *Katherine* and the *User name* and *Autho-*

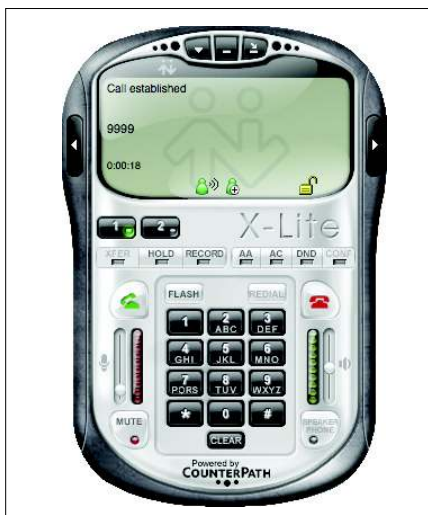


Figure 3: An X-Lite client calling extension 9999.

Table 2: Notable Configuration Files

Subdirectory/File	Description
<code>\$CONF/vars.xml</code>	Defines a number of customizable system variables, like default passwords and codec preference lists
<code>\$CONF/dialplan/default.xml</code>	The bulk of the default Dialplan is configured here
<code>\$CONF/directory/default/*.xml</code>	Individual users are defined here, e.g., <code>1000.xml</code>
<code>\$CONF/sip_profiles/internal.xml</code>	A “SIP profile,” or SIP UA to service an IP/port (usually port 5060)
<code>\$CONF/sip_profiles/external.xml</code>	Another SIP UA to handle outbound registrations to SIP providers (port 5080)
<code>\$CONF/autoload_configs/modules.conf.xml</code>	Modules loaded automatically when FreeSWITCH starts

ization user name fields to 1500 before saving the settings. Your client will momentarily say *registering* and then will say *Ready* in the display. Now you can make calls from your new extension. If you have a second phone, you can register it as 1000 and use it to call 1500.

Outside the Local Switch

One advantage of VoIP is that you can make free or very low cost phone calls on the Internet. FreeSWITCH can route calls to and from the Internet, even if it is behind a NAT device. NAT devices cause all sorts of trouble for SIP calls; however, FreeSWITCH does a good job of working around these issues.

In many cases, the default configuration will work without any changes. A simple test is to call the FreeSWITCH public conference server. The SIP URI of this server is `sip:888@conference.freemwitch.org`, and the default Dialplan will connect your phone if you simply dial 9888.

If your call is successful, you will hear a text-to-speech (TTS) voice announce your caller ID name or number. If you are the only person in the conference, you will also hear a recording saying so.

To call the conference with PortAudio, enter:

```
pa call 2
sip:888@conference.freemwitch.org
```

A number of service providers offer connection services to give you access to and from the public switched telephone network (PSTN) or simple SIP accounts. FreeSWITCH uses the concept of “gateways” to establish connections to these service providers. Gateways are used for

outbound registrations to providers. If you already have a SIP provider, you will want to use the settings for that provider. The FreeSWITCH wiki has sample configurations for a number of providers [5]. In this example, I will show you how to set up a gateway for `iptel.org`, which offers free SIP accounts.

First, create an account on the `iptel.org` website [6] and note your username and password. Next, create a new file called `iptel.org.xml` in `$CONF/sip_profiles/external`. Be sure to put your username and password in the appropriate fields. The file should look like Listing 2.

Now save the file, then at the FreeSWITCH prompt, issue the command:

```
sofia profile external 2
restart reloadxml
```

“Sofia” is the name of the SIP library [7] used by FreeSWITCH and is also the name of the command to perform SIP-related functions. (Just type `sofia` to see the syntax.) The above command tells FreeSWITCH to restart the *external* profile after reloading the XML data. The `reloadxml` is optional; however, if you’ve made changes to the XML data (by creating or editing a file), you will want those changes loaded when restarting the profile.

To see whether your system registered with `iptel.org`, issue the command:

```
sofia status
```

If your registration was successful, you will see a line like the following:

```
iptel gateway 2
sip:myusername@sip.iptel.org REGED
```

The *REGED* state means that the gateway is successfully registered to the provider. If you see *FAIL_WAIT*, then most likely you have a configuration problem.

The last step to receiving calls is to allow incoming calls in the Dialplan. The default configuration has a “public” Dialplan context that treats incoming calls from the Internet as inherently untrusted.

First you need to allow calls to your new `iptel.org` user, then you need to specify where to route those calls. To do

Listing 1: New User 1500.xml

```
01 <include>
02 <user id="1500" mailbox="1500">
03 <params>
04 <param name="password" value="`${default_password}`"/>
05 <param name="vm-password" value="1500"/>
06 </params>
07 <variables>
08 <variable name="toll_allow" value="domestic,international,local"/>
09 <variable name="accountcode" value="1500"/>
10 <variable name="user_context" value="default"/>
11 <variable name="effective_caller_id_name" value="Katherine"/>
12 <variable name="effective_caller_id_number" value="1500"/>
13 <variable name="outbound_caller_id_name" value="`${outbound_caller_
14 <variable name="outbound_caller_id_number" value="`${outbound_caller_
15 <variable name="callgroup" value="techsupport"/>
16 </variables>
17 </user>
18 </include>
```

so, create a new file called `$CONF/dialplan/public/00_inbound_ip.tel.xml` (or make a copy of the example file named `00_inbound_did.xml`). It should look like Listing 3.

Be sure to specify your `iptel.org` username in the regular expression. Your username is the *destination number*. (Destination numbers are not restricted to numeric characters.)

The call is routed to the extension specified in parameters that are fed to the transfer application.

In my case, I routed to the MOH extension for the sake of testing. Also, you can route to your SIP phone. If you are registered as 1000, then the transfer destination is `1000 XML default`. Save the file and then go to the FreeSWITCH prompt and press F6 (or type `reloadxml`) at the FreeSWITCH prompt.

To receive a call on your new account, you will need someone to make a SIP call to your new `iptel.org` SIP URI. In a pinch, you can test with PortAudio. At the FreeSWITCH prompt type:

```
pa call sip:myusername@sip.iptel.org
```

A successful call will ring the destination extension.

High-Quality Codecs

Traditional telephone calls are of marginal sound quality and mobile phone calls are worse. FreeSWITCH delivers

high-quality, voice-quality calls with free and open source codecs. The CELT [8] open source codec is high quality and lossless.

Also included with FreeSWITCH are the Siren [9] codecs from Polycom. These high-quality codecs are closed source but royalty free. The default configuration loads the Siren codecs.

To load the CELT codec, issue the `load mod_celt` command.

One thing to keep in mind is that higher quality codecs could expose limitations in your headset equipment. Most newer headsets and those that are listed as “Skype certified” will generally perform well with the higher quality codecs. Personally, I use a Logitech Clear Chat Pro USB headset with good success.

For testing, I recommend making a call to a friend who also has FreeSWITCH installed.

To start, make a PSTN or mobile phone call, then hang up and immediately make a call with a high-quality codec. With the use of PortAudio, it is easy to test. First, edit `$CONF/autoload_configs/portaudio.xml`. Locate the line `<param name="sample-rate" value="8000"/>`, change the value to `48000`, and save the file.

At the FreeSWITCH prompt issue these commands:

```
reloadxml
reload mod_portaudio
```

Now make a call to your friend or call the FreeSWITCH conference:

```
pa call 9888
```

The audio quality should be much better, especially if you have a fast Internet connection. FreeSWITCH conferences can mix calls of any codec and any sampling rate. In fact, you can call in to the FreeSWITCH conference via the PSTN by dialing +1-213-799-1400 and compare the difference between the high-quality VoIP codecs and the low quality of the PSTN.

Where to Go Next

This brief introduction to FreeSWITCH barely scratches the surface of what this open source software can do. One way to put FreeSWITCH to practical use is presented by IBM’s Rob Smart, who wrote a tutorial called “Home VOIP system using FreeSwitch and a Linksys 3102 voice gateway” [10] on how to set up FreeSWITCH as a home communications server in the UK.

A vibrant and growing community offers support to newcomers and veterans alike, so join this community and continue learning about the amazing possibilities of VoIP. ■

Listing 2: Setting Up a Gateway for a SIP Account

```
01 <include>
02 <gateway name="iptel"> <!-- used in dialstrings -->
03 <param name="username" value="YOUR_USERNAME_HERE"/>
04 <param name="realm" value="iptel.org"/>
05 <param name="password" value="YOUR_PASSWORD_HERE"/>
06 <param name="proxy" value="sip.iptel.org"/>
07 </gateway>
08 </include>
```

Listing 3: 00_inbound_ip.tel.xml

```
01 <include>
02 <extension name="iptel-inbound">
03 <condition field="destination_number" expression="^(YOUR_IPTEL_ORG_
    USERNAME)$">
04 <action application="set" data="domain_name=${domain}"/>
05 <action application="transfer" data="9999 XML default"/>
06 </condition>
07 </extension>
08 </include>
```

INFO

- [1] “How does FreeSWITCH compare to Asterisk?” <http://www.freeswitch.org/node/117>
- [2] FreeSWITCH website: <http://www.freeswitch.org>
- [3] FreeSWITCH downloads: <http://files.freeswitch.org/>
- [4] X-Lite: <http://www.counterpath.com/x-lite.html>
- [5] FreeSWITCH wiki: http://wiki.freeswitch.org/wiki/SIP_Provider_Examples
- [6] iptel.org website: <http://www.iptel.org>
- [7] Sofia-SIP library: <http://sofia-sip.sourceforge.net>
- [8] CELT codec: <http://www.celt-codec.org>
- [9] Siren codecs: http://www.polycom.com/company/about_us/technology/siren_g7221/index.html
- [10] FreeSWITCH tutorial: http://robsmart.co.uk/2009/06/02/freeswitch_linksys3102