Freedom Fone

Installation/testing
Guide 1.5

Part 1:
FreeSWITCH

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1. Changelog

2009-10-13	Change installation to use 1 interface
	Update to debian 5.03
	Correction of reported bugs
	Updated to freeswitch.trunk 15146
	Reordering of document structure
2009-10-14	Removing static dialplan information
	Adding all gsmopen sections
2009-10-15	Adding reference to SVN structure

2. Introduction

The Freedom Fone architecture combines several building blocks:

Voice Channels

A set of voice communication channels for FreeSWITCH that can receive voice calls. Currently we are supporting:

- Skype
- SIP (authenticated and unauthenticated)
- GSM via GSM low cost modem (Mobigater)
- GSM via GSM gateway (VoiceBlue)

FreeSWITCH

FreeSWITCH is the main communication engine that routes calls and executes voice related applications. For example IVRs and the Leave-a-Message application.

Spidermonkey

Spidermonkey is the Javascript implementation of Mozilla (ECMA Script). It supports all the standard Javascript language elements and several other elements that are specific to FreeSWITCH. We use Spidermonkey to run the more complex voice applications.

Event Socket

We use FreeSWITCH event socket in *inbound mode* to announce events to the Freedom Fone dispatcher. By default, Freedom Fone uses the CUSTOM event as a place holder for its own events. In the concrete case of arrival of SMSs, a MESSAGE type of event is generated.

Dispatcher

The dispatcher is a PHP5 daemon that connects to the event socket and listens to Freedom Fone specific events. The dispatcher is responsible for parsing each event, determining its type, and storing it in the application specific event spooler.

Event SQL spooler(s)

The SQL event spoolers store data temporarily between the dispatcher and the CakePHP application.

Cron

The cron engine is responsible for scheduling event processing. The cron engine fetches data from the dispatcher, applies application specific logic, and makes the data available to CakePHP in application specific SQL tables.

CakePHP

CakePHP is a PHP development framework that implements MVC (Model, View, Controller) separation. The Freedom Fone applications are written in CakePHP. All Freedom Fone applications use a common API to interact with their specific event spooler.

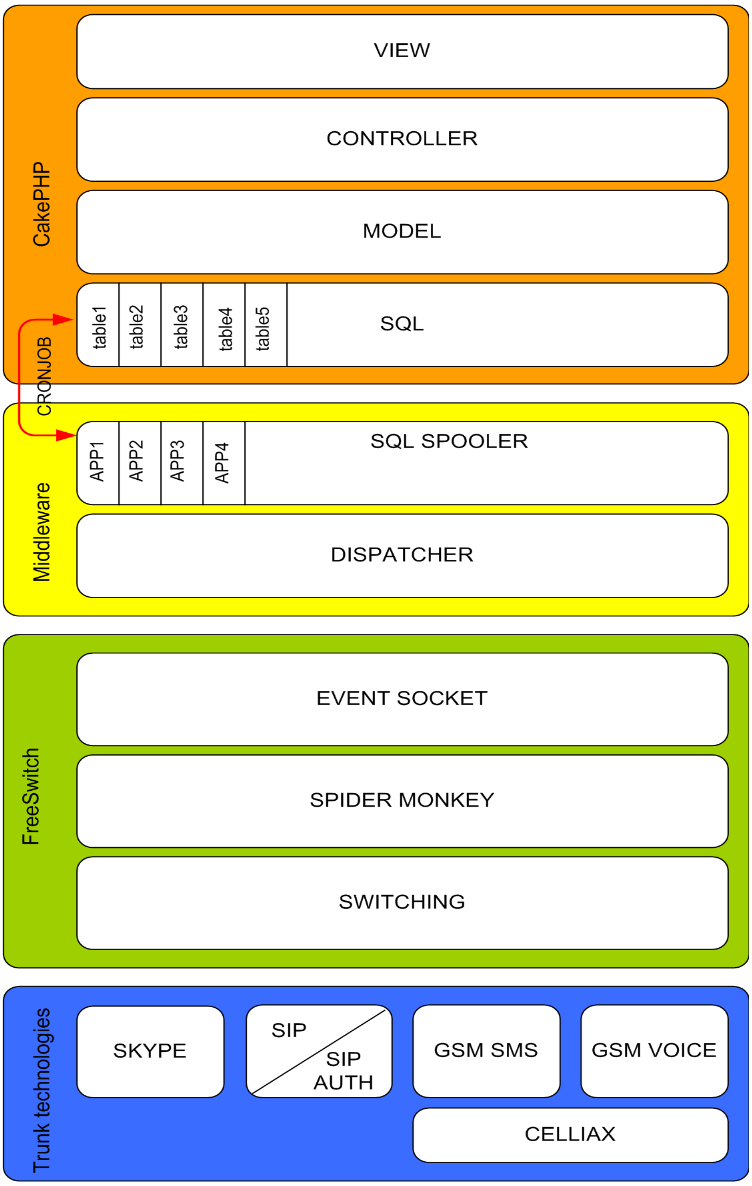

Image 1: The Freedom Fone architecture

3. Architecture walk-through

To illustrate the overall architecture, let us take the example of the ***Leave-a-Message*** application to show the interaction between the different components.

When a voice call is received by FreeSWITCH, the dialplan calls Spider Monkey to execute a Javascript program that manages all the logic of the Leave-a-Message IVR application. Once the application has recorded a message, a CUSTOM event is triggered.

The dispatcher is subscribed to and listens to a set of FreeSWITCH event types. When an event is captured by the dispatcher, the event is converted to an internal Freedom Fone XML format. The XML event is then processed by a set of application specific filters (XSL) and rules before it is finally parsed and inserted in the SQL database. The final result is that the event data is inserted into one or more application specific SQL spoolers.

The cron daemon periodically notifies the Leave-a-Message application(s) that the new events are waiting for processing.

The Leave-a-Message application is configured to run a crontab (every x min) to fetch new data from the spooler, apply logics to the data, and store the information in one or more application specific SQL tables.

The dispatcher acts as a middleware that glues FreeSWITCH events and any Freedom Fone application by a common API.

With this architecture we can allow several applications to process the same audio messages and decide what items of the event information are delivered to each application. For example, we could deliver the audio message to one application with the CallerID information and ignore this field in another application.

4. Installation Overview

We have built the testbed using Ubuntu 9.04 and Debian Lenny.

4.1 Hardware testbed

In our testbed, FreeSWITCH is running on a laptop with one network interface. The network interface (eth0) is connected to the public Internet via DHCP (192.168.46.0/24). The default router is 192.168.46.1 and has NAT enabled.

The Mobigater unit is connected to FreeSWITCH via USB.
By means of a hub, two VoIP phones, a VoiceBlue GSM gateway and a second laptop is connected to the same internal network. VoiceBlue uses a USB interface for configuration via a GUI (Windows operating system only).

The second laptop is used for troubleshooting, and requires Zoiper and Wireshark.

Finally, a third laptop is used to test the application from the public network.

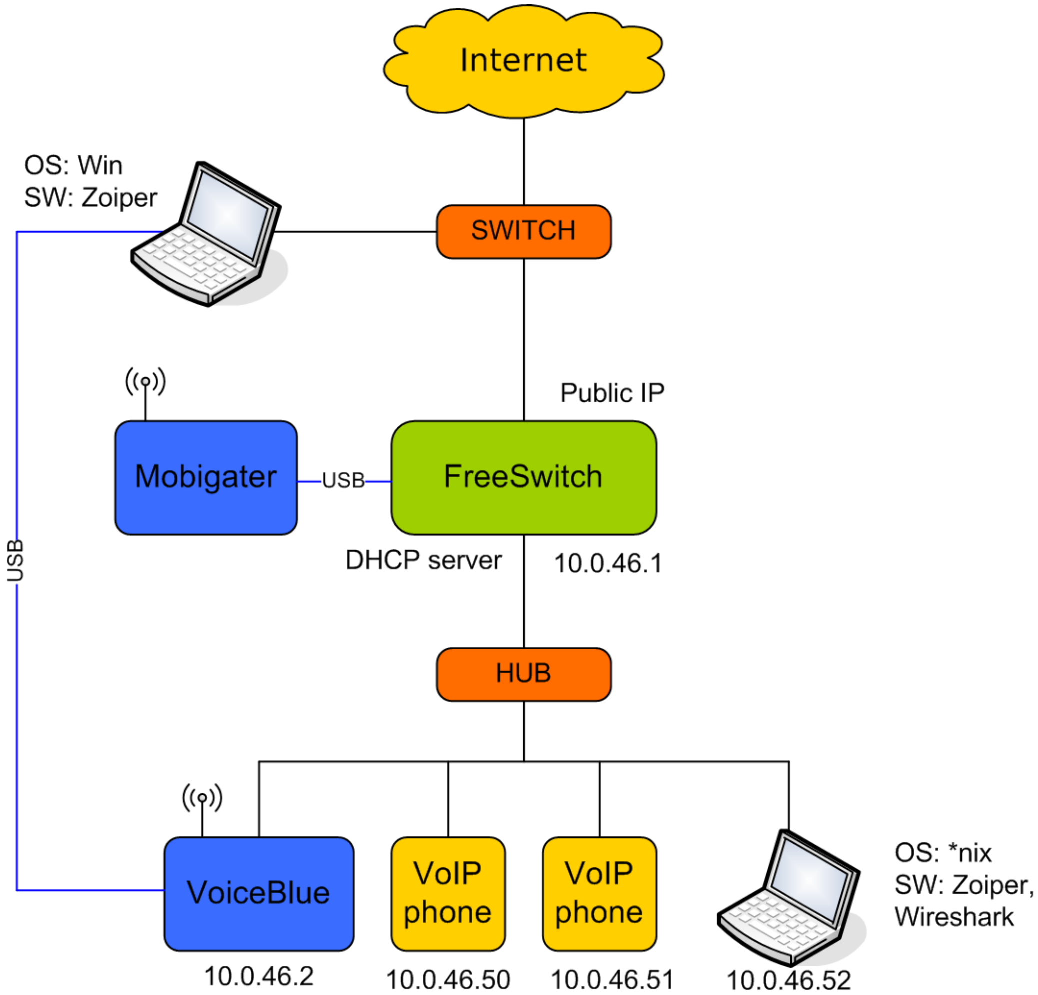

Image 2: Hardware setup of Freedom Fone testbed.

4.2 Installation Flow

Installing the development platform requires the following steps:

1. Install linux distribution and basic development packages
2. Compile FreeSWITCH (stable or trunk)
3. Download Cepstral Text to Speech SDK and register a licensed voice
4. Download Freedom Fone SVN code.
5. Re-compile FreeSWITCH to add XML Curl, gsmopen endpoint and Text to Speech extra modules
6. Enable Cepstral, XML Curl and gsmopen in FreeSWITCH
7. Configure a VoIP client and register to FreeSWITCH
8. Configure the webserver to serve the dialplans and IVRs
9. Install Freedom Fone applications (e.g. the Leave_a_Message Javascript application)
10. Install lame 3.97 to enable wav2mp3 (temporary)
11. Add Freedomfone special macros
12. (for VoiceBlue). Add a external profile to Sofia SIP.
13. (for Voiceblue). Configure VoiceBlue unit

5. Installing Linux and Development Environment

5.1 Install Lenny

Let us prepare a USB bootable image:

Download the boot image from:
```
wget http://http.us.debian.org/debian/dists/stable/main/installer-i386/current/images/hd-media/boot.img.gz
```

Download the net image from:
```
wget http://cdimage.debian.org/debian-cd/current/i386/iso-cd/debian-503-i386-netinst.iso
```

Insert the USB key/flash drive in your Linux box and binary write the boot image to (/dev/sdb). The name of the USB can change, so be sure that is /dev/sdb (dmesg)

```
dmesg | grep removable
[ 2247.207917] sd 2:0:0:0: [sdb] Attached SCSI removable disk


umount /dev/sdb1
zcat boot.img.gz > /dev/sdb
mount /dev/sdb /mnt
cp debian-503-i386-netinst.iso /mnt
umount /mnt
```

Change your BIOS settings and boot from the USB.
Install Debian 5.0 Lenny.

5.2 Install basic development tools

We are going to need a full development environment (gcc, autconf, make, etc) and a few extra tools. The current version uses lame (debian non-free) to convert files from wav to mp3.

Make sure that you have an apt source to download lame in your /etc/apt/sources.list

```
deb http://ftp.sunet.se/pub/os/Linux/distributions/debian-multimedia/
stable main
deb-src http://ftp.sunet.se/pub/os/Linux/distributions/debian-multimedia/
stable main
deb http://ftp.se.debian.org/debian/ lenny main non-free
deb-src http://ftp.se.debian.org/debian/ lenny main non-free

deb http://security.debian.org/ lenny/updates main
deb-src http://security.debian.org/ lenny/updates main
```

```
deb http://volatile.debian.org/debian-volatile lenny/volatile main
deb-src http://volatile.debian.org/debian-volatile lenny/volatile main
```

Download and install the development environment

```
apt-get update
apt-get upgrade
```

Development environment

```
apt-get install autoconf libtool build-essential libncurses5-dev

apt-get install subversion openssh-server vim wireshark
```

LAMP stack

```
apt-get install apache2 php5 php5-curl php5-cli php5-mysql php5-xsl mysql-
server libapache2-mod-php5
```

Freeswitch audio conversion, inode watcher

```
apt-get install lame iwatch
```

gsm open endpoint

```
apt-get install libasound2-dev libx11-dev
```

Approximately 75 MB download
You can press ENTER when asked for the mysql root password.

6. Freedom Fone Code

To build Freedom Fone you need the following components

- Linux and a development environment (gcc, autoconf, make, etc)
- Cepstral SDK
- FreeSWITCH
- Cake PHP SDK
- Freedom Fone specific architecture components and applications (SVN)

Freedom Fone architecture and applications are named based on the African Wild Dog or Lycaon. For those interested in this curious canid check:
http://en.wikipedia.org/wiki/African_Wild_Dog

There are five recognized subspecies of Lycaon that inspired Freedom Fone releases:

- Lycaon pictus pictus (release 1.0 – November 2009)
- Lycaon pictus lupinus
- Lycaon pictus manguensis
- Lycaon pictus sharicus
- Lycaon pictus somalicus

When checking out from the repository have in mind the following release tags: pictus, lupinus, manguensis, sharicus and somalicus

6.1 Getting all the software

Let us get all the software we need under /usr/src

Freeswitch
cd /usr/src
```
svn co http://svn.freeswitch.org:/svn/freeswitch/trunk pictus
```

Cepstral SDK
```
cd /usr/src
wget    http://downloads.cepstral.com/cepstral/i386-linux/Cepstral_Allison-8kHz_i386-linux_5.1.0.tar.gz

tar zxvf Cepstral_Allison-8kHz_i386-linux_5.1.0.tar.gz
```

Freedom Fone SVN

```
cd /usr/src
svn co https://dev.freedomfone.org/svn/freedomfone pictus
```

Cake PHP MVC

```
cd /usr/src
wget http://cakeforge.org/frs/download.php/734/cake_1.2.5.tar.gz
```

7. Installing FreeSWITCH

During the development we are building FreeSWITCH from trunk

7.1 Installing FreeSWITCH (trunk version 15162)

Check out the latest version using subversion

```
cd freeswitch.trunk
sh bootstrap.sh   (8 mins in a VIA C7D)
./configure       (12 mins in a VIA C7D)
make
make install      (30 mins in a VIA C7D)
make cd-sounds-install
make cd-moh-install
```

This will compile from trunk and leave you a working installation in /usr/local/freeswitch

Compilation can take as much as 15 minutes in a 3.x GHz CPU system, go for a big latte!

```
make current
```

Will make sure that you get the latest freeswitch compiled at anytime

7.2 Installing extra modules

Freedom Fone uses **three extra modules** that are not in the default configuration: mod_cepstral, mod_xml_curl and mod_gsmopen

1. mod_cepstral is used to convert text to speech.
2. xml_curl is used to retrieve dialplan and configuration information dynamically from a web site.
3. mod_gsmopen is used to support our gsm usb modems

7.2.1 Install Cepstral

Start by installing Cepstral. You will need to buy a license to get rid of the audio propaganda :). In the Freedom Fone testbed we have used Allison voice for telephone (notice 8 kHz voices!). Download Allison voice version 5.1.0 for telephony (8 kHz):

```
cd /usr/src/Cepstral_Allison-8kHz_i386-linux_5.1.0#
./install.sh
```

Configuration options follows:

```
Install into what directory? [/opt/swift]
/opt/swift does not exist.  Create it? ([n]/y) y

Swift will be installed in the following directories:

  Voices in /opt/swift/voices
  Shared libraries in /opt/swift/lib
  Binaries in /opt/swift/bin
  Configuration file in /opt/swift/etc
  Header files in /opt/swift/include
  Examples in /opt/swift/examples
  Sound effects filters in /opt/swift/sfx
  Documentation in /opt/swift/doc
```

Edit the file /etc/ld.so.conf.d/cepstral.conf and include /opt/swift/lib then run:

```
ldconfig
```

7.2.2 Registering/Licensing Cepstral

```
open46:~# swift --reg-voice

               Your Name: Alberto FreeSWITCHER

 Company (if applicable): IT46 HB

                   Voice: Allison-8kHz

             License Key: 52-1c5349-5c0f08-8dfbaf-9c38c6-xxxxxx

The information you have entered appears to be valid.
Thank you for purchasing Cepstral Allison-8kHz.
```

7.2.3 Enable Cepstral

Enable the Cepstral module by editing /usr/local/freeswitch/conf/autoload_configs/ modules.conf and uncommenting the line:

```
<load module="mod_cepstral"/>
```

7.2.4 Install gsm open

The "gsm open channel" (endpoint) is not part of FreeSWITCH trunk code (yet!). Make a symbolic link from freeswitch source code folder to Freedom Fone repository (pictus) as follows:

Add gsmopen endpoint channel

```
ln -s /usr/src/freeswitch.trunk/src/mod/endpoints/mod_gsmopen
 /usr/src/pictus/freeswitch/src/mod/endpoints/mod_gsmopen
```

7.2.5 Enable gsm open

Enable gsm endpoint by editing /usr/local/freeswitch/conf/autoload_configs/ modules.conf and adding the line:

```
<load module="mod_gsmopen"/>
```

7.2.6 Checking GSM modem

Make sure that Mobigater is properly connected:

```
#aplay -l
freedomfone:/usr/src/pictus/installer# aplay -l
**** List of PLAYBACK Hardware Devices ****
card 0: V8237 [VIA 8237], device 0: VIA 8237 [VIA 8237]
  Subdevices: 4/4
  Subdevice #0: subdevice #0
  Subdevice #1: subdevice #1
  Subdevice #2: subdevice #2
  Subdevice #3: subdevice #3
card 0: V8237 [VIA 8237], device 1: VIA 8237 [VIA 8237]
  Subdevices: 1/1
  Subdevice #0: subdevice #0
card 1: default [C-Media USB Headphone Set  ], device 0: USB Audio [USB
Audio]
  Subdevices: 1/1
  Subdevice #0: subdevice #0
```

Notice the card 1 that corresponds to the Audio Device

7.2.7 Build and install the modules

Let us compile the modules:

Edit /usr/src/freeswitch.trunk/modules.conf and remove the # from the following two lines

```
#asr_tts/mod_cepstral
...
#xml_int/mod_xml_curl
```

and add the new channel mod_gsmopen

```
endpoints/mod_gsmopen
```

```
(Re)compile freeswitch modules now:

cd /usr/src/freeswitch.trunk/ (or freeswitch-1.0.4)
make
make install
```

8. Creating dynamic dialplan/configuration

Instead of using the static dialplan from:
/usr/local/freeswitch/conf/dialplan/default.xml

Freedom Fone dialplans and configurations as IVR (ivr.conf) will be generated on the fly by means of XML Curl. In XML Curl FreeSWITCH will request from a webserver (by means of a specially crafted POST method) the dialplan and configurations.

8.1 Enabling XML Curl

Edit the /usr/local/freeswitch/conf/autoload_configs/xml_curl.conf.xml to set the webserver URL where the information is going to be requested and the bindings

```
<configuration name="xml_curl.conf" description="cURL XML Gateway">
  <bindings>
    <binding name="Freedomfone Dialplan Configuration Retrival">
<param name="gateway-url" value="http://localhost/freedomfone/xml_curl/"
bindings="configuration dialplan"/>
    </binding>
  </bindings>
</configuration>
```

Enable the XML Curl module buy editing /usr/local/freeswitch/conf/autoload_configs/modules.conf and uncommenting the line:

```
<load module="mod_xml_curl"/>
```

Notice the path: http://localhost/**freedomfone/xml_curl/.**

FreeSWITCH will request configuration from:
http://localhost/freedomfone/xml_curl/

This path needs to be point inside of our GUI component
/var/www/freedomfone/app/webroot/xml_curl

a) Go to your web server root directory /var/www and create the folder freedomfone/app/webroot/

Create a symbolic link to xml_curl in the SVN

```
ln -s /usr/src/pictus/xml_curl/ /var/www/freedomfone/app/webroot/xml_curl
```

b) Create a .htaccess file under /var/www/freedomfone

```
<IfModule mod_rewrite.c>
  RewriteEngine on
  RewriteRule    ^$ app/webroot/    [L]
  RewriteRule    (.*) app/webroot/$1 [L]
</IfModule>
```

c) Make sure that mod rewrite for apache is enabled

```
a2enmod rewrite
```

d) Copy the apache2 configuration file for the site

```
cp /usr/src/pictus/gui/apache2/freedomfone /etc/apache2/sites-available/
```

e) Restart Apache

```
a2enmod rewrite

a2dissite default

a2ensite freedomfone

/etc/init.d/apache2 reload
```

8.2 Adding the FreeSWITCH XML Curl index page

To evaluate the solution we have created a simple version of the dynamic configuration PHP engine that returns static files depending on the variables requested in the POST method.

```
<?php

header('Content-Type: text/xml');

$postvar1 = $_REQUEST['key_value'];
$postvar2 = $_REQUEST['section'];

if ( $postvar1 == "ivr.conf" )
        {
        $static_page = file_get_contents('ivr.xml');
        die($static_page);
        }
        elseif ( $postvar2 == "dialplan" )
        {
        $static_page = file_get_contents('dialplan.xml');
        die($static_page);
        }
        else
        {
        $static_page = file_get_contents('notfound.xml');
        die($static_page);
```

```
        }
?>
```

Examples of the static files are available in the svn in the xml_curl folder. Link the webserver folder /var/www/freedomfone/xml_curl → /usr/src/pictus/xml_curl as described in the previous section.

```
#ln -s /usr/src/pictus/xml_curl/ /var/www/freedomfone/app/webroot/xml_curl
```

9. First run

You can test that FreeSWITCH runs by typing:

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

You exit the application with the command:

```
 ...
```

In case you wonder, Yes! Three dot-dot-dot is a command.

You can always run freeswitch in background and/or force no NAT autodetection

```
/usr/local/freeswitch/bin/freeswitch -nc -nonat
```

/usr/local/freeswitch/bin/fs_cli

9.1 FreeSWITCH in background and client

You can run FreeSWITCH in background:

```
/usr/local/freeswitch/bin/freeswitch -nc
```

and connect to FreeSWITCH using the fs_cli cient

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

           _____ ____     ____ _     ___
          |  ___/ ___|   / ___| |   |_ _|
          | |_  \___ \  | |   | |    | |
          |  _|  ___) | | |___| |___ | |
          |_|   |____/   \____|_____|___|

*****************************************************

* Anthony Minessale II, Ken Rice, Michael Jerris    *

* FreeSWITCH (http://www.freeswitch.org)            *

* Brought to you by ClueCon http://www.cluecon.com/ *
```

```
*****************************************************

Type /help <enter> to see a list of commands
```

9.2 Setting up debug

In the development you will have to enable debug in:

1. FreeSWITCH console message (include debugging coming from our Javascript scripts)
2. SIP (known as Sofia in FreeSWITCH)
3. xml_curl (to debug the dynamic look ups of dialplans and configuration)

The following commands turn ON the debug in console, sip and xml_curl

```
freeswitch@internal> console loglevel 7

+OK console log level set to DEBUG

freeswitch@internal> sofia loglevel all 9

Sofia log level for component [all] has been set to [9]

freeswitch@internal> xml_curl debug_on
```

9.3 Debugging SIP messages

```
TPORT_LOG=1 ./freeswitch

Press F8 for console debug
```

9.4 Using Gdb

gdb bin/freeswitch core.XXX

```
gdb>bt
gdb>bt all
gdb>thread apply all bt
gdb>thread apply all bt full
```

9.5 Testing Cepstral

Configure a SIP client in the 192.168.46.0/24 network and register it, you can use extension 1000 with password 1234). If you do not have a VoIP Phone use a SIP client (zoiper, xlite, etc). **WARNING! Just make sure that STUN support is disable in Zoiper**

You can verify that the phone is registered by issuing the command:

```
freeswitch@internal> sofia status profile internal

=============================================================================================
====

Name                    internal

Domain Name             N/A

DBName                  sofia_reg_internal

Pres Hosts

Dialplan                XML

Context                 public

Challenge Realm         auto_from

RTP-IP                  192.168.46.238

Ext-RTP-IP              192.168.46.238

SIP-IP                  192.168.46.238

Ext-SIP-IP              192.168.46.238

URL                     sip:mod_sofia@192.168.46.238:5060

BIND-URL                sip:mod_sofia@192.168.46.238:5060

HOLD-MUSIC              local_stream://moh

OUTBOUND-PROXY          N/A

CODECS                  G7221@32000h,G7221@16000h,G722,PCMU,PCMA,GSM

TEL-EVENT               101

DTMF-MODE               rfc2833

CNG                     13

SESSION-TO              0

MAX-DIALOG              0

NOMEDIA                 false

LATE-NEG                false

PROXY-MEDIA             false

AGGRESSIVENAT           false
```

```
STUN-ENABLED            true
STUN-AUTO-DISABLE       false
CALLS-IN                0
FAILED-CALLS-IN         0
CALLS-OUT               0
FAILED-CALLS-OUT        0

Registrations:
=============================================================================================
Call-ID:       OGIxYThjMTgzZDBhNmU2NzdkY2ZkMTkwNDQ5ZTA1MTk.
User:          1003@192.168.46.238
Contact:       "user" <sip:1003@192.168.46.72:5060;rinstance=863c0dab613b6c09;transport=UDP>
Agent:         Zoiper rev.4688
Status:        Registered(UDP)(unknown) EXP(2009-10-14 14:06:20)
Host:          freedomfone
IP:            192.168.46.72
Port:          5060
Auth-User:     1003
Auth-Realm:    192.168.46.238
MWI-Account:   1003@192.168.46.238

Call-ID:       a016a029-57da6910@192.168.46.243
User:          1001@192.168.46.238
Contact:       1001 <sip:1001@192.168.46.243:5060>
Agent:         Sipura/SPA1001-2.0.13(SEg)
Status:        Registered(UDP)(unknown) EXP(2009-10-14 14:21:53)
Host:          freedomfone
IP:            192.168.46.243
Port:          5060
Auth-User:     1001
Auth-Realm:    192.168.46.238
MWI-Account:   1001@192.168.46.238

=============================================================================================

freeswitch@internal>
```

In the example you can see two phones registered with extension 1001 and 1003, coming from 192.168.46.72 and 192.168.46.243

Now you can place a call to extension "9000" and test Cepstral Text-to-Speech (TTS)

10. Adding applications to FreeSWITCH

The first release of Freedom Fone comes with four applications:

1. Leave a message application is composed of a Javascript that runs in FreeSWITCH and a CakePHP LAMP application
2. The tickle/callback service
3. The IVR service
4. A SMS pool application that talks with the dispatcher without any Javascript handling.

10.1 Install the scripts

Make a symbolic link to the script folder in Freedom Fone SVN

```
ln -s /usr/src/pictus/freeswitch/scripts/
/usr/local/freeswitch/scripts/freedomfone
```

scripts folders

```
|
|-- dtmf

|   `-- main.js

|-- echo

|   |-- README.mon

|   `-- main.js

|
|
```

```
|
|-- ivr
|   `-- 100
|       |-- conf
|       |-- ivr
|       `-- nodes
|-- leave_message
|   |-- 100
|   |   |-- audio_menu
|   |   |   `-- README
|   |   |-- conf
|   |   |   |-- 100.conf -> default.conf
|   |   |   |-- README
|   |   |   `-- default.conf
|   |   `-- messages
|   |       |-- 250cfbae-9329-11de-9565-7344fd34a3d8.meta
|   |       |-- 54df7258-9324-11de-9981-17076624b6e4.meta
|   |       |-- 5a7e0854-932a-11de-9565-7344fd34a3d8.meta
|   |       `-- ed91096a-9321-11de-9981-17076624b6e4.meta
|   `-- main.js
|-- sms
|   |-- README
|   `-- main.js
`-- tickle
    `-- main.js
```

Notice the tree structure, scripts are named "main.js" inside of every script folder

1.1 Add Freedom Fone macros

Freedom Fone's Leave-a-Message script uses one special text-to-speech macro.

Edit the file /usr/local/freeswitch/conf/lang/en/en.xml:

```
<include>

   <language name="en" sound-path="$${base_dir}/sounds/en/us/callie" tts-
engine="cepstral" tts-voice="allison">

    <X-PRE-PROCESS cmd="include" data="demo/*.xml"/>
    <X-PRE-PROCESS cmd="include" data="freedomfone/*.xml"/> <
 <X-PRE-PROCESS cmd="include" data="vm/sounds.xml"/>
  </language>

</include>
```

Notice the default TTS voice is set to Allison.
Create the file /usr/local/freeswitch/conf/lang/en/freedomfone/default.xml and include the macro “speak”

```
<macro name="speak">
     <input pattern="(.*)">
       <match>
         <action function="speak-text" data="$1"/>
       </match>
     </input>
</macro>
```

1.2 Add lame to handle wav2mp3 conversion

In the current version, files get recorded in .WAV format and are dynamically converted to MP3 within the Leave-a-Message Javascript. This functionality will be moved to an audio conversion engine in future versions.

Due to some limitations of FreeSWITCH stack size, it is necessary to use lame 3.97 to do this conversion!. Version 3.98 will Segfault!

```
cp /usr/src/pictus/extras/lame397 /usr/local/freeswitch/bin
```

FIXME! This is hack and next release will address the conversion in a separate component!

1.3 Ready to test

Call the extension "**2000**". The Javascript freedomfone/leave_message/main.js should be triggered with an identifier/instantiation "100".

Check that the audio files get recorded at:

```
cd /usr/local/freeswitch/scripts/freedomfone/leave_message/100/messages/
ls -al
```

```
messages/
|-- dc50dc9a-914e-11de-a33a-edf6d27cc3dd.meta
|-- dc50dc9a-914e-11de-a33a-edf6d27cc3dd.mp3
|-- dc50dc9a-914e-11de-a33a-edf6d27cc3dd.wav
|-- ed91096a-9321-11de-9981-17076624b6e4.meta
|-- ed91096a-9321-11de-9981-17076624b6e4.mp3
`-- ed91096a-9321-11de-9981-17076624b6e4.wav
```

1.4 Monitoring leave a message events

You can connect to the event socket to listen to our leave a message events

```
telnet localhost 8021
Trying 127.0.0.1...
Connected to localhost.
Escape character is '^]'.
Content-Type: auth/request


auth ClueCon

Content-Type: command/reply
Reply-Text: +OK accepted

events plain custom lmsm

Content-Type: command/reply
Reply-Text: +OK event listener enabled plain

Content-Length: 684
Content-Type: text/event-plain

Event-Subclass: lmsm
Event-Name: CUSTOM
Core-UUID: 880f63fa-9481-11de-8a74-3b9705bff4ae
FreeSWITCH-Hostname: open46
FreeSWITCH-IPv4: 85.225.10.37                 <---- FS Server
```

```
FreeSWITCH-IPv6: %3A%3A1
Event-Date-Local: 2009-08-29%2012%3A37%3A42
Event-Date-GMT: Sat,%2029%20Aug%202009%2010%3A37%3A42%20GMT
Event-Date-Timestamp: 1251542262283879
Event-Calling-File: mod_spidermonkey.c
Event-Calling-Function: event_construct
Event-Calling-Line-Number: 502
Event-Subclass: lmsm                            <--- Event subclass
FF-LmID: 100
FF-LmURI: http%3A//habibi.ath.cx/freedomfone/leave_message/100/messages/
FF-FileID: f216141e-9487-11de-8a74-3b9705bff4ae <--- File unique ID
FF-CallerID: 1000
FF-CallerName: aep
FF-StartTimeEpoch: 1251542261016
FF-FinishTimeEpoch: 1251542262244

exit

Content-Type: command/reply
Reply-Text: +OK bye

Content-Type: text/disconnect-notice
Content-Length: 67

Disconnected, goodbye.
See you at ClueCon!
```

You can see that the event carries lots of information including the CallerID, timestamps and a URI to the place where the application can find the message:

http://habibi.ath.cx/freedomfone/leave_message/100/messages/

You can also see that Freedom Fone fields are tagged with the prefix **FF-**

11. Notes

11.1 About Dialplan

11.2 Apache Security and Access Control

11.3 The Freedom Fone SVN code

12. Adding VoiceBlue Lite as GSM gateway

In the svn there is a configuration file that can be used as a reference to add VoiceBlue Lite to the Freedom Fone testbed.

The file to configure the unit is available in the freeswitch/channels/voiceblue/conf folder

Things to take into consideration:

1. The VoiceBlue unit needs to be configured in the private network 10.0.46.0. In the example we use 10.0.46.2
2. We register the unit using one of the internal directory accounts user:1004 pass:1234
3. We configure the VoiceBlue to add a DISA prefix "gsm1"
4. We configure the VoiceBlue not to add any CLIP
5. We make sure that numbers starting with 07 and followed by 8 digits are accepted by VoiceBlue 07/8. CALL REJECTED problems related to Table prefix information
6. Calls from IP (FS) -> GSM take place via a sofia SIP gateway, voiceblue does not request any Auth information to place the calls.

```
<include>
  <gateway name="2n">
     <param name="username" value="2n"/>    <--- this information is never
used
     <param name="password" value="2n"/>    <--- this information is never
used
    <param name="realm" value="85.225.10.37"/>
    <param name="register" value="false"/>
    <param name="caller-id-in-from" value="true"/>
    <param name="proxy" value="10.0.46.2"/>
  </gateway>
</include>
```

13. Known errors

Error with occupied port
[ERR] sofia.c:871 Error Creating SIP UA for profile: internal
Something is running in port 5060

Error **libswift.so.5: cannot open shared object file: No such file or directory**
You forgot to add /opt/swift/lib to /etc/ld.conf or run ldconfig

I do not hear anything!:)
Most of the audio problems we have experienced are related to NAT and RTP. This problems are difficult to troubleshoot because they are very dependent of your scenario. One simple tip, using sofia status commands, identify that the IP addresses for RTP and SIP make sense!

14. References

Javascript/Spidermonkey

http://wiki.freeswitch.org/wiki/Javascript

http://wiki.freeswitch.org/wiki/Category:Javascript

Event Socket

http://wiki.freeswitch.org/wiki/Event_Socket

http://wiki.freeswitch.org/wiki/Event_list#CUSTOM

What script should I use?

http://wiki.freeswitch.org/wiki/Which_scripting_language_should_I_use%3F

15. Annex 1: VoiceBlue Tickle Debug

This is how we debugged VoiceBlue Lite for the tickle application

```
REGISTER sip:85.225.10.37:5060 SIP/2.0
Max-Forwards: 70
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-79003596-CX04
From: <sip:1004@85.225.10.37:5060>;tag=0050C281FBCC-175014066
To: <sip:1004@85.225.10.37:5060>
Call-ID: 0050C281FBCC-158019090@10.0.46.2
CSeq: 2 REGISTER
Contact: <sip:1004@10.0.46.2:5060;user=phone;phone-context=1004>
Expires: 600
User-Agent: 2N VoiceBlue V-02.07.35b02
Content-Length: 0

SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-79003596-CX04
From: <sip:1004@85.225.10.37:5060>;tag=0050C281FBCC-175014066
To: <sip:1004@85.225.10.37:5060>;tag=m76m0p5e3SSUa
Call-ID: 0050C281FBCC-158019090@10.0.46.2
CSeq: 2 REGISTER
User-Agent: FreeSWITCH-mod_sofia/1.0.4-exported
Allow:  INVITE,  ACK,  BYE,  CANCEL,  OPTIONS,  PRACK,  MESSAGE,  SUBSCRIBE,
NOTIFY, REFER, UPDATE, REGISTER, INFO, PUBLISH
Supported: timer, precondition, path, replaces
WWW-Authenticate:  Digest  realm="85.225.10.37",  nonce="03c9a2e6-948f-11de-
a987-c77e6ca0cf56", algorithm=MD5, qop="auth"
Content-Length: 0

REGISTER sip:85.225.10.37:5060 SIP/2.0
Max-Forwards: 70
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-140013245-CX04
From: <sip:1004@85.225.10.37:5060>;tag=0050C281FBCC-175014066
To: <sip:1004@85.225.10.37:5060>
Call-ID: 0050C281FBCC-158019090@10.0.46.2
CSeq: 4 REGISTER
Contact: <sip:1004@10.0.46.2:5060;user=phone;phone-context=1004>
Expires: 600
User-Agent: 2N VoiceBlue V-02.07.35b02
Authorization:                                                         Digest
username="1004",realm="85.225.10.37",uri="sip:85.225.10.37",response="14c0
aeef94a7405f9f9fdcb18a698ab5",nonce="03c9a2e6-948f-11de-a987-
c77e6ca0cf56",qop="auth",cnonce="33bfc508",nc=00000001,algorithm=MD5
Content-Length: 0

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-140013245-CX04
From: <sip:1004@85.225.10.37:5060>;tag=0050C281FBCC-175014066
To: <sip:1004@85.225.10.37:5060>;tag=Ng0D2Hpj02Fep
Call-ID: 0050C281FBCC-158019090@10.0.46.2
CSeq: 4 REGISTER
Contact:                         <sip:1004@10.0.46.2:5060;user=phone;phone-
context=1004>;expires=600
Date: Sat, 29 Aug 2009 11:28:03 GMT
```

```
User-Agent: FreeSWITCH-mod_sofia/1.0.4-exported
Allow:  INVITE,  ACK,  BYE,  CANCEL,  OPTIONS,  PRACK,  MESSAGE,  SUBSCRIBE,
NOTIFY, REFER, UPDATE, REGISTER, INFO, PUBLISH
Supported: timer, precondition, path, replaces
Content-Length: 0

NOTIFY                            sip:1004@10.0.46.2:5060;user=phone;phone-
context=1004;fs_nat=yes;fs_path=sip:1004%4010.0.46.2:5060%3Buser%3Dphone
%3Bphone-context%3D1004 SIP/2.0
Via: SIP/2.0/UDP 85.225.10.37;rport;branch=z9hG4bKgZ1NZjgt9KX9B
Route: <sip:1004@10.0.46.2:5060>;user=phone;phone-context=1004
Max-Forwards: 70
From: <sip:1004@85.225.10.37>;tag=pSS63c7NXB60H
To: <sip:1004@85.225.10.37>
Call-ID: db34c4fa-0f31-122d-4c84-001372cbb7a9
CSeq: 119654305 NOTIFY
Contact: <sip:mod_sofia@85.225.10.37:5060>
User-Agent: FreeSWITCH-mod_sofia/1.0.4-exported
Allow:  INVITE,  ACK,  BYE,  CANCEL,  OPTIONS,  PRACK,  MESSAGE,  SUBSCRIBE,
NOTIFY, REFER, UPDATE, REGISTER, INFO, PUBLISH
Supported: timer, precondition, path, replaces
Event: message-summary
Allow-Events:  talk,  presence,  dialog,  call-info,  sla,  include-session-
description, presence.winfo, message-summary, refer
Subscription-State: terminated;reason=timeout
Content-Type: application/simple-message-summary
Content-Length: 64

Messages-Waiting: no
Message-Account: sip:1004@85.225.10.37

SIP/2.0 481 Call/Transaction Does Not Exist
Via: SIP/2.0/UDP 85.225.10.37;rport;branch=z9hG4bKgZ1NZjgt9KX9B
Route: <sip:1004@10.0.46.2:5060>;user=phone;phone-context=1004
Max-Forwards: 70
From: <sip:1004@85.225.10.37>;tag=pSS63c7NXB60H
To: <sip:1004@85.225.10.37>
Call-ID: db34c4fa-0f31-122d-4c84-001372cbb7a9
CSeq: 119654305 NOTIFY
Contact: <sip:mod_sofia@85.225.10.37:5060>
User-Agent: FreeSWITCH-mod_sofia/1.0.4-exported
Allow:  INVITE,  ACK,  BYE,  CANCEL,  OPTIONS,  PRACK,  MESSAGE,  SUBSCRIBE,
NOTIFY, REFER, UPDATE, REGISTER, INFO, PUBLISH
Supported: timer, precondition, path, replaces
Event: message-summary
Allow-Events:  talk,  presence,  dialog,  call-info,  sla,  include-session-
description, presence.winfo, message-summary, refer
Subscription-State: terminated;reason=timeout
Content-Type: application/simple-message-summary
Content-Length: 64

Messages-Waiting: no
Message-Account: sip:1004@85.225.10.37

===============================================
Registration as extension 1004 is now completed
===============================================
```

```
****************************************************
We place a call that is routed to the tickle service
The 2N Voice Gateway performs a DISA to extension URI gsm1
That is the way we can identify the incoming module
****************************************************



INVITE sip:gsm1@85.225.10.37:5060 SIP/2.0
Max-Forwards: 70
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-167011669-CX00
From: <sip:0702867989@85.225.10.37:5060>;tag=0050C281FBCC-73020710
To: <sip:gsm1@85.225.10.37:5060>
Call-ID: 0050C281FBCC-37007171@10.0.46.2
CSeq: 2 INVITE
Contact: <sip:0702867989@10.0.46.2:5060>
User-Agent: 2N VoiceBlue V-02.07.35b02
Allow: INVITE, BYE, ACK, CANCEL, OPTIONS, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: 239

v=0
o=VoiceBlue 11371 16836 IN IP4 10.0.46.2
s=GSM call
c=IN IP4 10.0.46.2
t=0 0
m=audio 16000 RTP/AVP 8 0 18 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11



****************************************************
Freeswitch requires authentication to route the call
****************************************************


SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-167011669-CX00
From: <sip:0702867989@85.225.10.37:5060>;tag=0050C281FBCC-73020710
To: <sip:gsm1@85.225.10.37:5060>
Call-ID: 0050C281FBCC-37007171@10.0.46.2
CSeq: 2 INVITE
User-Agent: FreeSWITCH-mod_sofia/1.0.4-exported
Content-Length: 0

SIP/2.0 407 Proxy Authentication Required
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-167011669-CX00
From: <sip:0702867989@85.225.10.37:5060>;tag=0050C281FBCC-73020710
To: <sip:gsm1@85.225.10.37:5060>;tag=Q2jZ57QStmvKD
Call-ID: 0050C281FBCC-37007171@10.0.46.2
CSeq: 2 INVITE
```

```
User-Agent: FreeSWITCH-mod_sofia/1.0.4-exported
Accept: application/sdp
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, PRACK, MESSAGE, SUBSCRIBE,
NOTIFY, REFER, UPDATE, REGISTER, INFO, PUBLISH
Supported: timer, precondition, path, replaces
Allow-Events: talk, presence, dialog, call-info, sla, include-session-
description, presence.winfo, message-summary, refer
Proxy-Authenticate: Digest realm="85.225.10.37", nonce="2ccf3e80-948f-
11de-a987-c77e6ca0cf56", algorithm=MD5, qop="auth"
Content-Length: 0

ACK sip:85.225.10.37:5060 SIP/2.0
Max-Forwards: 70
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-167011669-CX00
From: <sip:0702867989@85.225.10.37:5060>;tag=0050C281FBCC-73020710
To: <sip:gsm1@85.225.10.37:5060>
Call-ID: 0050C281FBCC-37007171@10.0.46.2
CSeq: 2 ACK
Contact: <sip:0702867989@10.0.46.2:5060>
User-Agent: 2N VoiceBlue V-02.07.35b02
Allow: INVITE, BYE, ACK, CANCEL, OPTIONS, REFER, NOTIFY
Content-Length: 0

*****************************************************
VoiceBlue sends the INVITE with Authentication
*****************************************************

INVITE sip:gsm1@85.225.10.37:5060 SIP/2.0
Max-Forwards: 70
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-4008835-CX00
From: <sip:0702867989@85.225.10.37:5060>;tag=0050C281FBCC-73020710
To: <sip:gsm1@85.225.10.37:5060>
Call-ID: 0050C281FBCC-37007171@10.0.46.2
CSeq: 4 INVITE
Contact: <sip:0702867989@10.0.46.2:5060>
User-Agent: 2N VoiceBlue V-02.07.35b02
Allow: INVITE, BYE, ACK, CANCEL, OPTIONS, REFER, NOTIFY
Proxy-Authorization: Digest
username="1004",realm="85.225.10.37",uri="sip:85.225.10.37",response="f2eb
5ed547a60980238db9c751fc4f1e",nonce="2ccf3e80-948f-11de-a987-
c77e6ca0cf56",qop="auth",cnonce="847c6ba1",nc=00000001,algorithm=MD5
Content-Type: application/sdp
Content-Length: 239

v=0
o=VoiceBlue 11371 16836 IN IP4 10.0.46.2
s=GSM call
c=IN IP4 10.0.46.2
t=0 0
m=audio 16000 RTP/AVP 8 0 18 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-4008835-CX00
From: <sip:0702867989@85.225.10.37:5060>;tag=0050C281FBCC-73020710
To: <sip:gsm1@85.225.10.37:5060>
Call-ID: 0050C281FBCC-37007171@10.0.46.2
CSeq: 4 INVITE
User-Agent: FreeSWITCH-mod_sofia/1.0.4-exported
Content-Length: 0

*****************************************************
The tickle starts to Ring! The service does not answer
the call and grabs the CallerID :-)
*****************************************************

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-4008835-CX00
From: <sip:0702867989@85.225.10.37:5060>;tag=0050C281FBCC-73020710
To: <sip:gsm1@85.225.10.37:5060>;tag=rBcr728vQXj6r
Call-ID: 0050C281FBCC-37007171@10.0.46.2
CSeq: 4 INVITE
Contact: <sip:gsm1@85.225.10.37:5060;transport=udp>
User-Agent: FreeSWITCH-mod_sofia/1.0.4-exported
Accept: application/sdp
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, PRACK, MESSAGE, SUBSCRIBE,
NOTIFY, REFER, UPDATE, REGISTER, INFO, PUBLISH
Supported: timer, precondition, path, replaces
Allow-Events: talk, presence, dialog, call-info, sla, include-session-
description, presence.winfo, message-summary, refer
Content-Length: 0

*****************************************************
The Caller Hangs within the 10 seconds...
*****************************************************
CANCEL sip:gsm1@85.225.10.37:5060 SIP/2.0
Max-Forwards: 70
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-4008835-CX00
From: <sip:0702867989@85.225.10.37:5060>;tag=0050C281FBCC-73020710
To: <sip:gsm1@85.225.10.37:5060>;tag=rBcr728vQXj6r
Call-ID: 0050C281FBCC-37007171@10.0.46.2
CSeq: 4 CANCEL
User-Agent: 2N VoiceBlue V-02.07.35b02
Allow: INVITE, BYE, ACK, CANCEL, OPTIONS, REFER, NOTIFY
Content-Length: 0

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-4008835-CX00
From: <sip:0702867989@85.225.10.37:5060>;tag=0050C281FBCC-73020710
To: <sip:gsm1@85.225.10.37:5060>;tag=rBcr728vQXj6r
Call-ID: 0050C281FBCC-37007171@10.0.46.2
CSeq: 4 CANCEL
Content-Length: 0

SIP/2.0 487 Request Terminated
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-4008835-CX00
From: <sip:0702867989@85.225.10.37:5060>;tag=0050C281FBCC-73020710
To: <sip:gsm1@85.225.10.37:5060>;tag=rBcr728vQXj6r
Call-ID: 0050C281FBCC-37007171@10.0.46.2
```

```
CSeq: 4 INVITE
User-Agent: FreeSWITCH-mod_sofia/1.0.4-exported
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, PRACK, MESSAGE, SUBSCRIBE,
NOTIFY, REFER, UPDATE, REGISTER, INFO, PUBLISH
Supported: timer, precondition, path, replaces
Allow-Events: talk, presence, dialog, call-info, sla, include-session-
description, presence.winfo, message-summary, refer
Content-Length: 0

ACK sip:gsm1@85.225.10.37:5060;transport=udp SIP/2.0
Max-Forwards: 70
Via: SIP/2.0/UDP 10.0.46.2:5060;branch=z9hG4bK-4008835-CX00
From: <sip:0702867989@85.225.10.37:5060>;tag=0050C281FBCC-73020710
To: <sip:gsm1@85.225.10.37:5060>;tag=rBcr728vQXj6r
Call-ID: 0050C281FBCC-37007171@10.0.46.2
CSeq: 4 ACK
Contact: <sip:0702867989@10.0.46.2:5060>
User-Agent: 2N VoiceBlue V-02.07.35b02
Allow: INVITE, BYE, ACK, CANCEL, OPTIONS, REFER, NOTIFY
Content-Length: 0

*****************************************************
The call back starts now... (WITHOUT ANY AUTH)
*****************************************************

INVITE sip:0702867989@10.0.46.2 SIP/2.0
Via: SIP/2.0/UDP 85.225.10.37:5080;rport;branch=z9hG4bKpZte656KSZ20p
Max-Forwards: 70
From: "FreeSWITCH" <sip:666@85.225.10.37>;tag=N7ea86yrKUFpB
To: <sip:0702867989@10.0.46.2>
Call-ID: 11586708-0f32-122d-4c84-001372cbb7a9
CSeq: 119654351 INVITE
Contact: <sip:gw+2n@85.225.10.37:5080;transport=udp>
User-Agent: FreeSWITCH-mod_sofia/1.0.4-exported
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, PRACK, MESSAGE, SUBSCRIBE,
NOTIFY, REFER, UPDATE, REGISTER, INFO
Supported: timer, precondition, path, replaces
Allow-Events: talk, refer
Content-Type: application/sdp
Content-Disposition: session
Content-Length: 291
Remote-Party-ID:                                              "FreeSWITCH"
<sip:666@85.225.10.37>;party=calling;screen=yes;privacy=off

v=0
o=FreeSWITCH 1251514166 1251514167 IN IP4 85.225.10.37
s=FreeSWITCH
c=IN IP4 85.225.10.37
t=0 0
m=audio 31208 RTP/AVP 0 8 3 101 13
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:3 GSM/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=rtpmap:13 CN/8000
a=ptime:20
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 85.225.10.37:5080;rport;branch=z9hG4bKpZte656KSZ20p
From: "FreeSWITCH" <sip:666@85.225.10.37>;tag=N7ea86yrKUFpB
To: <sip:0702867989@10.0.46.2>;tag=0050C281FBCC-212019576
Call-ID: 11586708-0f32-122d-4c84-001372cbb7a9
CSeq: 119654351 INVITE
Contact: <sip:0702867989@10.0.46.2:5060>
User-Agent: 2N VoiceBlue V-02.07.35b02
Allow: INVITE, BYE, ACK, CANCEL, OPTIONS, REFER, NOTIFY
Content-Length: 0

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 85.225.10.37:5080;rport;branch=z9hG4bKpZte656KSZ20p
From: "FreeSWITCH" <sip:666@85.225.10.37>;tag=N7ea86yrKUFpB
To: <sip:0702867989@10.0.46.2>;tag=0050C281FBCC-212019576
Call-ID: 11586708-0f32-122d-4c84-001372cbb7a9
CSeq: 119654351 INVITE
Contact: <sip:0702867989@10.0.46.2:5060>
User-Agent: 2N VoiceBlue V-02.07.35b02
Allow: INVITE, BYE, ACK, CANCEL, OPTIONS, REFER, NOTIFY
Content-Type: application/sdp
Content-Length: 213

v=0
o=VoiceBlue 32734 32411 IN IP4 10.0.46.2
s=GSM call
c=IN IP4 10.0.46.2
t=0 0
m=audio 16002 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-11

*****************************************************
And yes! We like it!
*****************************************************
```

16. Annex 2: Leave_a_Message state machine

The image below shows the state machine of the Leave-a-Message application, with default audio messages.

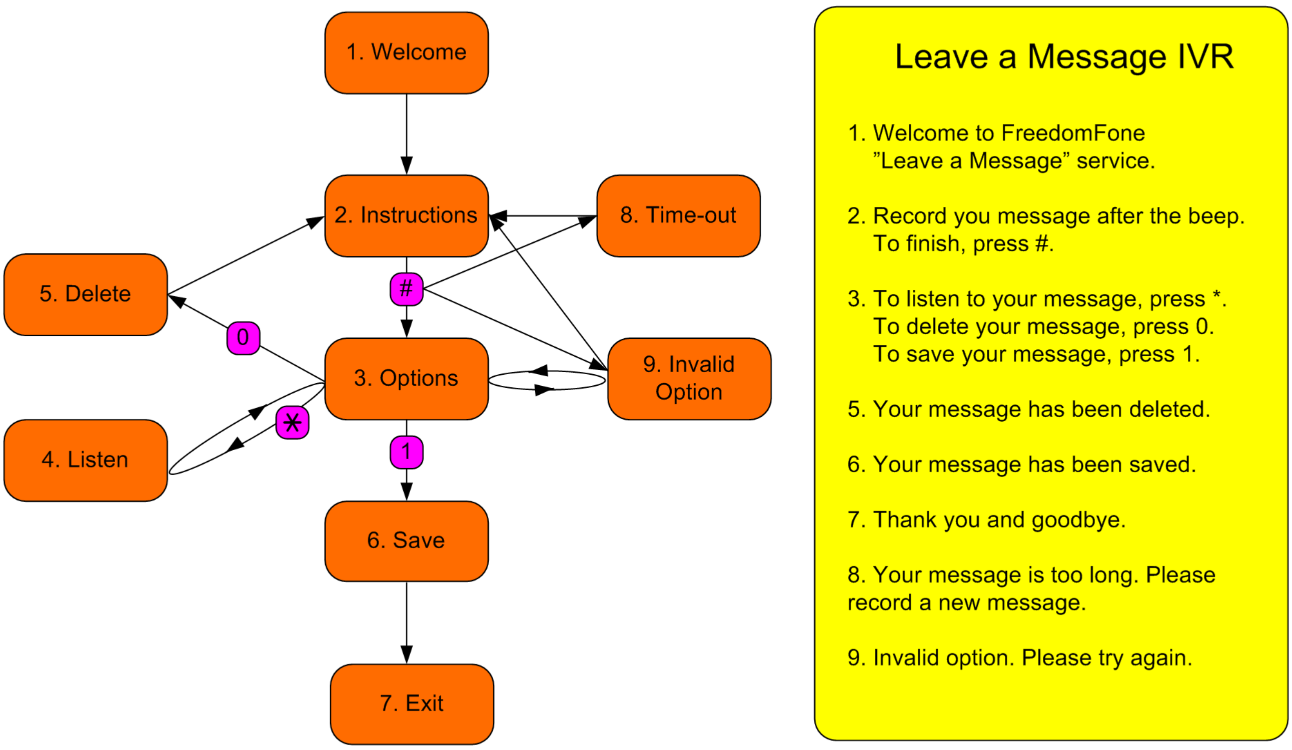

Image 3: Leave a message state machine.

17. Annex 3: Our development environment

There are two main SVN repositories involved in the testbed. FreeSWITCH and Freedomfone.

FreeSWITCH install under /usr/local/freedomfone

We checkout FreedomFone SVN under /usr/src/pictus

We make the following symbolic links:

Apache /var/www/freedomfone/app folder points to Freedom Fone GUI

The Cake PHP Development is placed under the /var/www/freedomfone

Freedomfone scripts need to be under:

/var/www/freedomfone/app/webroot/freedomfone

XML Curl needs to be under /var/www/freedomfone/app/webroot/xml_curl

```
/var/www/freedomfone/app -> /usr/src/pictus/gui


/var/www/freedomfone/app/webroot/freedomfone    ->
/usr/src/pictus/freeswitch/scripts/freedomfone/

/usr/local/freeswitch/scripts/
freedomfone -> /usr/src/pictus/freeswitch/scripts/freedomfone


/var/www/freedomfone/app/webroot/xml_curl -> /usr/src/pictus/xml_curl/
```

Permissions:
The webserver needs to be able to write configuration files and dynamic dialplans. Chown the files (www-data) under

```
/var/www/freedomfone/app/webroot/freedomfone
```