Freedom Fone Lycaon Suyo 2.D

Newfies-Dialer Technical Documentation

Version 1.1.4

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# Table of Contents

1.   Introduction

Newfies[1] is a Dialer written by Areski Belaid (Star2Billing[2]), which has been subsidized by the Freedom Fone project. The name Newfies refers to the Newfoundland Dog, which is known as a reliable dog that keeps his promises, often used by sea rescue services around the world.

Newfies is a stand-alone dialer that utilizes the FreeSWITCH telephony engine for outbound calls. Newfies is written in Python using the Django Framework. It uses Plivo[3], an XML based Communications Framework to communicate with FreeSWITCH.

Freedom Fone communicates with Newfies via REST API to issue outgoing calls, check call status and to manage campaigns.

The architecture of Newfies-Dialer and its communication with Freedom Fone and GSM users, is illustrated below. The IP addresses of Newfies-Dialer and the OfficeRoute device are example parameters that is being used throughout this manual.

This guide contains installation and configuration instructions for (1) OfficeRoute, (2) Newfies-Dialer and (3) Freedom Fone.
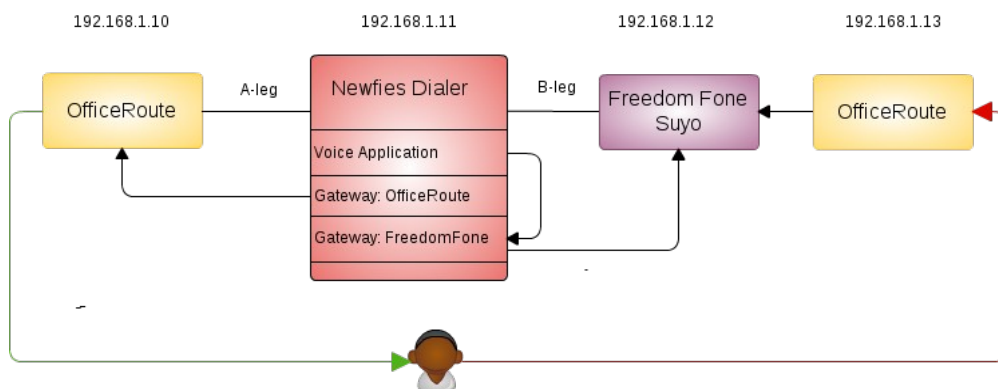
2.   2N OfficeRoute configuration: Outbound calls

This section describes how to configure the 2N OfficeRoute device to use GSM as a SIP gateway for outbound calls. The guide is based in OfficeRoute firmware version 2.3.13.

A default configuration file for 2N OfficeRoute (outbound calls) is available via the SVN[4]. The configuration file can be uploaded from *Administration → Configuration Backup → Upload config*

If you choose to use this configuration file (instead of manually configure the unit), do not forget to change the example IP address to the IP address of the OfficeRoute unit in use.

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[1]   Newfies official website: http://newfies-dialer.org/

[2]   Star2Billing: http://www.star2billing.com/

[3]   Plivo: http://www.plivo.org/

[4]   Filename: backup_officeroute_outbound_2.3.13.tgz

2.1 Change password

User management → Users →

Change the default password of the OfficeRoute to avoid unauthorized access to it. To do so, edit Admin, and replace default password with the password of your choice. This password will only be used to log in to the OfficeRoute GUI.

Logout and login again.

2.2 Network settings

Network → Main configuration

1. Disable DHCP server

2. Set network parameters

   - IP address: 192.168.1.10

   - Gateway: 192.168.1.1

   - DNS: 8.8.8.8

2.3 SIP line

Telephony services → Devices → SIP lines

Edit the default SIP line, and set:

   - SIP server address: 192.168.1.10 (the same IP address as the OfficeRoute)

   - Description: Newfies

Save changes.

2.4 Create OfficeRoute User

User Management → Users

Create an OfficeRoute user with the parameters shown below. The SIP client (Newfies-Dialer) will register to a SIP proxy of OfficeRoute using the credentials of this OfficeRoute user.

   User name: 1973

   Password: 1973

   Line number: 1973

Note that "1973" is used in this example, but can be replaced with any string. If you replace "1973" with another string, please do the corresponding changes to user name and password in the Freeswitch trunk (see Section 3.5).

2.5   Configure SIP proxy

Telephony services → SIP Proxy

1. Switch SIP proxy ON. When the SIP proxy is on, the following messages should be prompted: "SIP Proxy successfully switched on.

2. Delete existing SIP proxy routing rules (except for the default "else" routing rule)

3. Edit the "else" routing rules

   - If prefix: else
   - Strip: 0
   - Add: 0
   - Do action: connect to LCR
   - With parameter: SIP – Newfies

"SIP - Newfies" refers to the default SIP line in OfficeRoute (Section 2.3).

2.6   Configure LCR (least cost routing)

Telephony services → LCR

In the LCR table, we make sure that the (by the Wizard) predefined LCR route "To GSM" is enabled, and configured with the following parameters:

- Destination: To GSM
- Prefix: .*
- Route: GSM

The "To GSM" route implies that all calls to GSM (with any prefix) are routed to the route "GSM".

In the list of LCR Routes, we can see that the GSM route is forwarding outbound calls to all four GSM modules. If you wish to limit outbound calls to a sub-set of GSM modules, please edit the GSM route accordingly.

The OfficeRoute is now configured for outbound calls.

2.7   Make outbound test call

To make sure that your OfficeRoute is correct configured, you can test to make a SIP call to the device.

Install a SIP client in your computer (for example, Ekiga Softphone[5] for Linux).

Step 1: Create a SIP account using the credentials from the User your created in OfficeRoute (user name: 1973, password 1973). The SIP account should be listed as "Registered" in Ekiga if you have configured OfficeRoute and Ekiga correctly.

Step 2: Dial a phone number that you have access to. If the phone rings, you have configured OfficeRoute correctly for outgoing calls.

---

[5] Install Ekiga by running: *apt-get install ekiga*

Step 1a: Create SIP account.

Step 1b: Account status should be "Registered".

Step 2: Dial a phone number that you have access to. The phone number is in this case 5551234.

3.   2N OfficeRoute configuration: Inbound calls

The 2N OfficeRoute that is used for inbound calls should be configured with the *Freedom Fone 2.D configuration file*. The difference between the 2.D configuration file and the default Freedom Fone configuration file for inbound calls, is addition of the Tickle service. To allow Tickle calls, the OfficeRoute must be configured with (at least) one channel pointing to extension 3000 in Freedom Fone 2.D.

Illustration 1: GSM Operator pointing at extension 3000 in FreeSWITCH.

Illustration 2: Mapping the Tickle GSM operator to a GSM channel (device).

4. Installation of Newfies-Dialer

Newfies-Dialer (version 1.2) needs to be installed in a separate machine to Freedom Fone. The Newfies-Dialer machine needs to be connected to the Internet during the installation process. After a complete installation, the Newfies-Dialer machine only needs to be located on the same LAN as the Freedom Fone server.

This section is based on the Beginner's Guide[6] of Freedom Fone provided by Start2Billing.

4.1 Introduction

This is a simple step-by-step guide for installation of the necessary components of Newfies-Dialer. It covers the installation of the operating system, Freeswitch, Plivo and Newfies-Dialer along with all the dependencies.

4.2 Operating System Installation

We recommend that you use Ubuntu LTS 10.04 server edition, and this document is written with this in mind. To take advantage of modern hardware the 64 bit edition may be used. If you are testing on older hardware, it may be necessary to install the 32 bit edition. It is available at http://www.ubuntu.com/download/ubuntu/download

Please note that the Newfies-Dialer install scripts are only designed to work on Ubuntu LTS 10.04 and CentOS 6.2.

Burn the image to CD, and install onto your hardware. The installation of Ubuntu is straight forward and simple.

4.3 Ubuntu packages

The only packages to install are

1. wget (for downloading files from the Web from command line)

   ```
   sudo apt-get install wget
   ```

2. SSH Server (for managing the server remotely)

---

[6] Newfies-Dialer Beginner's Guide: http://www.newfies-dialer.org/documentation/beginners-guide/

```
sudo apt-get install openssh-server
```

When Ubuntu is installed, it maybe worthwhile installing the latest patches and updates. This is achieved with the following commands.

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

A reboot after major updates is recommended.

```
sudo reboot
```

4.4   Complete Installation

The installation script installs Newfies-Dialer on your Ubuntu Server and includes all the dependencies such as Freeswitch, Plivo, Celery and Redis.

- Freeswitch is the telephony engine used by Newfies-Dialer to make calls, as well as play voice applications
- Plivo is an open source communications framework to rapidly deploy voice based applications used in conjunction with Newfies-Dialer.
- Celery is the task queuing system.
- Redis is the task messaging system

Please note that the installation script should only be run on a clean installation of Ubuntu Server 10.04 LTS or CentOS 6.2

Once Ubuntu is installed, download and run the installation script. For those unused to Ubuntu, typing *"sudo su -"* at the console logs you in as root.

Once logged in as root, execute the following command.

```
wget https://raw.github.com/Star2Billing/newfies-dialer/master/install/install-all.sh
bash ./install-all.sh
```

Note!

The Newfies installation script detects the IP address of the server and provisions it to the config files of Celery[7]. If you change the IP address of your service (after running the Newfies installation script), you need to restart Celery so that the change of IP address takes affect.

```
/etc/init.d/newfies-celeryd restart
```

FreeSWITCH

The installation will prompt you to install Freeswitch (press enter). This will download and install Freeswitch with the modules appropriate for Newfies-Dialer. The installation will take some time, but does not require your interaction once started.

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[7]   Celery is an open source asynchronous task queue/job queue based on distributed message passing.

Plivo

When complete, the install-all.sh script will prompt you to install Plivo. Press enter to continue. This will download and install Plivo and all its dependencies.

NB: Please note that the Plivo script makes alterations to the Freeswitch dial-plan, so it should not be run on an existing working Freeswitch installation, as it will change your current configuration. If you wish to install Plivo on an existing version of Freeswitch, use the script as a guide, or edit it to suit your requirements.

Finally, the Newfies-Dialer selection menu will appear.

Newfies-Dialer

Newfies-Dialer is the management and control system providing a web interface for admin and customers alike, to manage the platform, subscribers and voice broadcast campaigns.

The installation script will prompt you as to what you want to install. For this guide, we chose the option to install all.

During the installation, a number of questions will be asked about your preferences. For the purposes of the this guide, we will go for the default installation using MySQL. When prompted for the database to use, simply press enter.

After some time, a pseudo graphical interface will appear prompting you for a MySQL password. Type in a password, press enter, and type it again to confirm. *Take a note of the password.*

You will be prompted for the location, name, port and password of Newfies-Dialer database. Go with the defaults, with the exception of the password, where you type in the password entered during the installation of MySQL.

You will be prompted to create a superuser. Accept root, enter your email address, and your chosen password (twice). This will be the user name and password you use to log on to the Newfies-Dialer web interface. *Take a note of the password.*

Once installed, the system will then prompt you to continue with the installation and install the backend processes. Press enter to continue. On completion, the script will return to the original Newfies-Dialer install menu. Exit the menu.

We can now reboot to ensure that, on startup, all required services are running.

        reboot

After reboot, log into the new system via your web browser to check that everything is working.

        Http://192.168.1.11:8008

If you have followed the instructions above, you should be able to login using the user name and password you created during the installation.

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5. Freeswitch Trunk configuration

In order for Newfies-Dialer to make outbound calls to its subscribers, and feed the outbound call with a Freedom Fone service, two SIP trunks need to be created in Freeswitch. These trunks are called A-leg (out-bound call) and B-leg (service to be delivered).

The Freeswitch wiki can provide more information on configuring trunks. However creating a trunk simply for Newfies-Dialer is straightforward.

Trunks or gateways, as they are known in Freeswitch, are configured using XML syntax.

5.1 A-leg

Create an XML file (officeroute.xml) in /usr/local/freeswitch/conf/sip_profiles/external/.

```
cd /usr/local/freeswitch/conf/sip_profiles/external/
vi  officeroute.xml
```

Place the following lines in the file (assuming that the IP address of the OfficeRoute is 192.168.1.10, and that you have created a user with the credentials 1973:1973 in OfficeRoute).

```
<include>
<gateway name="192.168.1.10">
<!--/// account username *required* ///-->
<param name="username" value="1973"/>
<!--/// auth realm: *optional* same as gateway name, if blank ///-->
<!--<param name="realm" value="asterlink.com"/>-->
<!--/// username to use in from: *optional* same as username, if blank ///-->
<param name="from-user" value="1973"/>
<!--/// domain to use in from: *optional* same as realm, if blank ///-->
<param name="from-domain" value="192.168.1.10"/>
<!--/// account password *required* ///-->
<param name="password" value="1973"/>
<!--/// extension for inbound calls: *optional* same as username, if blank ///-->
<!--<param name="extension" value="cluecon"/>-->
<!--/// proxy host: *optional* same as realm, if blank ///-->
<!--<param name="proxy" value="asterlink.com"/>-->
<!--/// send register to this proxy: *optional* same as proxy, if blank ///-->
<!--<param name="register-proxy" value="mysbc.com"/>-->
<!--/// expire in seconds: *optional* 3600, if blank ///-->
<!--<param name="expire-seconds" value="60"/>-->
<!--/// do not register ///-->
<param name="register" value="true"/>
<!-- which transport to use for register -->
<!--<param name="register-transport" value="udp"/>-->
<!--How many seconds before a retry when a failure or timeout occurs -->
<!--<param name="retry-seconds" value="30"/>-->
<!--Use the callerid of an inbound call in the from field on outbound calls via this
gateway -->
<!--<param name="caller-id-in-from" value="false"/>-->
<!--extra sip params to send in the contact-->
<!--<param name="contact-params" value="tport=tcp"/>-->
<!--send an options ping every x seconds, failure will unregister and/or mark it down-->
<!--<param name="ping" value="25"/>-->
</gateway>
</include>
```

5.2 B-leg

Create another XML file (freedomfone.xml) in the same folder.

```
cd /usr/local/freeswitch/conf/sip_profiles/external/
vi  freedomfone.xml
```

Place the following lines in the file (assuming that the IP address of Freedom Fone Suyo is 192.168.1.12).

```
<include>
<gateway name="192.168.1.12">
<!--/// account username *required* ///-->
<param name="username" value="1001"/>
<!--/// auth realm: *optional* same as gateway name, if blank ///-->
<!--<param name="realm" value="asterlink.com"/>-->
<!--/// username to use in from: *optional* same as username, if blank ///-->
<param name="from-user" value="1001"/>
<!--/// domain to use in from: *optional* same as realm, if blank ///-->
<param name="from-domain" value="192.168.1.12"/>
<!--/// account password *required* ///-->
<param name="password" value="1234"/>
<!--/// extension for inbound calls: *optional* same as username, if blank ///-->
<!--<param name="extension" value="cluecon"/>-->
<!--/// proxy host: *optional* same as realm, if blank ///-->
<!--<param name="proxy" value="asterlink.com"/>-->
<!--/// send register to this proxy: *optional* same as proxy, if blank ///-->
<!--<param name="register-proxy" value="mysbc.com"/>-->
<!--/// expire in seconds: *optional* 3600, if blank ///-->
<!--<param name="expire-seconds" value="60"/>-->
<!--/// do not register ///-->
<param name="register" value="true"/>
<!-- which transport to use for register -->
<!--<param name="register-transport" value="udp"/>-->
<!--How many seconds before a retry when a failure or timeout occurs -->
<!--<param name="retry-seconds" value="30"/>-->
<!--Use the callerid of an inbound call in the from field on outbound calls via this
gateway -->
<!--<param name="caller-id-in-from" value="false"/>-->
<!--extra sip params to send in the contact-->
<!--<param name="contact-params" value="tport=tcp"/>-->
<!--send an options ping every x seconds, failure will unregister and/or mark it down-->
<!--<param name="ping" value="25"/>-->
</gateway>
</include>
```

5.3 Load trunks

Finally we need to load the new configuration, and check the trunks are registered. The Freeswitch console is accessed by typing *"fs_cli"* at the command prompt. The logging level can be increased or decreased from level 0 to level 7.

For the greatest verbosity, type */log 7* followed by enter.

```
fs_cli
```

You should now see the Freeswitch CLI. To reload the Freeswitch configuration, use the following command: (hint; Tab auto-completes)

```
    sofia profile external restart reloadxml
```

When complete, check the trunk has registered with the following command.

```
    sofia status
```

If you have configured OfficeRoute and the Freeswitch trunk correctly, you should see the your trunk being listed as REGED (registered).

```
    external::192.168.1.10    gateway    sip:1973@192.168.1.10    REGED
    external::192.168.1.12    gateway    sip:1001@192.168.1.12    REGED
```

Please not that the OfficeRoute and Freedom Fone needs to be powered on, configured according to this manual and routable by the Newfies-Dialer, in order to be registered as trunks in Freeswitch.

To exit the Freeswitch CLI, type *CTRL+D,* or ... (three dots).

Freeswitch configuration is now complete.

6.   Newfies-Dialer Configuration

Next we are going to make the basic configure to Newfies-Dialer in order to work with Freedom Fone.

Log into the Newfies-Dialer administration interface (http://192.168.1.11/admin/) using your root username and password previously created in these instructions.

6.1   Hostname

Some of the features of Newfies-Dialer are dependent on the host name or IP address of the platform, so this parameter has to be set. In the Admin dashboard, locate the "Sites" link and click change.
By default, there will be an entry of "example.com". Edit this setting to reflect the host name or the IP address of the server, for example:

> http://www.my-newfies.org:8008

> or,

> http://192.168.1.11:8008

6.2   Dialer Gateway

We are going to create two dialer gateways, one A-leg gateway pointing at OfficeRoute, and one B-leg gateway pointing at Freedom Fone.

Create the A-leg gateway with the following parameters:

```
1. Name (for identification): OfficeRoute
```

```
2. Gateway: sofia/gateway/192.168.1.10/
```

Next, create the B-leg gateway with the following parameters:

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```
1. Name (for identification): FreedomFone

2. Gateway: sofia/gateway/192.168.1.12/
```

"192.168.1.10" and "192.168.1.12" are the names of the gateways which you can take from the gateway *name* when you typed "sofia status" in Freeswitch. The gateway name can either be an IP address or a domain name. Do not forget the trailing slash after the gateway name.

Finally, check that status is set to "active", and save. Two trunks should now appear in the gateway list.

6.3   Dialer Settings

Each customer, including the root user, needs to have Dialer Settings and trunks associated with their account. These set the limit to which each customer can utilize Newfies-Dialer in terms of trunks, calls per minute, call duration, subscribers and campaigns, as well as blacklisting or whitelisting phone numbers.

In the Admin Dashboard, locate the Dialer Settings, and click Add. As name, put "DS1" and, for the moment, leave the default settings as they are, then save them.

In the Admin dashboard, click Admins, then root. Scroll to the bottom of the page.

1. Select which Dialer Settings apply to this customer (DS1)
2. Select which gateway(s) this customer is allowed to use (officeroute and freedomfone)

Finally, click save.

6.4   Voice Application

Next we are going to configure the voice application that will be executed when Newfies-Dialer makes an outbound call. We want Newfies-Dialer to use the "freedomfone" gateway (B-leg), and execute a "dial" as user "root". The voice application of type "dial" will be associate with a FreeSWITCH extension (Freedom Fone service id), such as 2100 or 4100.

Click VoiP App, then click add.

```
Name: freedomfone
Type: Dial
B-leg (gateway): freedomfone
User: root
```

The configuration of Newfies-Dialer is now completed.

7.   Configuration of Freedom Fone

Open the main configuration[8] file of Freedom Fone and edit the Newfies-Dialer parameters marked in bold.

Host:	*Hostname or IP address of Newfies-Dialer (with http:// and trailing slash)*
User:	*User name of Newfier-Dialer (root)*
Pwd:	*Password of Newfies-Dialer GUI*
A-leg gateway:	*Id of the A-leg gateway created in Newfies-Dialer (by default 1)*
Object_id:	*Id for the voice application created in Newfies-Dialer (by default 1)*
Caller-id:	*The number to be displayed as caller-id by the platform.*

```
$config['DIALER'] = array(
                'host'                        => 'http://192.168.1.11:8008/',
                'user'                        => 'root',
                'pwd'                         => 'newfies',
                'path'                        => 'api/v1/',
                'campaign_POST'               => 'campaign/',
                'campaign_GET'                => 'campaign/',
                'campaign_PUT'                => 'campaign/',
                'campaign_DELETE'             => 'campaign_delete_cascade/',
                'campaign_subscriber_POST'    => 'campaignsubscriber/',
                'campaign_subscriber_PUT'     => 'campaignsubscriber/',
                'campaign_subscriber_GET'     => 'campaignsubscriber_per_campaign/',
                'account'                     => '',
                'a-leg_gateway'               => 1,
                'frequency'                   => 10,
                'call_timeout'                => 10,
                'object_id'                   => 1,
                'content_type'                => 'voice_app',
                'caller_id'                   => '123456'
                );
```

The configuration of Freedom Fone to use Newfies-Dialer is now completed.

8.   Dialer functionality in Freedom Fone

Freedom Fone utilizes Newfies-Dialer (aka Dialer in Freedom Fone) via two types of callback application:

1. Campaigns

2. Callback Services

All Dialer functionality can be reach from the tab "Dialer" in the main menu.

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[8]   Configuration file: /opt/freedomfone/gui/app/config/config.php

8.1 Campaigns

A Freedom Fone Campaign is a callback request for a certain service to a set of users. The users to be called are determined by the Phone book associated with the Campaign. The campaign has a starting time and a finishing time.

Create Campaign

A Campaign has the following parameters:

- Name: Name of campaign, unique in Freedom Fone and Newfies
- Phone Book: A set of Users that the Callback Campaign is targeting.
- Service: Type and instance of service to be delivered
- Max retries: Number of retries that Newfies does to each User.
- Retry interval: Interval between attempts to reach a User
- Max duration: Max duration of a Campaign call.
- Start time: Starting time of Campaign.
- End time: End time of Campaign (independent if all pending calls have been placed or not)

Freedom Fone is using the following API methods to create a new campaign:

- Campaign::write (POST)
- Campaign::read (GET)
- CampaignSubscriber::write (POST)

When a Campaign is created in Freedom Fone, the following actions takes place in Newfies:

- Campaign created
- Phone book created
- Contacts created
- Campaign Subscribers created
- Call requests created

Campaign overview

A drop-down list of Campaigns are provided to choose which Campaign to analyze.

After selecting a Campaign, the screen will display the current status of Campaign calls in percentage. The status[9] values can be:

1. Completed

2. Pending

3. Processing

4. Paused

---

[9]   See Section 5.3 (Call Status) for detailed description of status values.

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5. Failed

6. Aborted

This view offers the Administrator the possibility to apply the following Call actions on the Campaign: Start, Pause or Abort.

- A Campaign that has been Paused, can later be Started whenever needed.

- A Campaign that has been Aborted, cannot be Started later.

Call status

The Administrator can list Callback requests based on Campaign name and Call Status, and determine sorting order.

A Callback request can have any of these statuses:

1. Completed (call has successfully been placed and answered by User)

2. Pending (call is in the queue, one or more attempts to call User can have been made)

3. Processing (ongoing call attempt)

4. Paused (call has been put on hold by Freedom Fone administrator)

5. Failed (maximum number of retries have been made without success. No further attempts will be made

6. Aborted (call has been aborted by Freedom Fone administrator)

The Freedom Fone administrator can set the following statuses for a Callback request:

1. Start (Return to previous status, if the call has been Paused or Aborted)

2. Pause (Temporary set call on hold, the Dialer will make no attempts to call User)

3. Abort (Stop all further attempts to call User)

For each Call request listed, the Administrator can view:

1. Campaign overview (click on Campaign name)

2. User details (click on User name)

3. Service ID (service to be delivered to User)

4. Attempts: Number of attempts used to reach the User

5. Call status (Completed, Pending, Processing, Paused, Failed, Aborted)

6. Call action: (Start, Pause, Abort)

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8.2   Callback Services

A Callback Service is an SMS- or Call-based service that allows Users to access Freedom Fone services (Voice Menus, Leave-a-Message or Content) for free or at a very small cost.

A Callback Service can be reached either via SMS or by a so called "Tickle" (aka "Buzz", an unterminated call).

Create Callback Service

A Callback Service is created with the following parameters:

Service based parameters:

- SMS code: Alpha-numeric string to be sent to Freedom Fone via SMS
- Service: Type and instance of service to be delivered
- Max retries: Number of retries that Newfies does to each User.
- Retry interval: Interval between attempts to reach a User
- Max duration: Max duration of a Campaign call
- Start time: Starting time of Campaign.
- End time: End time of Campaign (independent if all Users have been reached or not)

User based parameters:

- Max calls per user: Maximum number of calls per user for this service.
- Max calls per user and day: Maximum number of calls per user/day (24h) for this service.
- Max calls: Maximum number of calls for this service.

Freedom Fone is using the following API methods to create a new Callback Service

- Campaign::write (POST)
- Contact::write (POST)

When a Callback Service is created in Freedom Fone, the following actions take place in Newfies:

- Campaign created
- Phone book created (empty phone book)

Edit Callback Service

A Callback Service can be edited at any time. All fields but the SMS code are editable.

Delete Callback Service

A Callback Service can be deleted at any time. When a Callback Service is deleted in Freedom Fone, the following actions take place in Newfies:

- Campaign deleted

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- Phone book deleted
- Contacts deleted (if any)
- Campaign subscribers deleted (if any)

Call Status

The Administrator can list Callback Service calls based on Call Status, Callback Service, Type (SMS or Tickle) and determine sorting order.

A Callback Service call can have any of these statuses:

1. Completed (call has successfully been placed and answered by User)

2. Pending (call is in the queue, one or more attempts to call User can have been made)

3. Processing (ongoing call attempt)

4. Paused (call has been put on hold by Freedom Fone administrator)

5. Failed (maximum number of retries have been made without success. No further attempts will be made

6. Aborted (call has been aborted by Freedom Fone administrator)

The Freedom Fone administrator can set the following statuses for a Callback Service call:

1. Start (Return to previous status, if the call has been Paused or Aborted)
2. Pause (Temporary set call on hold, the Dialer will make no attempts to call User)
3. Abort (Stop all further attempts to call User)

For each Call request listed, the Administrator can view:

1. Callback Service (SMS code)

2. Campaign status (Running, Aborted, Finished), hover over icon

3. User details (click on User name)

4. Service ID (service to be delivered to User)

5. Type: SMS or Tickle

6. Attempts: Number of attempts used to reach the User

7. Call status

8. Call action: (Start, Pause, Abort)

8.3 Tickle Service

The Tickle Service can be enabled or disabled according to your desire. To enable the Tickle service, the Administrator must mark one Callback Service as the "tickle" service. If the button "Disable tickle" is selected, the system will not allow any incoming Tickle requests to trigger outgoing calls.

If a Tickle service is selected, a caller that makes a Tickle to Freedom Fone, will automatically be redirected to the tickle service.

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8.4   User Access Control

Freedom Fone implements four different types of access control levels for the Callback Service to avoid abuse of the service. If at least one of the parameters are not met, the callback request is refused.

Callback Service time frame
Is the Callback Service open ? (start and end time)

Max calls per service
Is the total quota (for all users) exceeded for the Callback Service ? (total calls)

User based Access Control List
Is the User black listed in Freedom Fone ?

Max calls per user
Is the total quota for this user and this service exceeded ?

Max calls per user and day
Is the daily (24h) quota for this user and this service exceeded ?

8.5   Refresh methods

Freedom Fone 2.D implements two new levels of "refresh" methods to (1) poll data from the dispatcher (incoming SMS and tickle), and (2) to poll call status from Newfies-Dialer.

Dispatcher refresh

Incoming Dialer requests (SMS or Tickle) are polled from the Dispatcher via the following method:

- /freedomfone/callbacks/refresh

The method is executed via cronjob.

Newfies-Dialer refresh

Three methods are implemented to fetch status information on campaigns, and pending call requests in Newfies-Dialer. All three methods are executed from a cronjob.

The methods are:

1. Refresh Campaign status from Newfies-Dialer (Running, Paused, Finished)

   /freedomfone/campaigns/newfies_ refresh

2. Refresh status from calls originated from Campaigns

   /freedomfone/callbacks/newfies_ campaign_ refresh

3. Refresh status from calls originated from Callback Services

   /freedomfone/callbacks/newfies_ callback_ refresh

8.6 Testing Callback Services

Callback Services can be tested without the need of sending costly SMS or making Tickle calls to Freedom Fone.

By using a PHP based test script[10], Freeswitch events corresponding to (1) an SMS or (2) a Tickle, can be generated and replace a real SMS or call to a GSM gateway. The test scripts can be used the following way.

Generate SMS

```
php sendSMS.php <callback_service_code> <sender>
```

Where <callback_service_code> is the alpha-numeric code associated with the callback service, and <sender> is the (numeric) phone number of the sender.
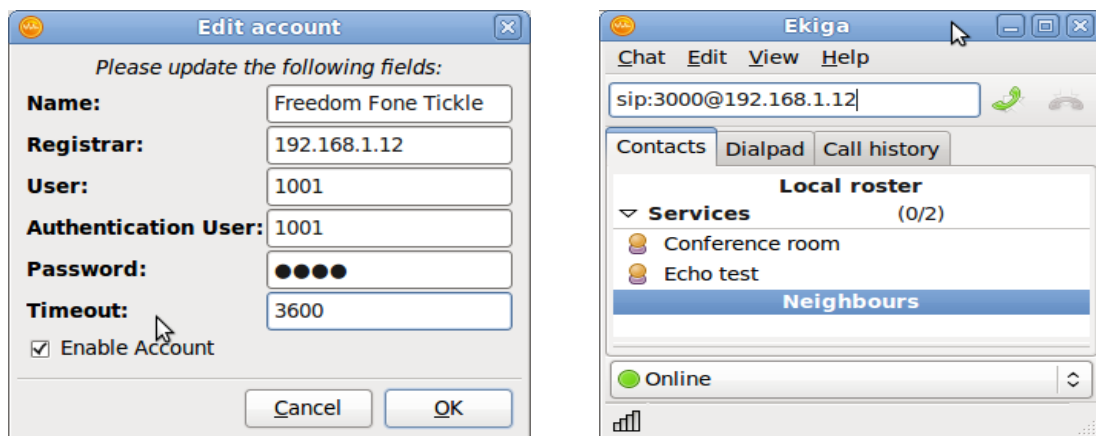
Generate Tickle

```
php sendTickle.php <sender>
```

Where sender is the (numeric) phone number of the sender.

Generate Tickle via SIP

Configure a SIP client (Eikiga for example) to register to your Freedom Fone server (192.168.1.12) to extension 3000 using account 1001 (password 1234).

Ring and hang up to create a tickle.

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[10] Test scripts are located at: /opt/freedomfone/dispatcher_in/tests/