

Web Configuration

Access

All the configuration is managed through a web interface based on the FreePBX project. To access to the web interface, open the link

http://your_server_address/admin.

It's protected by a login (the first login after the installation will request you a password).



Login

After installation there are 3 default login accounts, with different access permissions

- **user** : access to Voximal configuration and reports.
- **admin** : user access and network configuration
- **root** : full access (full freePBX features).

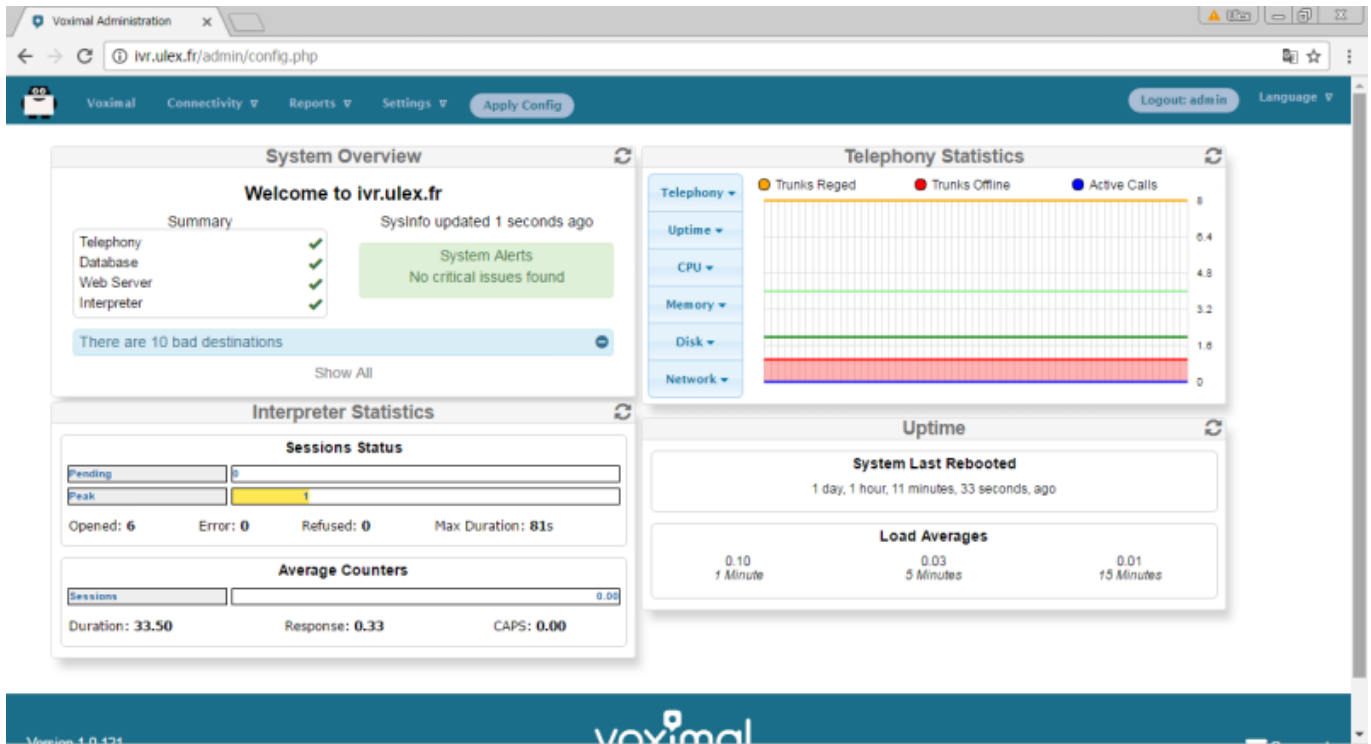
Each login has a default password (requested after the package installation, or the image activation), you can change them with the root account.



System Status

The home page after login show you the system status.

- Host name of the server
- Summary : Main modules status
- Interpreter statistics
- Telephony statistics
- Uptime / Load average



Main configuration steps

To configure the Voximal IVR you have 4 steps to do :

1. Configure the VoIP telephony access ([SIP accounts](#))
2. Create/edit a VoiceXML service ([Create a VoiceXML service](#))
3. Create Voximal accounts ([Voximal accounts](#))
4. Configure the relation between the DID and the VoiceXML service you want to use ([Routes configuration](#))

Note: On each configuration page, don't forget to click on **[Submit]** button at the bottom page before changing page. If you don't click on **[Submit]** you'll lose all your changes.

Note: All changes are not directly configure on the server. After one change, you'll see a button **[Apply Config]** in menu. To activate the changes on server you have to click on **[Apply Config]**.

1) Configure the telephony access

a) Use the test number and the PIN

If your server is connected to the internet you can use the free test access to place calls to your server. There is nothing to do. You should only check that the port 4569 is open from/to internet in UDP.

b) Connect a trunk SIP

Before you can make external calls or accept incoming calls from outside, you need to setup SIP Trunks. You can choose any VoIP Service providers. You can create a SIP trunk with an operator referenced in the page [Voip Providers](#). You have to enter informations :

- A trunk name : a string to identify you accounts
- The peers details : enter all trunk informations (host, username, password, type)

```
host=myprovider.com
username=0033123456789
type=peer
```

- Register informations : enter registry informations like :

```
0033123456789:password@myprovider.com/33123456789
```

- Click on **[submit]** button (don't remember to apply your modifications)

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c) Add an extension

After you setup your Voxibot, the first thing you do is to add extensions (connect a Phone). The integrated FreePbx allows you to add a couple of different Device types

- Generic SIP Device
- Generic IAX2 Device
- Generic DAHDi Device
- Other Custom Device

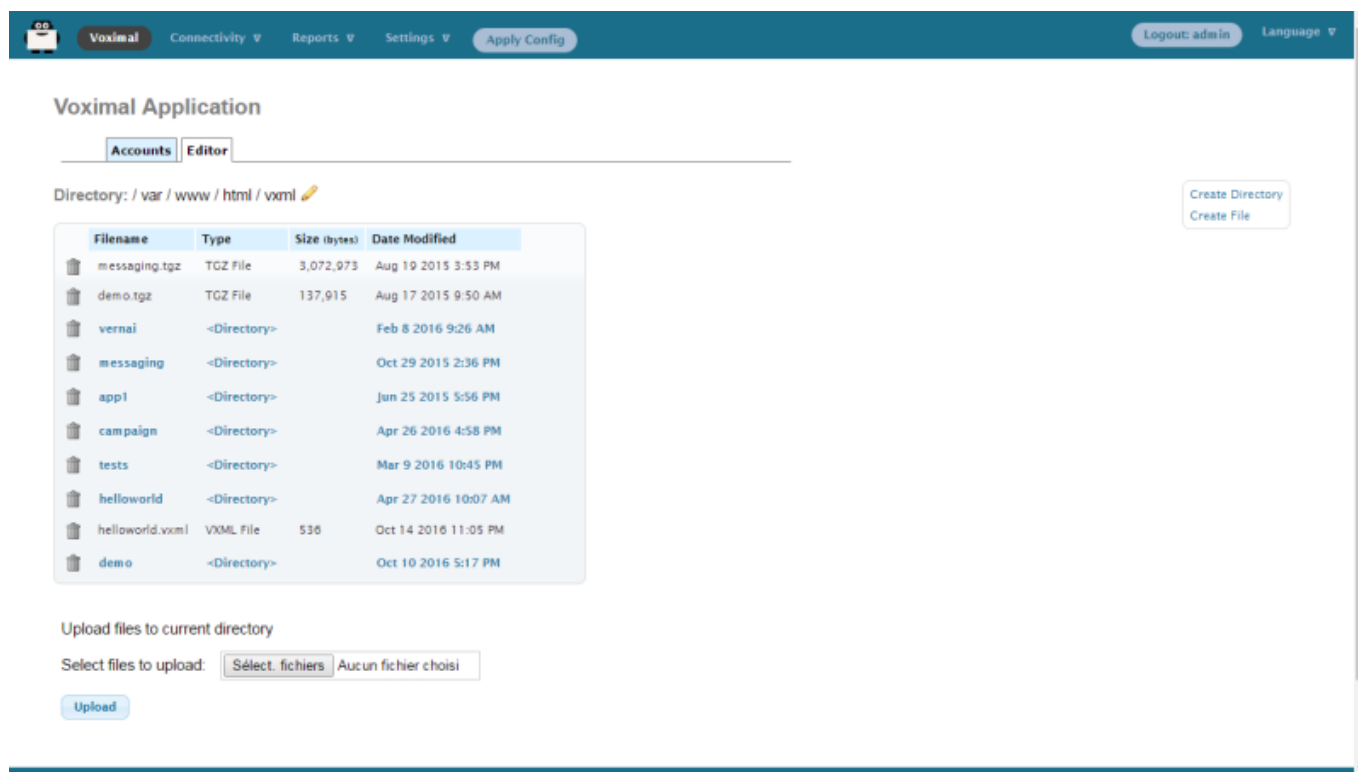
Among these types, SIP device is the most common and popular one.

You can give your extension any unique number, Display Name, password, whether allow this extension to accept inbound external calls or can make outbound external calls. can have voicemail or not etc.

2) Create a VoiceXML service

You need to create VoiceXML service, by uploading it or writing it directly with the embedded VoiceXML/PHP editor. Select the application menu **Voximal**.

Uploading VoiceXML service



In page Applications/Voximal click on **Editor** tab. At the end of the page, you have the way to upload files :

- Click on **Browse** and select your files (.vxml, .php, ...).
- Then click on **[Upload]**. Your files will appear in list.

Create VoiceXML service with editor

You can update VoiceXML file (extension .vxml), by clicking on the filename. The file content will appear, and you'll be able to update it, and check the syntax.

The editor page allow to create folders and files, and edit them.

To create folders or files, click on right items on top right corner (**Create Directory** or **Create File**).

Then enter filename, and click **[Create]**.



The screenshot shows the Voximal Application Editor interface. At the top, there is a navigation bar with 'Voximal', 'Connectivity', 'Reports', 'Settings', 'Apply Config', 'Logout: admin', and 'Language'. Below the navigation bar, the title 'Voximal Application' is displayed, followed by 'Accounts' and 'Editor' tabs. The main area shows 'Editing file: /helloworld.vxml'. A code editor displays the following VXML script:

```
1 <?xml version="1.0"?>
2 <vxml version="2.0" xmlns="http://www.w3.org/2001/vxml" xml:lang="en-US">
3   <form>
4     <block>
5       <var name="caller" expr="session.connection.remote.uri"/>
6       <var name="called" expr="session.connection.local.uri"/>
7       <var name="id" expr="telephone.id"/>
8       <var name="param" expr="telephone.param"/>
9     </block>
10    <prompt>
11      Welcome. You are on the Voximal IVR.
12      Your caller number is : <value expr="caller"/>.
13      You are calling the : <value expr="called"/>.
14      Goodbye
15    </prompt>
16  </form>
17 </vxml>
18
```

Below the code editor, there are 'Save', 'Back', and 'Check' buttons. A message below the buttons states 'The Vxml syntax is valid'. At the bottom of the interface, there is a footer with 'Version 1.0.121', '18 Oct 10:10:36', the 'voximal' logo with the tagline 'The voice interaction', and a 'Support' link.

3) Create a Voximal account

When your VoiceXML services is ready, you have to configure it in the server. To do it, add Voximal account. Select the **Voximal** menu item.

To create a Voximal Account, you have to define in minimum 2 required informations :

1. An application name, in field **Name**
2. The VoiceXML url of the service, in field **URL**. You can select a local one, in the list or writing a local or HTTP one.

There are several options :

1. **Max sessions** : you can define a specific limit, if you want lower maximum session than the license.
2. **Dial format** : you can define a specific dialout format for outgoing calls.
3. **Mark** : you can define a specific mark, that will appear in traces.
4. **Speech** : you can specify the use of the ASR. In case of using ASR server, the better way is to set **Automatic** choice.
5. **Max time** : you can set a maximum duration of call. If not setted or equals to 0, the duration is unlimited.
6. **Vxml parameter** : you can set a string to pass it to vxml script.
7. **Start delay** : you can set a time delay to start the service.

The screenshot shows the 'Voximal Application' configuration page. At the top, there is a navigation bar with 'Voximal', 'Connectivity', 'Reports', 'Settings', and 'Apply Config' buttons. On the right, there are 'Logout: admin' and 'Language' options. The main content area is titled 'Voximal Application' and has two tabs: 'Accounts' and 'Editor'. Below the tabs is a section titled 'Add Application' with a sub-header 'Add Application'. A note states: 'The fields marked with * can not be left in blank'. The form includes the following fields and options:

- Name* (text input)
- URL* (text input) with a 'Select ...' dropdown menu
- Max Sessions (text input)
- Dial Format (text input)
- Mark (text input)
- Speech (radio buttons: Emulation, No, Yes, Automatic)
- Max time (s) (text input)
- Session parameter (text input)
- Start delay (ms) (text input with value 2000)

A 'Create' button is located at the bottom left of the form. On the right side, there is a dropdown menu titled 'Add New Application' with the following options: Campaign, Demo, HelloWord, messaging-consult, messaging-depot, Saasivr, SiteVernal, Tests, and Voximal_Access.

4) Number/Routes configuration

You have your DID number and SIP Trunk set up (with the test number, the called number will be 4568 : "VXML" in the dialpad). If you set up the inbound rules, you define when people call your DID number (for example 555-555-555) how your Voxiboy handle such call, normally you set up a Voximal Account (where you define the first VoiceXML document of your voice portal).

You can to define which application you want to use by default for all incoming calls. It's done by defining the **any DID/any CID** or **All DIDs** settings in **Connectivity/Numbers** page.

1. Select **All DIDs** item (on top right corner)
2. Leave empty Field DID Number
3. Select the application to use : **Voximal Application**
 - You can select an existing application (for example Voximal)
 - Or create a new one by clicking "Add new **Voximal application**"
4. Click on **[submit]** button.

Call Detail Record

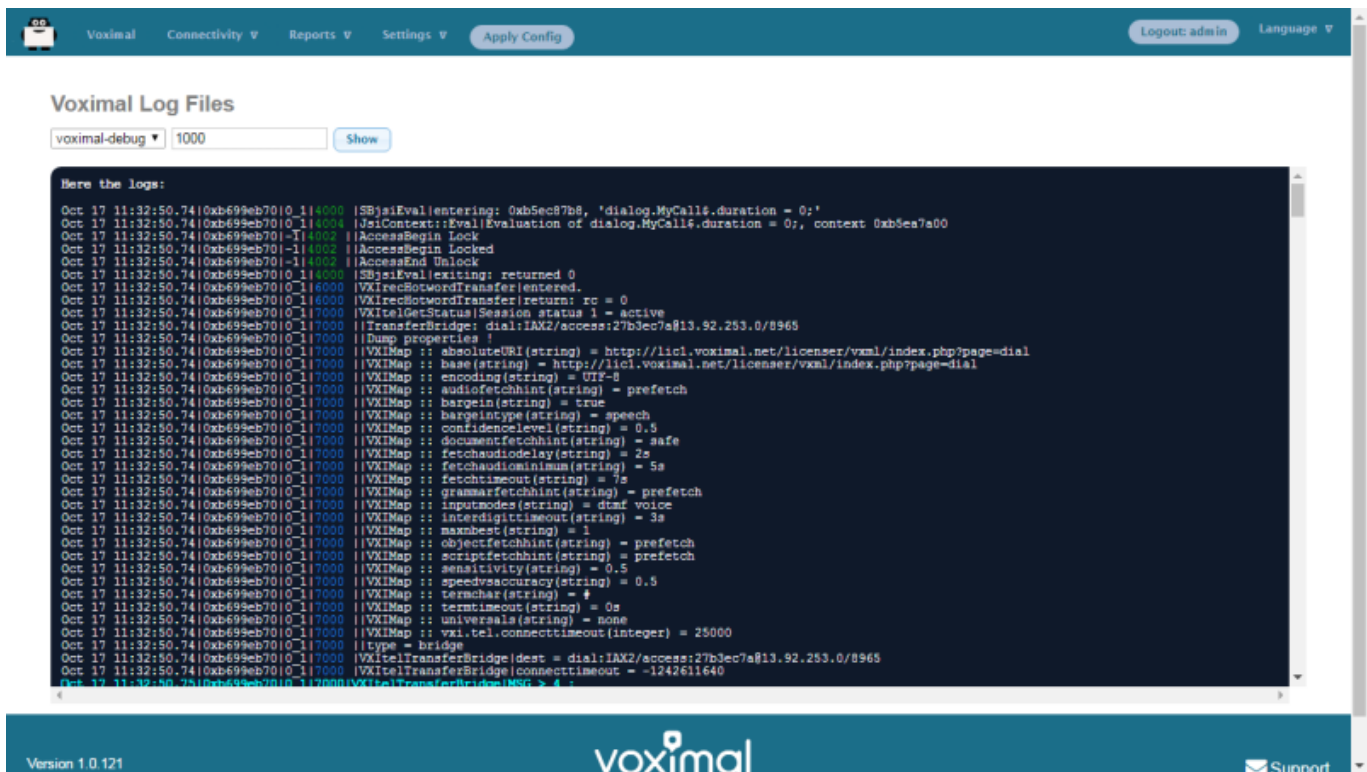
The CDR Reports allows you to view a report showing the telephone calls made from and received to your system. You can choose to view a complete history or calls, or to search by date, date range, number called, caller ID, etc.

Call Date	Recording	System	CallerID	Outbound CallerID	DID	App	Destination	Disposition	Duration	Userfield	Account
2016-10-18 00:21:13		1476742873.238	"Borje SIXTO" <11>	"Borje SIXTO" <11>		Dial	0170613300	ANSWERED	01:31		
2016-10-18 00:09:29		1476742106.222	"Borje SIXTO" <11>	"Borje SIXTO" <11>		Dial	0170613300	ANSWERED	00:38		
2016-10-18 00:02:35		1476741755.206	"Borje SIXTO" <11>	"Borje SIXTO" <11>		Dial	0170613300	ANSWERED	00:46		
2016-10-17 23:56:13		1476741373.190	"Borje SIXTO" <11>	"Borje SIXTO" <11>		Dial	0170613300	ANSWERED	00:46		
2016-10-17 23:50:50		1476741050.174	"Borje SIXTO" <11>	"Borje SIXTO" <11>		Dial	0170613300	ANSWERED	01:27		

Logs

The Asterisk Logfiles Module is an easy way to view portions of the Asterisk Log. However, this Module is only useful when you want to view a very recent event in the Asterisk Log.

You have similar Module for the Voximal log :



Settings

a) General

The screenshot shows the 'Voximal Settings' page with the 'License' tab selected. The settings are as follows:

UID	d2a9fd5e-6d96-11e5-b2c8-8d0ac9e2f828
License	Remote Local
Max sessions	5
TTS	Enable
ASR	Enable
License status	ok

Buttons: Save, Discard

Footer: Version 1.0.121, 18 Oct 10:10:03, voximal The voice interaction, Support

b) TextToSpeech

The screenshot shows the 'Voximal Settings' page with the 'TextToSpeech' tab selected. The settings are as follows:

API	HTTP or MRCP
URI	http://tts.voximal.net/tts/cereproc/tts.php
Method	POST GET ASTERISK
Format	wav wav16 pcm alaw ulaw raw s16 s1616
SSML	Yes No
Cache ageing	-1
Cut prompt	Yes No

Buttons: Save, Discard, Cache clear

Footer: Version 1.0.121, 18 Oct 10:10:47, voximal The voice interaction, Support

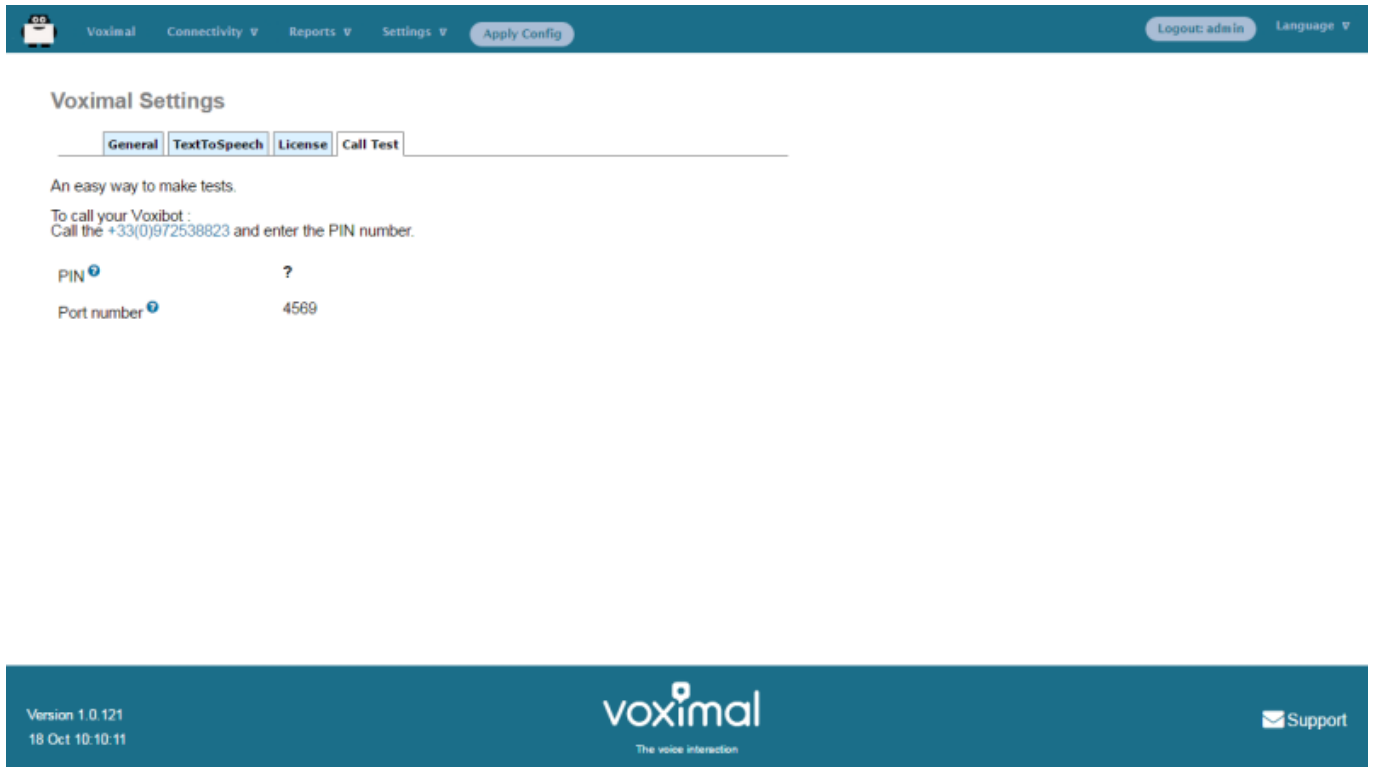
The home page after login show you the system status.

- Host name of the server
- Summary : Main modules status
- Interpreter statistics

- Telephony statistics
- Uptime / Load average

d) License

c) Test number



The screenshot shows the 'Voximal Settings' page with the 'License' tab selected. The page includes a navigation bar with 'Voximal', 'Connectivity', 'Reports', 'Settings', and 'Apply Config' buttons. The 'License' tab is active, and the 'Call Test' tab is also visible. The main content area contains instructions for calling a Voxibot and a table with fields for 'PIN' and 'Port number'.

Voximal Settings

General TextToSpeech License Call Test

An easy way to make tests.
To call your Voxibot :
Call the +33(0)972538823 and enter the PIN number.

PIN	?
Port number	4569

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Configuration files

- `/etc/voximald.conf` (not modify this file)
- `/etc/asterisk/voximal.conf` (not modify directly this file)

From:
<https://wiki.voximal.com/> - **Voximal documentation**

Permanent link:
https://wiki.voximal.com/doku.php?id=installation_guide:configuration:start&rev=1478553435

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