

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).

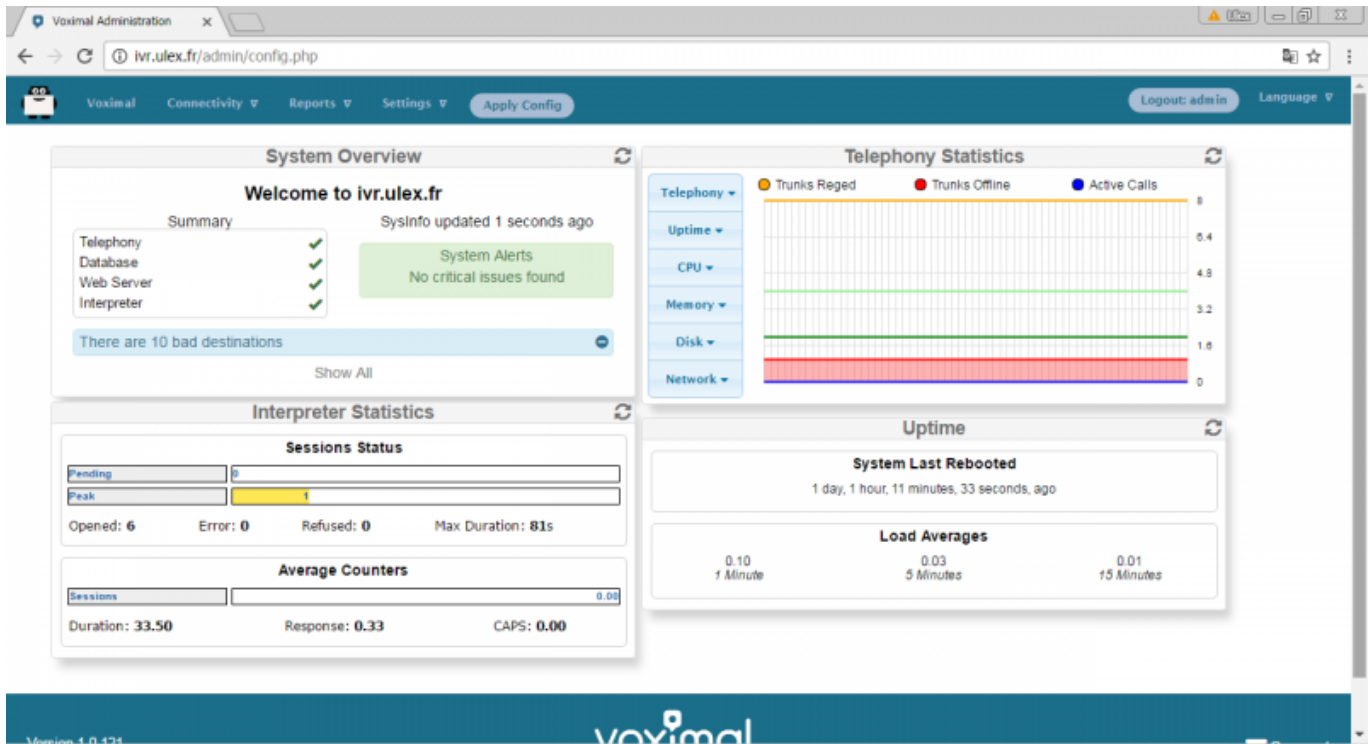
Each login has a default password (requested after the package installation, or the image activation), you can change them with the root account.



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)



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

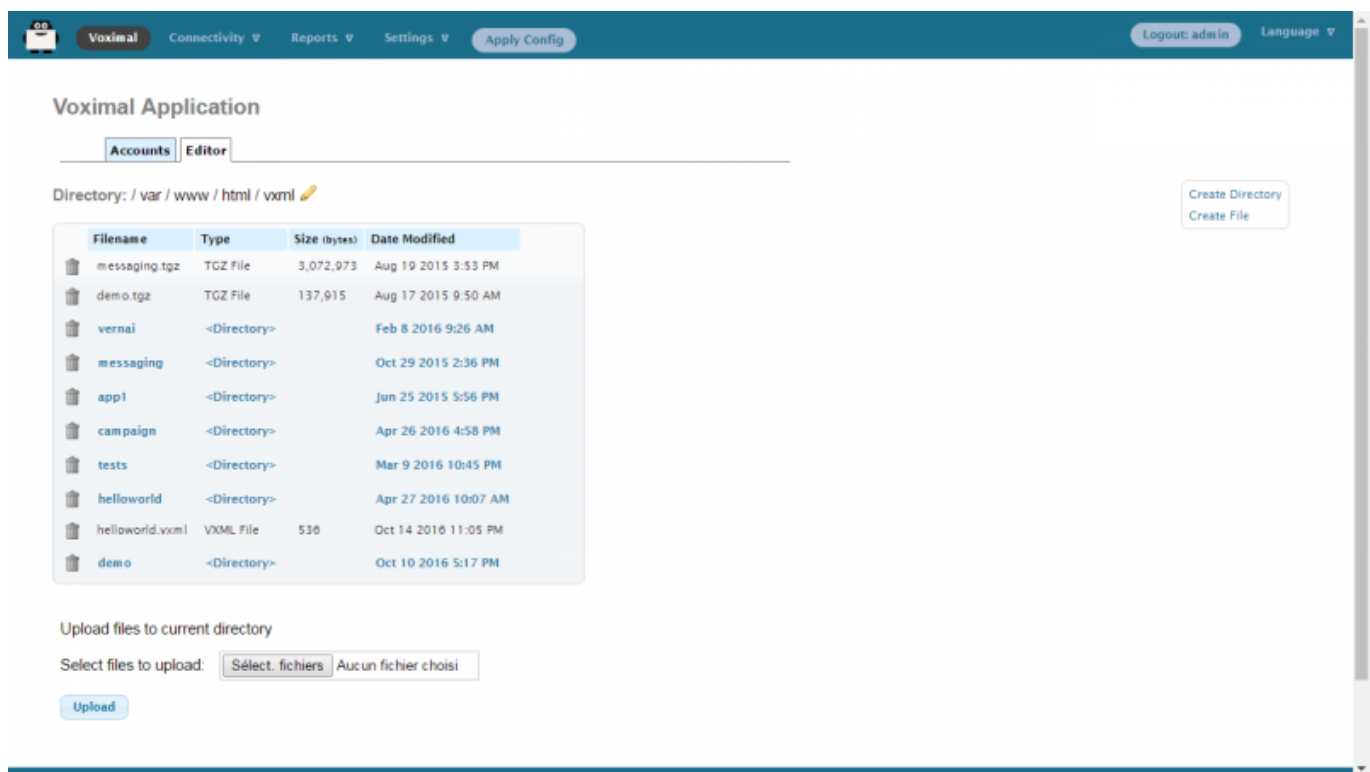
Among these types, SIP device is the most common and popular one.

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]**.



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.

Voximal Application

Accounts Editor

Add Application

The fields marked with * can not be left in blank

Name*

URL*

Max Sessions

Dial Format

Mark

Speech ☒ Emulation ☐ No ☐ Yes ☐ Automatic

Max time (s)

Session parameter

Start delay (ms)

Add New Application

- Campaign
- Demo
- HelloWord
- messaging-consult
- messaging-depot
- Saasivr
- SiteVernal
- Tests
- 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.

VoximalConnectivity▼Reports▼Settings▼Apply Config

Logout: adminLanguage▼

Add Incoming Route

Add Incoming Route

All DIDs (toggle sort)

User DIDs

General DIDs

Unused DIDs

any DID / any CID

12345

test / any CID

Demo

01 / any CID

Echo Test

700 / any CID

Coldorak

0034011413999 / any CID

HelloWorld

8905 / any CID

Tests

00 / any CID

Voximal Demo

33972538733 / any CID

Voximal Test

0033972538823 / any CID

Add Incoming Route

Description:

DID Number:

CallerID Number:

CID Priority Route: ☐

Options

Alert Info:

CID name prefix:

Signal RINGING: ☐

Reject Reverse Charges: ☐

Pause Before Answer: ☐

Privacy

Privacy Manager: No

Call Recording

Call Recording: Allow

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.

VoximalConnectivity▼Reports▼Settings▼Apply Config

Logout: adminLanguage▼

CDR Reports

Call Detail Record Search

Order By

Search conditions

Extra options

Call Date: 2016-10-18 00:21:13

CallerID Number: 1476742873.238

CallerID Name: "Borja SIXTO" <11>

Outbound CallerID Number: "Borja SIXTO" <11>

DID: "Borja SIXTO" <11>

Destination: 0170613300

Destination CallerID Name: "Borja SIXTO" <11>

Userfield: "Borja SIXTO" <11>

Account Code: "Borja SIXTO" <11>

Duration: 01:31

Disposition: ANSWERED

Between: 00:00 And: 23:59

Group By: Day

Search

Report type: CSV file

Result limit: 100

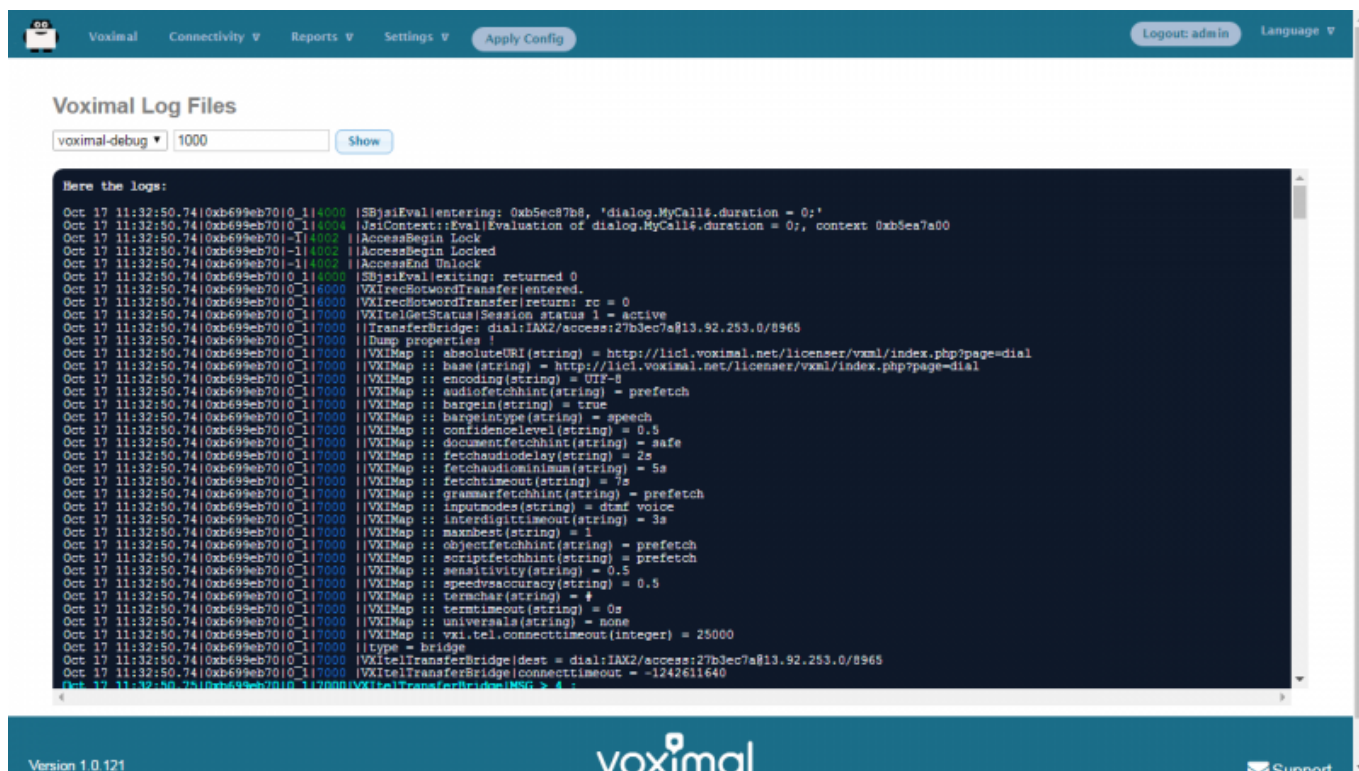
Call Detail Record - Search Returned 100 Calls

Call Date	Recording	System	CallerID	Outbound CallerID	DID	App	Destination	Disposition	Duration	Userfield	Account
2016-10-18 00:21:13		1476742873.238	"Borja SIXTO" <11>	"Borja SIXTO" <11>		Dial	0170613300	ANSWERED	01:31		
2016-10-18 00:08:29		1476742106.222	"Borja SIXTO" <11>	"Borja SIXTO" <11>		Dial	0170613300	ANSWERED	00:38		
2016-10-18 00:02:35		1476741755.206	"Borja SIXTO" <11>	"Borja SIXTO" <11>		Dial	0170613300	ANSWERED	00:46		
2016-10-17 23:56:13		1476741373.190	"Borja SIXTO" <11>	"Borja SIXTO" <11>		Dial	0170613300	ANSWERED	00:46		
2016-10-17 23:50:50		1476741050.174	"Borja SIXTO" <11>	"Borja 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.

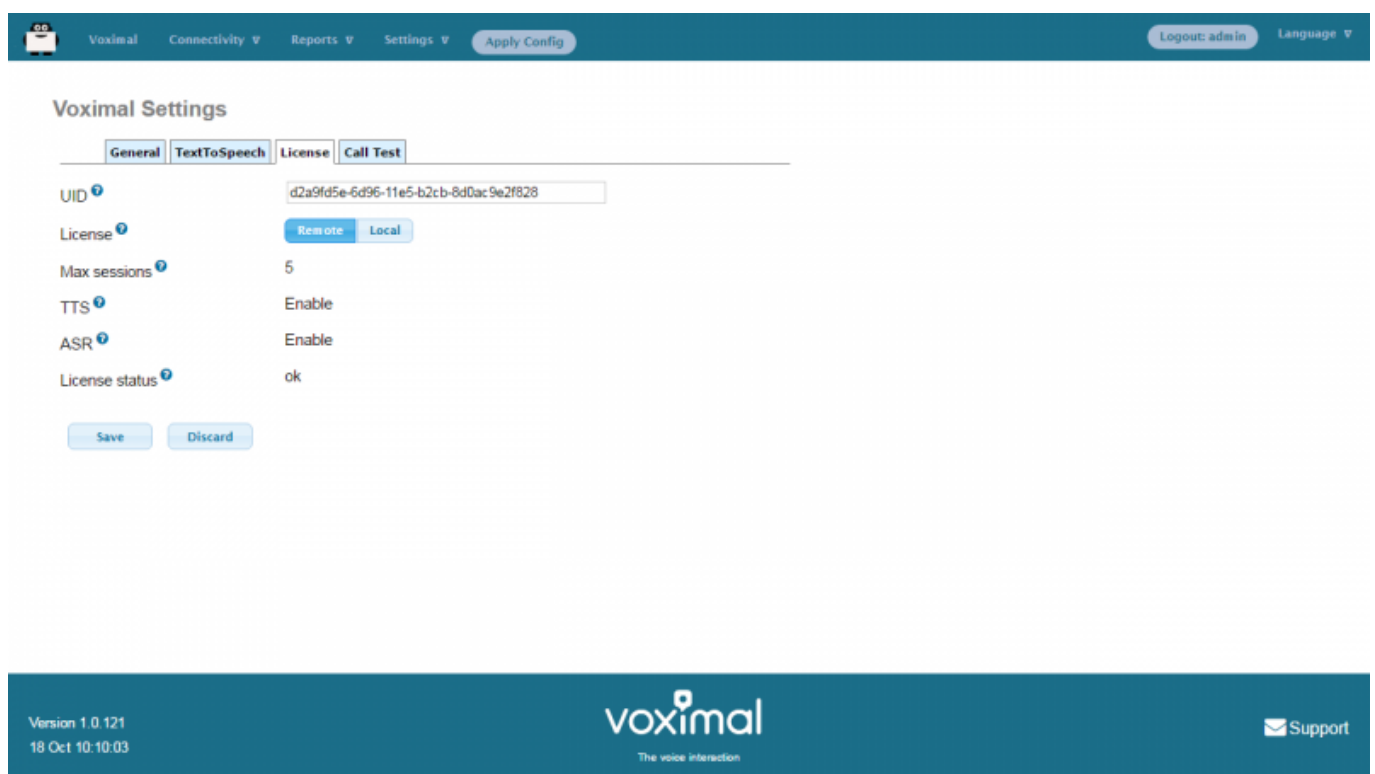
You have similar Module for the Voximal log :



The screenshot shows the Voximal web interface. At the top, there's a navigation bar with links for Voximal, Connectivity, Reports, Settings, and an Apply Config button. A user is logged in as 'admin'. Below the navigation bar, the 'Voximal Log Files' section is active. It features a dropdown menu set to 'voximal-debug' and a text input field containing '1000', with a 'Show' button next to it. The main area displays a list of log entries, each starting with a timestamp and a log level (e.g., 'Oct 17 11:32:50.7410xb699eb7010-114000'). The log entries include various system messages and configuration details, such as 'ISBjsiEval[entering: 0xbSec87b8, 'dialog.MyCall\$.duration = 0;'' and 'VXIMap :: base(string) = http://lic1.voximal.net/licenses/vxml/index.php?page=dial'. The interface also shows the version 'Version 1.0.121' and a 'Support' link in the bottom right corner.

Settings

a) General



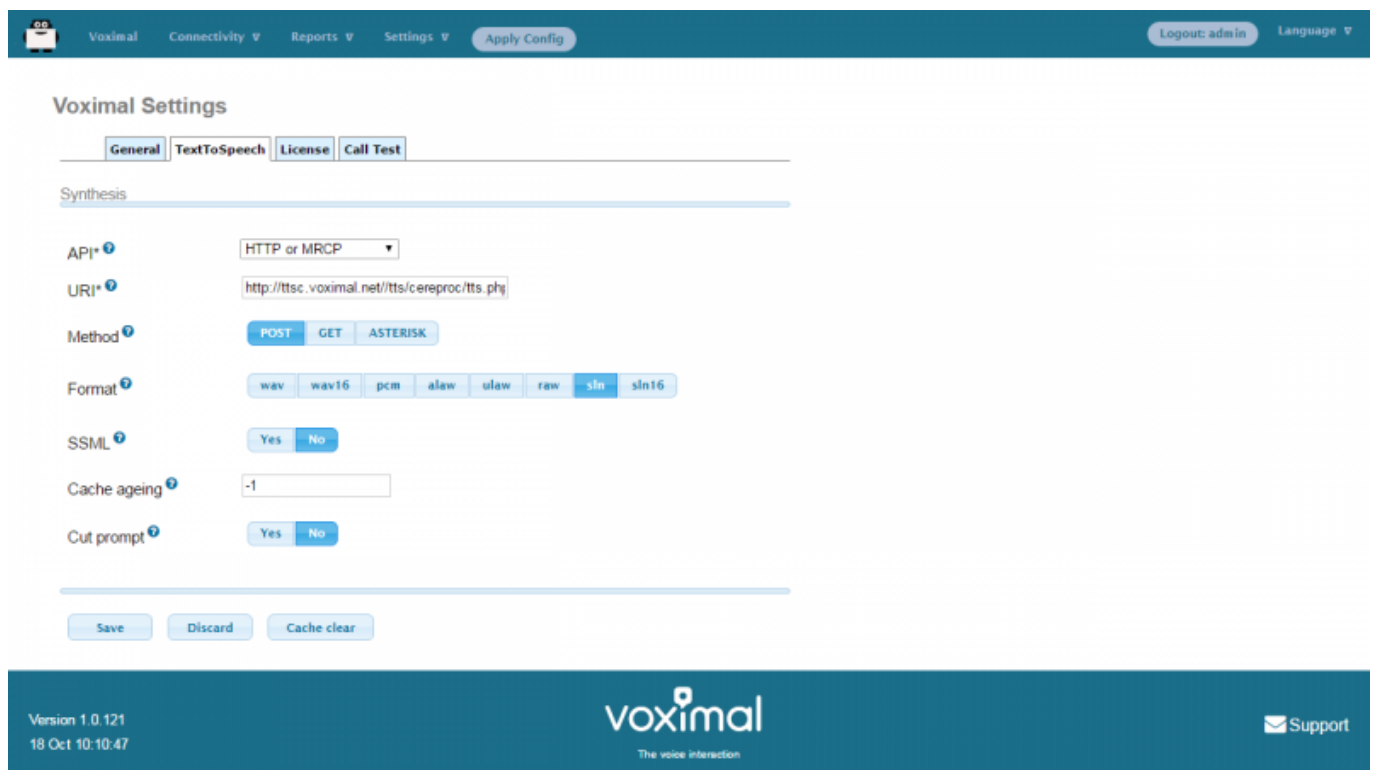
The screenshot shows the 'Voximal Settings' page with the 'License' tab selected. The 'General' tab is also visible. The 'License' tab contains the following fields and options:

- UID: d2a9fd5e-6d96-11e5-b2c8-8d0ac9e2f828
- License: Remote (selected), 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 'General' tab is also visible. The 'TextToSpeech' tab contains the following fields and options:

- Synthesis: HTTP or MRCP
- URI: http://tts.voximal.net/tts/cereproc/tts.php
- Method: POST (selected), GET, ASTERISK
- Format: wav, wav16, pcm, alaw, ulaw, raw, sin (selected), sin16
- 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

 Voximal Connectivity ▼ Reports ▼ Settings ▼ Apply Config Logout: admin Language ▼


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

Version 1.0.121
18 Oct 10:10:11


The voice interaction

 Support

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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