
XiVO Solutions Documentation

Release

Avencall

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XiVO Solutions is a complete solution for enterprise communications and contact centre infrastructure. XiVO is a PABX application, XiVO-CC is an application suite for contact centers.

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1.1 XiVO Documentation

XiVO is an application suite based on several free existing components including [Asterisk](#), and our own developments to provide communication services (IPBX, Unified Messaging, ...) to businesses.

XiVO is [free software](#). Most of its distinctive components, and XiVO as a whole, are distributed under the *GPLv3 license*.

You may also check the [XiVO blog](#) for more information.

XiVO documentation is also available as a downloadable HTML, EPUB or PDF file. See the [downloads page](#) for a list of available files or use the menu on the lower right.

1.1.1 Table of Contents

Introduction

XiVO is a PABX application based on several free existant components including Asterisk and our own developments. XiVO provides a solution for enterprises who wish to replace or add telephone services (PABX).

XiVO is free software. Most of its distinctive components, and XiVO as a whole, are distributed under the GPLv3 license.

XiVO History

XiVO was created in 2005 by Sylvain Boily (Proformatique SARL). The XiVO mark is now owned by Avencall SAS after a merge between Proformatique SARL and Avencall SARL in 2010. The XiVO core team works for Proformatique INC in Quebec City since 2010, after Sylvain Boily moved to Quebec city.

[XiVO 1.2](#) was released on February 3, 2012.

Installation

Installing the System

Please refer to the section [Troubleshooting](#) if ever you have errors during the installation.

There are two official ways to install XiVO:

- using the official ISO image
- using a minimal Debian installation and the XiVO installation script

XiVO can be installed on both virtual (QEMU/KVM, VirtualBox, ...) and physical machines. That said, since Asterisk is sensitive to timing issues, you might get better results by installing XiVO on real hardware.

Installing from the ISO image

- Download the ISO image. ([latest version](#)) ([all versions](#))
- Boot from the ISO image, select `Install` and follow the instructions. You must select a locale with charset UTF-8.
- At the end of the installation, you can continue by running the *configuration wizard*.

During the installation of Debian, only a proxy that supports proxying http/https requests may eventually be entered. Otherwise GPG key of XiVO repository will not be installed and must be added manually:

```
wget http://mirror.xivo.solutions/xivo_current.key -O - | apt-key add -
```

Installing from a minimal Debian installation XiVO can be installed directly over a **32-bit** or a **64-bit** Debian jessie. When doing so, you are strongly advised to start with a clean and minimal installation of Debian jessie.

The latest installation image for Debian jessie can be found at <https://www.debian.org/releases/jessie/debian-installer>.

Requirements The installed Debian must:

- use the architecture `i386` or `amd64`
- have a default locale with charset UTF-8

In case you want to migrate a XiVO from `i386` to `amd64`, see *Migrate XiVO from i386 (32 bits) to amd64 (64 bits)*.

Installation Once you have your Debian jessie properly installed, download the XiVO installation script and make it executable:

```
wget http://mirror.xivo.solutions/xivo_install.sh
chmod +x xivo_install.sh
```

And run it:

```
./xivo_install.sh
```

At the end of the installation, you can continue by running the *configuration wizard*.

Alternatives versions The installation script can also be used to install an *archive version* of XiVO (14.18 or later only). For example, if you want to install XiVO 16.03:

```
./xivo_install.sh -a 16.03
```

When installing an archive version, note that:

- versions 14.18 to 15.19 of XiVO can only be installed on a Debian 7 (wheezy) system
- the 64-bit versions of XiVO are only available starting from 15.16

You may also install development versions of XiVO with this script. These versions may be unstable and should not be used on a production server. Please refer to the usage of the script:

```
./xivo_install.sh -h
```

Other installation methods It's also possible to install XiVO by PXE. It is not documented here.

Running the Wizard

After the system installation, you must go through the wizard before being able to use your XiVO. Open your browser and enter your server's IP address in the navigation bar. (For example: <http://192.168.1.10>)

Language You first have to select the language you want to use for the wizard.

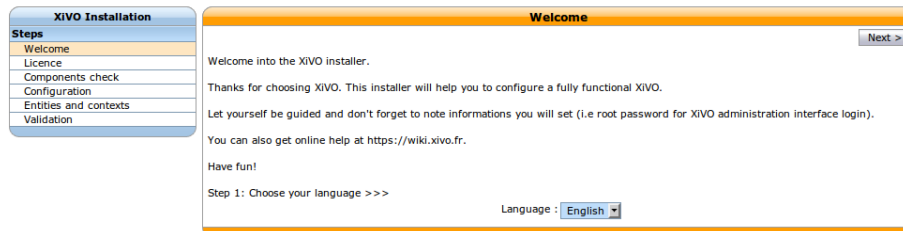


Fig. 1.1: Select the language

License You then have to accept the *GPLv3 License* under which XiVO is distributed.

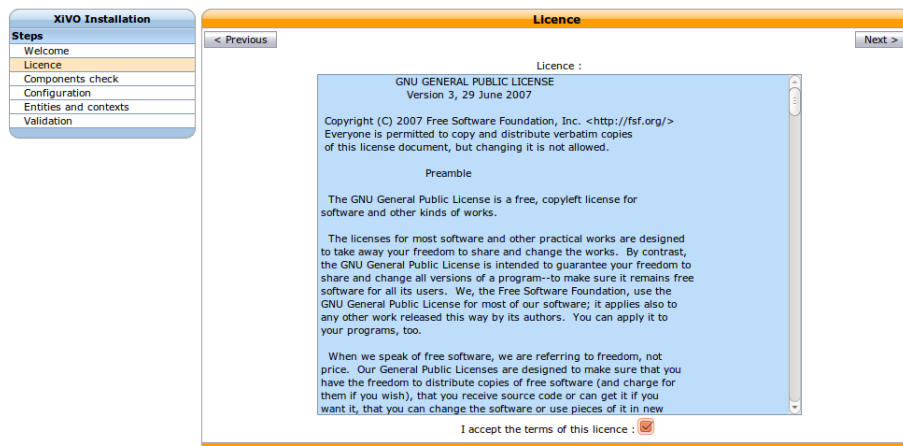


Fig. 1.2: Accept the license

Configuration

1. Enter the hostname (Allowed characters are : A-Z a-z 0-9 -)
2. Enter the domain name (Allowed characters are : A-Z a-z 0-9 - .)
3. Enter the password for the `root` user of the web interface,
4. Configure the IP address and gateway used by the VoIP interface
5. Finally, modify the DNS server information if needed.

Entities and Contexts Contexts are used for managing various phone numbers that are used by your system.

- The Internal calls context manages extension numbers that can be reached internally
- The Incalls context manages calls coming from outside of your system
- The Outcalls context manages calls going from your system to the outside

The screenshot shows the 'Configuration' screen in the XiVO installation wizard. On the left, a 'Steps' sidebar lists: Welcome, Licence, Configuration (highlighted), Entities and contexts, and Validation. The main area is titled 'Configuration' and has '< Previous' and 'Next >' buttons. It contains five numbered sections:

- 1. Hostname:** A text field with 'xivo' entered.
- 2. Domain name:** An empty text field.
- 3. WebInterface root password:** Two text fields for 'Password' and 'Re-enter password'.
- 4. Interface VoIP:** Two dropdown menus. 'Address' is set to '192.168.16.134 (eth0)' and 'Default gateway' is set to '192.168.16.254 (eth0)'.
- 5. DNS servers:** Two text fields for 'Primary' (192.168.16.128) and 'Secondary'.

Fig. 1.3: Basic configuration

The screenshot shows the 'Entities and contexts' screen in the XiVO installation wizard. The 'Steps' sidebar on the left lists: Welcome, Licence, Components check, Configuration, Entities and contexts (highlighted), and Validation. The main area is titled 'Entities and contexts' and has '< Previous' and 'Next >' buttons. It contains four numbered sections:

- 1. Entity:** A text field for '* Printed name'.
- 2. Internal calls context:** Three text fields: '* Printed name' (Default), '* Numbers interval start', and '* Numbers interval end'.
- 3. Incalls context:** Four text fields: '* Printed name' (Incalls), 'Numbers interval start', 'Numbers interval end', and 'DID length' (4).
- 4. Outcalls context:** A text field for '* Printed name' (Outcalls).

Fig. 1.4: Entities and Contexts

1. Enter the entity name (e.g. your organization name) (Allowed characters are : A-Z a-z 0-9 - .)
2. Enter the number interval for you internal context. The interval will define the users's phone numbers for your system (you can change it afterwards)
3. Enter the DID range and DID length for your system.
4. You may change the name of your outgoing calls context.

Validation Finally, you can validate your configuration by clicking on the `Validate` button. Note that if you want to change one of the settings you can go backwards in the wizard by clicking on the `Previous` button.

Warning: This is the last time the `root` password will be displayed. Take care to note it.

Congratulations, you now have a fully functional XiVO server.

To start configuring XiVO, see [Getting Started](#).

Post Installation

Here are a few configuration options that are commonly changed once the installation is completed. Please note that these changes are optional.

Display called name on internal calls When you call internally another phone of the system you would like your phone to display the name of the called person (instead of the dialed number only). To achieve this you must change the following SIP options:

- *Services* → *IPBX* → *General settings* → *SIP Protocol* → *Default*:
 - Trust the Remote-Party-ID: yes,
 - Send the Remote-Party-ID: select `PAI`

Incoming caller number display The caller ID number on incoming calls depends on what is sent by your operator. You can modify it via the file `/etc/xivo/asterisk/xivo_in_callerid.conf`.

Note: The reverse directory lookup use the caller ID number after it has been modified by `xivo_in_callerid.conf`

Examples:

- If you use a prefix to dial outgoing numbers (like a 0) you should add a 0 to all the `add =` sections,
- You may want to display incoming numbers in E.164 format. For example, you can change the `[national1]` section to:

```
callerid = ^0[1-9]\d{8}$
strip = 1
add = +33
```

To enable the changes you have to restart `xivo-agid`:

```
service xivo-agid restart
```

Time and date

- Configure your locale and default time zone device template => *Configuration* → *Provisioning* → *Template Device* by editing the default template
- Configure the timezone in => *Services* → *IPBX* → *General settings* → *Advanced* → *Timezone*

- If needed, reconfigure your timezone for the system:

```
dpkg-reconfigure tzdata
```

Codecs You should also select default codecs. It obviously depends on the telco links, the country, the phones, the usage, etc. Here is a typical example for Europe (the main goal in this example is to select *only* G.711 A-Law instead of both G.711 A-Law and G.711 μ -Law by default):

- SIP : *Services* \rightarrow *IPBX* \rightarrow *General settings* \rightarrow *SIP Protocol* \rightarrow *Signaling*:
 - Customize codec : enabled
 - Codec list:

```
G.711 A-Law
G.722
G.729A
H.264
```

- IAX2 : *Services* \rightarrow *IPBX* \rightarrow *General settings* \rightarrow *IAX Protocol* \rightarrow *Default*:
 - Customize : enabled
 - Codec list:

```
G.711 A-Law
G.722
G.729A
H.264
```

Getting Started

This section will show you how to create a user with a SIP line. This simple use case covers what a lot of people need to start using a phone. You can use these steps for configuring a phone (e.g a softphone, an Analog-to-Digital switch or a SIP phone).

This tutorial doesn't cover how to automatically provision a *supported device*. For this, consult the *provisioning section*.

We first need to log into the XiVO web interface. The web interface is where you can administer the whole system.

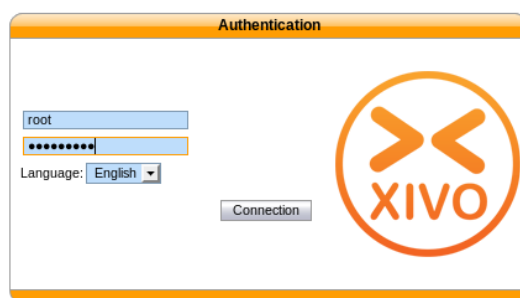


Fig. 1.5: Logging into the XiVO

When logged in, you will see a page with all the status information about your system. This page helps you monitor the health of your system and gives you information about your network. Please note the IP address of your server, you will need this information later on when you will configure your device (e.g. phone)

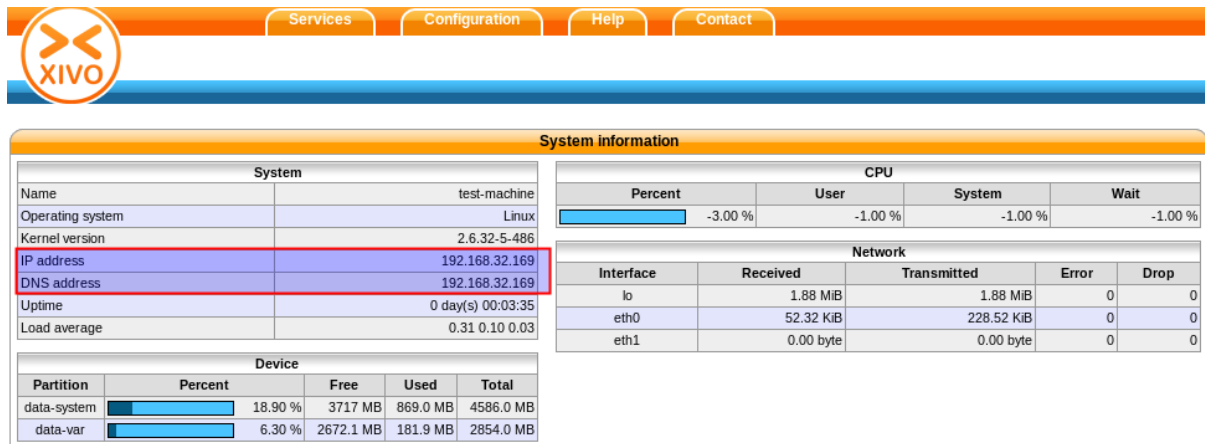


Fig. 1.6: System informations

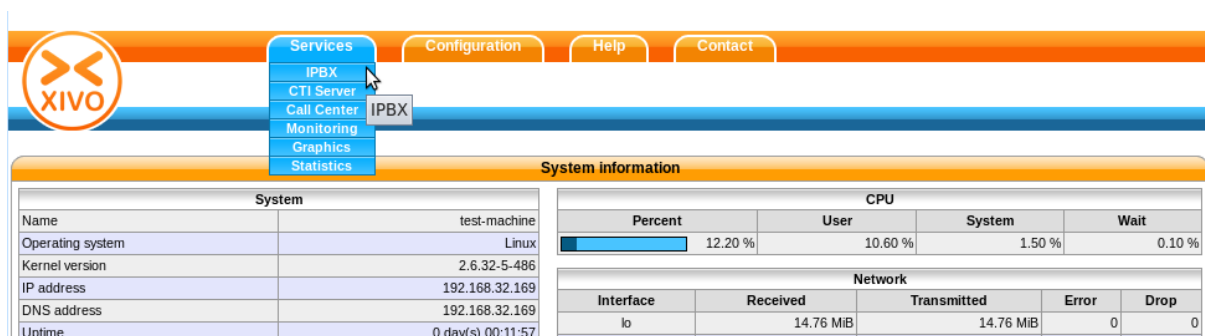


Fig. 1.7: Menu IPBX

To configure a device for a user, start by navigating to the IPBX menu. Hover over the *Services* tab, a dropdown menu will appear. Click on *IPBX*.

Select the *Users* setting in the left menu.

From here, press on the “plus” sign. A pop up will appear where you can click on *Add*.

We now have the form that will allow us to create a new user. The three most important fields are ‘First name’, ‘Last name’ and ‘Language’. Fill in the fields and click on *Save* at the bottom. For our example, we will create a user called ‘Alice Wonderland’.

Afterwards, click on the “Lines” tab.

Enter a number for your phone. If you click inside the field, you will see the range of numbers you can use. For our example, we will use ‘1000’.

By default, the selected protocol is SIP, which is what we want for now. Click on *Save* to create the line.

We now have a user named ‘Alice Wonderland’ with the phone number ‘1000’.

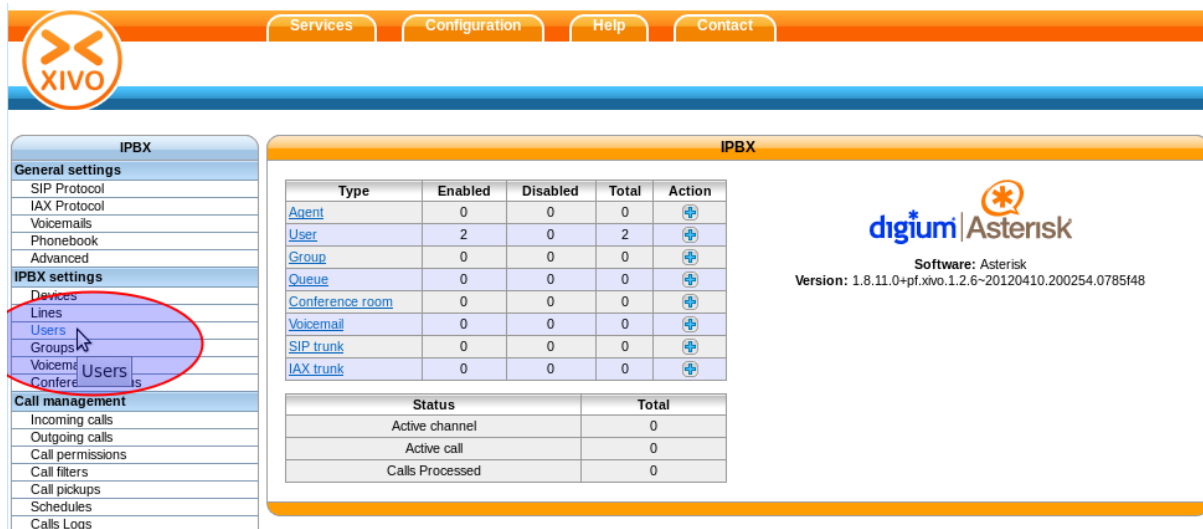
Now we need to go get the SIP username and password to configure our phone. Go back to the IPBX menu on the left, and click on ‘Lines’.

You will see a line associated with the user we just created. Click on the pencil icon to edit the line.

We can now see the username and password for the SIP line. you can configure your phone using the IP for your server, the username and the password.

Upgrading

Upgrading a XiVO is done by executing commands through a terminal on the server. You can connect to the server either through SSH or with a physical console.



IPBX

General settings

- SIP Protocol
- IAX Protocol
- Voicemails
- Phonebook
- Advanced

IPBX settings

- Devices
- Lines
- Users**
- Groups
- Voicemail
- Conferences

Call management

- Incoming calls
- Outgoing calls
- Call permissions
- Call filters
- Call pickups
- Schedules
- Calls Logs

IPBX

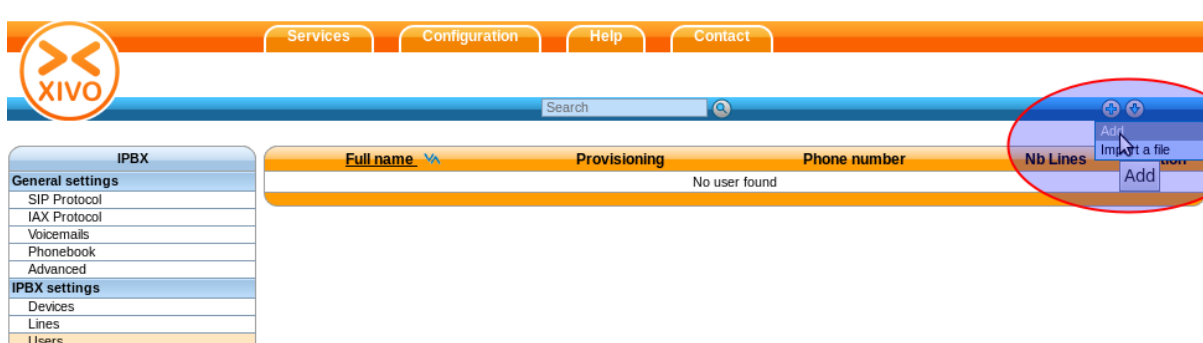
Type	Enabled	Disabled	Total	Action
Agent	0	0	0	+
User	2	0	2	+
Group	0	0	0	+
Queue	0	0	0	+
Conference room	0	0	0	+
Voicemail	0	0	0	+
SIP trunk	0	0	0	+
IAX trunk	0	0	0	+

digium|Asterisk

Software: Asterisk
Version: 1.8.11.0+pf.xivo.1.2.6~20120410.200254.0785f48

Status	Total
Active channel	0
Active call	0
Calls Processed	0

Fig. 1.8: Users settings



IPBX

General settings

- SIP Protocol
- IAX Protocol
- Voicemails
- Phonebook
- Advanced

IPBX settings

- Devices
- Lines
- Users**

Search

Full name	Provisioning	Phone number	Nb Lines
No user found			

Add

Import a file

Add

Fig. 1.9: Adding a new line

Users > Add

General Lines No answer Services Voicemail Groups Func Keys

First name: Alice
Last name: Wonderland
User picture: Browse...

Mobile phone number:

Create a schedules
Ringing time: 30 seconds
Simultaneous calls: 5
On-Hold Music: default
Language: en_US
Timezone:

Caller ID: Alice Wonderland
Outgoing Caller ID: Default
Preprocess subroutine:

User field:

XIVO Client

Enable XIVO Client: ☐
Login:
Password:
Profile: Client

Description:

Save

Fig. 1.10: User information

Users > Add

General Lines No answer Services Voicemail Groups Func Keys

Entity: machine-test

	Protocol	Name	Context	Number	Site	Device	Line (N°)	
	No line							

Save

Fig. 1.11: Lines menu

Users > Add

General Lines No answer Services Voicemail Groups Func Keys

Entity: machine-test

	Protocol	Name	Context	Number	Site	Device	Line (N°)	
1	SIP		Default	1000 - 1500	local			

Save

Fig. 1.12: Line information

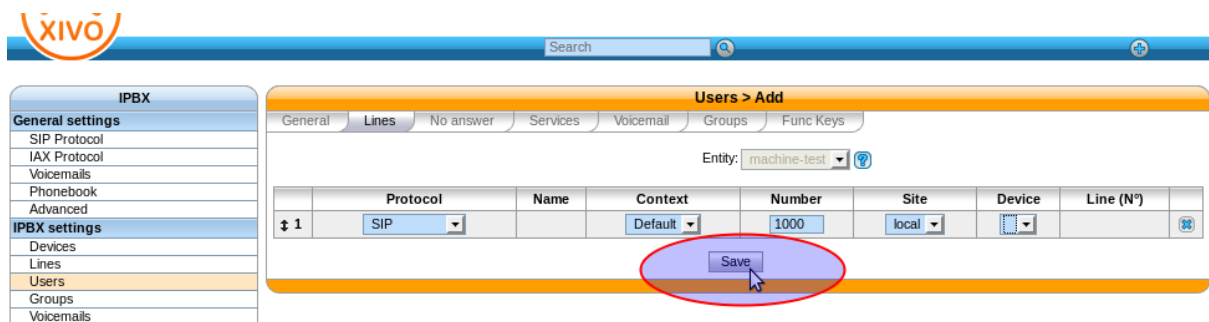


Fig. 1.13: Save

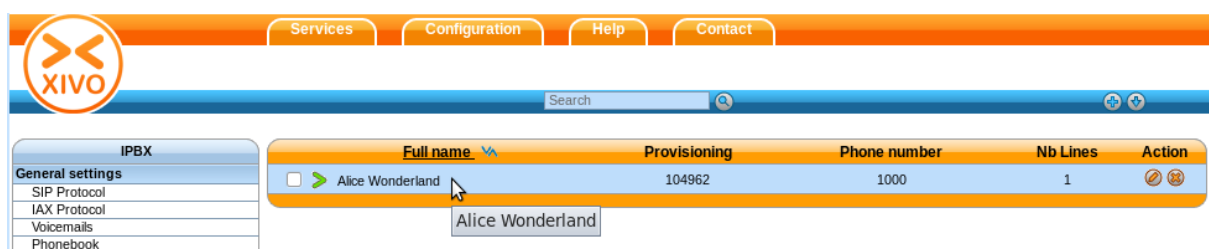


Fig. 1.14: User added information

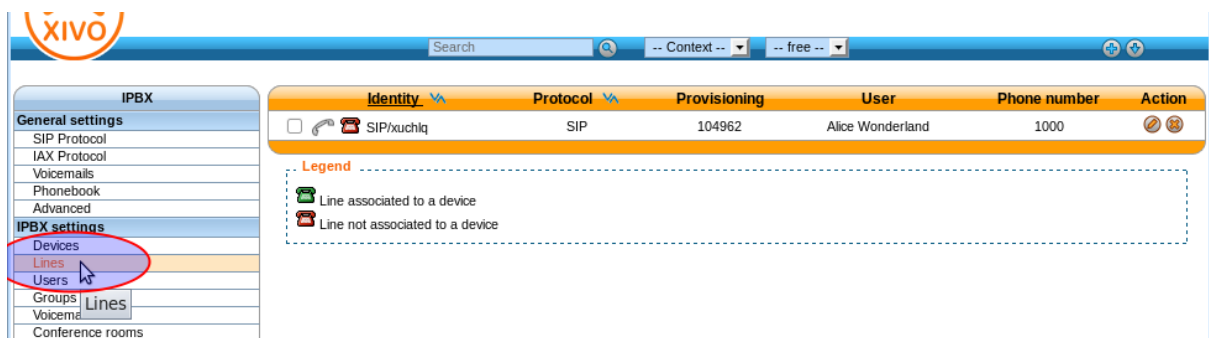


Fig. 1.15: Lines information

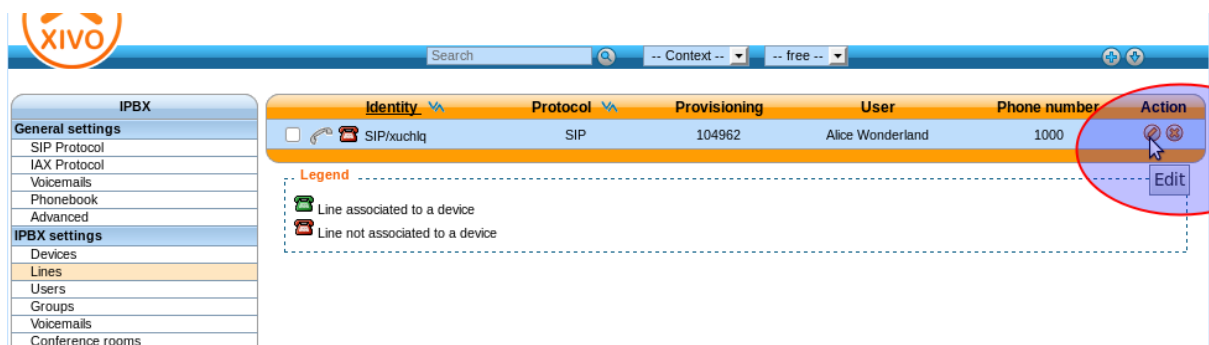


Fig. 1.16: Edit line

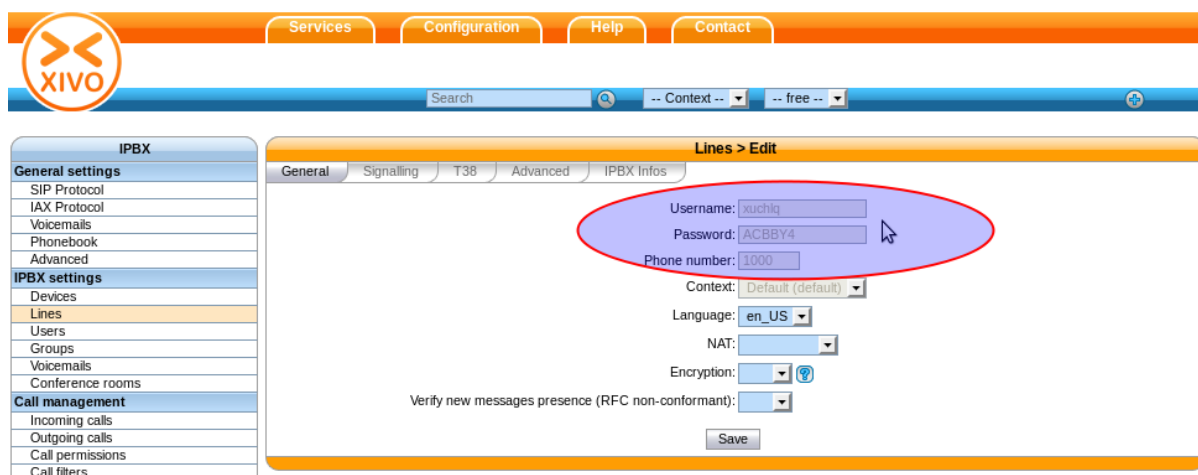


Fig. 1.17: General line information

To upgrade your XiVO to the latest version, you **must** use the `xivo-upgrade` script. You can start an upgrade with the command:

```
xivo-upgrade
```

Note:

- You can't use `xivo-upgrade` if you have not run the wizard yet
- Upgrading from a version prior to XiVO 1.2 is not supported.
- When upgrading XiVO, you **must** also upgrade **all** associated XiVO Clients. There is currently no retro-compatibility on older XiVO Client versions.

This script will update XiVO and restart all services.

There are 2 options you can pass to `xivo-upgrade`:

- `-d` to only download packages without installing them. **This will still upgrade the package containing `xivo-upgrade` and `xivo-service`.**
- `-f` to force upgrade, without asking for user confirmation

`xivo-upgrade` uses the following environment variables:

- `XIVO_CONFD_PORT` to set the port used to query the *HTTP API of xivo-confd* (default is 9486)

Upgrade procedure

- Consult the [roadmaps](#) starting from your current version to the current prod version.
- Read all existing Upgrade Notes (see below) starting from your version to the latest version.
- For custom setups, follow the required procedures described below (e.g. HA cluster).
- To download the packages beforehand, run `xivo-upgrade -d`. This is not mandatory, but it does not require stopping any service, so it may be useful to reduce the downtime of the server while upgrading.
- When ready, run `xivo-upgrade` which will start the upgrade process. **Telephony services will be stopped during the process**
- When finished, check that all services are running (the list is displayed at the end of the upgrade).
- Check that services are correctly working like SIP registration, ISDN link status, internal/incoming/outgoing calls, XiVO Client connections etc.

Version-specific upgrade procedures

Switch to xivo.solutions To follow Avencall's official releases you must switch to xivo.solutions mirrors. In order to do that follow the following procedure:

1. Download the switch-to-xivo-solutions script:

```
wget http://mirror.xivo.solutions/debian/tools/migration-tools/switch-to-xivo-solutions.sh
chmod +x ./switch-to-xivo-solutions.sh
```

2. Execute the script:

```
./switch-to-xivo-solutions.sh

...

Your XiVO has been switched to xivo.solutions successfully.
Votre XiVO a été basculé vers xivo.solutions avec succès.
```

3. Update the sources list:

```
apt-get update
```

Upgrading from XiVO 14.11 and before When upgrading from XiVO 14.11 or earlier, you must do the following, before the normal upgrade:

```
sed -i 's/xivo\.fr/xivo.solutions/g' /etc/apt/sources.list.d/*.list
```

Upgrading from XiVO 14.01, 14.02, 14.03, 14.04 installed from the ISO In those versions, xivo-upgrade keeps XiVO on the same version. You must do the following, before the normal upgrade:

```
echo "deb http://mirror.xivo.solutions/debian/ xivo-five main" > /etc/apt/sources.list.d/xivo-upg
&& apt-get update \
&& apt-get install xivo-fai \
&& rm /etc/apt/sources.list.d/xivo-upgrade.list \
&& apt-get update
```

Upgrading from XiVO 13.24 and before When upgrading from XiVO 13.24 or earlier, you must do the following, before the normal upgrade:

1. Ensure that the file `/etc/apt/sources.list` is *not* configured on `archive.debian.org`. Instead, it must be configured with a non-archive mirror, but still on the `squeeze` distribution, even if it is not present on this mirror. For example:

```
deb http://ftp.us.debian.org/debian squeeze main
```

2. Add `archive.debian.org` in another file:

```
cat > /etc/apt/sources.list.d/squeeze-archive.list <<EOF
deb http://archive.debian.org/debian/ squeeze main
EOF
```

And after the upgrade:

```
rm /etc/apt/sources.list.d/squeeze-archive.list
```

Upgrading from XiVO 13.03 and before When upgrading from XiVO 13.03 or earlier, you must do the following, before the normal upgrade:

```
wget http://mirror.xivo.solutions/xivo_current.key -O - | apt-key add -
```

Upgrading from XiVO 12.13 and before When upgrading from XiVO 12.13 or earlier, you must do the following, before the normal upgrade:

```
apt-get update
apt-get install debian-archive-keyring
```

Upgrading from XiVO 1.2.1 and before Upgrading from 1.2.0 or 1.2.1 requires a special procedure before executing `xivo-upgrade`:

```
apt-get update
apt-get install xivo-upgrade
/usr/bin/xivo-upgrade
```

Upgrading a cluster

Here are the steps for upgrading a cluster, i.e. two XiVO with *High Availability (HA)*:

1. On the master : deactivate the database replication by commenting the cron in `/etc/cron.d/xivo-ha-master`
2. On the slave, deactivate the `xivo-check-master-status` script cronjob by commenting the line in `/etc/cron.d/xivo-ha-slave`
3. On the slave, start the upgrade:

```
xivo-slave:~$ xivo-upgrade
```

4. When the slave has finished, start the upgrade on the master:

```
xivo-master:~$ xivo-upgrade
```

5. When done, launch the database replication manually:

```
xivo-master:~$ xivo-master-slave-db-replication <slave ip>
```

6. Reactivate the cronjobs (see steps 1 and 2)

Upgrading to/from an archive version

Upgrade involving archive version of XiVO

Introduction

What is an archive version? An archive version refers to a XiVO installation whose version is frozen: you can't upgrade it until you manually change the upgrade server.

What is the point? Using archive versions enable you to upgrade your XiVO to a specific version, in case you don't want to upgrade to the latest (which is not recommended, but sometimes necessary). You will then be able to upgrade your newer archive version to the latest version or to an even newer archive version.

Prerequisites

Warning: These procedures are *complementary* to the upgrade procedure listed in *Version-specific upgrade procedures*. You must follow the version-specific procedure *before* running the following procedures.

Archive package names Archive packages are named as follow:

XiVO version	Archive package name
1.2 to 1.2.12	pf-fai-xivo-1.2-skaro-1.2.1
12.14 to 13.24	xivo-fai-skaro-13.04
13.25 to 14.17	xivo-fai-14.06
14.18+	<i>packages removed</i>

Upgrade from an archive to the latest version Archive version < 13.25:

```
apt-get update
apt-get install {xivo-fai,xivo-fai-skaro}/squeeze-xivo-skaro-$(cat /usr/share/pf-xivo/XIVO-VERSION)
sed -i 's/xivo\.fr/xivo.solutions/g' /etc/apt/sources.list.d/*.list
xivo-upgrade
```

Archive version >= 13.25 and < 14.18:

```
apt-get update
apt-get install xivo-fai
sed -i 's/xivo\.fr/xivo.solutions/g' /etc/apt/sources.list.d/*.list
xivo-upgrade
```

Archive version >= 14.18:

```
xivo-dist xivo-five
xivo-upgrade
```

As a result, xivo-upgrade will upgrade XiVO to the latest stable version.

Upgrade from an older non-archive version to a newer archive version Non-archive version means any “normal” way of installing XiVO (ISO install, script install over pre-installed Debian, xivo-upgrade).

Downgrades are not supported: you can only upgrade to a greater version.

We only support upgrades to archive versions >= 13.25, e.g. you can upgrade a 12.16 to 14.16, but not 12.16 to 13.16

Current version before 14.18 (here 13.25)

```
apt-get install xivo-fai-13.25
sed -i 's/xivo\.fr/xivo.solutions/g' /etc/apt/sources.list.d/*.list
```

You are now considered in an archived version, see the section [Upgrade from an older archive version to a newer archive version](#) below.

Current version after 14.18

```
xivo-dist xivo-15.12
apt-get update
apt-get install xivo-upgrade/xivo-15.12
xivo-upgrade
```

Upgrade from an older archive version to a newer archive version Downgrades are not supported: you can only upgrade to a greater version.

We only support upgrades to archive versions >= 13.25, e.g. you can upgrade a 12.16 to 14.16, but not 12.16 to 13.16

1.2 - 13.24 to 13.25 - 14.17 (here 1.2.3 to 14.16)


```

cat > /etc/apt/sources.list.d/squeeze-archive.list <<EOF
deb http://archive.debian.org/debian/ squeeze main
EOF

apt-get update
apt-get install {xivo-fai,xivo-fai-skaro}/squeeze-xivo-skaro-1.2.3
sed -i 's/xivo\.fr/xivo.solutions/g' /etc/apt/sources.list.d/*.list
apt-get update
apt-get install xivo-fai-14.16
sed -i 's/xivo\.fr/xivo.solutions/g' /etc/apt/sources.list.d/*.list
apt-get update
apt-get install xivo-upgrade/xivo-14.16

cat > /etc/apt/preferences.d/50-xivo-14.16.pref <<EOF
Package: *
Pin: release a=xivo-five
Pin-Priority: -10

Package: *
Pin: release a=xivo-14.16
Pin-Priority: 700
EOF

xivo-upgrade
rm /etc/apt/preferences.d/50-xivo-14.16.pref
rm /etc/apt/sources.list.d/squeeze-archive.list
apt-get update

```

13.25 - 14.16 to 13.25 - 14.17 (here 13.25 to 14.16)

```

apt-get update
apt-get install xivo-fai
apt-get purge xivo-fai-13.25
sed -i 's/xivo\.fr/xivo.solutions/g' /etc/apt/sources.list.d/*.list
apt-get update
apt-get install xivo-fai-14.16
sed -i 's/xivo\.fr/xivo.solutions/g' /etc/apt/sources.list.d/*.list
apt-get update
apt-get install xivo-upgrade/xivo-14.16

cat > /etc/apt/preferences.d/50-xivo-five.pref <<EOF
Package: *
Pin: release a=xivo-five
Pin-Priority: -10
EOF

xivo-upgrade
rm /etc/apt/preferences.d/50-xivo-five.pref

```

13.25 - 14.17 to 14.18+ (here 14.05 to 15.11)

```

apt-get update
apt-get install xivo-fai
sed -i 's/xivo\.fr/xivo.solutions/g' /etc/apt/sources.list.d/*.list
apt-get update
apt-get install xivo-dist
xivo-dist xivo-15.11
apt-get purge 'xivo-fai*'
apt-get update
apt-get install xivo-upgrade/xivo-15.11
xivo-upgrade

```

14.18+ to 14.19+ (here 14.18 to 15.12)

```
xivo-dist xivo-15.12
apt-get update
apt-get install xivo-upgrade/xivo-15.12
xivo-upgrade
```

Upgrading from i386 (32 bits) to amd64 (64 bits)

Migrate XiVO from i386 (32 bits) to amd64 (64 bits) There is no fully automated method to migrate XiVO from i386 to amd64.

The procedure is:

1. *Upgrade* your i386 machine to XiVO >= 15.13
2. *Install* a XiVO amd64 **using the same version as the upgraded XiVO i386**
3. Make a backup of your XiVO i386 by following the *backup procedure*
4. Copy the backup tarballs to the XiVO amd64
5. Restore the backup by following the *restore procedure*

Before starting the services after restoring the backup on the XiVO amd64, you should ensure that there won't be a conflict between the two machines, e.g. two DHCP servers on the same broadcast domain, or both XiVO fighting over the same SIP trunk register. You can disable the XiVO i386 by running:

```
xivo-service stop
```

But be aware the XiVO i386 will be enabled again after you reboot it.

Troubleshooting

Postgresql When upgrading XiVO, if you encounter problems related to the system locale, see *PostgreSQL localization errors*.

xivo-upgrade If xivo-upgrade fails or aborts in mid-process, the system might end up in a faulty condition. If in doubt, run the following command to check the current state of xivo's firewall rules:

```
iptables -nvL
```

If, among others, it displays something like the following line (notice the DROP and 5060):

```
0      0 DROP      udp  --  *      *      0.0.0.0/0      0.0.0.0/0      udp dpt:5060
```

Then your XiVO will not be able to register any SIP phones. In this case, you must delete the DROP rules with the following command:

```
iptables -D INPUT -p udp --dport 5060 -j DROP
```

Repeat this command until no more unwanted rules are left.

Upgrade Notes

2016.04 Consult the [2016.04 Roadmap](#)

Upgrading from 2016.03:

```
xivo-dist xivo-2016.04
xivo-upgrade
```

For upgrade from older versions see *Version-specific upgrade procedures*.

2016.03 To upgrade to this version, use the `switch-to-xivo-solutions` script:

```
wget http://mirror.xivo.solutions/debian/tools/migration-tools/switch-to-xivo-solutions.sh
chmod +x ./switch-to-xivo-solutions.sh
./switch-to-xivo-solutions.sh
```

At the end the script should say:

```
Your XiVO has been switched to xivo.solutions successfully.
Votre XiVO a été basculé vers xivo.solutions avec succès.
```

Then you have to points towards this version:

```
xivo-dist xivo-2016.03
apt-get update
```

And finally do the normal upgrade procedure.

16.08.2 Consult the [16.08.2 release post](#)

This release fixes a number of known issues that were present in the 16.08. Mainly it:

- ships with asterisk 13.10.0 which fixes a number of crashes, instabilities and memory leak,
- fixes a number of upgrade problems,
- fixes a status (presence, phone) update problem in XiVO Client,
- add the possibility to create a XiVO Client login in the format *name@domain.tld*

16.08 Consult the [16.08 Roadmap](#)

- *CTI Protocol* is now in version 2.2
- Some *security features have been added to the XiVO provisioning server*. To benefit from these new features, you'll need to *update your xivo-provd plugins to meet the system requirements*.

If you have many phones that are connected to your XiVO through a NAT equipment, you should review the default configuration to make sure that the IP address of your NAT equipment don't get banned unintentionally by your XiVO.

- Newly created groups and queues now ignore call forward requests from members by default. Previously, call forward requests from members were always followed. This only applies to call forward configured directly on the member's phone: call forward configured via *21 have always been ignored in these cases.

Note that during the upgrade, the previous behaviour is kept for already existing queues and groups.

This behaviour is now configurable per queue/group, via the "Ignore call forward requests from members" option under the "Application" tab. We recommend enabling this option.

16.07 Consult the [16.07 Roadmap](#)

- If you were affected by the [bug #6213](#), i.e. if your agent login time statistics were incorrect since your upgrade to XiVO 15.20 or later, and you want to fix your statistics for that period of time, you'll need to [manually apply a fix](#).

16.06 Consult the [16.06 Roadmap](#)

16.05 Consult the [16.05 Roadmap](#)

- The `view`, `add`, `edit`, `delete` and `deleteall` actions of the "lines" web service provided by the web interface have been removed. As a reminder, note that the web services provided by the web interface are deprecated.

16.04 Consult the [16.04 Roadmap](#)

- [CTI Protocol](#) is now in version 2.1
- The field *Rightcall Code* from *Services* -> *IPBX* -> *IPBX Settings* -> *Users* under *Services* tab will overwrite all password call permissions for the user.
- Faxes stored on FTP servers are now converted to PDF by default. See [Using the FTP backend](#) if you want to keep the old behavior of storing faxes as TIFF files.

16.03 Consult the [16.03 Roadmap](#)

- The new section *Services* → *Statistics* → *Switchboard* in the web interface will only be visible by a non-root administrator after adding the corresponding permissions in the administrator configuration.
- Update the switchboard configuration page for the statistics in [Configuration for multiple switchboards](#).
- The API for associating a line to a device has been replaced. Consult the [xivo-confd REST API changelog](#) for further details
- The configuration parameters of *xivo_ldap_user* plugin of *xivo-auth* has been changed. See [xivo_ldap plugin](#).
- The user's email is now a unique constraint. Every duplicate email will be deleted during the migration. (This does not apply to the voicemail's email)

16.02 Consult the [16.02 Roadmap](#)

- The experimental *xivo_ldap_voicemail* plugin of *xivo-auth* has been removed. Use the new [xivo_ldap plugin](#).
- Bus messages in the *xivo* exchange are now sent with the content-type *application/json*. Some libraries already do the message conversion based the content-type. Kombu users will receive a python dictionary instead of a string containing json when a message is received.
- [xivo-ctid encryption](#) is automatically switched on for every XiVO server and XiVO Client >= 16.02. If you really don't want encryption, you must disable it manually on the server after the upgrade. In that case, XiVO Clients will ask whether to accept the connection the first time.

16.01 Consult the [16.01 Roadmap](#)

- The page *Configuration* → *Management* → *Web Services Access* → *Acces rights* has been removed. Consequently, every Web Services Access has now all access rights on the web services provided by the web interface. These web services are deprecated and will be removed soon.
- During the upgrade, if no CA certificates were trusted at the system level, all the CA certificates from the ca-certificates package will be added. This is done to resolve an issue with installations from the ISO and PXE. In the (rare) case you manually configured the ca-certificates package to trust no CA certificates at all, you'll need to manually reconfigure it via `dpkg-reconfigure ca-certificates` after the upgrade.
- *xivo-ctid* uses *xivo-auth* to authenticate users. See [Authentication](#).
- the *service_discovery* section of the *xivo-ctid* configuration has changed. If you have set up [Contact and Presence Sharing](#), you should update your xivo-ctid configuration.
- the [CTI Protocol](#) is now versioned and a message will be displayed if the server and a client have incompatible protocol versions.

15.20 Consult the [15.20 Roadmap](#)

- Debian has been upgraded from version 7 (wheezy) to 8 (jessie).
- CSV webservices in the web interface have been removed. Please use the [xivo-confd REST API](#) instead.
- The CSV import format has been changed. Consult [CSV Migration](#) for further details.

- xivo-ctid now uses STARTTLS for the client connections.
 - For users already using the CTIS protocol the client can be configured to use the default port (5003)

Please consult the following detailed upgrade notes for more information:

Debian 8 (jessie) Upgrade Notes The upgrade to XiVO 15.20 or later will take longer than usual, because the whole Debian system will be upgraded.

The database management system (postgresql) will also be upgraded from version 9.1 to version 9.4 at the same time. This will upgrade the database used by XiVO. This operation should take at most a few minutes.

After the upgrade, the system will need to be rebooted.

Before the upgrade

- If you are upgrading from XiVO 13.24 or earlier, you'll need to first upgrade to Debian 7 (wheezy) before being able to upgrade to Debian 8 (jessie). To do so, you'll have to:
 - Run `xivo-upgrade` a first time, which will upgrade your XiVO to version 15.19 (Debian 7)
 - Reboot your system
 - Run `xivo-upgrade` a second time, which will upgrade your XiVO to the latest version (Debian 8)
 - Reboot your system

Consult the [Debian 7 \(wheezy\) Upgrade Notes](#) for more information on the first upgrade.

- Make sure you have sufficient space for the upgrade. You might run into trouble if you have less than 2 GiB available in the file system that holds the `/var` and `/` directories.
- If you have customized the Debian system of your XiVO in some nontrivial way, you might want to review the [official Debian release notes](#) before the upgrade. Most importantly, you should:
 - Make sure you don't have any unofficial sources in your `/etc/apt/sources.list` or `/etc/apt/sources.list.d` directory. If you were using the wheezy-backports source, you must remove it.
 - Remove packages that were automatically installed and are not needed anymore, by running `apt-get autoremove --purge`.
 - Purge removed packages. You can see the list of packages in this state by running `dpkg -l | awk '/^rc/ { print $2 }'` and purge all of them with `apt-get purge $(dpkg -l | awk '/^rc/ { print $2 }')`
 - Remove `.dpkg-old`, `.dpkg-dist` and `.dpkg-new` files from previous upgrade. You can see a list of these files by running `find /etc -name '*.dpkg-old' -o -name '*.dpkg-dist' -o -name '*.dpkg-new'`.

After the upgrade

- Check that customization to your configuration files is still effective.

During the upgrade, new version of configuration files are going to be installed, and these might override your local customization. For example, the vim package provides a new `/etc/vim/vimrc` file. If you have customized this file, after the upgrade you'll have both a `/etc/vim/vimrc` and `/etc/vim/vimrc.dpkg-old` file, the former containing the new version of the file shipped by the vim package while the later is your customized version. You should merge back your customization into the new file, then delete the `.dpkg-old` file.

You can see a list of affected files by running `find /etc -name '*.dpkg-old'`. If some files shows up that you didn't modify by yourself, you can ignore them.

- Purge removed packages. You can see the list of packages in this state by running `dpkg -l | awk '/^rc/ { print $2 }'` and purge all of them with `apt-get purge $(dpkg -l | awk '/^rc/ { print $2 }')`

- If you had customizations in one of these files:

- `/etc/default/asterisk`
- `/etc/default/consul`
- `/etc/default/xivo-ctid`

Then you'll need to review your customizations to make sure they still work with systemd. This is necessary since these 3 files aren't read under systemd.

For `/etc/default/asterisk`, only the `CONFD_*` options are automatically migrated to `/etc/systemd/system/asterisk.service.d/auto-sysv-migration.conf`.

For `/etc/default/consul`, only the `WAIT_FOR_LEADER` and `CONFIG_DIR` options are automatically migrated to `/etc/systemd/system/consul.service.d/auto-sysv-migration.conf`.

For `/etc/default/xivo-ctid`, only the `XIVO_CTID_AMI_PROXY` option is automatically migrated to `/etc/systemd/system/xivo-ctid.service.d/auto-sysv-migration.conf`.

- Reboot your system. It is necessary for the upgrade to the Linux kernel and init system (systemd) to be effective.

Changes Here's a non-exhaustive list of changes that comes with XiVO on Debian 8:

- In Debian 7, the `halt` command powered off the machine. In Debian 8, the command halts the system, but does not power off the machine. To halt the machine and turn it off, use the `poweroff` or `shutdown` command.
- With the init system switch from SysV to systemd, you should now use the `systemctl` command to manage services (i.e. `start/stop/status`) instead of the `service` command or `/etc/init.d/<service>`, although these two methods should still work fine.

If you are new to systemd, you can find some basic usage on the [systemd page of the Debian Wiki](#).

- The `bootlogd` package is not installed by default anymore, since it is not needed with systemd. If you want to see the boot messages, use the `journalctl -b` command instead.
- The virtual terminals (`tty1` to `tty6`) now shows up earlier during the boot, before all services have been started.
- The way the *ami-proxy is configured* for `xivo-ctid` has changed. If your XiVO was using the `ami-proxy`, the configuration will be automatically upgraded.
- Customization to `asterisk` and `consul` startup is now done by customizing the systemd unit file (by creating a drop-in file for example) instead of editing the `/etc/default/asterisk` and `/etc/default/consul` files. These files are not used anymore.

List of Known Bugs And Limitations

- If your system is using a swap partition or file and is using more memory than it can fit in the RAM, then system power-off or reboot might hangs indefinitely. This is due to a limitation in the current systemd version.

If you find yourself in this case, you should try allocating more RAM to your system. Otherwise, you can try stopping the `xivo` services using `xivo-service stop` before rebooting to lessen the likelihood of this problem.

See <http://projects.xivo.io/issues/6016>

External Links

- [Official Debian 8 release notes](#)

CSV Migration This page describes how to migrate CSV files from the legacy format to the new format. Consult the [API documentation](#) on user imports for further details.

CSV Data

- Only data encoded as UTF-8 will be accepted
- The pipe delimiter (|) has been replaced by a comma (,)
- Double-quotes (") must be escaped by writing them twice (e.g Robert "Bob" Jenkins)

Field names Fields have been renamed in the new CSV format. Use the following table to rename your fields. Fields marked as **N/A** are no longer supported.

Old name	New name
entityid	entity_id
firstname	firstname
lastname	lastname
language	language
outcallerid	outgoing_caller_id
mobilephonenumber	mobile_phone_number
agentnumber	N/A
bosssecretary	N/A
callerid	N/A
enablehint	supervision_enabled
enablexfer	call_transfer_enabled
enableclient	cti_profile_enabled
profileclient	cti_profile_name
username	username
password	password
phonenumber	exten
context	context
protocol	line_protocol
linename	sip_username
linesecret	sip_secret
incallexten	incall_exten
incallcontext	incall_context
incallringseconds	incall_ring_seconds
voicemailname	voicemail_name
voicemailnumber	voicemail_number
voicemailcontext	voicemail_context
voicemailpassword	voicemail_password
voicemailemail	voicemail_email
voicemailattach	voicemail_attach_audio
voicemaildelete	voicemail_delete_messages
voicemailaskpassword	voicemail_ask_password

15.19 Consult the 15.19 Roadmap

- The sound file `/usr/share/asterisk/sounds/fr_FR/une.wav` has been moved to `/usr/share/asterisk/sounds/fr_FR/digits/1F.wav`.
- If you would like to use the new “[transfer to voicemail](#)” feature from the People Xlet, you’ll need to update your directory definition and your directory display, i.e.:

- edit your “internal” directory definition (Services / CTI server / Directories / Definitions) and add a field “voicemail” with value “voicemail_number”
 - edit your display (Services / CTI server / Directories / Display filters) and add a row with title “Voice-mail”, field type “voicemail” and field name “voicemail”
 - restart xivo-dird
- It is now possible to send an email to a user with a configured email address in the *people* xlet. See [Views](#) to add the appropriate field to your configured displays.
 - The *Contacts* xlet (aka. *Search*) has been removed in favor of the *People Xlet*. You may need to do some manual configuration in the directories for the People Xlet to be fully functional. See [the detailed upgrade notes](#) for more details.
 - If you need context separation in the People Xlet, you will have to **manually configure** xivo-dird to keep it working, see [Context separation](#). This procedure is only temporary, later versions will handle the context separation automatically.
 - xivo-agentd now uses mandatory token authentication for its REST API. If you have custom development using this service, update your program accordingly.
 - Some actions that used to be available in the *contact* xlets are not implemented in the *people* xlet yet.
 - Cancel transfer is only available using the *switchboard* xlet
 - Hanging up a call is only possible using the *switchboard* xlet
 - Call interception is not available anymore
 - Conference room invitation is not available anymore

Please consult the following detailed upgrade notes for more information:

People Xlet features Upgrade Notes When upgrading your XiVO to 15.19, there are some features in the directories that could not be upgraded automatically, because it risked breaking some manual configurations.

After you upgrade your XiVO, your CTI displays in *Services* → *CTI Server* → *Directories* → *Displays* may look like this:

Field title	Field type	Default value	Field name
Nom	text	nom	nom
Numéro	number	phone	phone
Entreprise	text	Inconnue	company
E-mail	text	mail	mail
Source	text	directory	directory

Description

Affichage par défaut

You need to restart the Dird server to apply changes.

Save

You should update your displays to make them look like:

This will give you a Xlet People looking like this:

You can find more details about the field types in [Integration of XiVO dird with the rest of XiVO](#).

CTI Server

General settings

Profiles

Status

Presences

Phone hints

Directories

Definitions

Reverse directories

Direct directories

Display filters

Sheets

Models

Events

Update displays

Name:

Field title	Field type	Default value	Field name
<input type="text" value="Nom"/>	<input type="text" value="name"/>	<input type="text"/>	<input type="text" value="name"/>
<input type="text" value="Numéro"/>	<input type="text" value="number"/>	<input type="text"/>	<input type="text" value="phone"/>
<input type="text" value="Entreprise"/>	<input type="text"/>	<input type="text"/>	<input type="text" value="company"/>
<input type="text" value="Mobile"/>	<input type="text" value="callable"/>	<input type="text"/>	<input type="text" value="mobile"/>
<input type="text" value="Source"/>	<input type="text"/>	<input type="text"/>	<input type="text" value="directory"/>
<input type="text" value="E-mail"/>	<input type="text" value="email"/>	<input type="text"/>	<input type="text" value="email"/>
<input type="text" value="Favori"/>	<input type="text" value="favorite"/>	<input type="text"/>	<input type="text" value="favorite"/>
<input type="text" value="Personnel"/>	<input type="text" value="personal"/>	<input type="text"/>	<input type="text"/>

Description

You need to restart the Dird server to apply changes.

Save

Liste de contacts

TOUS · FAVORIS · MES CONTACTS

rechercher

NOM	NUMÉRO	ENTREPRISE	SOURCE	FAVORI
Davy Crockett	+14185555555	Crockett Inc.		★
● Bernard Marx	● 102		internal	★
● Charliez Chaplin	● 103		internal	★

Liste de contacts

TOUS · FAVORIS · MES CONTACTS

rechercher

NOM	NUMÉRO	ENTREPRISE	SOURCE	FAVORI	PERSONNEL
Davy Crockett	APPELER	Crockett Inc.		★	 
<div>E-MAIL - davy.crockett@example.com</div> <div>MOBILE - +14185556666</div>					

Context separation Without context separation, you only need one contact source for all the users of your XiVO.

However, if you need context separation, each context is considered as a separate independant source of contacts, each with a different context filter. For this, you need:

- one contact source per context (a file in `/etc/xivo-dird/sources.d`), so that we have a source containing only the contacts from one context
- one profile per context (equivalent to *Services → CTI Server → Directories → Direct directories*) so that users in one context only see people from the same context.

Each source should look like this one, e.g. the context is named `INSIDE`:

```
confd_config:
  host: localhost
  https: false
  port: 9487
  timeout: 4
  verify_certificate: false
  version: '1.1'
first_matched_columns: [exten]
format_columns:
  directory: "R\xE9pertoire XiVO Interne"
  location: '{description}'
  mobile: '{mobile_phone_number}'
  name: '{firstname} {lastname}'
  number: '{exten}'
  sda: '{userfield}'
  voicemail: '{voicemail_number}'
searched_columns: [firstname, lastname, userfield, description]
type: xivo
unique_column: id
name: internal_INSIDE # <--- each source has a different name, one per context
extra_search_params:
  context: INSIDE # <--- each source filters users according to one context
```

The parameters in this file have the same effect than *Configuration → Directories* and *Services → CTI Server → Directories → Direct directories* put together.

You may generate these config files from `xivo-confgen dird/sources.yml`. Be sure to have `name` and `extra_search_params` correct for each source file.

Now that we have our contact sources, we need our search profiles.

Create a new file to override the profiles generated by `xivo-confgen`. You only need one file, which will define all your profiles at once.

```
xivo-confgen dird/services.yml >> /etc/xivo-dird/conf.d/001-context-separation.yml
```

In this file, there is a list of services (favorites, lookup, ...) where each profile has a set of sources. You need to match one profile to the right internal source for each service. For example, to have context separation between contexts `INSIDE` and `INDOORS`:

```
services:
  favorites:
    __default_phone:
      sources: [xivodir, internal, ldaptest, personal]
    __switchboard_directory:
      sources: [xivodir, ldaptest, personal]
  INSIDE:
    sources: [xivodir, internal_INSIDE, ldaptest, personal] # <--- profile INSIDE uses the so
  INDOORS:
    sources: [xivodir, internal_INDOORS, ldaptest, personal] # <--- profile INDOORS uses the s
  lookup:
    __default_phone:
```

```

sources: [xivodir, internal, ldaptest, personal]
__switchboard_directory:
sources: [xivodir, ldaptest, personal]
INSIDE:
sources: [xivodir, internal_INSIDE, ldaptest, personal] # <--- same HERE
INDOORS:
sources: [xivodir, internal_INDOORS, ldaptest, personal] # <--- and HERE

```

15.18 Consult the 15.18 Roadmap

- The `provd_pycli` command (deprecated in 15.06) has been removed in favor of `xivo-provd-cli`. If you have custom scripts referencing `provd_pycli`, you'll need to update them.
- The `xivo-agentctl` command (deprecated in 15.06) has been removed in favor of `xivo-agentd-cli`. If you have custom scripts referencing `xivo-agentctl`, you'll need to update them.
- `xivo-agentd` now uses HTTPS. If you have custom development using this service, update your configuration accordingly. The `xivo-agentd-client` library, used to interact with `xivo-agentd`, has also been updated to use HTTPS by default.
- `xivo-confd` ports 50050 and 50051 have been removed. Please use 9486 and 9487 instead

Configuration File Upgrade Notes

The file format of configuration files for daemons exposing an HTTP/S API has changed. The following services have been affected :

- `xivo-agentd`
- `xivo-amid`
- `xivo-auth`
- `xivo-confd`
- `xivo-ctid`
- `xivo-dird`
- `xivo-dird-phoned`

Ports and listening addresses are now organised in the following fashion:

```

rest_api:
  https:
    enabled: true
    port: 9486
    listen: 0.0.0.0
    certificate: /usr/share/xivo-certs/server.crt
    private_key: /usr/share/xivo-certs/server.key
    ciphers: "ALL:!aNULL:!eNULL:!LOW:!EXP:!RC4:!3DES:!SEED:+HIGH:+MEDIUM"
  http:
    enabled: true
    port: 9487
    listen: 127.0.0.1

```

If you have any custom configuration files for these daemons, please modify them accordingly. Consult [Network](#) for further details on which network services are available for each daemon.

15.17 Consult the 15.17 Roadmap

- Online call recording is now done via `automixmon` instead of `automon`. This has no impact unless you have custom dialplan that is passing directly the “w” or “W” option to the Dial or Queue application. In these cases, you should modify your dialplan to pass the “x” or “X” option instead.

- The remote directory service available from *supported phones* is now provided by the new unified directory service, i.e. `xivo-dird`. Additional upgrade steps are required to get the full benefit of the new directory service; see the *detailed upgrade notes*.
- The field `enableautomon` has been renamed to `enableonlinerec` in the users web services provided by the web-interface (these web services are deprecated).
- The agent status dashboard now shows that an agent is calling or receiving a non ACD call while in wrapup or paused.
- SIP endpoints created through the REST API will not appear in the web interface until they have been associated with a line
- Due to limitations in the database, only a limited number of optional parameters can be configured on a SIP endpoint. Consult the *xivo-confd REST API changelog* for further details

Please consult the following detailed upgrade notes for more information:

Phone Remote Directory Upgrade Notes If you are not using the remote directory from your phones, you can safely skip this page.

Starting from XiVO 15.17, the remote directory used by the phones is now provided by the new directory service, composed principally of *xivo-dird* and *xivo-dird-phoned*. It was previously provided by the XiVO web interface.

This brings a few changes for the administrators, the biggest one being that lookup from both the XiVO client and phones are now configured at the same place, namely the (incorrectly named) *Services* → *CTI Server* → *Directories* section, with some advanced configuration only available in the configuration files. This means that lookup from the phones can now also display results from CSV or web services directories. For details on how to configure directories, refer to the *Directories* page.

For users, the biggest change is that they can now consult their personal contacts (that they added from their XiVO client) when doing a search from their phone.

Changes

Web Interface - LDAP Filters The following options have been removed from the web interface, in the *Services* → *IPBX* → *LDAP filters* page:

- the `Phone number type` field
- the `Attributes` tab

The phone number type is now configurable on a per source basis (and for all type of source, not just LDAP), in *Services* → *CTI Server* → *Directories*. For example, if you have LDAP records with the attribute `telephoneNumber` that you want to be displayed on your phone with the suffix “(Office)”, just make sure that your directory definition is configured with a field named `phone_office` with the value `{telephoneNumber}`.

By default, the following fields are available:

- `phone`: doesn't add a suffix
- `phone_office`: add a “(Office)” suffix
- `phone_mobile`: add a “(Mobile)” suffix
- `phone_home`: add a “(Home)” suffix
- `phone_other`: add a “(Other)” suffix

Note: These fields will automatically be added in your LDAP directory definitions during the upgrade, so you may only need to *review your directory configuration*.

This list of fields and the suffix associated to it is currently only configurable in the *xivo-dird configuration files*, in the *views/displays_phone* section.

This is causing 2 functional changes:

- Previously, the suffix displayed was translated in function of the phone's language. This is not possible anymore, and you'll have to edit the configuration files if you want the suffix to be in a different language than english.
- For "custom" phone number type, you'll have to add a new entry in the configuration files and add the correspond field in the directory definition.

In XiVO 15.16, the *Attributes* tab would allow a "fallback" mechanism, where if an LDAP attribute for a record was missing/empty, another attribute would be used. In XiVO 15.17, this mechanism is available (for all type of sources) by mapping the first attribute to a field name *phone*, the second to a field name *phone1*, etc. The fallback mechanism is available on the fields *phone*, *phone_office*, *phone_mobile*, *phone_home*, *phone_other* and *display_name*.

Web Interface - Phonebook The following options have been removed from the web interface, in the *Services* → *IPBX* → *Phonebook* page:

- the LDAP *filters* tab

LDAP sources used for lookup from the phone are now selected in the same place as for the XiVO client, i.e. in *Services* → *CTI Services* → *Direct directories*. A consequence of that is that it's not possible anymore to have sources only used for lookup from phone and other sources only used for lookup from the XiVO client.

Note: The LDAP filters that were used for phone lookup will be automatically added to all the profiles during the upgrade.

Additional Upgrade Steps After upgrading your XiVO to 15.17 or later, you should do the following steps.

Upgrade Your Provisioning Plugins This step is optional, although strongly recommended.

For the users to be able to search their personal contacts from their phone, the phone configuration needs to be updated. This means:

1. Installing new xivo-provd plugins or upgrading existing plugins
2. Restarting all affected phones

See the *provisioning* section for more information on installing or upgrading plugins.

Here's the list of plugins which have received modifications to be compatible with the new directory service:

Name	Version
xivo-aastra-3.3.1-SP4	1.5
xivo-aastra-4.1.0	1.5
xivo-cisco-sccp-9.0.3	0.8
xivo-cisco-sccp-cipc-2.1.2	0.8
xivo-cisco-sccp-legacy	0.8
xivo-cisco-sccp-wireless-1.4.5	0.8
xivo-cisco-spa-7.5.5	0.12
xivo-cisco-spa-legacy	0.12
xivo-polycom-4.0.4	1.4
xivo-polycom-5.3.0	1.5
xivo-snom-8.7.5.17	1.5
xivo-technicolor-ST2022-4.78-1	0.4
xivo-technicolor-ST2030-2.74	0.3
xivo-technicolor-TB30-1.74.0	0.3
xivo-yealink-v70	1.24
xivo-yealink-v72	1.24
xivo-yealink-v73	1.24
xivo-yealink-v80	1.24

Plugins with greater version number or greater firmware-version number are also compatible.

If the xivo-provd plugins are not updated or the phone are not rebooted, the user will by default only be able to search in the “internal” and “xivodir” directory definitions. If you want to add or remove sources for these phones, you’ll need to edit xivo-dird configuration files. More precisely, you’ll need to edit the sources associated to the profile named `default_phone`.

Update Your Firewall Rules If there’s a firewall (or a NAT equipment) between your XiVO and your phones, you must know that the port used for the directory lookup from the phone has changed from port TCP/80 to port TCP/9498. The new port is going to be used only by phones which are using a compatible plugins (see list above) and have been rebooted; otherwise, the port TCP/80 will still be used.

Review Your Directory Configuration During the upgrade, new LDAP directory definitions might be created and fields to existing one might be added.

For example, if you had an LDAP filter which was used for directory lookup from your phones, then a corresponding LDAP directory definition will be created if nonexistent, and otherwise be updated to make sure the `display_name` and `phone_office` (or another field, depending on the phone number type of your LDAP filter) fields are defined. The directory definition will also be added to all the direct directories entries, i.e. added to all items in the *Services* → *CTI Server* → *Direct directories* page.

If you were using LDAP filters with custom phone number types, the custom part will be lost, and to get back the same behaviour, you’ll need to modify xivo-dird configuration files and update the field’s name in your directory definition.

Also, if you have other directory definitions that you now want to use from your phones (e.g. CSV directories), make sure that their configuration is working, i.e. that they have a `display_name` and `phone` fields. During the upgrade, these fields are automatically added to the directory definition “xivodir”, “internal” and for LDAP source, like described above.

15.16 Consult the 15.16 Roadmap

- The directory column type “mobile” was removed in favor of the new “callable” type. If you have hand-written configuration files for xivo-dird, in section “views”, subsection “displays”, all keys “type” with value “mobile” must be changed to value “callable”.
- The xivo-auth backend interface has changed, `get_acls` is now `get_consul_acls`. All unofficial back ends must be adapted and updated. No action is required for “normal” installations.
- Voicemails can now be deleted even if they are associated to a user.

15.15 Consult the [15.15 Roadmap](#)

Voicemail Upgrade Notes

- Voicemail webservice in the web interface have been removed. Please use the *xivo-confd REST API* instead.
- Voicemail IMAP configuration has been migrated to the new Advanced tab.
- Voicemail option `Disable password checking` has been converted to `Ask password`. The value has also been inverted. (e.g. If `Disable password checking` was `false`, `Ask password` is `true`.) `Ask password` is activated by default.
- After an upgrade, if ever you have errors when searching for voicemails, please try clearing cookies in your web browser.
- A voicemail must be dissociated from any user prior to being deleted. Voicemail are dissociated by editing the user and clicking on the `Delete voicemail` button in the Voicemail tab. This constraint will disappear in future versions.
- Deleting a user will dissociate any voicemail that was attached, but will not delete it nor any messages.
- Creating a line is no longer necessary when attaching a voicemail to a user.
- The following fields have been modified when importing a CSV file:

Old name	New name	Required ?	New default value
voicemailmailbox	voicemailnumber	yes	
voicemailsippass	voicemailaskpassword	no	1
	voicemailcontext	yes	

Directories

- Concatenated fields in directories are now done in the directory definitions instead of the displays
- The field column in directory displays are now field names from the directory definition. No more `{db-*}` are required
- In the directory definitions fields can be modified using a python format string with the fields coming from the source.
- Most of the configuration for `xivo-dird` is now generated from `xivo-confgen` using the values in the web interface.
- The *remote directory* xlet has been removed in favor of the new *people* xlet.

See [Directories](#) and [Integration of XiVO dird with the rest of XiVO](#) for more details

15.14

- Consult the [15.14 Roadmap](#)
- Default password for `xivo-polycom-4.0.4` plugin version ≥ 1.3 is now **9486** (i.e. the word “xivo” on a telephone keypad).
- Default password for `xivo-polycom-5.3.0` plugin version ≥ 1.4 is now **9486**.
- Caller id management for users in `confd` has changed. Consult the *[xivo-confd REST API changelog](#)*.
- The Local Directory Xlet is replaced with the People Xlet. Contacts are automatically migrated to the server. Note that the CSV format for importing contacts has changed (see [People Xlet](#) for more information).

15.13

- Consult the [15.13 Roadmap](#)
- Asterisk has been upgraded from version 11.17.1 to 13.4.0, which is a major Asterisk upgrade.

- An [ARI](#) user has been added to `/etc/asterisk/ari.conf`. If you have configured Asterisk HTTP server to bind on a publicly reachable address (in `/etc/asterisk/http.conf`), then you should update your configuration to prevent unauthorized access on your Asterisk.
- The xivo-dird configuration option `source_to_display_columns` has been removed in favor of the new option `format_columns`. All source configuration using the `source_to_display_columns` must be updated. A migration script will automatically modify source configuration in the `/etc/xivo-dird/sources.d` directory.

Please consult the following detailed upgrade notes for more information:

Asterisk 11 to 13 Upgrade Notes You might be impacted by the upgrade to Asterisk 13 if you have:

- custom dialplan
- custom Asterisk configuration
- custom application using AGI, AMI or any other Asterisk interface
- custom application exploiting CEL or queue_log
- custom Asterisk modules (e.g. `codec_g729a.so`)
- customized Asterisk in some other way
- DAHDI trunks using SS7 signaling

If you find yourself in one of these cases, you should make sure that your customizations still work with Asterisk 13.

If you are upgrading from Asterisk 1.8, you should also check the [Asterisk 1.8 to 11 upgrade notes](#).

Changes Between Asterisk 11 and 13 Some of the more common changes to look for:

- SS7 support is not available in the Asterisk package of XiVO between version 15.13 and 16.08 inclusively.
- All channel and global variable names are evaluated in a case-sensitive manner. In previous versions of Asterisk, variables created and evaluated in the dialplan were evaluated case-insensitively, but built-in variables and variable evaluation done internally within Asterisk was done case-sensitively.
- The SetMusicOnHold dialplan application was deprecated and has been removed. Users of the application should use the CHANNEL function's `musicclass` setting instead.
- The WaitMusicOnHold dialplan application was deprecated and has been removed. Users of the application should use MusicOnHold with a `duration` parameter instead.
- The SIPPEER dialplan function no longer supports using a colon as a delimiter for parameters. The parameters for the function should be delimited using a comma.
- The SIPCHANINFO dialplan function was deprecated and has been removed. Users of the function should use the CHANNEL function instead.
- For SIP, the codec preference order in an SDP during an offer is slightly different than previous releases. Prior to Asterisk 13, the preference order of codecs used to be:

1. Our preferred codec
2. Our configured codecs
3. Any non-audio joint codecs

Now, in Asterisk 13, the preference order of codecs is:

1. Our preferred codec
2. Any joint codecs offered by the inbound offer
3. All other codecs that are not the preferred codec and not a joint codec offered by the inbound offer

- Queue strategy `rrmemory` (Round robin memory) now has a predictable order. Members will be called in the order that they are added to the queue. For agents, this means they will be called in the order they are logged.
- When performing queue pause/unpause on an interface without specifying an individual queue, the `PAUSE-ALL/UNPAUSEALL` event will only be logged if at least one member of any queue exists for that interface. This has an impact on the agent performance statistics; an agent must be a member of at least 1 queue for its pause time to show up in the statistics.

You can see the complete list of changes from the Asterisk website:

- <https://wiki.asterisk.org/wiki/display/AST/Upgrading+to+Asterisk+12>
- <https://wiki.asterisk.org/wiki/display/AST/Upgrading+to+Asterisk+13>
- <http://git.asterisk.org/gitweb/?p=asterisk/asterisk.git;a=blob;f=CHANGES;h=d0363f7c3b03cec5f71b3806535c4f9d2b2baa02>

The AGI protocol did not change between Asterisk 11 and Asterisk 13; if you have custom AGI applications, you only need to make sure that the dialplan applications and functions you are using from the AGI are still valid.

List of Known Bugs And Limitations List of known bugs and limitations for Asterisk 13 in XiVO:

- When direct media is active and DTMF are sent using SIP INFO, DTMF are not working properly. It is also impossible to do an attended transfer from the XiVO client in these conditions.
See <http://projects.xivo.io/issues/5692>.

15.12

- Consult the [15.12 Roadmap](#)
- The certificate used for HTTPS in the web interface will be regenerated if the default certificate was used. Your browser will complain about the new certificate, and it is safe to accept it (see [#3656](#)). See also [HTTPS certificate](#).
- If you have an [HA configuration](#), then you should run `xivo-sync -i` on the master node to setup file synchronization between the master and the slave. File synchronization will then be done automatically every hour via `rsync` and `ssh`.
- `xivo-auth` and `xivo-dird` now use HTTPS, if you have custom development using these services, update your configuration accordingly.

15.11

- Consult the [15.11 Roadmap](#)
- The call records older than 365 days will be periodically removed. The first automatic purge will occur in the night after the upgrade. See [Purge Logs](#) for more details.

15.10

- Consult the [15.10 Roadmap](#)

15.09

- Consult the [15.09 Roadmap](#)

15.08

- Consult the [15.08 Roadmap](#)
- The Dialer Xlet has been integrated in Identity Xlet.

15.07

- Consult the [15.07 Roadmap](#)

15.06

- Consult the [15.06 Roadmap](#)
- The provd client has been moved into a new python package, xivo_provd_client. If you have custom scripts using this client, you'll need to update them. See <http://projects.xivo.io/issues/5469> for more information.
- The provd_pycli command name has been deprecated in favor of xivo-provd-cli. These 2 commands do the same thing, the only difference being the name of the command. The provd_pycli command name will be removed in 15.18, so if you have custom scripts referencing provd_pycli, you'll need to update them.
- The xivo-agentctl command name has been deprecated in favor of xivo-agentd-cli. These 2 commands do the same thing, the only difference being the name of the command. The xivo-agentctl command name will be removed in 15.18, so if you have custom scripts referencing xivo-agentctl, you'll need to update them.

15.05

- Consult the [15.05 Roadmap](#)
- The Xlet identity has been modified to follow the new XiVO Client design which implies the removal of some details.

15.04

- Consult the [15.04 Roadmap](#)

15.03

- Consult the [15.03 Roadmap](#)

15.02

- Consult the [15.02 Roadmap](#)

15.01

- Consult the [15.01 Roadmap](#)
- The *confd REST API* is now more restrictive on HTTP headers. Particularly, the headers Accept and Content-Type must be set to (typically) application/json.
- The following configuration files have been created:
 - /etc/xivo-agid/config.yml
 - /etc/xivo-call-logd/config.yml
 - /etc/xivo-amid/config.yml
 - /etc/xivo-agentd/config.yml

Archives

Archived Upgrade Notes

2014

14.24

- Consult the [14.24 Roadmap](#)

The following security vulnerability has been fixed:

- [XIVO-2014-01](#): Queues and groups permit callers to make unwanted calls

14.23

- Consult the [14.23 Roadmap](#)
- The “waiting calls / logged agents ratio” *queue diversion scenario* has been renamed to “number of waiting calls per logged agents”.
- A new *community* section was added to the official documentation for all user-contributed documentation.

14.22

- Consult the [14.22 Roadmap](#)
- The sheet event *Dial* on queues is now only sent to the ringing agent. The sheet is also sent a little later during the call, when the ringing agent is known.

14.21

- Consult the [14.21 Roadmap](#)
- The *confd REST API* is now accessible via HTTPS on port 9486 and via HTTP on port 9487 (localhost only). These ports are replacing the 50051 and 50050 ports respectively. It will still be possible to access the confd REST API via the 50051 and 50050 ports for the next year, but you are advised to update your confd REST API clients as soon as possible.
- The old (unsupported) ami-proxy is now replaced by an ami-proxy built in xivo-ctid. You must **uninstall the old ami-proxy** before activating the built-in version. See [troubleshooting xivo-ctid](#) to learn how to activate.

14.20

- Consult the [14.20 Roadmap](#)
- Default parameters for all Cisco SPA ATA plugins have changed to be better suited for european faxes.
- Following the [POODLE attack](#) (CVE-2014-3566), SSL 3.0 has been disabled for the web interface and the xivo-confd REST API.

If you have Aastra phones and are using the remote directory on them, consult the following detailed upgrade notes:

Aastra Remote Directory Upgrade Notes Starting from XiVO 14.20, it is not possible anymore to use SSL 3.0 when connecting to XiVO using HTTPS.

This has the unfortunate consequence of breaking the remote directory on Aastra phones configured by the `xivo-aastra` provisioning plugins in version 1.2 and earlier.

Upgrade procedure To be able to use the remote directory on your Aastra phones on XiVO 14.20 or later, you'll need to take one of the following actions:

Upgrade to the Latest Plugin This is the recommended solution. This can be done either before or after the upgrade. You'll have to:

1. Upgrade your xivo-aastra plugin to version 1.3 or later
2. Restart/synchronize all your phones

The correction is only available for plugin xivo-aastra-3.3.1-SP2 and later. If you are using an older plugin (xivo-aastra-3.2.2-SP3 for example), then you'll need to install a newer plugin and *update all your phones to use the new plugin*.

If you were already using custom templates, make sure to update them so that the phones access the remote directory via HTTP instead of HTTPS. This can be done using the following command:

```
find /var/lib/xivo-provd/plugins/xivo-aastra* -name '*.tpl' -exec sed -i '/X_xivo_phonebook_ip/s/'
```

Update the Templates If you can't or don't want to update to a newer plugin, you can instead update the templates used by the plugin. This can be done either before or after the upgrade. You'll have to:

1. Update the templates so that the directory is accessed via HTTP
2. Restart/synchronize all your phones

In this specific case, it is safe to directly modify the templates used by the plugin instead of *creating custom templates*. To update the templates, you can use the following command:

```
find /var/lib/xivo-provd/plugins/xivo-aastra* -name '*.tpl' -exec sed -i '/X_xivo_phonebook_ip/s/'
```

Re-enable SSL 3.0 If you can't restart/synchronize your phones, the last solution is to re-enable SSL 3.0 on your XiVO. This should only be used as a temporary solution to give you more time to plan a firmware upgrade for your phones. This can be done only after the upgrade. You'll have to:

1. Update nginx configuration
2. Reload nginx

This can be done using the following commands:

```
sed -i 's/ssl_protocols ./ssl_protocols SSLv3 TLSv1 TLSv1.1 TLSv1.2;/' /etc/nginx/sites-available/
service nginx reload
```

14.19

- Consult the [14.19 Roadmap](#)

14.18

- Consult the [14.18 Roadmap](#)
- xivo-fai packages were replaced with xivo-dist : a new tool to handle repositories sources. Upon upgrade, xivo-dist is installed and run and all xivo-fai packages are purged. *Consult xivo-dist use cases*

14.17

- Consult the [14.17 Roadmap](#)
- DAHDI configuration file /etc/dahdi/modules is no more created by default and must now be maintained manually. No action is needed upon upgrade but be aware that the upstream sample file is now available in /usr/share/dahdi/modules.sample. See [dahdi modules documentation](#) for detailed info.
- The new *CCSS feature* will not be enabled upon upgrade, you must explicitly enable it in the *IPBX* → *IPBX Services* → *Extensions* menu.

14.16

- Consult the [14.16 Roadmap](#)
- See the [changelog](#) for xivo-confd's REST API
- DAHDI is upgraded to 2.10.0. If the upgrade process asks about `/etc/dahdi/modules`, we recommend that you keep the old version of the file.
- Asterisk now inserts CEL and queue log entries via the ODBC asterisk modules instead of the pgsql modules.

14.15

- Consult the [14.15 Roadmap](#)
- Duplicate function keys will be deleted upon upgrade. If multiple function keys pointing to the same destination are detected for a given user, only the one with the lowest position will be kept. To see the list of deleted function keys, check the xivo-upgrade log file such as:

```
grep MIGRATE_FK /var/log/xivo-upgrade.log
```

DAHDI 2.9.2 Upgrade Notes These notes only apply to:

- Digium TE133/TE131 cards that are in firmware version 780017 or earlier
- Digium TE435/TE235 cards that are in firmware version e0017 or earlier

Warning: The system will need to be power cycled after the upgrade. Your cards will not be usable until then.

After the upgrade First, you need to install the latest firmware for your TE133/TE131 or TE435/TE235 cards:

```
xivo-fetchfw install digium-te133
xivo-fetchfw install digium-te435
```

Then stop all the services and reload the DAHDI modules. Reloading the DAHDI module might take up to 30 seconds:

```
xivo-service stop
service dahdi stop
service dahdi start
```

Following this manipulation, you should see something similar at the end of the `/var/log/messages` file:

```
dahdi: Telephony Interface Unloaded
dahdi: Version: 2.9.2
dahdi: Telephony Interface Registered on major 196
wctel3xp 0000:03:0c.0: Firmware version 780017 is running, but we require version 780019.
wctel3xp 0000:03:0c.0: firmware: agent loaded dahdi-fw-tel133.bin into memory
wctel3xp 0000:03:0c.0: Found dahdi-fw-tel133.bin (version: 780019) Preparing for flash
wctel3xp 0000:03:0c.0: Uploading dahdi-fw-tel133.bin. This can take up to 30 seconds.
wctel3xp 0000:03:0c.0: Delaying reset. Firmware load requires a power cycle
wctel3xp 0000:03:0c.0: Running firmware version: 780017
wctel3xp 0000:03:0c.0: Loaded firmware version: 780019 (Will load after next power cycle)
wctel3xp 0000:03:0c.0: FALC version: 5
wctel3xp 0000:03:0c.0: Setting up global serial parameters for T1
wctel3xp 0000:03:0c.0: VPM450: firmware dahdi-fw-oct6114-032.bin not available from userspace
```

For the firmware update to complete, you **must halt** the machine (a reboot won't be enough) before restarting it.

14.14

- Consult the [14.14 Roadmap](#)
- See the [changelog](#) for REST API
- Upon an important freeze of Asterisk, Asterisk will be restarted. See the [associated ticket](#) for more information.

14.13

- Consult the [14.13 Roadmap](#)
- See the [changelog](#) for REST API
- Skills-based routing: for an agent which doesn't have the skill X, the rule $X < 10$ was previously evaluated to true, since not having the skill X was equivalent to having it with a value of 0. This behaviour has changed, and the same expression is now evaluated to false. If you are using skills-based routing, you'll need to check that your rules are still doing what you expect. See [skill evaluation](#) for more information.

14.12

- Consult the [14.12 Roadmap](#)
- All provisioning plugins were modified. Although not mandatory, it is strongly advised to update all used plugins.
- The function key 'Activate voicemail' was removed as it was a duplicate of existing function key 'Enable voicemail'. All users having the 'Activate voicemail' function key will have to be reconfigured with a 'Enable voicemail' function key in order to keep the equivalent feature.
- Log files have changed for the following daemons (previously in `/var/log/daemon.log`):
 - xivo-provd: `/var/log/xivo-provd.log`
 - xivo-agid: `/var/log/xivo-agid.log`
 - xivo-sysconfd: `/var/log/xivo-sysconfd.log`

14.11

- Consult the [14.11 Roadmap](#)
- The API URL `/lines/<id>/extension` is now deprecated. Use `/lines/<id>/extensions` instead.

14.10

- Consult the [14.10 Roadmap](#)
- Custom MOH have been [fixed](#), but can not be used for playing uploaded files anymore. See [Music on Hold](#).

14.09

- Consult the [14.09 Roadmap](#)
- REST API 1.0 is no more. All code, tests and documentation was removed from XiVO. All code developed for REST API 1.0 must now be adapted to use REST API 1.1.

14.08

- Consult the [14.08 Roadmap](#)
- The `xivo` database has been merged into the `asterisk` database. The database schema has also been altered in a way that it might make the upgrade longer than usual.

Please consult the following detailed updated notes for more information:

Databases Merge Upgrade Notes The `xivo` database has been merged into the `asterisk` database in XiVO 14.08. This has an impact on:

- The [restore](#) procedure. There's only one database to restore now. Also, the procedure to restore the data while keeping the system configuration has been updated.
- The data that is replicated between the master and the slave in a [high availability](#) cluster.

Previously, all the configuration that was under the “Configuration” menu of the web interface was not replicated between the master and slave. This is now replicated, except for:

- HA settings
- All the network configuration (i.e. everything under the *Configuration* → *Network* section)
- All the support configuration (i.e. everything under the *Configuration* → *Support* section)

The call center statistics have also been excluded from the replication.

The way the replication is done has also been updated, which makes it faster.

Optional Upgrade Procedure When upgrading to XiVO 14.08, the database schema will be altered.

This will result in a longer upgrade time if you have a lots of rows in the `queue_log` table.

You can see the number of rows in your `queue_log` table with:

```
sudo -u postgres psql -c "SELECT count(*) FROM queue_log" asterisk
```

On ordinary hardware, you can expect that it will take ~10 minutes for every 2.5 million of rows. So if you have 5 million of rows in your `queue_log` table, you can expect that the upgrade will take an extra 20 minutes.

It is possible to reduce the amount of additional time the upgrade will take by either removing rows from the table or altering the table before the upgrade.

Both these commands can be run while the XiVO services are up.

For example, if you want to remove all the rows before march 2014, you can use:

```
sudo -u postgres psql -c "DELETE FROM queue_log WHERE \"time\" < '2014-03-01'" asterisk
```

If you want to alter the table before the upgrade, you can use:

```
sudo -u postgres psql -c "ALTER TABLE queue_log ADD COLUMN id SERIAL PRIMARY KEY; GRANT ALL ON SE
```

Note: It is recommended to execute this command when there's no activity on the system.

More Technical Information The way the database is initially provisioned and the way it is altered during an upgrade has also been changed.

In XiVO 14.07 and earlier, the database was provisioned by executing the `/usr/share/xivo-manage-db/datastorage/asterisk.sql` SQL script. Starting with XiVO 14.08, the `xivo-init-db` is responsible for provisioning the database. This script should not be used by an administrator in normal circumstance.

Starting with XiVO 14.08, database migration are done with the help of [alembic](#) instead of the asterisk-XXX.sql and xivo-XXX.sql scripts. The alembic migration scripts can be found inside the `/usr/share/xivo-manage-db` directory.

Otherwise, the `xivo-check-db` and `xivo-update-db` commands have been updated to work with both the old and the new systems and are still the official way to check the database state and update the database respectively.

14.07

- Consult the [14.07 Roadmap](#)
- Configuration for phones used for the switchboard has changed.

Please consult the following detailed updated notes for more information:

Switchboard Phone Configuration Upgrade Notes The `xivo-aastra-switchboard` and `xivo-snom-switchboard` plugins have been removed and their functionalities are now provided by the generic `xivo-aastra` and `xivo-snom` plugins respectively.

The upgrade is not done automatically, so please follow the [Upgrade Procedure](#) section below.

Although you are strongly advised to upgrade your switchboard phone configuration, backwards compatibility with the old system will be maintained.

Note that if you need to install a switchboard for a previous version of XiVO, the old `xivo-aastra-switchboard` and `xivo-snom-switchboard` plugins can be found in [the archive repository](#).

Upgrade Procedure This procedure should be executed after the upgrade to 14.07 or later: the options used in this procedure are not available in versions before 14.07.

The following upgrade procedure suppose that you are using an Aastra phone as your switchboard phone. The same upgrade procedure apply for Snom phones, with the only difference being the different plugin name.

1. Update the list of installable plugins.
2. Install the latest `xivo-aastra` plugin, or upgrade it to the latest version if it is already installed.
3. Install the needed language files and firmware files.
4. For each phone used for the switchboard, [change the plugin and activate the switchboard option](#):
 - Select the generic `xivo-aastra` plugin.
 - Check the “switchboard” checkbox.
 - Synchronize the phone.
5. Once this is completed, you can uninstall the `xivo-aastra-switchboard` plugin.

An unofficial script that automates this procedure is also available on github:

```
cd /tmp
wget --no-check-certificate https://raw.githubusercontent.com/xivo-pbx/xivo-tools/master/scripts/
python migrate_switchboard_1407.py
```

14.06

- Consult the [14.06 Roadmap](#)
- The XiVO client now uses Qt 5 instead of Qt 4. There is nothing to be aware of unless you are [building your own version](#) of it.

14.05

- Consult the [14.05 Roadmap](#)
- The *CTI Protocol* has been updated.
- The specification of the ‘answered-rate’ queue statistic has changed to exclude calls on a closed queue
- The switchboard can now choose which incoming call to answer
- The package versions do not necessarily contain the current XiVO version, it may contain older versions. Only the package `xivo` is guaranteed to have the current XiVO version.

Please consult the following detailed updated notes for more information:

DAHDI 2.9.0 Upgrade Notes These notes only apply to Digium TE133 or TE134 cards that are in firmware version 770017 or earlier.

Warning: The system will need to be power cycled after the upgrade. Your cards will not be usable until then.

After the upgrade First, you need to install the latest firmware for your TE133 or TE134 cards:

```
xivo-fetchfw install digium-te133
xivo-fetchfw install digium-te134
```

Then stop all the services and reload the DAHDI modules. Reloading the DAHDI module might take up to 30 seconds:

```
xivo-service stop
service dahdi stop
service dahdi start
```

Following this manipulation, you should see something similar at the end of the `/var/log/messages` file:

```
dahdi: Telephony Interface Unloaded
dahdi: Version: 2.9.0
dahdi: Telephony Interface Registered on major 196
wctel3xp 0000:03:0c.0: Firmware version 6f0017 is running, but we require version 780017.
wctel3xp 0000:03:0c.0: firmware: agent loaded dahdi-fw-te134.bin into memory
wctel3xp 0000:03:0c.0: Found dahdi-fw-te134.bin (version: 780017) Preparing for flash
wctel3xp 0000:03:0c.0: Uploading dahdi-fw-te134.bin. This can take up to 30 seconds.
wctel3xp 0000:03:0c.0: Delaying reset. Firmware load requires a power cycle
wctel3xp 0000:03:0c.0: Running firmware version: 6f0017
wctel3xp 0000:03:0c.0: Loaded firmware version: 780017 (Will load after next power cycle)
wctel3xp 0000:03:0c.0: FALC version: 5
wctel3xp 0000:03:0c.0: Setting up global serial parameters for T1
wctel3xp 0000:03:0c.0: VPM450: firmware dahdi-fw-oct6114-032.bin not available from userspace
wctel3xp 0000:03:0c.0: Found a Wildcard TE132/TE134 (SN: 1TE134F - DF05132600690 - B1 - 20130702)
```

For the firmware update to complete, you **must halt** the machine (a reboot won’t be enough) before restarting it.

SCCP Upgrade Notes Important modification have been made to the internal structure of the SCCP channel driver, `xivo-libsccp`.

The modifications mostly affect administrators; users are not affected.

Major changes are:

- Improved support for live modifications; no more manual intervention in the asterisk CLI is needed.
- Improved handling of concurrency; crash and deadlock due to concurrency problems should not occur anymore.

CLI The following commands have been removed because they were not needed:

- `sccp resync`
- `sccp set directmedia`
- `sccp show lines`
- `sccp update config`

The behavior of the following commands have been changed:

- `module reload chan_sccp` reloads the module configuration, without interrupting the telephony service. A device will only be reset/restarted if needed, and only once the device is idle. Some changes don't even require the device to be reset.
- `sccp show config` output format has been changed a little.
- `sccp show devices` only show the connected devices instead of all the devices. This might change in the future. To get a list of all the devices, use `sccp show config`.

Configuration File The format of the `sccp.conf` configuration file has been changed. This will only impact you if you are using `xivo-libscdp` without using XiVO.

The format has been changed because the module is now using the ACO module from asterisk, which expect configuration file to have a specific format.

See [sccp.conf.sample](#) for a configuration file example.

Other Each SCCP session/connection now use 3 file descriptors instead of 1 previously. On XiVO, the file descriptor limit for the asterisk process is 8192, which means that the increase in used file descriptors should not be a problem, even on a large installation.

14.04

- Consult the [14.04 Roadmap](#)
- Live reload of the configuration can be enabled and disabled using the REST API
- The generation of call logs for unanswered calls from the XiVO client have been improved.

14.03

- Consult the [14.03 Roadmap](#)
- A migration script adds an index on the `linkedid` field in the `cel` table. Tests have shown that this operation can last up to 11.5 minutes on a XiVO Corporate with 18 millions CELs. `xivo-upgrade` will thus be slightly longer.
- Two new daemons are now operationnal, `xivo-amid` and `xivo-call-logd`:
 - `xivo-amid` constantly reads the AMI and sends AMI events to the RabbitMQ bus
 - `xivo-call-logd` generates call-logs in real time based on AMI `LINKEDID_END` events read on the bus
- An increase in load average is expected with the addition of these two new daemons.
- The cron job calling `xivo-call-logs` now runs once a day at 4:25 instead of every 5 minutes.

14.02

- Consult the [14.02 Roadmap](#)
- PHP Web services has been removed from documentation
- REST API 1.0 Web services has been removed from documentation

- REST API 1.1 User-Line-Extension service is replaced by User-Line and Line-Extension services

14.01

- Consult the [14.01 Roadmap](#)
- The following paths have been renamed:
 - `/etc/pf-xivo` to `/etc/xivo`
 - `/var/lib/pf-xivo` to `/var/lib/xivo`
 - `/usr/share/pf-xivo` to `/usr/share/xivo`

You must update any dialplan or configuration file using these paths

2013

13.25

- Consult the [13.25 Roadmap](#)
- Debian has been upgraded from version 6 (squeeze) to 7 (wheezy).

Please consult the following detailed upgrade notes for more information:

Debian 7 (wheezy) Upgrade Notes

Before the upgrade

- The upgrade will take longer than usual, because the whole Debian system will be upgraded
- The system must be restarted after the upgrade, because the Linux kernel will also be upgraded

LDAPS In case XiVO is using a LDAP server through SSL/TLS (LDAPS), the documentation instructed you to append the certificate to `/etc/ssl/certs/ca-certificates.crt`. However, this is the wrong way to add a new certificate, because it will be erased by the upgrade.

To keep your certificate installed through the upgrade, you must follow the instructions given in the [LDAP documentation](#).

After the upgrade

GRUB (Cloned Virtual Machines only) GRUB installations on cloned virtual machines may lead to unbootable systems, if not fixed properly before restarting the system. If xivo-upgrade detects your system is in a broken state, it will display a few commands to repair the GRUB installation.

13.24

- Consult the [13.24 Roadmap](#)
- Default Quality of Service (QoS) settings have been changed for SCCP. The IP packets containing audio media are now marked with the EF DSCP.

13.23

- Consult the [13.23 Roadmap](#)
- The *New call* softkey has been removed from SCCP phones in *connected* state. To start a new call, the user will have to press *Hold* then *New call*. This is the same behavior as a *Call Manager*.
- Some softkeys have been moved on SCCP phones. We tried to keep the keys in the same position at any given time. As an example, the *transfer* key will not become *End call* while transferring a call. Note that this is a work in progress and some models still need some tweaking.

13.22

- Consult the [13.22 Roadmap](#)
- PostgreSQL will be upgraded from 9.0 to 9.1. The upgrade of XiVO will take longer than usual, depending on the size of the database. Usually, the database grows with the number of calls processed by XiVO. The upgrade will be stopped if not enough space is available on the XiVO server.

13.21

- Consult the [13.21 Roadmap](#)
- It is no more possible to delete a device associated to a line using REST API.

13.20

- Consult the [13.20 Roadmap](#)
- xivo-libscdp now supports direct media on wifi phone 7920 and 7921
- xivo-confd now implements a voicemail list

13.19

- Since XiVO 13.18 was not released, the 13.19 release contains all developments of both 13.18 and 13.19, therefore please consult both Roadmaps :
- Consult the [13.19 Roadmap](#)
- Consult the [13.18 Roadmap](#)
- Call logs are now generated automatically, incrementally and regularly. Call logs generated before 13.19 will be erased one last time.
- The database was highly modified for everything related to devices : table `devicefeatures` does not exist anymore and now relies on information from `xivo-provd`.

13.17

- Consult the [13.17 Roadmap](#)
- There is a major change to call logs. They are no longer available as a web report but only as a csv export. See the [call logs documentation](#). Furthermore, call logs are now fetched from xivo-confd REST API.
- Paging group numbers are now exclusively numeric. All non-numeric paging group numbers are converted to their numeric-only equivalent while upgrading to XiVO 13.17 (*58 becomes 58, for example).

13.16

- Consult the [13.16 Roadmap](#)
- A migration script modifies the user and line related-tables and the way users, lines and extensions are associated. As a consequence of this script, it is not possible any more to associate a user and a line without extensions. Existing associations between users and one or more lines having no extensions will be removed. Users and lines will still exist unassociated.
- The *call logs* page is able to display partial results of big queries, instead of displaying a blank page.
- Two new CEL messages are now enabled : LINKEDID_END and BRIDGE_UPDATE. Those events will only exist in CEL for calls passed after upgrading to XiVO 13.16.
- The new REST API now makes possible to associate multiple user to a given line and/or extension. There are currently some limitations on how those users and lines can be manipulated using the web interface.

13.15

- There was no production release of XiVO 13.15. All 13.15 developments are included in the official 13.16 release.

13.14

- Consult the [13.14 Roadmap](#)
- The latest Polycom plugin enables the phone lock feature with a default user password of '123'. All Polycom phones used with XiVO also have a default admin password. In order for the phone lock feature to be secure, one should change every phone's admin AND user passwords.
- WebServices for SIP trunks/lines: field nat: value yes changed to force_rport, comedia
- The database has been updated in order to remove deprecated tables (generalfeatures, extennumbers, exten-hash, cost_center).

13.13

- Consult the [13.13 Roadmap](#)

13.12

- Consult the [13.12 Roadmap](#)
- CTI protocol: Modified values of agent availability. Read *CTI Protocol changelog*
- Clean-up was made related to the minimization of the XiVO Client. Some visual differences have been observed on Mac OS X that do not affect the XiVO Client in a functional way.

13.11

- Consult the [13.11 Roadmap](#)
- Asterisk has been upgraded from version 11.3.0 to 11.4.0

API changes:

- Dialplan variable XIVO_INTERFACE_0 is now XIVO_INTERFACE
- Dialplan variable XIVO_INTERFACE_NB and XIVO_INTERFACE_COUNT have been removed
- The following fields have been removed from the lines and users web services
 - line_num
 - roles_group

- rules_order
- rules_time
- rules_type

13.10

- Consult the [13.10 Roadmap](#)

API changes:

- CTI protocol: for messages of class `getlist` and function `updateconfig`, the `config` object/dictionary does not have a `rules_order` key anymore.

13.09

- Consult the [13.09 Roadmap](#)
- The *Restart CTI server* link has been moved from *Services* → *CTI Server* → *Control* to *Services* → *IPBX* → *Control*.
- The Agent Status Dashboard has been optimized.
- The Directory xlet can now be used to place call.

13.08

- Consult the [13.08 Roadmap](#)
- asterisk has been upgraded from version 1.8.21.0 to 11.3.0, which is a major asterisk upgrade.
- The switchboard's queue now requires the *xivo_subr_switchboard* preprocess subroutine.
- A fix to bug [#4296](#) introduced functional changes due to the order in which sub-contexts are included. Please refer to [ticket](#) for details.

Please consult the following detailed upgrade notes for more information:

Asterisk 1.8 to 11 Upgrade Notes Table of modules that were available in the asterisk 1.8 package but that are not available anymore in the asterisk 11 package:

Name	Description	Loaded in AST1.8	Asterisk Status	Replaced By
app_dahdibarge	Barge in on DAHDI channel application	Yes	Deprecated	app_chanspy
app_readfile	Stores output of file into a variable	Yes	Deprecated	func_env (FILE())
app_saycountpl	Say polish counting words	Yes	Deprecated	say.conf
app_setcallerid	Set CallerID Presentation Application	Yes	Deprecated	func_callerid
cdr_sqlite	SQLite CDR Backend	No	Removed	cdr_sqlite3_custom
chan_gtalk	Gtalk Channel Driver	No	Deprecated	chan_motif
chan_jingle	Jingle Channel Driver	No	Deprecated	chan_motif
chan_vpb	Voicetronix API driver	No	Supported	
format_sln16	Raw Signed Linear 16KHz Audio support	Yes	Removed	format_sln
res_ais	SAForum AIS	No	Removed	res_corosync
res_jabber	AJI - Asterisk Jabber Interface	No	Deprecated	res_xmpp

List of modules that were loaded in asterisk 1.8 but that are not loaded anymore in asterisk 11 (see modules.conf):

- res_calendar.so

- `res_calendar_caldav.so`
- `res_calendar_ews.so`
- `res_calendar_exchange.so`
- `res_calendar_icalendar.so`
- `res_config_sqlite.so`
- `res_stun_monitor.so`

List of debian packages that are not available anymore for asterisk 11:

- `asterisk-config`
- `asterisk-mysql`
- `asterisk-web-vmail`

Note: These packages were not installed by default for asterisk 1.8.

If you are using some custom dialplan or AGIs, it is your responsibility to make sure it still works with asterisk 11. See the [External Links](#) for more information.

External Links

- <http://svnview.digium.com/svn/asterisk/branches/11/UPGRADE-10.txt>
- <http://svnview.digium.com/svn/asterisk/branches/11/UPGRADE.txt>
- <https://wiki.asterisk.org/wiki/display/AST/New+in+10>
- <https://wiki.asterisk.org/wiki/display/AST/New+in+11>

The switchboard's queue preprocess subroutine The switchboard's queue now uses a preprocess subroutine named *xivo_subr_switchboard*. This preprocess subroutine will be associated with all queues named *__switchboard* that have no preprocess subroutine defined before the upgrade.

If your switchboard queue is named anything other than *__switchboard* you should add the preprocess subroutine manually.

If your switchboard queue already has a preprocess subroutine, you should add a `Gosub(xivo_subr_switchboard)` to your preprocess subroutine.

<p>Warning: This change is only applied to the switchboard distribution queue, not the queue for calls on hold.</p>
--

13.07

- Consult the [13.07 Roadmap](#)
- Agent Status Dashboard has more features and less limitations. See related [agent status dashboard documentation](#)
- XiVO call centers have no more notion of 'disabled agents'. All previously disabled agents in web interface will become active agents after upgrading.
- asterisk has been upgraded from version 1.8.20.1 to 1.8.21.0. Please note that in XiVO 13.08, asterisk will be upgraded to version 11.
- DAHDI has been upgraded from version 2.6.1 to 2.6.2.
- libpri has been upgraded from version 1.4.13 to 1.4.14.
- PostgreSQL upgraded from version 9.0.4 to 9.0.13

13.06

- Consult the [13.06 Roadmap](#)
- The new Agent Status Dashboard has a few known limitations. See related [dashboard xlet known issues section](#)
- Status Since counter in xlet list of agents has changed behavior to better reflect states of agents in queues as seen by asterisk. See [Ticket #4254](#) for more details.

13.05

- Consult the [13.05 Roadmap](#)
- The bug [#4228](#) concerning BS filter only applies to 13.04 servers installed from scratch. Please upgrade to 13.05.
- The order of softkeys on SCCP phones has changed, e.g. the *Bis* button is now at the left.

13.04

- Consult the [13.04 Roadmap](#)
- Upgrade procedure for HA Cluster has changed. Refer to [Specific Procedure : Upgrading a Cluster](#).
- Configuration of switchboards has changed. Since the directory xlet can now display any column from the lookup source, a display filter has to be configured and assigned to the `__switchboard_directory` context. Refer to [Directory xlet documenttion](#).
- There is no more context field directly associated with a call filter. Boss and secretary users associated with a call filter must necessarily be in the same context.

2012

12.24

- Consult the [12.24 Roadmap](#)
- XiVO 12.24 has some limitations mainly affecting the contact center features due to the rewriting of the code handling agents.

Please consult the following detailed upgrade notes for more information:

Contact Center XiVO 12.24 In order to fix problems related to Asterisk freezing through the `chan_agent` module, XiVO 12.24 implements a new way of managing agents.

Warning The contact center XiVO 12.24 does not implement all the features available in 12.22. Therefore, you must not upgrade your XiVO if you depend on these features. These features will be reimplemented in the future starting with version 13.01.

Missing Features

- Skill-based routing
- Penalties
- Call listening

Live reload via the web interface Agents must be logged out for the following operations:

- Adding or removing agents from the queues
- When changing the name of a queue (only the name, not the displayed name)

You can logoff all the agents with the following command:

```
xivo-agentctl -c "logoff all"
```

Preprocess subroutines Subroutines on users are currently no longer executed when an agent receives a call from the queue

High availability (HA) HA for the contact center is not supported for the moment. When switching from a master to a slave, you must relog all your agents.

SCCP Devices The “Available” / “In use” statuses for agents that are logged in do not work for the moment.

Changes in behavior

“In use” indicator in the XiVO client In XiVO 12.22, an agent is seen as “In use” when:

- The agent’s phone is ringing or has answered a call coming only from a queue

In XiVO 12.24:

- The agent’s phone is “In use” no matter where the call comes from

“Available” indicator in the XiVO client In XiVO 12.22, an agent is seen as “Available” when:

- The agent is not in pause/wrapup and his phone isn’t ringing/in conversation for a call coming from a queue

In XiVO 12.24:

- The agent is not in pause/wrapup and his phone is in the “idle” state

“Agent linked” / “Agent unlinked” Events The “Agent linked” event no longer exists in XiVO 12.24. xivo-upgrade will automatically migrate “Agent linked” / “Agent unlinked” sheets to the “linked” / “unlinked” event.

“Static” Agent VS “Dynamic” agent in the XiVO client There is no longer a difference between a “static” or “dynamic” membership. All agent memberships are now considered “static”. Membership changes between the web interface and the XiVO client are now synchronized.

Updates Please note that when upgrading, the following actions will take place automatically:

- All agents will be logged off before migrating
- All agents with a “dynamic” membership will be removed from their queues

Another change is in effect beginning with XiVO 12.24: the field `profileclient` in the CSV user import sees its values change.

Old value	New value
client	Client
agent	Agent
switchboard	Switchboard
agentsup	Supervisor
oper	<i>removed</i>
clock	<i>removed</i>

XiVO Client

What is the XiVO Client

The XiVO Client is an application that you install on your computer and is connected to the XiVO server. This application offers the following features:

- search contacts and show their presence, phone status
- make calls through your phone (the XiVO Client is **NOT** a softphone, it is complementary to the phone)
- access your voicemail through your phone
- enable call forwards, call filtering
- show the history of your calls
- list conference rooms and members
- send faxes

It also offers some call center features:

- show screen popups or open URLs when you receive/answer a call
- list agents, queues, calls in queues
- login/logoff, pause/unpause other agents (for supervisors)
- listen/whisper to agents through you phone (for supervisors)

A lot of those features are modular and may be enabled for each user by choosing which *Xlets* they can see.

Getting the XiVO client

Binaries of the XiVO Client are available on our mirror. ([latest version](#)) ([all versions](#))

Warning: The installed version of the XiVO Client must match the XiVO server's version installation. With our current architecture, there is no way to guarantee that the XiVO server will be retro-compatible with older versions of the XiVO Client. Non-matching XiVO server and XiVO Clients versions might lead to unexpected behaviour.

Choose the version you want and in the right directory, get :

- the `.exe` file for Windows
- the `.deb` file for Ubuntu or Debian (i386 or amd64, depending on your computer)
- the `.dmg` file for Mac OS

For Windows, double-click on the file and follow the instructions. You can also install it silently:

```
xivoclient-14.XX-x86.exe /S
```

For Ubuntu/Debian, double-click on the file or execute the following command:

```
$ gdebi xivoclient-*.deb
```

For Mac OS, double-click on the file and drag-and-drop the inner file on the Application entry of the Finder.

The XiVO Client should then be available in the applications menu of each platform.

If you want to build your own XiVO Client, see [Building the XiVO Client](#).

Connection to the server

To connect to the server using the XiVO client you need a user name, a password and the server's address. Optionally, it is possible to login an agent while connecting to the server.

Xlets

Xlets are features of the XiVO Client. It is the contraction of XiVO applets. To select which xlets are displayed in your client, see *CTI profiles*.

Conference Xlet

Overview The conference xlet allow the user to join conferences and view conference room statuses.

Usage The *Conference room list* tab show all available conference rooms configured on the XiVO. The user can click on a conference number to join the conference. When a user joins a conference, his phone will ring and the conference will be joined when the user answers the phone.

When clicking on a conference room a new tab is opened for the selected conference room. The new tab contains information about the members of the conference. The name and number of the member will be displayed when available. Users can also mute and unmute themselves using the microphone icon on the left.

Directory Xlet







Overview **Warning:** This xlet should only be used with a Switchboard profile. It is not meant to be used alone.

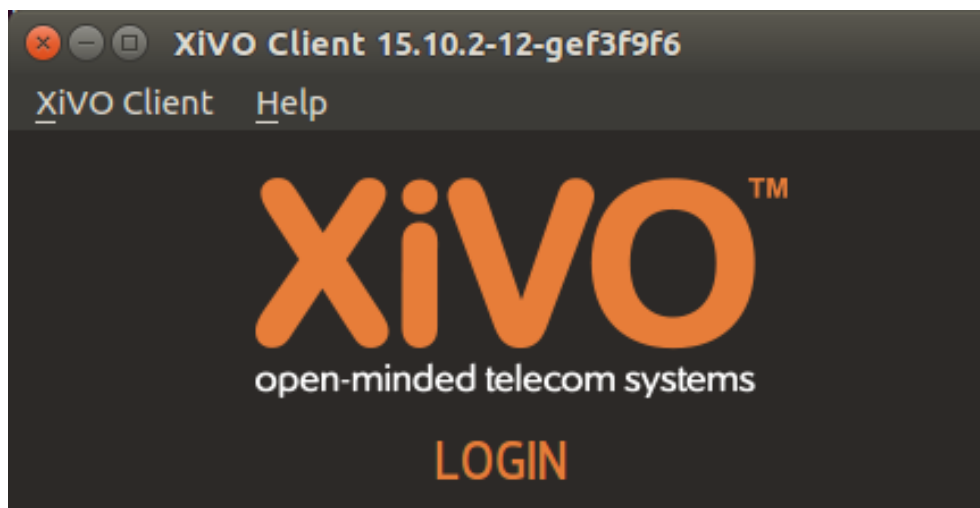
The goal of the directory xlet is to allow the user to search through XiVO users, directory entries and arbitrary numbers to be able to call and transfer calls to these destinations.

Usage The list of entries in the xlet is searched using the top field. Entries are filtered by column content. The entry list will initially appear as empty.

If the current search term is a valid number, it will be displayed in the result list with no name to allow transfer to numbers that are not currently in the phonebook or configured on the XiVO.

Legend

- Users available 
- Users ringing 
- Users talking 
- Users 
- Mobile phone 
- External contacts 

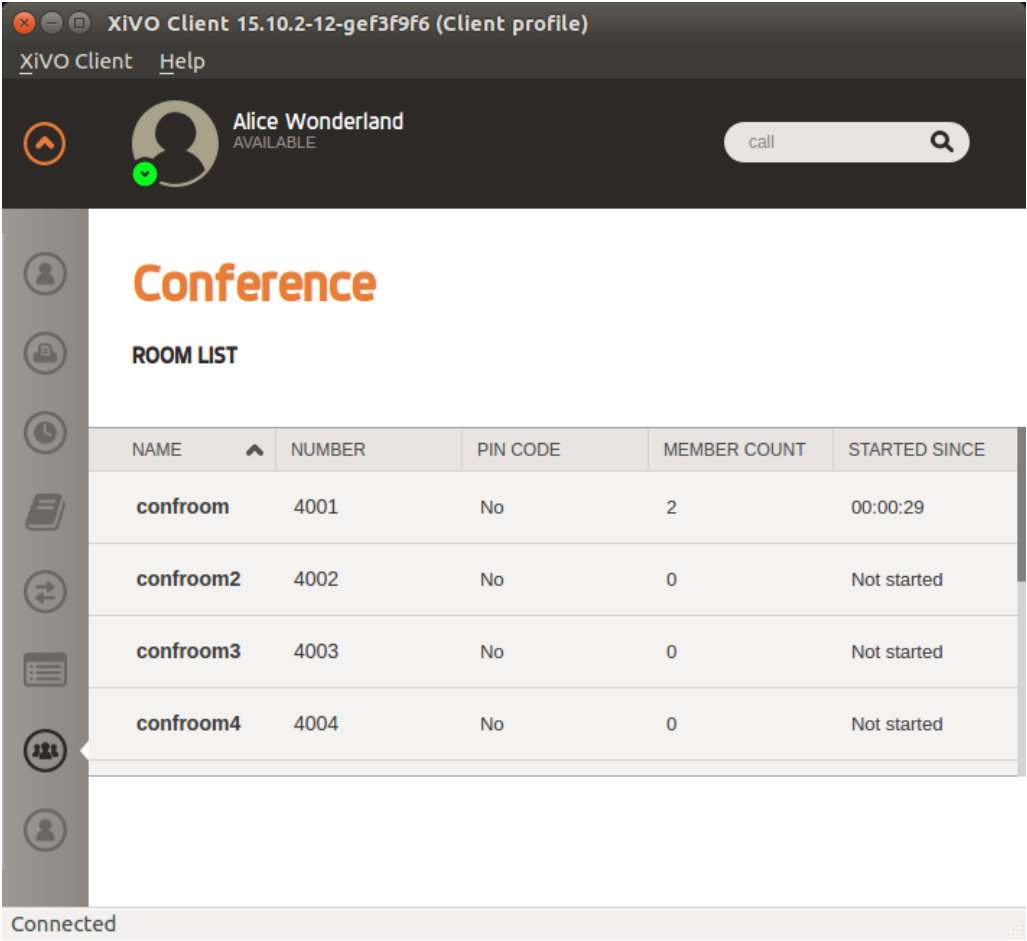


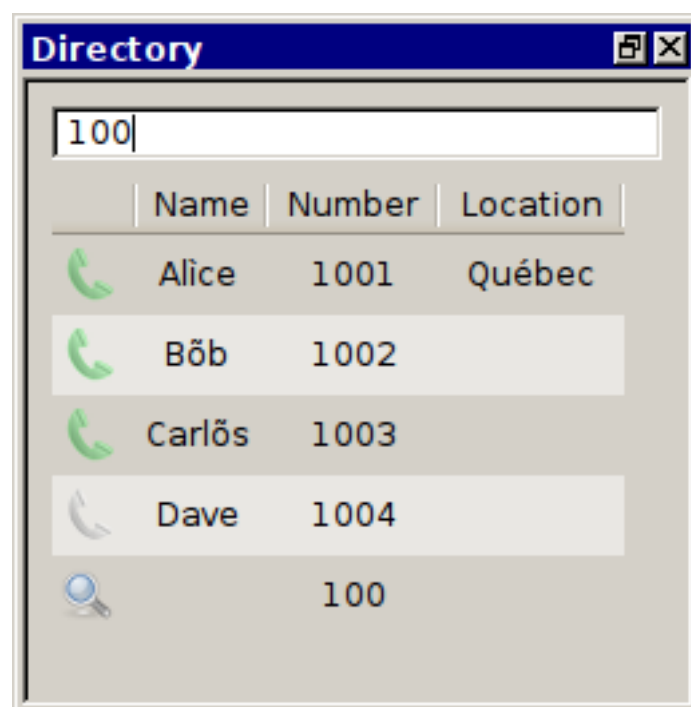
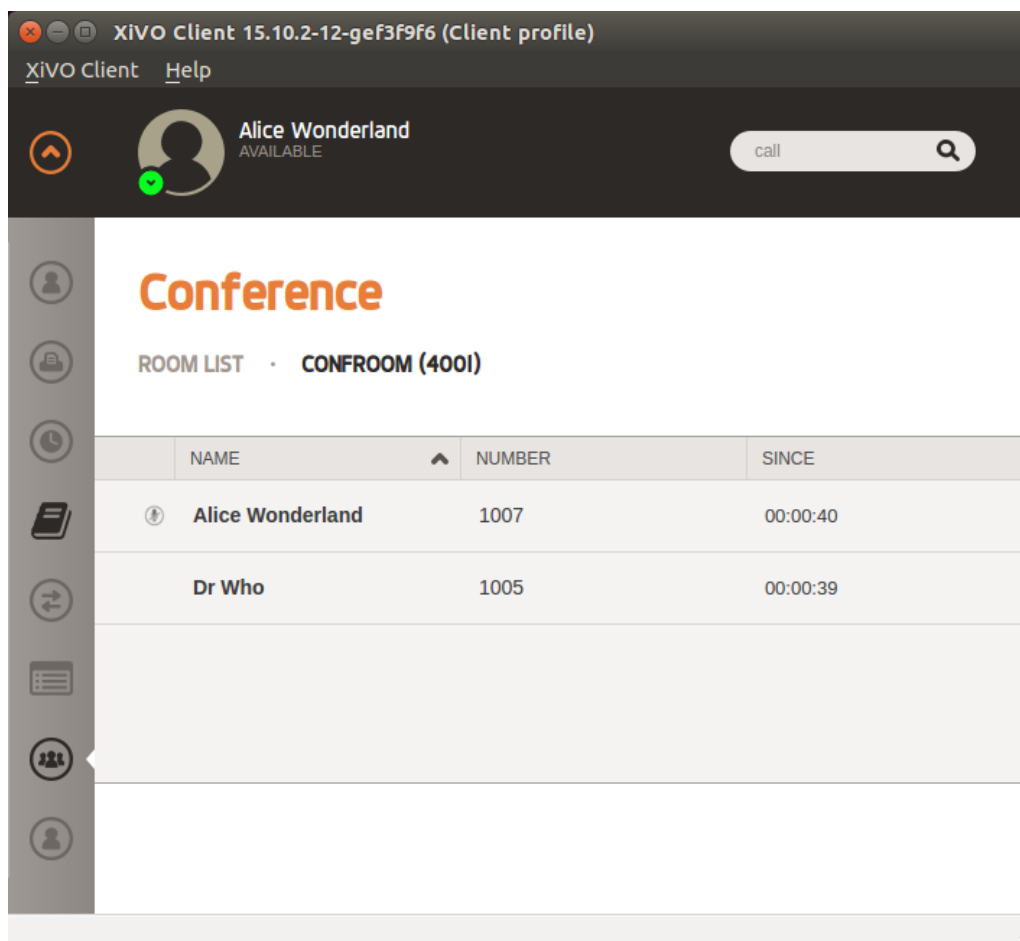
XiVO is a unified communication system that connects phones inside an organization with public and mobile telephone networks.

☒ Remember me

Agent (logged in) ▼

CONNECT





- Current search (not a contact)



Phonebook Phonebook searches are triggered after the user has entered 3 characters. Results from remote directories will appear after 1 second.

If a directory entry has the same number as a mobile or a phone configured on the XiVO, its extra columns will be added to the corresponding entry instead of creating a new line in the search result.

For example:

If *User 1* has number *1000* and is also in a configured LDAP with a location in “Québec”, if the display filter contains the *Location* column, the entry for *User 1* will show “Québec” in the *Location* column after the search results are received.

Configuration

Context The directory xlet needs a special context named `__switchboard_directory`. In *Services* → *IPBX* → *IPBX configuration* → *Contexts* add a new context with the following parameters :

- Name : `__switchboard_directory`
- Type of context : **Other**
- Display name : Switchboard

Name	Displayed name	Context type	Entity	Action
<input type="checkbox"/> > default	default	Internal	pcm-dev (pcm-dev)	
<input type="checkbox"/> > from-extern	Incalls	Incall	pcm-dev (pcm-dev)	
<input type="checkbox"/> > invalid	invalid	Incall	pcm-dev (pcm-dev)	
<input type="checkbox"/> > pcm-dev	pcm-dev	Internal	pcm-dev (pcm-dev)	
<input type="checkbox"/> > statscenter	statscenter	Internal	pcm-dev (pcm-dev)	
<input type="checkbox"/> > __switchboard_directory	Switchboard	Other	pcm-dev (pcm-dev)	
<input type="checkbox"/> > to-extern	Outcalls	Outcall	pcm-dev (pcm-dev)	

Display filter A new display filter must be created for the directory xlet.

Update displays

Name:

Field title	Field type	Default value	Field name	
<input type="text" value=""/>	<input type="text" value="status"/>	<input type="text" value=""/>	<input type="text" value=""/>	
<input type="text" value="Number"/>	<input type="text" value="number_office"/>	<input type="text" value=""/>	<input type="text" value="number"/>	
<input type="text" value="Name"/>	<input type="text" value="name"/>	<input type="text" value=""/>	<input type="text" value="name"/>	
<input type="text" value="Source"/>	<input type="text" value=""/>	<input type="text" value=""/>	<input type="text" value="source"/>	
<input type="text" value="Number"/>	<input type="text" value="number_mobile"/>	<input type="text" value=""/>	<input type="text" value="mobile"/>	

Description

You need to restart the Dird server to apply changes.

Save

The following fields must be configured with the correct value for the *Field type* column in order for entries to be displayed in the xlet:

1. *status* is the column that will be used to display the status icon, the title can be empty
2. *name* is displayed in the *Name* column of the xlet
3. *number_office* is displayed in the *Number* column with a phone icon in the xlet
4. *number_mobile* is displayed in the *Number* column with a mobile icon in the xlet
5. *number_...* any other field starting with *number_* will be displayed in the *Number* column of the xlet with a generic directory icon
6. Any other field will be displayed in their own column of the directory xlet

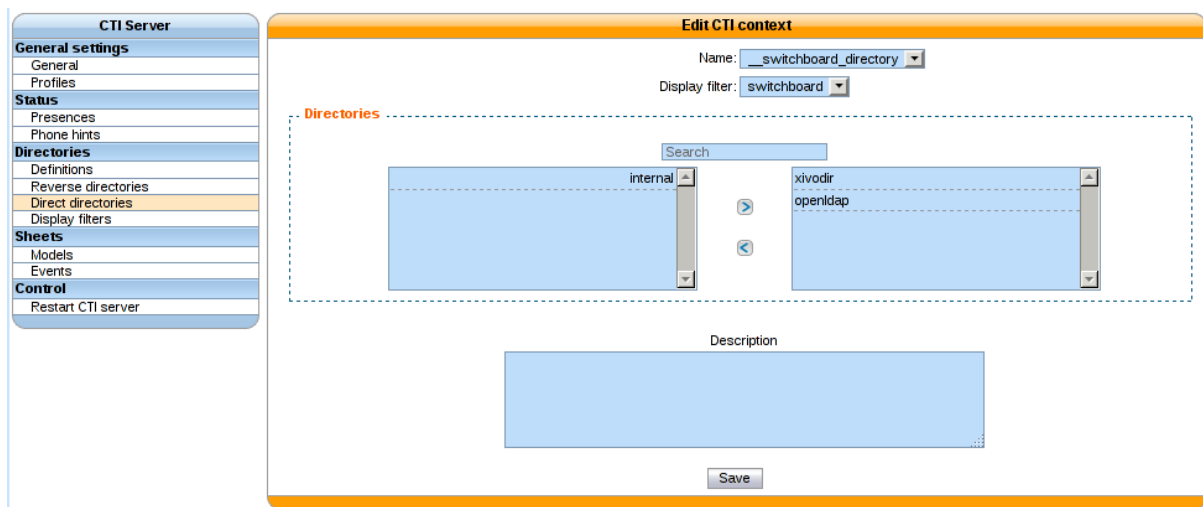
The values in the *Field name* column must contain values that were created in the *Directory definition*.

The title used for the *Number* column is the title of the first field whose type starts with *number_*.

Note: The field title of the first number column will be used for the header title in the xlet.

Warning: Make sure that the fields entered in the display format are also available in the directory definition, otherwise the filter will not return any results

Context and filter association The new *Display filter* has to be assigned to the `__switchboard_directory` context



You can then choose which directories will be searched by the Xlet.

Warning: You must **not select internal** directory, as it is already handled.

LDAP Configuration To search in ldap directories, you must have an LDAP server configured. See [LDAP](#) for more details.

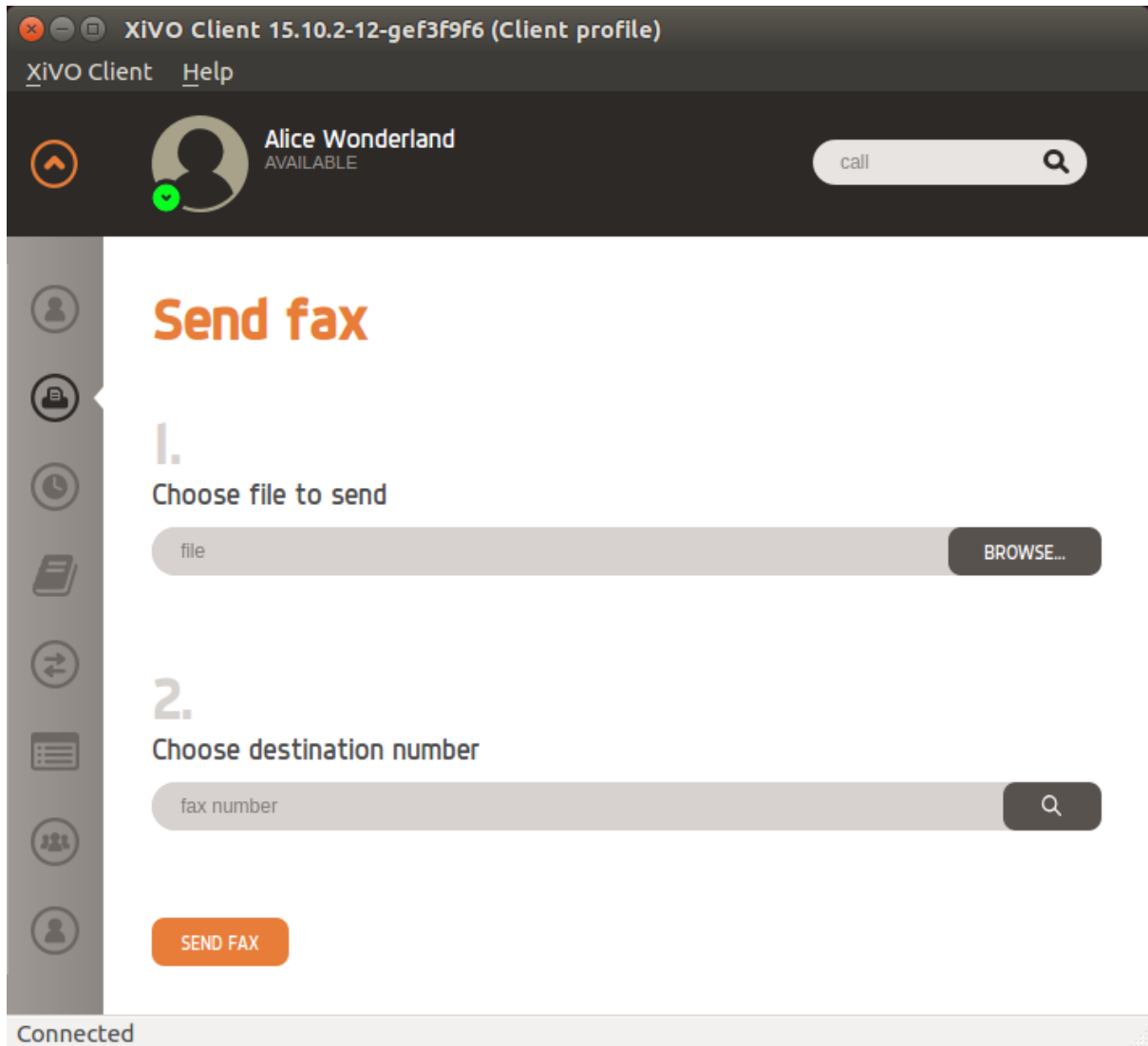
LDAP filter If you already have an LDAP filter configured for the *Remote directory* Xlet, you can use it.

If not, please refer to [Add a LDAP Filter](#).

Include the new directory for lookup You must use the new LDAP filter in the *Context and filter association* step.

Fax Xlet

Overview The Fax xlet allows the user to send faxes from his XiVO client.



Usage The *Choose a file to send* field is used to select which file you want to send. Only PDF documents are supported.

The *Choose destination number* field is the fax destination, directory search can be used to find the fax number in available directories.

History Xlet

Overview The history xlet allow the user to view his last calls. The user can filter by sent, received and missed calls.

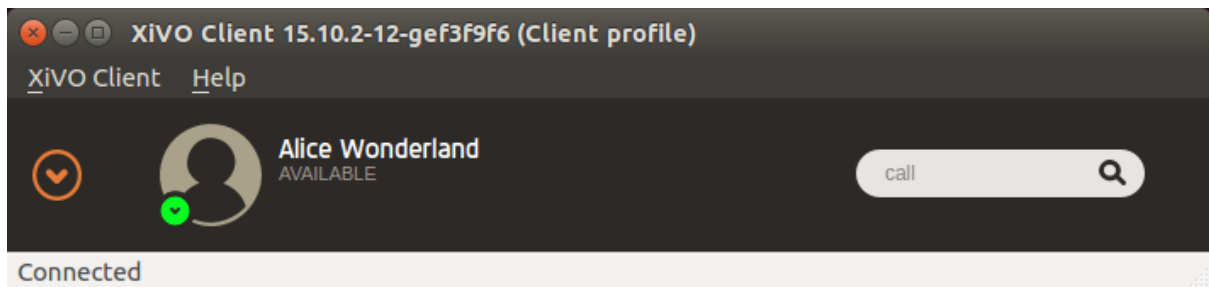
Usage The user can click on the number to initiate a new call with a given correspondent.

Warning:

- The column content is only refreshed when moving from one view to the other.
- The *Sent calls* tab displays only the phone number of the called party (the name column will be void).

Identity Xlet

Overview The Identity Xlet allows you to make calls from your computer, via your phone. This means that you can enter the number that you want to dial on your computer, then your phone rings and when you answer it, the called phone will ring.



Usage You can enter the number you want to dial in the text box and then click the button or press enter to dial it.

If you dial an invalid extension (a number is an extension), your phone will ring and you will be told that the extension is not valid.

People Xlet

Overview The People Xlet lists the people of your company and personal contacts, giving you access to their phone, status and other information configured by the administrator.

1. Display results of the search
2. Display favorite contacts
3. Search contacts
4. Call a contact
5. Transfer a call to a contact
6. Transfer a call to a contact's voicemail
7. Chat with a contact
8. Send an email to a contact
9. Bookmark/unmark the contact as a favorite
1. View all personal contacts
2. Edit or remove a personal contact
3. Create a personal contact
4. Import personal contacts from a CSV file
5. Export personal contacts to a CSV file

1

2

ALL

FAVORITES

MY CONTACTS

search

3

NAME	NUMBER	FAVORITE
Alice Antonia	1001	★ 9
Bob Barker	4	★
Esteban	<div> <div>CALL</div> <div> <div>EMAIL - bob@example.com 8</div> <div>MOBILE - 5555557676</div> <div>DID - 5555551234</div> <div>BLIND TRANSFER</div> <div>ATTENDED TRANSFER 5</div> <div>Send a message 7</div> </div> </div>	★
Zia		★

NUMBER - 1002

MOBILE - 5555557676

DID - 5555551234

VOICEMAIL - 1002 6

1

ALL

FAVORITES

MY CONTACTS

search

NAME	NUMBER	FAVORITE	PERSONAL
Esteban	4185555479	★	✎ 🗑 2
Sancho	5145555555	☆	✎ 🗑
Tao		☆	✎ 🗑
Zia	<div> <div>CALL</div> <div>▼</div> </div>	★	✎ 🗑

3

4

5

6

NEW CONTACT

IMPORT

EXPORT

DELETE ALL CONTACTS

60

Chapter 1. Table of Contents

1 ● Gaspard Gomez	2 ● 1005	3 ☹️	★
● Mendoza Spa	● 1006	😊	★
● Pichu T	● 1002		★

6. Delete all personal contacts

1. XiVO Client status (see [Presence Option](#))
2. Phone status (see [Services](#) → [CTI Server](#) → [Status](#) → [Phone hints](#) page)
3. Agent status (logged in or logged out)

Note: Most information (e.g. columns displayed, allowed actions, searched directories, etc.) is configurable through the [web interface](#).

Importing contacts via CSV file Imported files should have the following structure:

```
firstname,lastname,number,email,company,fax,mobile
John,Doe,5555551111,my@email,xivo,5555552222,5555553333
```

- The field order is not important.
- The file must be encoded in UTF-8.
- Invalid lines of the CSV file will be skipped and an error will be displayed in the import report.

Exporting contacts via CSV file The file has the same structure as the import file, with a supplementary field: `id`, which is the internal contact ID from XiVO.

- The first line (the list of field names) is ordered in alphabetical order.
- The file will be encoded in UTF-8.

Copying the number or email address It is possible to copy a contact's number or email address to the system's clipboard. To do so, right click on a contact's action menu and select the value you wish to copy.

Note: When using a mac without a right mouse button use *ctrl-Left click* to show the copy menu.

Service Xlet

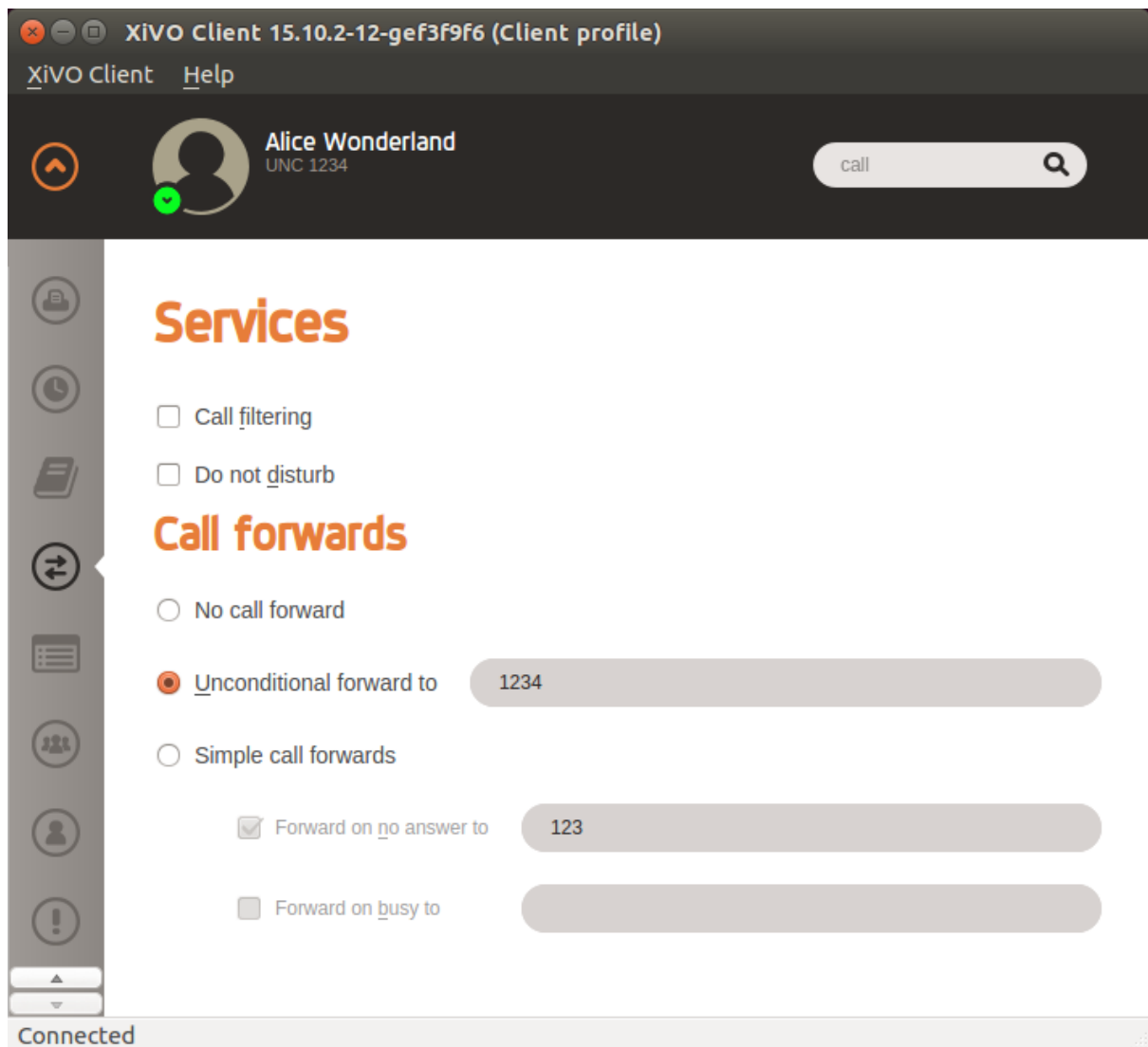
Overview The service xlet allows the user to enable and disable telephony services such as call forwarding, call filter and do not disturb.

Configuration The available service list is configured from the web interface in [Services](#) → [CTI Server](#) → [General settings](#) → [Profiles](#).

The right side of the [Services](#) section contains services that are available to a given profile.

Configuration

The XiVO Client configuration options can be accessed under [XiVO Client](#) → [Configure](#).



Edit CTI profile

General Xlets Preferences

Name:

Display name:
A more understandable name

Max. GUI:
Put -1 for no limit

Status

Presence:

Phonehints:

Agent:

Services

Search

Enable voicemail

Call filter

Enable DND

Unconditional transfer to a number

Transfer on busy

Save

Connection Configuration This page allows the user to set his network information to connect to the xivo-ctid server.

- *Server* is the IP address of the server.
- *Backup server* is the IP address of the backup server.
- *Port* is the port on which xivo-ctid is listening for connections. (default: 5003)
- *STARTTLS* is used to specify that a secure connect should be used

Note: To use STARTTLS, the server needs to be configured to *accept encrypted connection*.

Handling callto: and tel: URLs

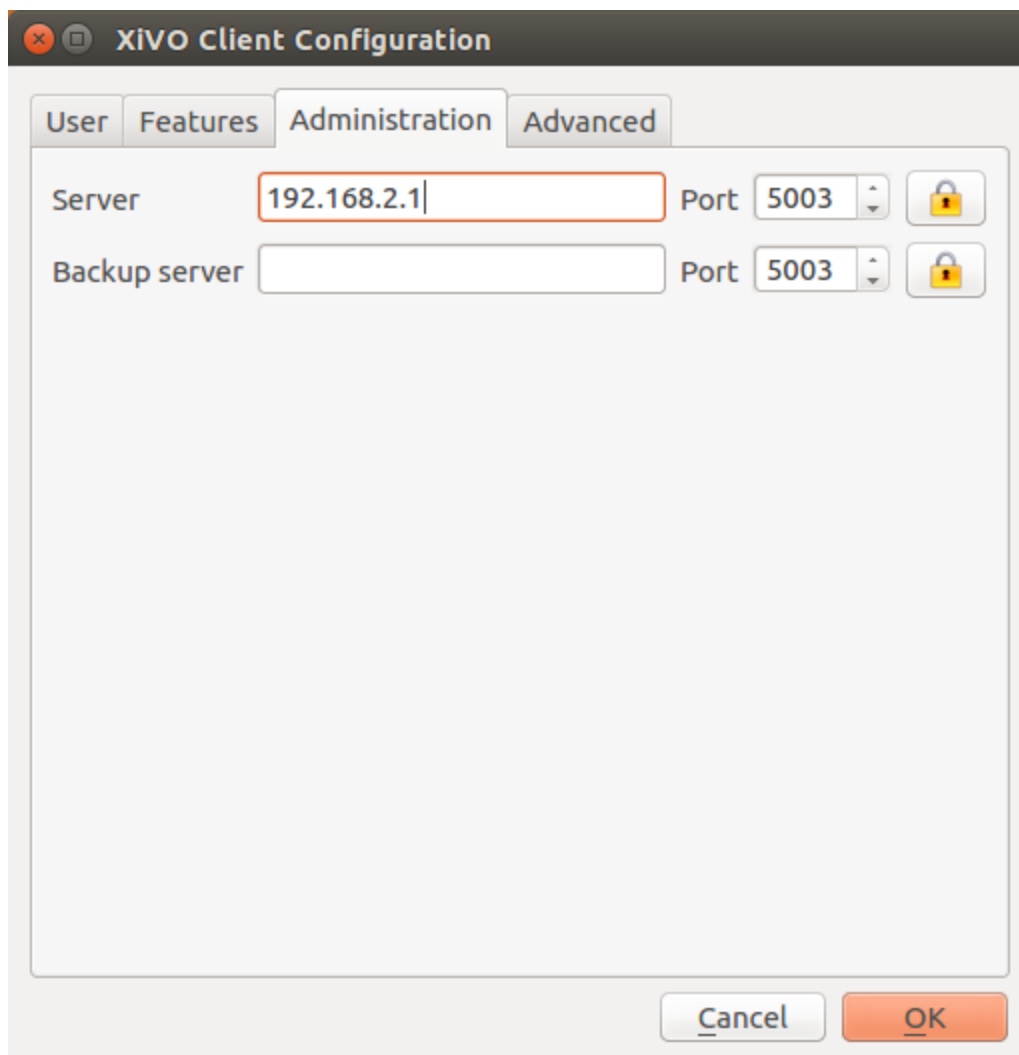
The XiVO Client can handle telephone number links that appear in web pages. The client will automatically dial the number when you click on a link.

Note: You must already be logged in for automatic dialing to work, otherwise the client will simply start up and wait for you to log in.

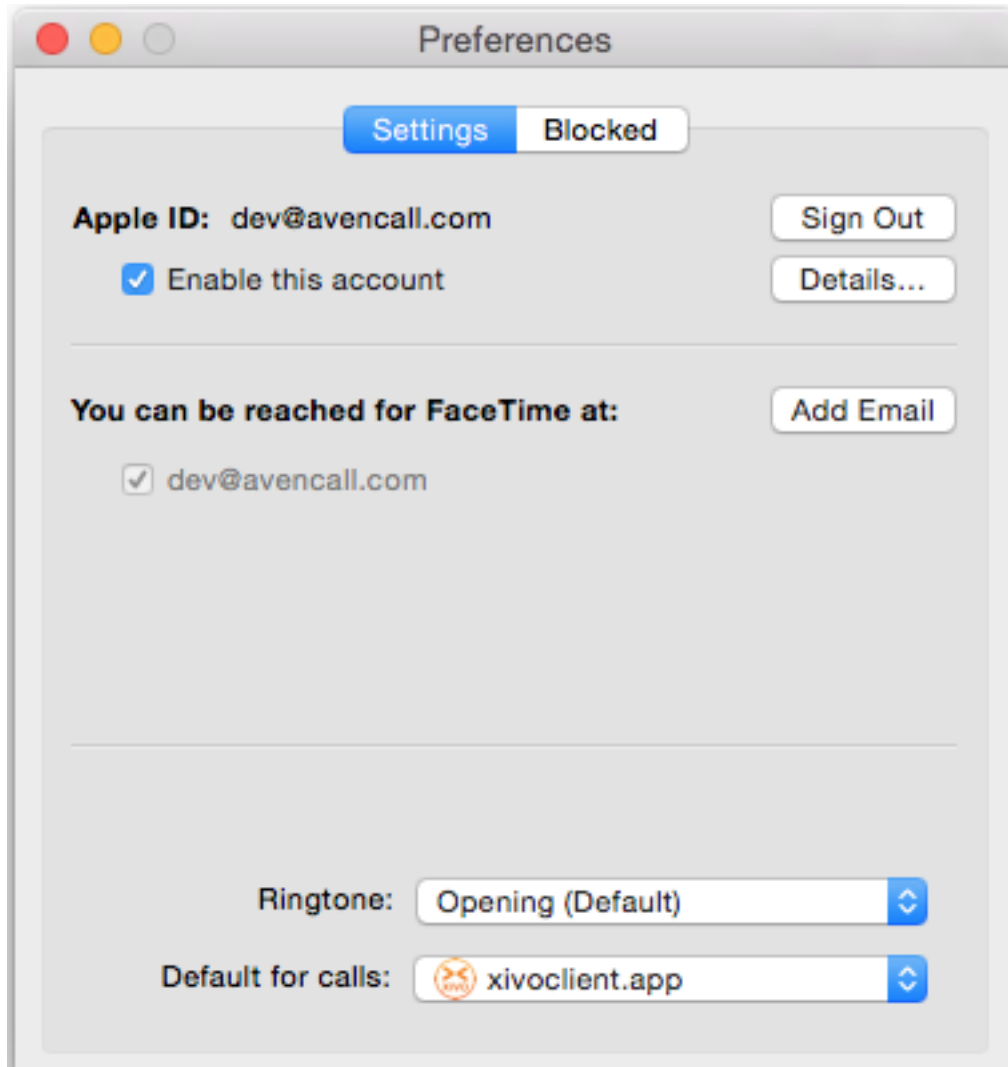
Warning: The option in the XiVO Client *GUI Options* → *Allow multiple instances of XiVO Client* must be disabled, else you will launch one new XiVO Client with every click.

Mac OS `callto:` links will work out-of-the-box in Safari and other web browsers after installing the client.

`tel:` links will open FaceTime after installing the client. To make the XiVO Client the default application to open `tel:` URLs in Safari.



1. Open the FaceTime application
2. Connect using your apple account
3. Open the FaceTime preferences
4. Change the *Default for calls* entry to *xivoclient.app*



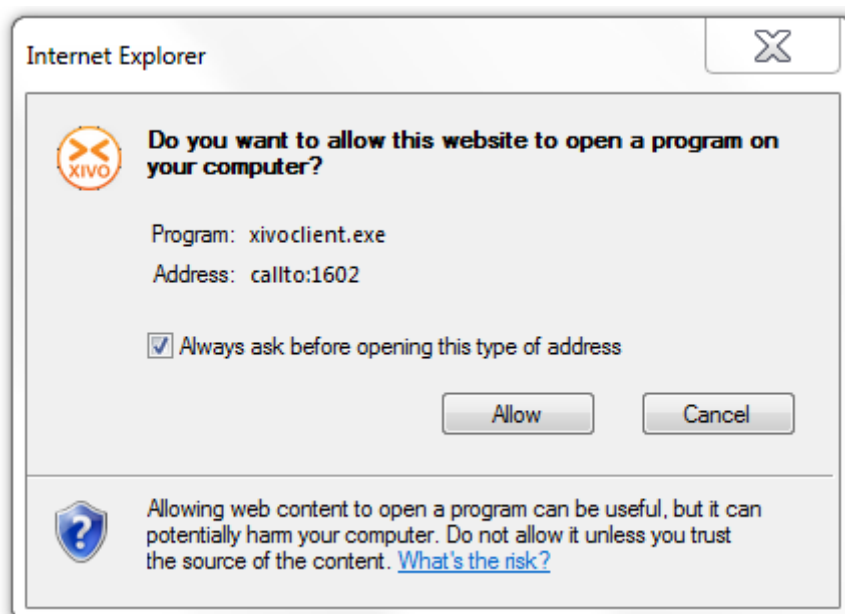
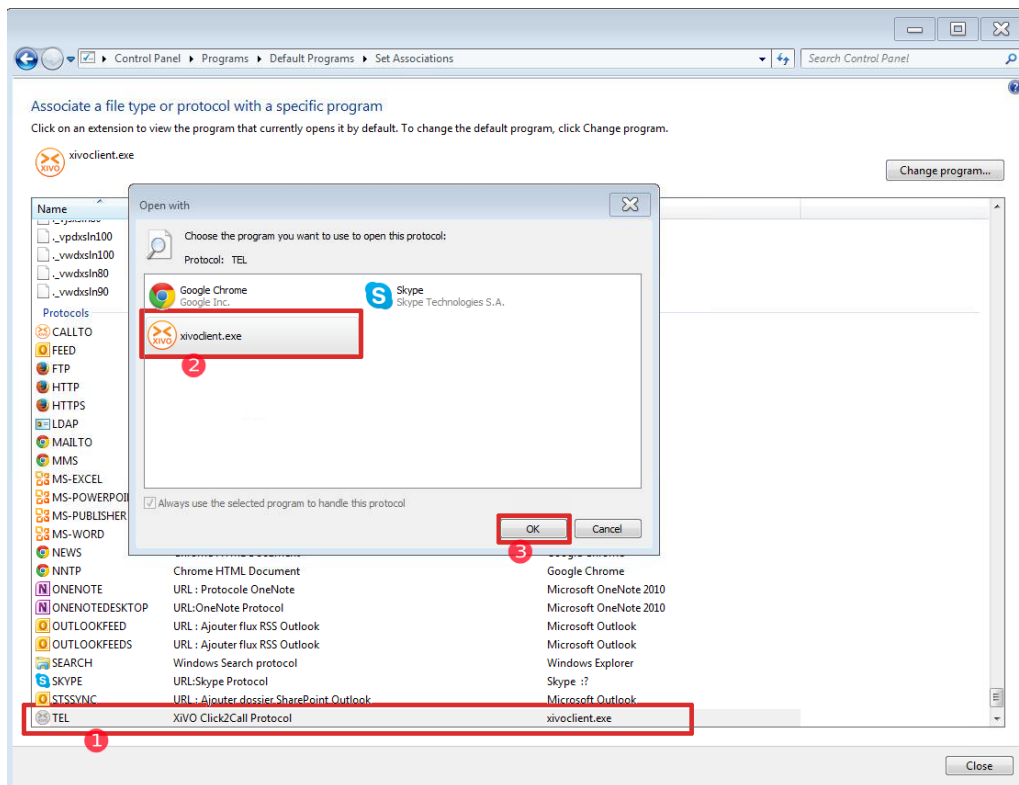
Note: The `tel:` URL works out-of-the-box in versions of mac osx before 10.10.

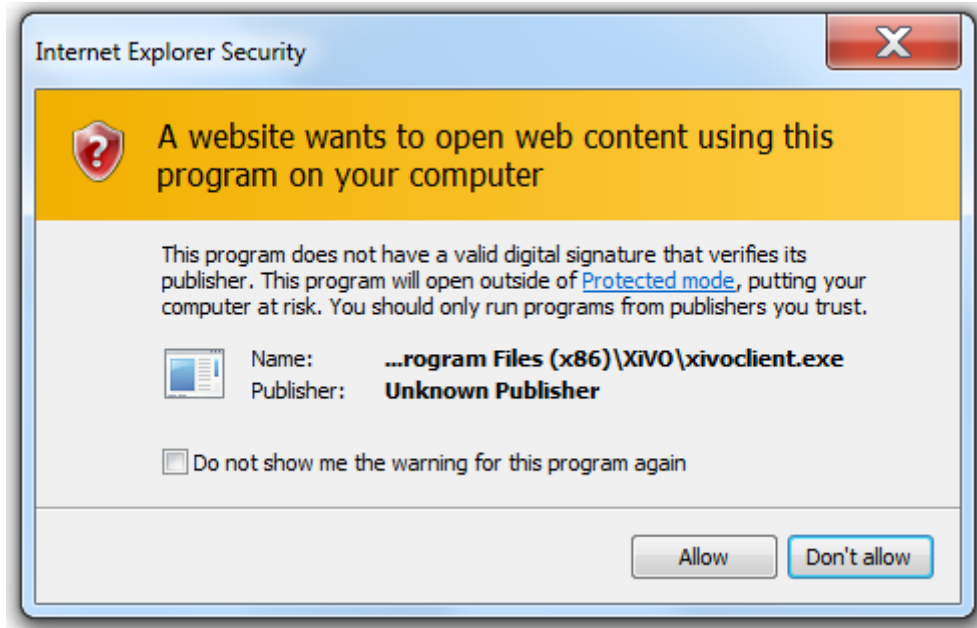
Windows XiVO Client is associated with `callto:` and `tel:` upon installation. Installing other applications afterward could end up overriding these associations. Starting with Windows Vista, it is possible to configure these associations via the Default Programs. Users can access Default Programs from Control Panel or directly from the Start menu.

The following popups might appear when you open a `callto:` or `tel:` link for the first time in Internet Explorer:

Simply click on *allow* to dial the number using the XiVO Client.

Note: If you do not want these warnings to appear each time, do not forget to check/uncheck the checkbox at the bottom of the popups.





Ubuntu Currently, `callto:` or `tel:` links are only supported in Firefox. There is no configuration needed.

GNU/Linux Debian Currently, `callto:` or `tel:` links are only supported in Firefox. If the XiVO Client is not listed in the proposition when you open the link, browse your files to find `/usr/bin/xivoclient`.

Manual association in Firefox If, for some reason, Firefox does not recognize `callto:` or `tel:` URIs you can manually associate them to the XiVO Client using the following steps:

1. Type `about:config` in the URL bar
2. Click the *I'll be careful, I promise !* button to close the warning
3. Right-click anywhere in the list and select *New -> Boolean*
4. Enter `network.protocol-handler.external.callto` as preference name
5. Select `false` as value
6. Repeat steps 3 to 6, but replace `callto` by `tel` at step 4

The next time that you click on a telephone link, Firefox will ask you to choose an application. You will then be able to choose the XiVO client for handling telephone numbers.

System

DHCP Server

XiVO includes a DHCP server used for assisting in the provisioning of phones and other devices. (See [Basic Configuration](#) for the basic setup). This section contains additional notes on how to configure more advanced options that may be helpful when integrating the server with different VOIP subnets.

Activating DHCP on another interface DHCP Server can be activated through the XiVO Web Interface *Configuration* → *Network* → *DHCP* :

By default, it will only answer to DHCP requests coming from the VoIP subnet (defined in the *Configuration* → *Network* → *Interfaces* section). If you need to activate DHCP on an other interface, you have to fill in the *Extra network interfaces* field with the interface name , for example : `eth0`

Fig. 1.18: *Configuration → Network → DHCP*

After saving your modifications, click on *Apply system configuration* so that the new settings can take effect.

Changing default DHCP gateway By default, the XiVO DHCP server uses the XiVO's IP address as the routing address. To change this you must create a custom-template:

1. Create a custom template for the `dhcpd_subnet.conf.head` file:

```
mkdir -p /etc/xivo/custom-templates/dhcp/etc/dhcp/
cd /etc/xivo/custom-templates/dhcp/etc/dhcp/
cp /usr/share/xivo-config/templates/dhcp/etc/dhcp/dhcpd_subnet.conf.head .
```

2. Edit the custom template:

```
vim dhcpd_subnet.conf.head
```

3. In the file, replace the string `#XIVO_NET4_IP#` by the routing address of your VoIP network, for example:

```
option routers 192.168.2.254;
```

4. Re-generate the dhcp configuration:

```
xivo-update-config
```

DHCP server should have been restarted and should now use the new routing address.

Configuring DHCP server to serve unknown hosts By default, the XiVO DHCP server serves only known hosts. That is:

- either hosts which MAC address prefix (the **OUI**) is known
- or hosts which Vendor Identifier is known

Known OUIs and Vendor Class Identifiers are declared in `/etc/dhcp/dhcpd_update/*` files.

If you want your XiVO DHCP server to serve also unknown hosts (like PCs) follow these instructions:

1. Create a custom template for the `dhcpd_subnet.conf.tail` file:

```
mkdir -p /etc/xivo/custom-templates/dhcp/etc/dhcp/
cd /etc/xivo/custom-templates/dhcp/etc/dhcp/
cp /usr/share/xivo-config/templates/dhcp/etc/dhcp/dhcpd_subnet.conf.tail .
```

2. Edit the custom template:

```
vim dhcpd_subnet.conf.tail
```

3. And add the following line at the head of the file:

```
allow unknown-clients;
```

4. Re-generate the dhcp configuration:

```
xivo-update-config
```

DHCP server should have been restarted and should now serve all network equipments.

DHCP-Relay If your telephony devices aren't located on the same site and the same broadcast domain as the XiVO DHCP server, you will have to add the option *DHCP Relay* to the site's router. This parameter will allow the DHCP requests from distant devices to be transmitted to the IP address you specify as DHCP Relay.

Warning: Please make sure that the IP address used as DHCP Relay is the same as one of XiVO's interfaces, and that this interface is configured to listen to DHCP requests (as described in previous part). Also verify that routing is configured between the distant router and the chosen interface, otherwise DHCP requests will never reach the XiVO server.

Configuring DHCP server for other subnets This section describes how to configure XiVO to serve other subnets than the VOIP subnet. As you can't use the Web Interface to declare other subnets (for example to address DATA subnet, or a VOIP subnet that isn't on the same site than XiVO server), you'll have to do the following configuration on the Command Line Interface.

Creating "extra subnet" configuration files First thing to do is to create a directory and to copy into it the configuration files:

```
mkdir /etc/dhcp/dhcpd_sites/
cp /etc/dhcp/dhcpd_subnet.conf /etc/dhcp/dhcpd_sites/dhcpd_siteXXX.conf
cp /etc/dhcp/dhcpd_subnet.conf /etc/dhcp/dhcpd_sites/dhcpd_lanDATA.conf
```

Note: In this case we'll create 2 files for 2 different subnets. You can change the name of the files, and create as many files as you want in the folder `/etc/dhcp/dhcpd_sites/`. Just adapt this procedure by changing the name of the file in the different links.

After creating one or several files in `/etc/dhcp/dhcpd_sites/`, you have to edit the file `/etc/dhcp/dhcpd_extra.conf` and add the according include statement like:

```
include "/etc/dhcp/dhcpd_sites/dhcpd_siteXXX.conf";
include "/etc/dhcp/dhcpd_sites/dhcpd_lanDATA.conf";
```

Adjusting Options of the DHCP server Once you have created the subnet in the DHCP server, you must edit each configuration file (the files in `/etc/dhcp/dhcpd_sites/`) and modify the different parameters. In section **subnet**, write the IP subnet and change the following options (underlined fields in the example):

```
subnet 172.30.8.0 netmask 255.255.255.0 {
```

- subnet-mask:

```
option subnet-mask 255.255.255.0;
```

- broadcast-address:

```
option broadcast-address 172.30.8.255;
```

- routers (specify the IP address of the router that will be the default gateway of the site):

```
option routers 172.30.8.1;
```

In section **pool**, modify the options:

```
pool {
```

- log (add the name of the site or of the subnet):

```
log(concat("[", binary-to-ascii(16, 8, ":", hardware), "] POOL VoIP Site XXX"));
```

- range (it will define the range of IP address the DHCP server can use to address the devices of that subnet):

```
range 172.30.8.10 172.30.8.200;
```

Warning: XiVO only answers to DHCP requests from *supported devices*. In case of you need to address other equipment, use the option *allow unknown-clients*; in the `/etc/dhcp/dhcpd_sites/` files

At this point, you can apply the changes of the DHCP server with the command:

```
service isc-dhcp-server restart
```

After that, XiVO will start to serve the DHCP requests of the devices located on other sites or other subnets than the VOIP subnet. You will see in `/var/log/daemon.log` all the DHCP requests received and how they are handled by XiVO.

Mail

This section describes how to configure the mail server shipped with XiVO (Postfix) and the way XiVO handles mails.

In *Configuration* → *Network* → *Mail*, the following options can be configured:

- *Domain Name messaging* : the server's displayed domain. Will appear in "Received" mail headers.
- *Source address of the server* : domain part of headers "Return-Path" and "From".
- *Relay SMTP* and *FallBack relay SMTP* : relay mail servers.
- *Rewriting shipping addresses* : Canonical address Rewriting. See [Postfix canonical documentation](#) for more info.

Warning: Postfix, the mail server shipped with XiVO, should be stopped on an installed XiVO with no valid and reachable DNS servers configured. If Postfix is not stopped, messages will bounce in queues and could end up affecting core pbx features.

If you need to disable Postfix here is how you should do it:

```
systemctl stop postfix
systemctl disable postfix
```

If you ever need to enable Postfix again:

```
systemctl enable postfix
systemctl start postfix
```

Alternatively, you can empty Postfix's queues by issuing the following commands on the XiVO server:

```
postsuper -d ALL
```

Network

This section describes how to configure additional network devices that may be used to better accomodate more complex network infrastructures. Network interfaces are managed in the XiVO web interface via the page *Configuration* → *Network* → *Interfaces*.

XiVO offers 2 types of interfaces: *VoIP* and *Data*. The *VoIP* interface is used by the DHCP server, provisioning server, and phone devices connected to your XiVO. These services will use the information provided on the *VoIP* interface for their configuration. For example, the DHCP server will only listen on the *VoIP* interface by default.

To change these settings, you must either create a new interface or edit an existing one and change its type. When adding a new *VoIP* interface, the type of the old one will automatically be changed to *Data*.

Configuring a physical interface In this example, we'll add and configure the **eth1** network interface on our XiVO.

First, we see there's already an unconfigured network interface named **eth1** on our system:

Interface	Mac address	Type	Method	Address	Gateway	VLAN ID	Action
<input type="checkbox"/> ➔ eth0	08:00:27:ea:b0:75	VoIP	Static	192.168.32.137	192.168.32.254	-	
<input type="checkbox"/> ➔ eth1	08:00:27:29:b7:e9	Data	-	-	-	-	

Legend
 ➔ Enable ➔ Disable ➔ Not apply

Fig. 1.19: Configuration → Network → Interfaces

To add and configure it, we click on the small plus button next to it, and we get to this page:

Interfaces > Add

Interface:

Type:

Method:

Address:

Netmask:

Default gateway:

Description:

Fig. 1.20: Configuration → Network → Interfaces → eth1 → Add

In our case, since we want to configure this interface with static information (i.e. not via DHCP), we fill the following fields:

Note that since our **eth0** network interface already has a default gateway, we do not enter information in the Default gateway field for our **eth1** interface.

Once the changes have been saved, the action **Apply network configuration** will appear in bold. This action must be clicked in order for the changes to take effect.


Adding a VLAN interface In this example, the XiVO already has 2 network interfaces configured:

Listing the network interfaces

To add and configure a new VLAN interface, we click on the small plus button in the top right corner,

Interfaces > Add

Interface:

Type: Data 

Method: Static



Address:

Netmask:

Default gateway:


Description:

Fig. 1.21: Configuration → Network → Interfaces → eth1 → Add

Web Services Access
Certificates
Network
Interfaces
Resolver
Mail
DHCP
Support
XiVO
Alerts
Limits
Provisioning
General
Template line
Template device
Plugin
Control System
 Apply network configuration
 Apply system configuration

Legend

➤ Enable
 ➤ Disable
 ➤ Not apply






Fig. 1.22: Apply after modify interface

Interface	Mac address	Type	Method	Address	Gateway	Action
<input type="checkbox"/> > eth0	08:00:27:6a:49:e5	Data	Static	192.168.32.51	192.168.32.254	
<input type="checkbox"/> > eth1	08:00:27:e9:fa:f4	VoIP	Static	10.97.5.2	-	

Legend

> Enable > Disable > Not apply

Fig. 1.23: Configuration → Network → Interfaces



Fig. 1.24: Configuration → Network → Interfaces → Add button

Interfaces > Add

Physical Interface of VLAN : eth0

ID of VLAN :

Type: Data

Method: Static

Address:

Netmask:

Default gateway:

Description:

Save

Fig. 1.25: Configuration → Network → Interfaces → Add

and we get to this page:

In our case, since we want to configure this interface with static information:

Interfaces > Add

Physical Interface of VLAN :

ID of VLAN :

Type:

Method:

Address:

Netmask:

Default gateway:

Description:

Fig. 1.26: *Configuration → Network → Interfaces → Add*

Click on **Save** list the network interfaces:

Interface	Mac address	Type	Method	Address	Gateway	Action
<input type="checkbox"/> eth0	08:00:27:6a:49:e5	Data	Static	192.168.32.51	192.168.32.254	
<input type="checkbox"/> eth0.101	-	Data	Static	10.97.6.2	-	
<input type="checkbox"/> eth1	08:00:27:e9:fa:f4	VoIP	Static	10.97.5.2	-	

Legend

Enable Disable Not apply

Fig. 1.27: *Configuration → Network → Interfaces*

- The new virtual interface has been successfully created.

Note: Do not forget after you finish the configuration of the network to apply it with the button: **Apply network configuration**

After applying the network configuration:

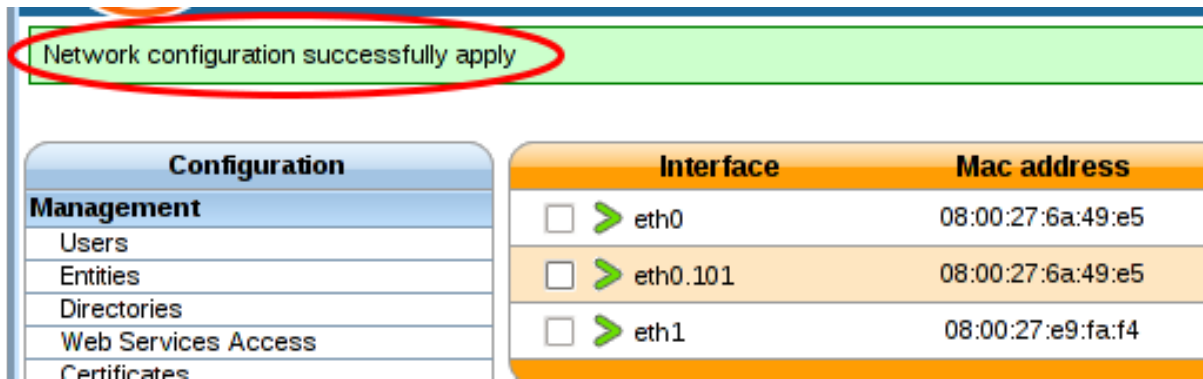


Fig. 1.28: Network configuration successfully apply

Add static network routes Static routes cannot be added via the web interface. However, you may add static routes to your XiVO by following the steps described below. This procedure will ensure that your static routes are applied at startup (i.e. each time the network interface goes up).

1. Create the file `/etc/network/if-up.d/xivo-routes`:

```
touch /etc/network/if-up.d/xivo-routes
chmod 755 /etc/network/if-up.d/xivo-routes
```

2. Insert the following content:

```
#!/bin/sh

if [ "${IFACE}" = "<network interface>" ]; then
    ip route add <destination> via <gateway>
    ip route add <destination> via <gateway>
fi
```

3. Fields `<network interface>`, `<destination>` and `<gateway>` should be replaced by your specific configuration. For example, if you want to add a route for `192.168.50.128/25` via `192.168.17.254` which should be added when `eth0` goes up:

```
#!/bin/sh

if [ "${IFACE}" = "eth0.2" ]; then
    ip route add 192.168.50.128/25 via 192.168.17.254
fi
```

Note: The above check is to ensure that the route will be applied only if the correct interface goes up. This check should contain the actual name of the interface (i.e. `eth0` or `eth0.2` or `eth1` or ...). Otherwise the route won't be set up in every cases.

Change interface MTU

Warning: Manually changing the MTU is risky. Please only proceed if you are aware of what you are

These steps describe how to change the MTU:

```
#. Create the file :file:`/etc/network/if-up.d/xivo-mtu`::
```

```
touch /etc/network/if-up.d/xivo-mtu chmod 755 /etc/network/if-up.d/xivo-mtu
```

1. Insert the following content:

```
#!/bin/sh
```

```
# Set MTU per iface
if [ "${IFACE}" = "<data interface>" ]; then
    ip link set ${IFACE} mtu <data mtu>
elif [ "${IFACE}" = "<voip interface>" ]; then
    ip link set ${IFACE} mtu <voip mtu>
fi
```

2. Change the *<data interface>* to the name of your interface (e.g. eth0), and the *<data mtu>* to the new MTU (e.g. 1492),
3. Change the *<voip interface>* to the name of your interface (e.g. eth1), and the *<voip mtu>* to the new MTU (e.g. 1488)

Note: In the above example you can set a different MTU per interface. If you don't need a per-interface MTU you can simply write:

```
#!/bin/sh

ip link set ${IFACE} mtu <my mtu>
```

Backup

Periodic backup A backup of the database and the data are launched every day with a logrotate task. It is run at 06:25 a.m. and backups are kept for 7 days.

Logrotate task:

```
/etc/logrotate.d/xivo-backup
```

Logrotate cron:

```
/etc/cron.daily/logrotate
```

Retrieve the backup You can retrieve the backup from the web-interface in *Services* → *IPBX* → *IPBX Configuration* → *Backup Files* page.

Otherwise, with shell access, you can retrieve them in `/var/backups/xivo`. In this directory you will find `db.tgz` and `data.tgz` files for the database and data backups.

Backup scripts:

```
/usr/sbin/xivo-backup
```

Backup location:

```
/var/backups/xivo
```

What is actually backed-up?

Data Here is the list of folders and files that are backed-up:

- `/etc/asterisk/`
- `/etc/consul/`
- `/etc/crontab`
- `/etc/dahdi/`
- `/etc/dhcp/`
- `/etc/hostname`

- /etc/hosts
- /etc/ldap/
- /etc/network/if-up.d/xivo-routes
- /etc/network/interfaces
- /etc/ntp.conf
- /etc/profile.d/xivo_uuid.sh
- /etc/resolv.conf
- /etc/ssl/
- /etc/systemd/
- /etc/wanpipe/
- /etc/xivo-agentd/
- /etc/xivo-agid/
- /etc/xivo-amid/
- /etc/xivo-auth/
- /etc/xivo-call-logd/
- /etc/xivo-confd/
- /etc/xivo-configend-client/
- /etc/xivo-ctid/
- /etc/xivo-ctid-ng/
- /etc/xivo-dird/
- /etc/xivo-dird-phoned/
- /etc/xivo-dxtora/
- /etc/xivo-purge-db/
- /etc/xivo-websocketd/
- /etc/xivo/
- /usr/local/bin/
- /usr/local/sbin/
- /usr/share/xivo/XIVO-VERSION
- /var/lib/asterisk/
- /var/lib/consul/
- /var/lib/xivo-provd/
- /var/lib/xivo/
- /var/log/asterisk/
- /var/spool/asterisk/
- /var/spool/cron/crontabs/

The following files/folders are excluded from this backup:

- folders:
 - /var/lib/consul/checks
 - /var/lib/consul/raft

- /var/lib/consul/serf
- /var/lib/consul/services
- /var/lib/xivo-provd/plugins/*/var/cache/*
- /var/spool/asterisk/monitor/
- /var/spool/asterisk/meetme/
- files
 - /var/lib/xivo-provd/plugins/xivo-polycom*/var/tftpboot/*.ld
- log files, coredump files
- audio recordings
- and, files greater than 10 MiB or folders containing more than 100 files if they belong to one of these folders:
 - /var/lib/xivo/sounds/
 - /var/lib/asterisk/sounds/custom/
 - /var/lib/asterisk/moh/
 - /var/spool/asterisk/voicemail/
 - /var/spool/asterisk/monitor/

Database The database asterisk from PostgreSQL is backed up. This include almost everything that is configured via the web interface.

Creating backup files manually

Warning: A backup file may take a lot of space on the disk. You should check the free space before creating one.

Database You can manually create a *database* backup file named `db-manual.tgz` in `/var/tmp` by issuing the following commands:

```
xivo-backup db /var/tmp/db-manual
```

Files You can manually create a *data* backup file named `data-manual.tgz` in `/var/tmp` by issuing the following commands:

```
xivo-backup data /var/tmp/data-manual
```

Restore

Introduction A backup of both the configuration files and the database used by a XiVO installation is done automatically every day. These backups are created in the `/var/backups/xivo` directory and are kept for 7 days.

Limitations

- You must restore a backup on the **same version** of XiVO that was backed up (though the architecture – i386 or amd64 – may differ)
- You must restore a backup on a machine with the **same hostname and IP address**
- Be aware that this procedure applies **only to XiVO >= 14.08** (see [14.08](#)).

Before Restoring the System

Warning: Before restoring a XiVO on a fresh install you have to setup XiVO using the wizard ([the Wizard](#) section).

Stop monit and all the xivo services:

```
xivo-service stop
```

Restoring System Files System files are stored in the data.tgz file located in the /var/backups/xivo directory.

This file contains for example, voicemail files, musics, voice guides, phone sets firmwares, provisioning server configuration database.

To restore the file

```
tar xvfp /var/backups/xivo/data.tgz -C /
```

Restoring the Database

Warning:

- This will destroy all the current data in your database.
- You have to check the free space on your system partition before extracting the backups.

Database backups are created as db.tgz files in the /var/backups/xivo directory. These tarballs contains a dump of the database used in XiVO.

In this example, we'll restore the database from a backup file named db.tgz placed in the home directory of root.

First, extract the content of the db.tgz file into the /var/tmp directory and go inside the newly created directory:

```
tar xvf db.tgz -C /var/tmp
cd /var/tmp/pg-backup
```

Drop the asterisk database and restore it with the one from the backup:

```
sudo -u postgres dropdb asterisk
sudo -u postgres pg_restore -C -d postgres asterisk-*.dump
```

To finalize the restore, see [After Restoring The System](#).

Troubleshooting When restoring the database, if you encounter problems related to the system locale, see [PostgreSQL localization errors](#).

Alternative: Restoring and Keeping System Configuration System configuration like network interfaces is stored in the database. It is possible to keep this configuration and only restore xivo data.

Rename the asterisk database to asterisk_previous:

```
sudo -u postgres psql -c 'ALTER DATABASE asterisk RENAME TO asterisk_previous'
```

Restore the asterisk database from the backup:

```
sudo -u postgres pg_restore -C -d postgres asterisk-*.dump
```

Restore the system configuration tables from the asterisk_previous database:

```
sudo -u postgres pg_dump -c -t dhcp -t netiface -t resolvconf asterisk_previous | sudo -u postgres
```

Drop the asterisk_previous database:

```
sudo -u postgres dropdb asterisk_previous
```

Warning: Restoring the data.tgz file also restores system files such as host hostname, network interfaces, etc. You will need to reapply the network configuration if you restore the data.tgz file.

After Restoring The System Resynchronize the xivo-auth keys:

```
xivo-update-keys
```

Update systemd runtime configuration:

```
source /etc/profile.d/xivo_uuid.sh
systemctl set-environment XIVO_UUID=$XIVO_UUID
systemctl daemon-reload
```

Restart the services you stopped in the first step:

```
xivo-service start
```

You may also reboot the system.

HTTPS certificate

X.509 certificates are used to authorize and secure communications with the server. They are mainly used for HTTPS, but can also be used for SIPS, CTIS, WSS, etc.

There are two categories of certificates in XiVO:

- the default certificate, used for HTTPS in the web interface, REST APIs and WebSockets
- the certificates created and managed via the web interface

This article is about the former. For the latter, see *Telephony certificates*.

Default certificate XiVO uses HTTPS where possible. The certificates are generated at install time (or during the *upgrade to 15.12+*). The main certificate is placed in `/usr/share/xivo-certs/server.crt`.

However, this certificate is self-signed, and HTTP clients (browser or REST API client) will complain about this default certificate because it is not signed by a trusted Certification Authority (CA).

The default certificate is untrusted To make the HTTP client accept this certificate, you have two choices:

- configure your HTTP client to trust the self-signed XiVO certificate by adding a new trusted CA. The CA certificate (or bundle) is the file `/usr/share/xivo-certs/server.crt`.
- replace the self-signed certificate with your own trusted certificate.

Use your own certificate For this, follow these steps:

1. Replace the following files with your own private key/certificate pair:
 - Private key: `/usr/share/xivo-certs/server.key`
 - Certificate: `/usr/share/xivo-certs/server.crt`
2. Change the hostname of XiVO for each XiVO component: the different processes of XiVO heavily use HTTPS for internal communication, and for these connection to establish successfully, all hostnames used must match the Common Name (CN) of your certificate. Basically, you must replace all occurrences of `localhost` (the default hostname) with your CN in the *configuration of the XiVO services*. For example:


```

mkdir /etc/xivo/custom
cat > /etc/xivo/custom/custom-certificate.yml << EOF
consul:
  host: xivo.example.com
auth:
  host: xivo.example.com
confd:
  host: xivo.example.com
dird:
  host: xivo.example.com
ajam:
  host: xivo.example.com
agentd:
  host: xivo.example.com
EOF
for config_dir in /etc/xivo-*/conf.d/ ; do
  ln -s "/etc/xivo/custom/custom-certificate.yml" "$config_dir/010-custom-certificate.yml"
done

```

Also, you must replace `localhost` in the definition of your directories in the web interface under *Configuration* → *Directories*.

3. If your certificate is not self-signed, and you obtained it from a third-party CA that is trusted by your system, you must enable the system-based certificate verification. By default, certificate verification is set to consider `/usr/share/xivo-certs/server.crt` as the only CA certificate.

The options are the following:

- Consul: `verify: True`
- Other XiVO services: `verify_certificate: True`

The procedure is the same as 2. with more configuration for each service. For example:

```

cat > /etc/xivo/custom/custom-certificate.yml << EOF
consul:
  host: xivo.example.com
  verify: True
auth:
  host: xivo.example.com
  verify_certificate: True
dird:
  host: xivo.example.com
  verify_certificate: True
...

```

Setting `verify_certificate` to `False` will disable the certificate verification, but the connection will still be encrypted. This is pretty safe as long as XiVO services stay on the same machine, however, this is dangerous when XiVO services are separated by an untrusted network, such as the Internet.

4. Ensure your CN resolves to a valid IP address with either:
 - a DNS entry
 - an entry in `/etc/hosts` resolving your CN to `127.0.0.1`. Note that `/etc/hosts` will be rewritten occasionally by `xivo-sysconfd`. To make the change persistent, you can:
 - (a) modify `/usr/share/xivo-sysconfd/templates/resolvconf/hosts` instead (which will be rewritten when `xivo-sysconfd` is upgraded...)
 - (b) then add a script in `/usr/share/xivo-upgrade/pre-start.d` to re-apply the modification to `/usr/share/xivo-sysconfd/templates/resolvconf/hosts` after each `xivo-upgrade`.
5. Restart all XiVO services:

```
xivo-service restart all
```

Configuration Files

This section describes some of the XiVO configuration files.

Configuration priority Usually, the configuration is read from two locations: a configuration file `config.yml` and a configuration directory `conf.d`.

Files in the `conf.d` extra configuration directory:

- are used in alphabetical order and the first one has priority
- are ignored when their name starts with a dot
- are ignored when their name does not end with `.yml`

For example:

`.01-critical.yml:`

```
log_level: critical
```

`02-error.yml.dpkg-old:`

```
log_level: error
```

`10-debug.yml:`

```
log_level: debug
```

`20-nodebug.yml:`

```
log_level: info
```

The value that will be used for `log_level` will be `debug` since:

- `10-debug.yml` comes before `20-nodebug.yml` in the alphabetical order.
- `.01-critical.yml` starts with a dot so is ignored
- `02-error.yml.dpkg-old` does not end with `.yml` so is ignored

File configuration structure Configuration files for every service running on a XiVO server will respect these rules:

- Default configuration directory in `/etc/xivo-{service}/conf.d` (e.g. `/etc/xivo-agentd/conf.d/`)
- Default configuration file in `/etc/xivo-{service}/config.yml` (e.g. `/etc/xivo-agentd/config.yml`)

The files `/etc/xivo-{service}/config.yml` should not be modified because **they will be overridden during upgrades**. However, they may be used as examples for creating additional configuration files as long as they respect the *Configuration priority*. Any exceptions to these rules are documented below.

xivo-agentd

- Default configuration directory: `/etc/xivo-agentd/conf.d`
- Default configuration file: `/etc/xivo-agentd/config.yml`

xivo-amid

- Default configuration directory: `/etc/xivo-amid/conf.d`
- Default configuration file: `/etc/xivo-amid/config.yml`

xivo-auth

- Default configuration directory: `/etc/xivo-auth/conf.d`
- Default configuration file: `/etc/xivo-auth/config.yml`

xivo-confgend

- Default configuration directory: `/etc/xivo-confgend/conf.d`
- Default configuration file: `/etc/xivo-confgend/config.yml`
- Default templates directory: `/etc/xivo-confgend/templates`

xivo-ctid

- Default configuration directory: `/etc/xivo-ctid/conf.d`
- Default configuration file: `/etc/xivo-ctid/config.yml`

xivo-dao

- Default configuration directory: `/etc/xivo-dao/conf.d`
- Default configuration file: `/etc/xivo-dao/config.yml`

This configuration is read by many XiVO programs in order to connect to the Postgres database of XiVO.

xivo-dird-phoned

- Default configuration directory: `/etc/xivo-dird-phoned/conf.d`
- Default configuration file: `/etc/xivo-dird-phoned/config.yml`

xivo-websocketd

- Default configuration directory: `/etc/xivo-websocketd/conf.d`
- Default configuration file: `/etc/xivo-websocketd/config.yml`

xivo_ring.conf

- Path: `/etc/xivo/asterisk/xivo_ring.conf`
- Purpose: This file can be used to change the ringtone played by the phone depending on the origin of the call.

Warning: Note that this feature has not been tested for all phones and all call flows. This page describes how you can customize this file but does not intend to list all validated call flows or phones.

This file `xivo_ring.conf` consists of :

- profiles of configuration (some examples for different brands are already included: `[aastra]`, `[snom]` etc.)
- one section named `[number]` where you apply the profile to an extension or a context etc.

Here is the process you should follow if you want to use/customize this feature :

1. Create a new profile, e.g.:

```
[myprofile-aastra]
```

2. Change the phonetype accordingly, in our example:

```
[myprofile-aastra]
phonetype = aastra
```

3. Chose the ringtone for the different type of calls (note that the ringtone names are brand-specific):

```
[myprofile-aastra]
phonetype = aastra
intern = <Bellcore-dr1>
group = <Bellcore-dr2>
```

4. Apply your profile, in the section [number]

- to a given list of extensions (e.g. 1001 and 1002):

```
1001@default = myprofile-aastra
1002@default = myprofile-aastra
```

- or to a whole context (e.g. default):

```
@default = myprofile-aastra
```

5. Restart xivo-agid service:

```
service xivo-agid restart
```

ipbx.ini

- Path: /etc/xivo/web-interface/ipbx.ini
- Purpose: This file specifies various configuration options and paths related to Asterisk and used by the web interface.

Here is a partial glimpse of what can be configured in file ipbx.ini :

1. Enable/Disable modification of SIP line username and password:

```
[user]
readonly-idpwd = "true"
```

When editing a SIP line, the username and password fields cannot be modified via the web interface. Set this option to false to enable the modification of both fields. This option is set to “true” by default.

Warning: This feature is not fully tested. It should be used only when absolutely necessary and with great care.

Consul

The default [consul](#) installation in XiVO uses the configuration file in /etc/consul/xivo/*.json. All files in this directory are installed with the package and *should not* be modified by the administrator. To use a different configuration, the administrator can add it's own configuration file at another location and set the new configuration directory by creating a systemd unit drop-in file in the /etc/systemd/system/consul.service.d directory.

The default installation generates a master token that can be retrieved in /var/lib/consul/master_token. This master token will not be used if a new configuration is supplied.

Variables The following environment variables can be overridden in a systemd unit drop-in file:

- `CONFIG_DIR`: the consul configuration directory
- `WAIT_FOR_LEADER`: should the “start” action wait for a leader ?

Example, in `/etc/systemd/system/consul.service.d/custom.conf`:

```
[Service]
Environment=CONFIG_DIR=/etc/consul/agent
Environment=WAIT_FOR_LEADER=no
```

Agent mode It is possible to run consul on another host and have the local consul node run as an agent only.

To get this kind of setup up and running, you will need to follow the following steps.

Downloading Consul For a 32 bits system

```
wget --no-check-certificate https://releases.hashicorp.com/consul/0.5.2/consul_0.5.2_linux_386.zip
```

For a 64 bits system

```
wget --no-check-certificate https://releases.hashicorp.com/consul/0.5.2/consul_0.5.2_linux_amd64.zip
```

Installing Consul on a new host

```
unzip consul_0.5.2_linux_386.zip
```

Or

```
unzip consul_0.5.2_linux_amd64.zip
```

```
mv consul /usr/bin/consul
mkdir -p /etc/consul/xivo
mkdir -p /var/lib/consul
adduser --quiet --system --group --no-create-home \
        --home /var/lib/consul consul
```

Copying the consul configuration from the XiVO to a new host On the new consul host, modify `/etc/consul/xivo/config.json` to include to following lines.

```
"bind_addr": "0.0.0.0",
"client_addr": "0.0.0.0",
"advertise_addr": "<consul-host>"
```

```
# on the consul host
scp root@<xivo-host>:/lib/systemd/system/consul.service /lib/systemd/system
systemctl daemon-reload
scp -r root@<xivo-host>:/etc/consul /etc
scp -r root@<xivo-host>:/usr/share/xivo-certs /usr/share
consul agent -data-dir /var/lib/consul -config-dir /etc/consul/xivo/
```

Note: To start consul with the systemd unit file, you may need to change owner and group (consul:consul) for all files inside `/etc/consul`, `/usr/share/xivo-certs` and `/var/lib/consul`

Adding the agent configuration Create the file `/etc/consul/agent/config.json` with the following content

```
{
  "acl_datacenter": "<node_name>",
  "datacenter": "xivo",
  "server": false,
  "bind_addr": "0.0.0.0",
  "advertise_addr": "<xivo_address>",
  "client_addr": "127.0.0.1",
  "bootstrap": false,
  "rejoin_after_leave": true,
  "data_dir": "/var/lib/consul",
  "enable_syslog": true,
  "disable_update_check": true,
  "log_level": "INFO",
  "ports": {
    "dns": -1,
    "http": -1,
    "https": 8500
  },
  "retry_join": [
    "<remote_host>"
  ],
  "cert_file": "/usr/share/xivo-certs/server.crt",
  "key_file": "/usr/share/xivo-certs/server.key"
}
```

- `node_name`: Arbitrary name to give this node, `xivo-paris` for example.
- `remote_host`: IP address of your new consul. Be sure the host is accessible from your XiVO and check the firewall. See the documentation [here](#).
- `xivo_address`: IP address of your xivo.

This file should be owned by consul user.

```
chown -R consul:consul /etc/consul/agent
```

Enabling the agent configuration Add or modify `/etc/systemd/system/consul.service.d/custom.conf` to include the following lines:

```
[Service]
Environment=CONFIG_DIR=/etc/consul/agent
```

Restart your consul server.

```
service consul restart
```

Updating the consul section of xivo-ctid Add a file in `/etc/xivo-ctid/conf.d/remote_consul.yml` with the following content

```
rest_api:
  http:
    listen: 0.0.0.0

service_discovery:
  advertise_address: <xivo-ctid-host>
  check_url: http://<xivo-ctid-host>:9495/0.1/infos
```

- `xivo-ctid-host`: Hostname to reach xivo-ctid

Log Files

Every XiVO service has its own log file, placed in `/var/log`.

asterisk The Asterisk log files are managed by logrotate.

Its configuration files `/etc/logrotate.d/asterisk` and `/etc/asterisk/logger.conf`

The message log level is enabled by default in `logger.conf` and contains notices, warnings and errors. The full log entry is commented in `logger.conf` and should only be enabled when verbose debugging is required. Using this option in production would produce VERY large log files.

- Files location: `/var/log/asterisk/*`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-agentd

- File location: `/var/log/xivo-agentd.log`
- Rotate configuration: `/etc/logrotate.d/xivo-agentd`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-agid

- File location: `/var/log/xivo-agid.log`
- Rotate configuration: `/etc/logrotate.d/xivo-agid`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-amid

- File location: `/var/log/xivo-amid.log`
- Rotate configuration: `/etc/logrotate.d/xivo-amid`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-auth

- File location: `/var/log/xivo-auth.log`
- Rotate configuration: `/etc/logrotate.d/xivo-auth`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-call-logd

- File location: `/var/log/xivo-call-logd.log`
- Rotate configuration: `/etc/logrotate.d/xivo-call-logd`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-confd

- File location: `/var/log/xivo-confd.log`
- Rotate configuration: `/etc/logrotate.d/xivo-confd`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-confgend The `xivo-confgend` daemon output is sent to the file specified with the `--logfile` parameter when launched with `twistd`.

The file location can be changed by customizing the `xivo-confgend.service` unit file.

- File location: `/var/log/xivo-confgend.log`
- Rotate configuration: `/etc/logrotate.d/xivo-confgend`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-ctid

- File location: `/var/log/xivo-ctid.log`
- Rotate configuration: `/etc/logrotate.d/xivo-ctid`
- Number of archived log files: 15
- Rotation frequency: Daily

xivo-ctid-ng

- File location: `/var/log/xivo-ctid-ng.log`
- Rotate configuration: `/etc/logrotate.d/xivo-ctid-ng`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-dird

- File location: `/var/log/xivo-dird.log`
- Rotate configuration: `/etc/logrotate.d/xivo-dird`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-dird-phoned

- File location: `/var/log/xivo-dird-phoned.log`
- Rotate configuration: `/etc/logrotate.d/xivo-dird-phoned`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-dxtora

- File location: `/var/log/xivo-dxtora.log`
- Rotate configuration: `/etc/logrotate.d/xivo-dxtora`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-provd

- File location: `/var/log/xivo-provd.log`
- Rotate configuration: `/etc/logrotate.d/xivo-provd`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-purge-db

- File location: `/var/log/xivo-purge-db.log`
- Rotate configuration: `/etc/logrotate.d/xivo-purge-db`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-stat

- File location: `/var/log/xivo-stat.log`
- Rotate configuration: `/etc/logrotate.d/xivo-stat`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-sysconfd

- File location: `/var/log/xivo-sysconfd.log`
- Rotate configuration: `/etc/logrotate.d/xivo-sysconfd`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-upgrade

- File location: `/var/log/xivo-upgrade.log`
- Rotate configuration: `/etc/logrotate.d/xivo-upgrade`
- Number of archived files: 15
- Rotation frequency: Daily

xivo-web-interface

- File location: `/var/log/xivo-web-interface/*.log`
- Rotate configuration: `/etc/logrotate.d/xivo-web-interface`
- Number of archived files: 21
- Rotation frequency: Daily

xivo-websocketd

- File location: `/var/log/xivo-websocketd.log`
- Rotate configuration: `/etc/logrotate.d/xivo-websocketd`
- Number of archived files: 15
- Rotation frequency: Daily

Nginx

XiVO use nginx as a web server and reverse proxy.

In its default configuration, the nginx server listens on port TCP/80 and TCP/443 and allows these services to be used:

- web interface (xivo-web-interface)
- API documentation (xivo-swagger-doc)

Starting from XiVO 16.13, an administrator can easily modify the configuration to allow additional services to be used (e.g. xivo-auth or xivo-confd).

To do so, an administrator only has to create a symbolic link inside the `/etc/nginx/locations/http-enabled` directory to the corresponding file in the `/etc/nginx/locations/http-available` directory, and then reload nginx with `systemctl reload nginx`. A similar operation must be done for HTTPS.

For example, to enable all the available services:

```
ln -sf /etc/nginx/locations/http-available/* /etc/nginx/locations/http-enabled
ln -sf /etc/nginx/locations/https-available/* /etc/nginx/locations/https-enabled
systemctl reload nginx
```

To disable all the services other than the web interface:

```
rm /etc/nginx/locations/http-enabled/* /etc/nginx/locations/https-enabled/*
ln -s /etc/nginx/locations/http-available/xivo-web-interface /etc/nginx/locations/http-enabled
ln -s /etc/nginx/locations/https-available/xivo-web-interface /etc/nginx/locations/https-enabled
systemctl reload nginx
```

NTP

XiVO has a NTP server, that must be synchronized to a reference server. This can be a public one or customized for specific target networking architecture. XiVO's NTP server is used by default as NTP server for the devices time reference.

Usage Show NTP service status:

```
service ntp status
```

Stop NTP service:

```
service ntp stop
```

Start NTP service:

```
service ntp start
```

Restart NTP service:

```
service ntp restart
```

Show NTP synchronization status:

```
ntpq -p
```

Configuring NTP service

1. Edit `/etc/ntp.conf`
2. Give your NTP reference servers:

```
server 192.168.0.1                # LAN existing NTP Server
server 0.debian.pool.ntp.org iburst dynamic # default in ntp.conf
server 1.debian.pool.ntp.org iburst dynamic # default in ntp.conf
```

3. If no reference server to synchronize to, add this to synchronize locally:

```
server 127.127.1.0                # local clock (LCL)
fudge 127.127.1.0 stratum 10      # LCL is not very reliable
```

4. Restart NTP service
5. Check NTP synchronization status.

Warning: If #5 shows that NTP doesn't use NTP configuration in `/etc/ntp.conf`, maybe have you done a `dhclient` for one of your network interface and the `dhcp` server that gave the IP address also gave a NTP server address. Thus you might check if the file `/var/lib/ntp/ntp.conf.dhcp` exists, if yes, this is used for NTP configuration prior to `/etc/ntp.conf`. Remove it and restart NTP, check NTP synchronization status, then it should work.

Proxy Configuration

If you use XiVO behind an HTTP proxy, you must do a couple of manipulations for it to work correctly.

apt Create the `/etc/apt/apt.conf.d/90proxy` file with the following content:

```
Acquire::http::Proxy "http://domain\username:password@proxyip:proxyport";
```

provd Proxy information is set via the *Configuration* → *Provisioning* → *General* page.

dhcp-update This step is needed if you use the DHCP server of the XiVO. Otherwise the DHCP configuration won't be correct.

Proxy information is set via the `/etc/xivo/dhcpd-update.conf` file.

Edit the file and look for the `[proxy]` section.

xivo-fetchfw *This step is not needed if you don't use xivo-fetchfw.*

Proxy information is set via the `/etc/xivo/xivo-fetchfw.conf` file.

Edit the file and look for the `[proxy]` section.

Service Discovery

Overview XiVO uses `consul` for service discovery. When a daemon is started, it registers itself on the configured consul node.

`Consul template` may be used to generate the configuration files for each daemons that requires the availability of another service. Consul template can also be used to reload the appropriate service.

Service Authentication

XiVO services expose more and more resources through REST API, but they also ensure that the access is restricted to the authorized programs. For this, we use an *authentication daemon* who delivers authorizations via tokens.

Call flow Here is the call flow to access a REST resource of a XiVO service:

1. Create a username/password (also called `service_id/service_key`) with the right *ACLs*, via *Web Services Access*.
2. *Create a token* with these credentials and the backend *xivo-service*.
3. *Use this token* to access the REST resource defined by the *ACL*.

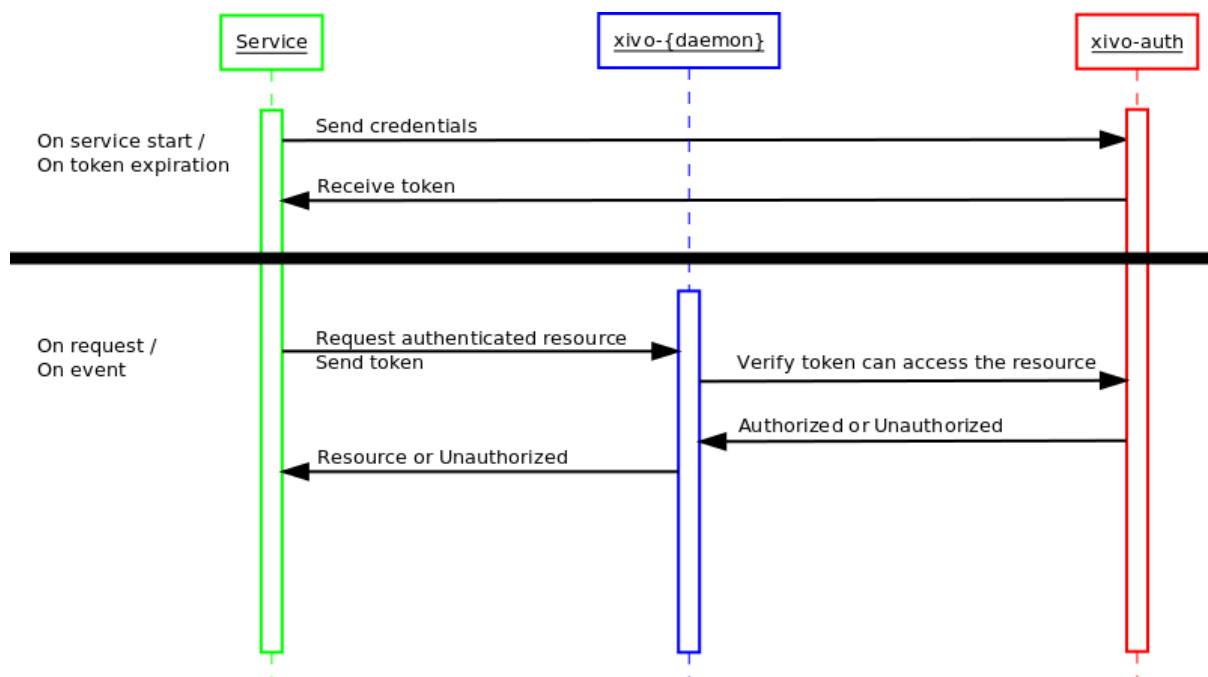


Fig. 1.29: Call flow of service authentication

Service Service who needs to access a REST resource.

xivo-{daemon} Server that exposes a REST resource. This resource must have an attached ACL.

xivo-auth Server that authenticates the *Service* and validates the required ACL with the token.

XiVO services directly use this system to communicate with each other, as you can see in their Web Services Access.

xivo-auth

xivo-auth is a scalable, extendable and configurable authentication service. It uses an HTTP interface to emit tokens to users who can then use those tokens to identify and authenticate themselves with other services compatible with xivo-auth.

The HTTP API reference is at <http://api.xivo.io>.

xivo-auth HTTP API Changelog

16.02

- POST /0.1/token, field `expiration`: only integers are accepted, floats are now invalid.
- Experimental backend `ldap_user_voicemail` has been removed.
- New backend `ldap_user` has been added.

15.19

- POST /0.1/token do not accept anymore argument `backend_args`

15.17

- New backend `ldap_user_voicemail` has been added. **WARNING** this backend is **EXPERIMENTAL**.

15.16

- HEAD and GET now take a new `scope` query string argument to check ACLs
- Backend interface method `get_acls` is now named `get_consul_acls`
- Backend interface method `get_acls` now returns a list of ACLs
- HEAD and GET can now return a 403 if an ACL access is denied

15.15

- POST /0.1/token accept new argument `backend_args`
- Signature of backend method `get_ids()` has a new argument `args`
- New method `get_acls` for backend has been added
- New backend `service` has been added

xivo-auth Developer's Guide

Architecture xivo-auth contains 4 major components, an HTTP interface, a celery worker, authentication backends and a consul client. All operations are made through the HTTP interface, tokens are stored by consul as well as the persistence for some of the data attached to tokens. The celery worker is used to schedule tasks that outlive the lifetime of the xivo-auth process. Backends are used to test if a supplied username/password combination is valid and provide the xivo-user-uuid.

xivo-auth is made of the following modules and packages.

plugins the plugin package contains the xivo-auth backends that are packaged with xivo-auth.

http The http module is the implementation of the HTTP interface.

- Validate parameters
- Calls the backend to check the user authentication
- Forward instructions to the *token_manager*
- Handle exceptions and return the appropriate *status_code*

controller The controller is the plumbin of xivo-auth, it has no business logic.

- Start the HTTP application
- Start the celery worker
- Load all enabled plugins
- Instantiate the *token_manager*

token The token module contains the business logic of xivo-auth.

- Creates and delete tokens
- Creates ACLs for XiVO
- Schedule token expiration
- Read/write token data to consul

tasks The tasks module contains implementation of celery tasks that are executed by the worker.

- Called by the celery worker
- Forwards instructions to the *token manager*

extension This is a place holder for a global variable for the celery app. It will be removed and should not be used.

Other modules that should not need documentation are *helpers*, *config*, *interfaces*

Plugins xivo-auth is meant to be easy to extend. This section describes how to add features to xivo-auth.

Backends xivo-auth allows its administrator to configure one or many sources of authentication. Implementing a new kind of authentication is quite simple.

1. Create a python module implementing the [backend interface](#).
2. Install the python module with an entry point *xivo_auth.backends*

An example backend implementation is available [here](#).

Stock Plugins Documentation

Backends Plugins

XiVO Admin Backend name: `xivo_admin`

Purpose: Authenticate a XiVO administrator. The login/password is configured in *Configuration* → *Management* → *Users*.

XiVO Service Backend name: `xivo_service`

Purpose: Authenticate a XiVO *Web Services Access*. The login/password is configured in *Configuration → Management → Web Service Access*.

XiVO User Backend name: `xivo_user`

Purpose: Authenticate a XiVO user. The login/password is configured in *IPBX → Services → PBX Settings → Users* in the CTI client section.

LDAP Backend name: `ldap_user`

Purpose: Authenticate with an LDAP user.

For example, with the given configuration:

```
ldap:
  uri: ldap://example.org
  bind_dn: cn=xivo,dc=example,dc=org
  bind_password: bindpass
  user_base_dn: ou=people,dc=example,dc=org
  user_login_attribute: uid
  user_email_attribute: mail
```

When an authentication request is received for username `alice` and password `userpass`, the backend will:

1. Connect to the LDAP server at `example.org`
2. Do an LDAP “bind” operation with bind DN `cn=xivo,dc=example,dc=org` and password `bindpass`
3. Do an LDAP “search” operation to find an LDAP user matching `alice`, using:
 - the base DN `ou=people,dc=example,dc=org`
 - the filter `(uid=alice)`
 - a SUBTREE scope
4. If the search returns exactly 1 LDAP user, do an LDAP “bind” operation with the user’s DN and the password `userpass`
5. If the LDAP “bind” operation is successful, search in XiVO a user with an email matching the `mail` attribute of the LDAP user
6. If a XiVO user is found, success

To use an anonymous bind instead, the following configuration would be used:

```
ldap:
  uri: ldap://example.org
  bind_anonymous: True
  user_base_dn: ou=people,dc=example,dc=org
  user_login_attribute: uid
  user_email_attribute: mail
```

The backend can also works in a “no search” mode, for example with the following configuration:

```
ldap:
  uri: ldap://example.org
  user_base_dn: ou=people,dc=example,dc=org
  user_login_attribute: uid
  user_email_attribute: mail
```

When the server receives the same authentication request as above, it will directly do an LDAP “bind” operation with the DN `uid=alice,ou=people,dc=example,dc=org` and password `userpass`, and continue at step 5.

Note: User's email and voicemail's email are two separate things. This plugin only use the user's email.

Configuration

uri the URI of the LDAP server. Can only contain the scheme, host and port of an LDAP URL.

user_base_dn the base dn of the user

user_login_attribute the attribute to login a user

user_email_attribute (optional) the attribute to match with the XiVO user's email (default: mail)

bind_dn (optional) the bind DN for searching for the user DN.

bind_password (optional) the bind password for searching for the user DN.

bind_anonymous (optional) use anonymous bind for searching for the user DN (default: false)

Usage xivo-auth is used through HTTP requests, using HTTPS. Its default port is 9497. As a user, the most common operation is to get a new token. This is done with the POST method.

Alice retrieves a token using her username/password:

```
$ # Alice creates a new token, using the xivo_user backend, expiring in 10 minutes
$ curl -k -X POST -H 'Content-Type: application/json' -u 'alice:s3cre7' "https://localhost:9497/0.1/token"
{"data": {"issued_at": "2015-06-05T10:16:58.557553", "token": "1823c1ee-6c6a-0cdc-d869-964a7f08a7"}}
```

In this example Alice used here XiVO CTI client login `alice` and password `s3cre7`. The authentication source is determined by the *backend* in the POST data.

Alice could also have specified an expiration time on her POST request. The expiration value is the number of seconds before the token expires.

After retrieving her token, Alice can query other services that use xivo-auth and send her token to those service. Those services can then use this token on Alice's behalf to access her personal storage.

If Alice wants to revoke her token before its expiration:

```
$ curl -k -X DELETE -H 'Content-Type: application/json' "https://localhost:9497/0.1/token/1823c1ee-6c6a-0cdc-d869-964a7f08a7"
```

See <http://api.xivo.io> for more details about the HTTP API.

See *Service Authentication* for details about the authentication process.

Usage for services using xivo-auth A service that requires authentication and identification can use xivo-auth to externalise the burden of authentication. The new service can then accept a token as part of its operations to authenticate the user using the service.

Once a service receives a token from one of its user, it will need to check the validity of that token. There are 2 forms of verification, one that only checks if the token is valid and the other returns information about this token's session if it is valid.

Checking if a token is valid:

```
$ curl -k -i -X HEAD -H 'Content-Type: application/json' "https://localhost:9497/0.1/token/1823c1ee-6c6a-0cdc-d869-964a7f08a7"
HTTP/1.1 204 NO CONTENT
Content-Type: text/html; charset=utf-8
Content-Length: 0
Date: Fri, 05 Jun 2015 14:49:50 GMT
Server: pcm-dev-0

$ # get more information about this token
$ curl -k -X GET -H 'Content-Type: application/json' "https://localhost:9497/0.1/token/1823c1ee-6c6a-0cdc-d869-964a7f08a7"
{"data": {"issued_at": "2015-06-05T10:16:58.557553", "token": "1823c1ee-6c6a-0cdc-d869-964a7f08a7"}}
```


Launching xivo-auth

```
usage: xivo-auth [-h] [-c CONFIG_FILE] [-u USER] [-d] [-f] [-l LOG_LEVEL]

optional arguments:
  -h, --help            show this help message and exit
  -c CONFIG_FILE, --config-file CONFIG_FILE
                        The path to the config file
  -u USER, --user USER User to run the daemon
  -d, --debug           Log debug messages
  -f, --foreground      Foreground, don't daemonize
  -l LOG_LEVEL, --log-level LOG_LEVEL
                        Logs messages with LOG_LEVEL details. Must be one of:
                        critical, error, warning, info, debug. Default: None
```

HTTP API Reference The complete HTTP API documentation is at <http://api.xivo.io>.

See also the *xivo-auth HTTP API Changelog*.

Development See *xivo-auth Developer's Guide*.

xivo-confd

xivo-confd is a HTTP server that provides a RESTful API service for configuring and managing basic resources on a XiVO server.

The HTTP API reference is available at <http://api.xivo.io>.

Developer's Guide (xivo-confd) xivo-confd resources are organised through a plugin mechanism. There are 2 main plugin categories:

Resource plugins A plugin that manages a resource (e.g. users, extensions, voicemails, etc). A resource plugin exposes the 4 basic CRUD operations (Create, Read, Update, Delete) in order to operate on a resource in a RESTful manner.

Association plugins A plugin for associating or dissociating 2 resources (e.g a user and a line). An association plugin exposes an HTTP action for associating (either POST or PUT) and another for dissociating (DELETE)

The following diagram outlines the most important parts of a plugin:

Resource Class that receives and handles HTTP requests. Resources use [flask-restful](#) for handling requests.

There are 2 kinds of resources: *ListResource* for root URLs and *ItemResource* for URLs that have an ID. *ListResource* will handle creating a resource (POST) and searching through a list of available resources (GET). *ItemResource* handles fetching a single item (GET), updating (PUT) and deleting (DELETE).

Service Class that handles business logic for a resource, such as what to do in order to get, create, update, or delete a resource. *Service* classes do not manipulate data directly. Instead, they coordinate what to do via other objects.

There are 2 kinds of services: *CRUDService* for basic CRUD operations in *Resource plugins*, and *AssociationService* for association/dissociation operations in *Association plugins*.

Dao Data Access Object. Knows how to get data and how to manipulate it, such as SQL queries, files, etc.

Notifier Sends events after an operation has completed. An event will be sent in a messaging queue for each CRUD operation. Certain resources also need to send events to other daemons in order to reload some configuration data. (i.e. asterisk needs to reload the dialplan when an extension is updated)

Validator Makes sure that a resource's data does not contain any errors before doing something with it. A *Validator* can be used for validating input data or business rules.

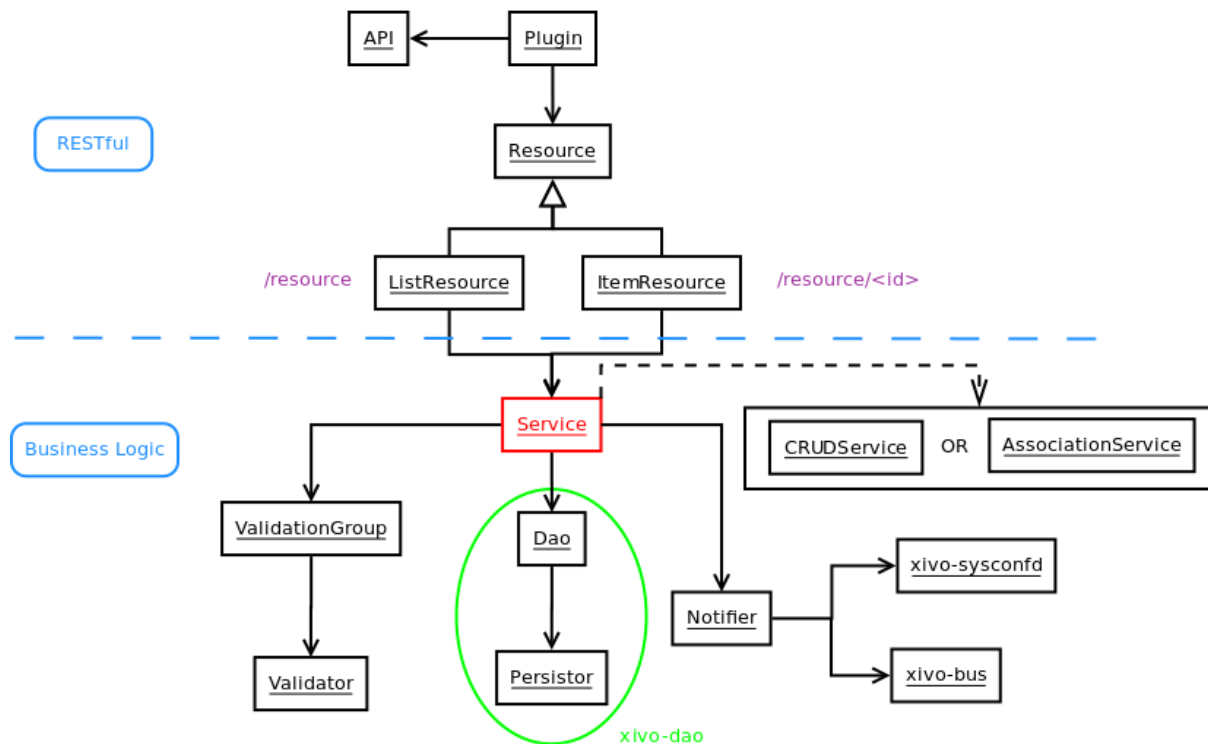


Fig. 1.30: Plugin architecture of xivo-confd

XiVO confgend

xivo-confgend is a configuration file generator. It is mainly used to generate the Asterisk configuration files.

XiVO confgend developer's guide xivo-confgend uses drivers to implement the logic required to generate configuration files. It uses [stevedore](#) to do the driver instantiation and discovery.

Plugins in xivo-confgend use `setuptools`' entry points. That means that installing a new plugin to xivo-confgend requires an entry point in the plugin's `setup.py`.

Drivers Driver plugin are classes that are used to generate the content of a configuration file.

The implementation of a plugin should have the following properties.

1. It's `__init__` method should take one argument
2. It should have a `generate` method which will return the content of the file
3. A `setup.py` adding an entry point

The `__init__` method argument is the content of the configuration of xivo-confgend. This allows the driver implementor to add values to the configuration in `/etc/xivo-confgend/conf.d/*.yaml` and these values will be available in the driver.

The `generate` method has no argument, the configuration provided to the `__init__` should be sufficient for most cases. `generate` is called within a `scoped_session` of xivo-dao, allowing the usage of xivo-dao without prior setup in the driver.

The namespaces used for entry points in xivo-confgend have the following form:

```
xivo_confgend.<resource>.<filename>
```

as an example, a generator for `sip.conf` would have the following namespace:

xivo_confgend.asterisk.sip.conf

Example Here is a typical setup.py:

```

1  #!/usr/bin/env python
2  # -*- coding: utf-8 -*-
3  # Copyright 2016 by Avencall
4  # SPDX-License-Identifier: GPL-3.0+
5
6  from setuptools import setup
7  from setuptools import find_packages
8
9
10 setup(
11     name='XiVO confgend driversample',
12     version='0.0.1',
13
14     description='An example driver',
15
16     packages=find_packages(),
17
18     entry_points={
19         'xivo_confgend.asterisk.sip.conf': [
20             'my_driver = src.driver:MyDriver',
21         ],
22     }
23 )

```

With the following package structure:

```

.
-- setup.py
-- src
    -- driver.py

```

driver.py:

```

1  # -*- coding: utf-8 -*-
2  # Copyright 2016 by Avencall
3  # SPDX-License-Identifier: GPL-3.0+
4
5
6  class MyDriver(object):
7
8      def __init__(self, config):
9          self._config = config
10
11      def generate(self):
12          return 'Hello World!'

```

To enable this plugin, you need to:

1. Install the plugin with:

```
python setup.py install
```

2. Create a config file in /etc/xivo-confgend/conf.d:

```
plugins:
    asterisk.sip.conf: my_driver
```

3. Restart xivo-confgend:

```
systemctl restart xivo-confgend
```

XiVO dird

xivo-dird is the directory server for XiVO. It offers a simple REST interface to query all directories that are configured. xivo-dird is extendable with plugins.

xivo-dird changelog

16.12

- Added phonebook imports
 - POST `0.1/tenants/<tenant>/phonebooks/<phonebook_id>/contacts/import`

16.11

- Added a new internal phonebook with a CRUD interface
- Added a new backend to do lookups in the new phonebook

15.20

- The ldap plugins `ldap_network_timeout` default value has been incremented from 0.1 to 0.3 seconds

15.19

- Added the voicemail type in *Views* configuration
- Removed reverse endpoints in REST API:
 - GET `/0.1/directories/reverse/<profile>/me`

15.18

- Added reverse endpoints in REST API:
 - GET `/0.1/directories/reverse/<profile>/<xivo_user_uuid>`
 - GET `/0.1/directories/reverse/<profile>/me`

15.17

- Added directories endpoints in REST API:
 - GET `/0.1/directories/input/<profile>/aastra`
 - GET `/0.1/directories/lookup/<profile>/aastra`
 - GET `/0.1/directories/input/<profile>/polycom`
 - GET `/0.1/directories/lookup/<profile>/polycom`
 - GET `/0.1/directories/input/<profile>/snom`
 - GET `/0.1/directories/lookup/<profile>/snom`
 - GET `/0.1/directories/lookup/<profile>/thomson`
 - GET `/0.1/directories/lookup/<profile>/yealink`

15.16

- Added more cisco endpoints in REST API:
 - GET /0.1/directories/input/<profile>/cisco
- Endpoint /0.1/directories/lookup/<profile>/cisco accepts a new limit and offset query string arguments.

15.15

- Added cisco endpoints in REST API:
 - GET /0.1/directories/menu/<profile>/cisco
 - GET /0.1/directories/lookup/<profile>/cisco

15.14

- Added more personal contacts endpoints in REST API:
 - GET /0.1/personal/<contact_id>
 - PUT /0.1/personal/<contact_id>
 - POST /0.1/personal/import
 - DELETE /0.1/personal
- Endpoint /0.1/personal accepts a new format query string argument.

15.13

- Added personal contacts endpoints in REST API:
 - GET /0.1/directories/personal/<profile>
 - GET /0.1/personal
 - POST /0.1/personal
 - DELETE /0.1/personal/<contact_id>
- Signature of backend method list() has a new argument args
- Argument args for backend methods list() and search() has a new key token_infos
- Argument args for backend method load() has a new key main_config
- Methods __call__() and lookup() of service plugin lookup take a new token_infos argument

15.12

- Added authentication on all REST API endpoints
- Service plugins receive the whole configuration, rather than only their own section

XIVO dird configuration There are three sources of configuration for xivo-dird:

- the *command line options*
- the main configuration file
- the sources configuration directory

The command-line options have priority over the main configuration file options.

Main Configuration File Default location: `/etc/xivo-dird/config.yml`. Format: YAML

The default location may be overwritten by the command line options.

Here's an example of the main configuration file:

```
1 debug: False
2 foreground: False
3 log_filename: /var/log/xivo-dird.log
4 log_level: info
5 pid_filename: /var/run/xivo-dird/xivo-dird.pid
6 source_config_dir: /etc/xivo-dird/sources.d
7 user: www-data
8
9 rest_api:
10     wsgi_socket: /var/run/xivo-dird/xivo-dird.sock
11
12 enabled_plugins:
13     backends:
14         - csv
15         - ldap
16         - phonebook
17     services:
18         - lookup
19     views:
20         - cisco_view
21         - default_json
22
23 views:
24     displays:
25         switchboard_display:
26             -
27                 title: Firstname
28                 default: Unknown
29                 field: firstname
30                 type: name
31             -
32                 title: Lastname
33                 default: Unknown
34                 field: lastname
35                 type: name
36         default_display:
37             -
38                 title: Firstname
39                 field: fn
40                 type: name
41             -
42                 title: Location
43                 default: Canada
44                 field: country
45             -
46                 title: Number
47                 field: number
48                 type: number
49     displays_phone:
50         default:
51             name:
52                 - display_name
53             number:
54                 -
55                     field:
56                         - phone
57                 -
58                     field:
```

```

59         - phone_mobile
60         name_format: "{name} (Mobile)"
61     profile_to_display:
62         default: default_display
63         switchboard: switchboard_display
64     profile_to_display_phone:
65         default: default
66
67 services:
68     lookup:
69         default:
70             sources:
71                 - my_csv
72                 - ldap_quebec
73             timeout: 0.5
74     switchboard:
75         sources:
76             - my_csv
77             - xivo_phonebook
78             - ldap_quebec
79         timeout: 1
80
81 sources:
82     my_source:
83         name: my_source
84         type: ldap
85         ldap_option1: value
86         ldap_option2: value
87         ...

```

Root section

debug Enable log debug messages. Overrides `log_level`. Default: `False`.

foreground Foreground, don't daemonize. Default: `False`.

log_filename File to write logs to. Default: `/var/log/xivo-dird.log`.

log_level Logs messages with LOG_LEVEL details. Must be one of: `critical`, `error`, `warning`, `info`, `debug`. Default: `info`.

pid_filename File used as lock to avoid multiple xivo-dird instances. Default: `/var/run/xivo-dird/xivo-dird.pid`.

source_config_dir The directory from which sources configuration are read. See [Sources Configuration](#). Default: `/etc/xivo-dird/sources.d`.

user The owner of the process. Default: `www-data`.

enabled_plugins section This section controls which plugins are to be loaded at xivo-dird startup. All plugin types must have at least one plugin enabled, or xivo-dird will not start. For back-end plugins, sources using a back-end plugin that is not enabled will be ignored.

views section

displays A dictionary describing the content of each display. The key is the display's name, and the value are the display's content.

The display content is a list of fields. Each field is a dictionary with the following keys:

- **title**: The label of the field
- **default**: The default value of the field

- **type:** An arbitrary identifier of the field. May be used by consumers to identify the field without matching the label. For meaningful values inside XiVO, see *Integration of XiVO dird with the rest of XiVO*.
- **field:** the key of the data from the source that will be used for this field.

The display may be used by a plugin view to configure which fields are to be presented to the consumer.

displays_phone A dictionary describing the content of phone-related displays. Like `displays`, the key is the display's name and the value is the display's content. These displays are used by phone-related view plugins, like the `cisco_view` plugin.

The display content contains 2 keys, `name` and `number`.

The value of the `name` key is a list of source result fields. For a given source result, the first field that will return a non-empty value will be used as the display name on the phone. For example, if `name` is configured with `["display_name", "name"]` and you have a source result with fields `{"display_name": "", "name": "Bob"}`, then "Bob" will be displayed on the phone.

The value of the `number` key is a list of number item. Each item is composed of a dictionary containing at least a `field` key, and optionally a `name_format` key. For example, if you have the following number configuration:

```
name:
  - display_name
number:
  -
    field:
      - phone
  -
    field:
      - phone_mobile
    name_format: "{name} (Mobile)"
```

and you have a source result `{"display_name": "Bob", "phone": "101", "phone_mobile": "102"}`, then 2 results will be displayed on your phone:

1. "Bob", with number "101"
2. "Bob (Mobile)", with number "102"

The `name_format` value is a python format string. There's two substitution variables available, `{name}` and `{number}`.

profile_to_display A dictionary associating a profile to a display. It allows xivo-dird to use the right display when a consumer makes a query with a profile. The key is the profile name and the value is the display name.

profile_to_display_phone: A dictionary associating a profile to a phone display. This is similar to `profile_to_display`, but only used by phone-related view plugins.

services section This section is a dictionary whose keys are the service plugin name and values are the configuration of that service. Hence the content of the value is dependent of the service plugin. See the documentation of the service plugin (*Stock Plugins Documentation*).

sources section This section is a dictionary whose keys are the source name and values are the configuration for that source. See the *Sources Configuration* section for more details about source configuration.

Sources Configuration There are two ways to configure sources:

- in the sources section of the main configuration
- in files of a directory, one file for each source:
 - Default directory location `/etc/xivo-dird/sources.d`

- Files format: YAML
- File names are ignored
- Each file listed in this directory will be read and used to create a data source for xivo-dird.

Here is an example of a CSV source configuration in its own file:

```

1 type: csv
2 name: my_contacts_in_a_csv_file
3 file: /usr/local/share/my_contacts.csv
4 unique_column: id
5 searched_columns:
6   - fn
7   - ln
8 format_columns:
9   name: "{fn} {ln}"
10  number: "{num}"

```

This is strictly equivalent in the main configuration file:

```

1 sources:
2   my_contacts_in_a_csv_file:
3     type: csv
4     name: my_contacts_in_a_csv_file
5     file: /usr/local/share/my_contacts.csv
6     unique_column: id
7     searched_columns:
8       - fn
9       - ln
10    source_to_display_columns:
11      ln: lastname
12      fn: firstname
13      num: number

```

type the type of the source. It must be the same than the name of one of the enabled back-end plugins.

name is the name of this given configuration. The name is used to associate the source to profiles. The value is arbitrary, but it must be unique across all sources.

Warning: Changing the name of the source will make all favorites in that source disappear. There is currently no tool to help you migrate favorites between source names, so choose your source names carefully.

The other options are dependent on the source type (the back-end used). See the documentation of the back-end plugin (*Stock Plugins Documentation*). However, the following keys should be present in all source configurations:

first_matched_columns (optional) the columns used for the reverse lookup. Any column having the search term will be a reverse lookup result.

format_columns (optional) a mapping between result fields and a format string. The new key will be added to the result, if this name already exists in the result, it will be replaced with the new value. The syntax is a python format string. See <https://docs.python.org/2/library/string.html#formatspec> for a complete reference.

searched_columns (optional) the columns used for the lookup. Any column containing the search term substring will be a lookup result.

unique_column (optional) This column is what makes an entry unique in this source. The `unique_column` is used to build the `uid` that is passed to the `list` method to fetch a list of results by unique ids. This is necessary for listing and identifying favorites.

XiVO dird developer's guide The XiVO dird architecture uses plugins as extension points for most of its job. It uses *stevedore* to do the plugin instantiation and discovery and *ABC* classes to define the required interface.

Plugins in xivo-dird use *setuptools*' entry points. That means that installing a new plugin to xivo-dird requires an entry point in the plugin's `setup.py`. Each entry point's *namespace* is documented in the appropriate documentation

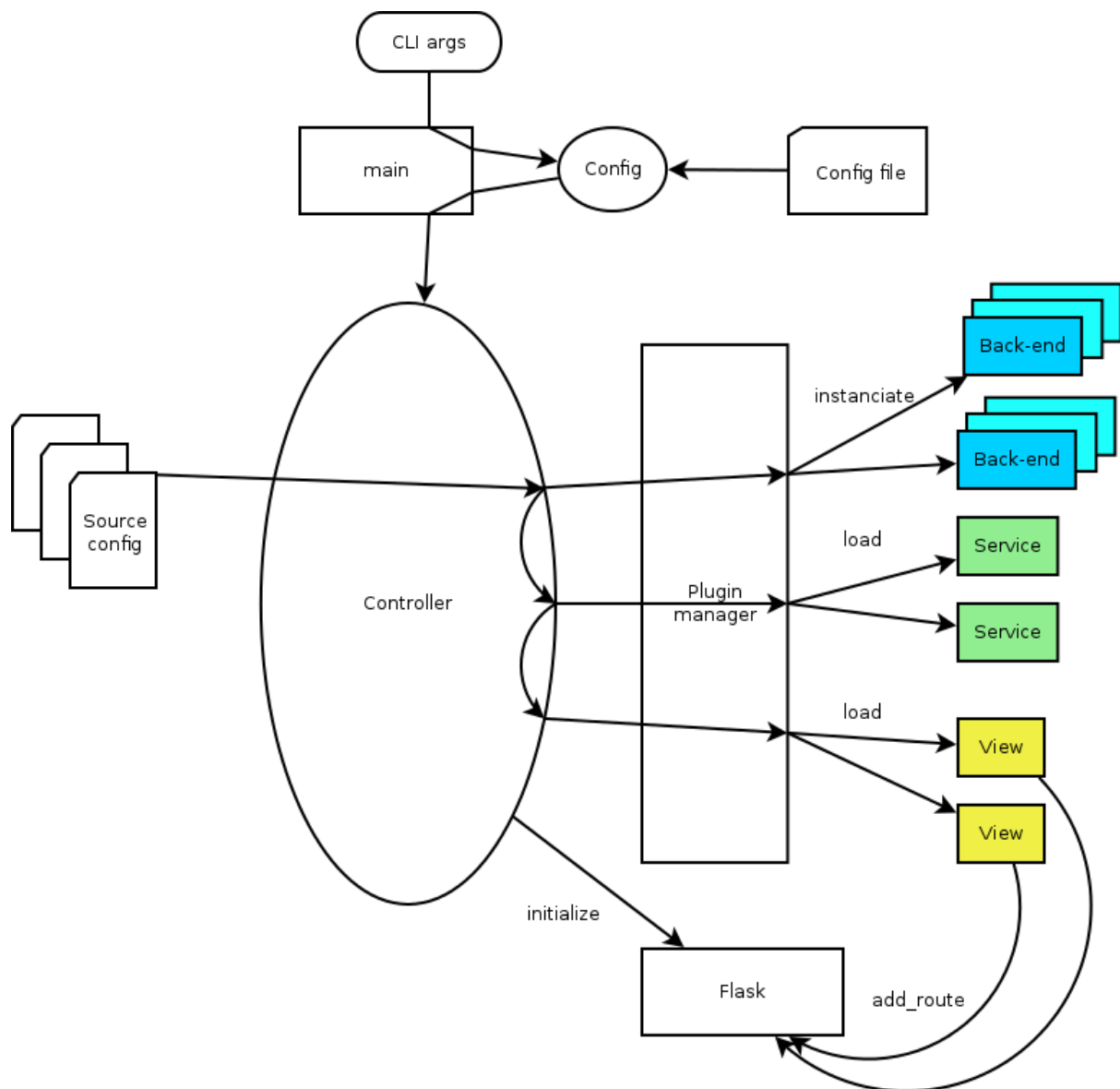


Fig. 1.31: xivo-dird startup flow

section. These entry points allow xivo-dird to be able to discover and load extensions packaged with xivo-dird or installed separately.

Each kind of plugin does a specific job. There are three kinds of plugins in dird.

1. *Back-End*
2. *Service*
3. *View*

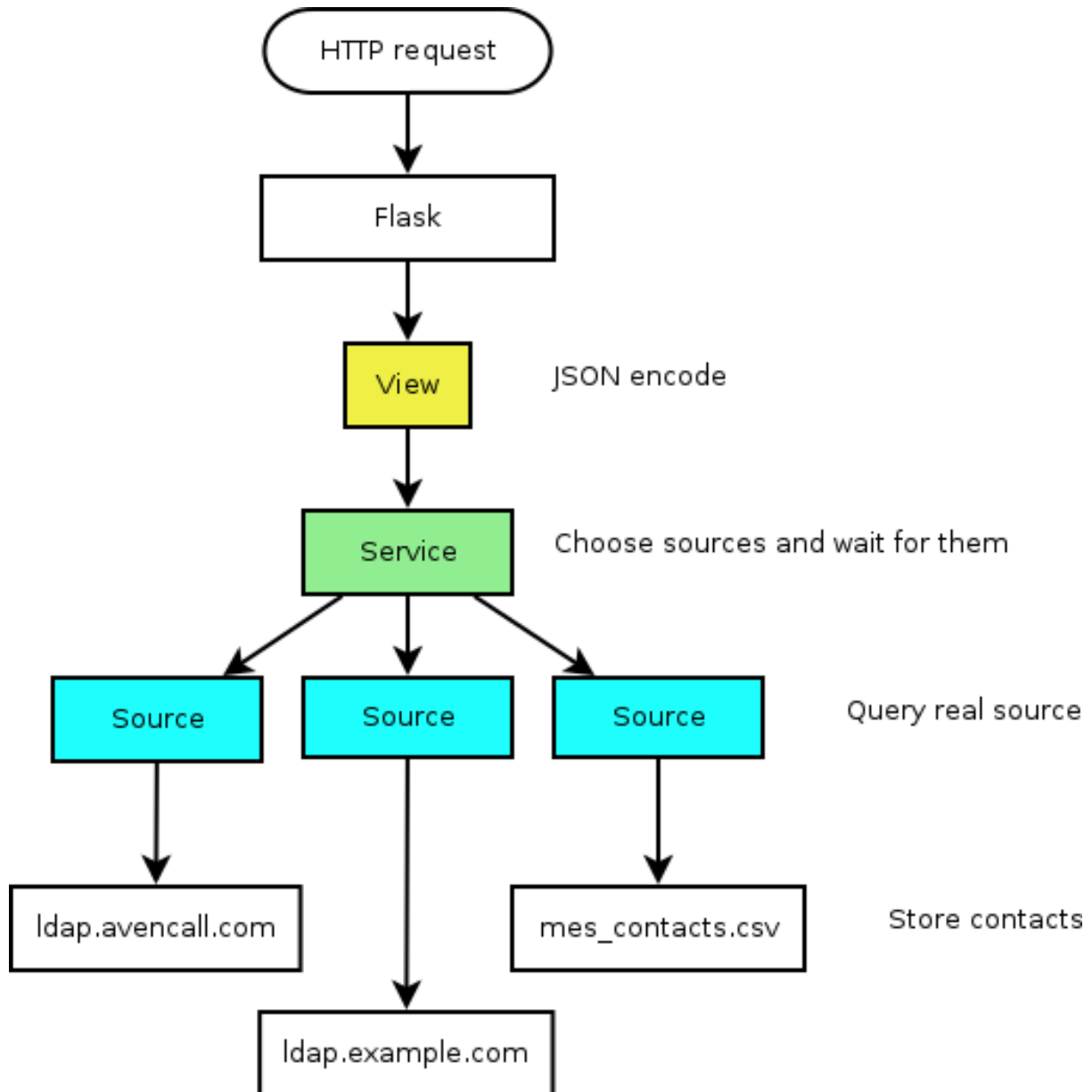


Fig. 1.32: xivo-dird HTTP query

All plugins are instantiated by the core. The core then keeps a catalogue of loaded extensions that can be supplied to other extensions.

The following `setup.py` shows an example of a python library that add a plugin of each kind to xivo-dird:

```

1  #!/usr/bin/env python
2  # -*- coding: utf-8 -*-
3
4  from setuptools import setup

```

```

5 from setuptools import find_packages
6
7
8 setup(
9     name='XiVO dird plugin sample',
10    version='0.0.1',
11
12    description='An example program',
13
14    packages=find_packages(),
15
16    entry_points={
17        'xivo_dird.services': [
18            'my_service = dummy:DummyServicePlugin',
19        ],
20        'xivo_dird.backends': [
21            'my_backend = dummy:DummyBackend',
22        ],
23        'xivo_dird.views': [
24            'my_view = dummy:DummyView',
25        ],
26    }
27 )

```

Back-End Back-ends are used to query directories. Each back-end implements a way to query a given directory. Each instance of a given back-end is called a source. Sources are used by the services to get results from each configured directory.

Given one LDAP back-end, I can configure a source from the LDAP at alpha.example.com and another source from the other LDAP at beta.example.com. Both of these sources use the LDAP back-end.

Implementation details

- Namespace: `xivo_dird.backends`
- Abstract source plugin: `BaseSourcePlugin`
- Methods:
 - `name`: the name of the source, typically retrieved from the configuration injected to `load()`
 - `load(args)`: set up resources used by the plugin, depending on the config. `args` is a dictionary containing:
 - * `key config`: the source configuration for this instance of the back-end
 - * `key main_config`: the whole configuration of xivo-dird
 - `unload()`: free resources used by the plugin.
 - `search(term, args)`: The search method returns a list of dictionary.
 - * Empty values should be `None`, instead of empty string.
 - * `args` is a dictionary containing:
 - `key token_infos`: data associated to the authentication token (see [xivo-auth](#))
 - `first_match(term, args)`: The `first_match` method returns a dictionary.
 - * Empty values should be `None`, instead of empty string.
 - * `args` is a dictionary containing:
 - `key token_infos`: data associated to the authentication token (see [xivo-auth](#))

- `list(uids, args)`: The list method returns a list of dictionary from a list of uids. Each uid is a string identifying a contact within the source.
- * `args` is a dictionary containing:
 - `key token_infos`: data associated to the authentication token (see [xivo-auth](#))

See [Sources Configuration](#). The implementation of the back-end should take these values into account and return results accordingly.

Example The following example add a backend that will return random names and number.

`dummy.py`:

```

1  # -*- coding: utf-8 -*-
2
3  import logging
4
5  logger = logging.getLogger(__name__)
6
7  class DummyBackendPlugin(object):
8
9      def name(self):
10         return 'my_local_dummy'
11
12     def load(self, args):
13         logger.info('dummy backend loaded')
14
15     def unload(self):
16         logger.info('dummy backend unloaded')
17
18     def search(self, term, args):
19         nb_results = random.randint(1, 20)
20         return _random_list(nb_results)
21
22     def list(self, unique_ids):
23         return _random_list(len(unique_ids))
24
25     def _random_list(self, nb_results):
26         columns = ['Firstname', 'Lastname', 'Number']
27         return [_random_entry(columns) for _ in xrange(nb_results)]
28
29     def _random_entry(self, columns):
30         random_stuff = [_random_string() for _ in xrange(len(columns))]
31         return dict(zip(columns, random_stuff))
32
33     def _random_string(self):
34         return ''.join(random.choice(string.lowercase) for _ in xrange(5))

```

Service Service plugins add new functionality to the dird server. These functionalities are available to views. When loaded, a service plugin receives its configuration and a dictionary of available sources.

Some service examples that come to mind include:

- A lookup service to search through all configured sources.
- A reverse lookup service to search through all configured sources and return a specific field of the first matching result.

Implementation details

- Namespace: `xivo_dird.services`
- Abstract service plugin: [BaseServicePlugin](#)

- Methods:

- `load(args)`: set up resources used by the plugin, depending on the config. `args` is a dictionary containing:
 - * `key config`: the whole configuration file in dict form
 - * `key sources`: a dictionary of source names to sources`load` must return the service object, which is any kind of python object.
- `unload()`: free resources used by the plugin.

Example The following example adds a service that will return an empty list when used.

`dummy.py`:

```
1  # -*- coding: utf-8 -*-
2
3  import logging
4
5  from xivo_dird import BaseServicePlugin
6
7  logger = logging.getLogger(__name__)
8
9  class DummyServicePlugin(BaseServicePlugin):
10     """
11     This plugin is responsible for instantiating and returning the
12     DummyService. It manages its life time and should take care of
13     its cleanup if necessary
14     """
15
16     def load(self, args):
17         """
18         Ignores all provided arguments and instantiate a DummyService that
19         is returned to the core
20         """
21         logger.info('dummy loaded')
22         self._service = DummyService()
23         return self._service
24
25     def unload(self):
26         logger.info('dummy unloaded')
27
28
29  class DummyService(object):
30     """
31     A very dumb service that will return an empty list every time it is used
32     """
33
34     def list(self):
35         """
36         This function must be called explicitly from the view, `list` is not a
37         special method name for xivo-dird
38         """
39         return []
```

View View plugins add new routes to the HTTP application in `xivo-dird`, in particular the REST API of `xivo-dird`: they define the URLs to which `xivo-dird` will respond and the formatting of data received and sent through those URLs.

For example, we can define a REST API formatted in JSON with one view and the same API formatted in XML with another view. Supporting the directory function of a phone is generally a matter of adding a new view for the format that the phone consumes.

Implementation details

- Namespace: `xivo_dird.views`
- Abstract view plugin: [BaseViewPlugin](#)
- Methods:
 - `load(args)`: set up resources used by the plugin, depending on the config. Typically, register routes on Flask. Those routes would typically call a service. `args` is a dictionary containing:
 - * key `config`: the section of the configuration file for all views in dict form
 - * key `services`: a dictionary of services, indexed by name, which may be called from a route
 - * key `http_app`: the [Flask application](#) instance
 - * key `rest_api`: a [Flask-RestFul Api](#) instance
 - `unload()`: free resources used by the plugin.

Example The following example adds a simple view: GET `/0.1/directories/ping` answers `{"message": "pong"}`.

`dummy.py`:

```

1  # -*- coding: utf-8 -*-
2
3  import logging
4
5  from flask_restful import Resource
6
7  logger = logging.getLogger(__name__)
8
9
10 class PingViewPlugin(object):
11
12     name = 'ping'
13
14     def __init__(self):
15         logger.debug('dummy view created')
16
17     def load(self, args):
18         logger.debug('dummy view args: %s', args)
19
20         args['rest_api'].add_resource(PingView, '/0.1/directories/ping')
21
22     def unload(self):
23         logger.debug('dummy view unloaded')
24
25
26 class PingView(Resource):
27     """
28     Simple API using Flask-Restful: GET /0.1/directories/ping answers "pong"
29     """
30
31     def get(self):
32         return {'message': 'pong'}

```

Stock Plugins Documentation

View Plugins

default_json View name: default_json

Purpose: present directory entries in JSON format. The format is detailed in <http://api.xivo.io>.

headers View name: headers

Purpose: List headers that will be available in results from default_json view.

personal_view View name: personal_view

Purpose: Expose REST API to manage personal contacts (create, delete, list).

phonebook_view View name: phonebook_view

Purpose: Expose REST API to manage xivo-dird's internal phonebooks.

aastra_view View name: aastra_view

Purpose: Expose REST API to search in configured directories for Aastra phone.

cisco_view View name: cisco_view

Purpose: Expose REST API to search in configured directories for Cisco phone (see [CiscoIP-Phone_XML_Objects](#)).

polycom_view View name: polycom_view

Purpose: Expose REST API to search in configured directories for Polycom phone.

snom_view View name: snom_view

Purpose: Expose REST API to search in configured directories for Snom phone.

thomson_view View name: thomson_view

Purpose: Expose REST API to search in configured directories for Thomson phone.

yealink_view View name: yealink_view

Purpose: Expose REST API to search in configured directories for Yealink phone.

Service Plugins

lookup Service name: lookup

Purpose: Search through multiple data sources, looking for entries matching a word.

Configuration Example (excerpt from the main configuration file):

```
1 services:
2     lookup:
3         default:
4             sources:
5                 - my_csv
6             timeout: 0.5
```


The configuration is a dictionary whose keys are profile names and values are configuration specific to that profile.

For each profile, the configuration keys are:

sources The list of source names that are to be used for the lookup

timeout The maximum waiting time for an answer from any source. Results from sources that take longer to answer are ignored. Default: no timeout.

favorites Service name: favorites

Purpose: Mark/unmark contacts as favorites and get the list of all favorites.

personal Service name: personal

Purpose: Add, delete, list personal contacts of users.

phonebook Service name: phonebook

Purpose: Add, delete, list phonebooks and phonebook contacts.

Configuration Example (excerpt from the main configuration file):

```

1 services:
2     favorites:
3         default:
4             sources:
5                 - my_csv
6                 timeout: 0.5

```

The configuration is a dictionary whose keys are profile names and values are configuration specific to that profile.

For each profile, the configuration keys are:

sources The list of source names that are to be used for the lookup

timeout The maximum waiting time for an answer from any source. Results from sources that take longer to answer are ignored. Default: no timeout.

reverse Service name: reverse

Purpose: Search through multiple data sources, looking for the first entry matching an extension.

Configuration Example:

```

1 services:
2     reverse:
3         default:
4             sources:
5                 - my_csv
6                 timeout: 1

```

The configuration is a dictionary whose keys are profile names and values are configuration specific to that profile.

For each profile, the configuration keys are:

sources The list of source names that are to be used for the reverse lookup

timeout The maximum waiting time for an answer from any source. Results from sources that take longer to answer are ignored. Default: 1.

Back-end Configuration This sections completes the *Sources Configuration* section.

csv Back-end name: csv

Purpose: read directory entries from a CSV file.

Limitations:

- the CSV delimiter is not configurable (currently: , (comma)).

Configuration Example (a file inside `source_config_dir`):

```
1 type: csv
2 name: my_csv
3 file: /var/tmp/test.csv
4 unique_column: id
5 searched_columns:
6   - fn
7   - ln
8 first_matched_columns:
9   - num
10 format_columns:
11   lastname: "{ln}"
12   firstname: "{fn}"
13   number: "{num}"
```

With the CSV file:

```
1 id,fn,ln,num
2 1,Alice,Abrams,55553783147
3 2,Bob,Benito,5551354958
4 3,Charles,Curie,5553132479
```

file the absolute path to the CSV file

CSV web service Back-end name: csv_ws

Purpose: search using a web service that returns CSV formatted results.

Given the following configuration, *xivo-dird* would call “<https://example.com:8000/ws-phonebook?firstname=alice&lastname=alice>” for a lookup for the term “alice”.

Configuration Example (a file inside `source_config_dir`):

```
1 type: csv_ws
2 name: a_csv_web_service
3 lookup_url: "https://example.com:8000/ws-phonebook"
4 list_url: "https://example.com:8000/ws-phonebook"
5 verify_certificate: False
6 searched_columns:
7   - firstname
8   - lastname
9 first_matched_columns:
10   - exten
11 delimiter: ","
12 timeout: 16
13 unique_column: id
14 format_columns:
15   number: "{exten}"
```

lookup_url the URL used for directory searches.

list_url (optional) the URL used to list all available entries. This URL is used to retrieve favorites.

verify_certificate (optional) whether the SSL cert will be verified. A `CA_BUNDLE` path can also be provided. Defaults to True.

delimiter (optional) the field delimiter in the CSV result. Default: ‘,’

timeout (optional) the number of seconds before the lookup on the web service is aborted. Default: 10.

dird_phonebook back-end name: dird_phonebook

Purpose: search the xivo-dird’s internal phonebooks

Configuration:

```

1 type: dird_phonebook
2 name: phonebook
3 db_uri: 'postgresql://asterisk:proformatique@localhost/asterisk'
4 tenant: default
5 phonebook_id: 42
6 phonebook_name: main
7 first_matched_columns:
8   - number
9 searched_columns:
10  - firstname
11  - lastname
12 format_columns:
13   name: "{firstname} {lastname}"

```

db_uri the URI of the DB used by xivo-dird to store the phonebook.

tenant the tenant of the phonebook to query.

phonebook_name the *name* of the phonebook used by this source.

phonebook_id (deprecated, use phonebook_name) the *id* of the phonebook used by this source.

ldap Back-end name: ldap

Purpose: search directory entries from an LDAP server.

Configuration Example (a file inside `source_config_dir`):

```

1 type: ldap
2 name: my_ldap
3 ldap_uri: ldap://example.org
4 ldap_base_dn: ou=people,dc=example,dc=org
5 ldap_username: cn=admin,dc=example,dc=org
6 ldap_password: foobar
7 ldap_custom_filter: (l=québec)
8 unique_column: entryUUID
9 searched_columns:
10  - cn
11 first_matched_columns:
12  - telephoneNumber
13 format_columns:
14   firstname: "{givenName}"
15   lastname: "{sn}"
16   number: "{telephoneNumber}"

```

ldap_uri the URI of the LDAP server. Can only contains the scheme, host and port part of an LDAP URL.

ldap_base_dn the DN of the entry at which to start the search

ldap_username (optional) the user’s DN to use when performing a “simple” bind.

Default to an empty string.

When both `ldap_username` and `ldap_password` are empty, an anonymous bind is performed.

ldap_password (optional) the password to use when performing a “simple” bind.

Default to an empty string.

ldap_custom_filter (optional) the custom filter is used to add more criteria to the filter generated by the back end.

Example:

- ldap_custom_filter: (l=québec)
- searched_columns: [cn,st]

will result in the following filter being used for searches. (& (l=québec) (| (cn=%Q*) (st=%Q*)))

If only the custom filter is to be used, leave the `searched_columns` field empty.

This must be a valid [LDAP filter](#), where the string %Q will be replaced by the (escaped) search term when performing a search.

Example: (& (o=ACME) (cn=%Q*))

ldap_network_timeout (optional) the maximum time, in second, that an LDAP network operation can take. If it takes more time than that, no result is returned.

Defaults to 0.3.

ldap_timeout (optional) the maximum time, in second, that an LDAP operation can take.

Defaults to 1.0.

unique_column (optional) the column that contains a unique identifier of the entry. This is necessary for listing and identifying favorites.

For OpenLDAP, you should set this option to “entryUUID”.

For Active Directory, you should set this option to “objectGUID” and also set the “unique_column_format” option to “binary_uuid”.

unique_column_format (optional) the unique column’s type returned by the queried LDAP server. Valid values are “string” or “binary_uuid”.

Defaults to “string”.

phonebook Back-end name: phonebook

Purpose: search directory entries from a XiVO *phone book*.

Configuration Example (a file inside `source_config_dir`):

```

1 type: phonebook
2 name: my_phonebook
3 phonebook_url: https://example.org/service/ipbx/json.php/restricted/pbx_services/phonebook
4 phonebook_username: admin
5 phonebook_password: foobar
6 first_matched_columns:
7     - phonebooknumber.office.number
8     - phonebooknumber.mobile.number
9 format_columns:
10     firstname: "{phonebook.firstname}"
11     lastname: "{phonebook.lastname}"
12     number: "{phonebooknumber.office.number}"

```

phonebook_url (optional) the phonebook’s URL.

Default to `http://localhost/service/ipbx/json.php/private/pbx_services/phonebook`.

The URL to use differs depending on if you are accessing the phone book locally or remotely:

- Local: `http://localhost/service/ipbx/json.php/private/pbx_services/phonebook`
- Remote: `https://example.org/service/ipbx/json.php/restricted/pbx_services/phonebook`

phonebook_username (optional) the username to use in HTTP requests.

No HTTP authentication is tried when `phonebook_username` or `phonebook_password` are empty.

phonebook_password (optional) the password to use in HTTP requests.

phonebook_timeout (optional) the HTTP request timeout, in seconds.

Defaults to 1.0.

To be able to access the phone book of a remote XiVO, you must create a web services access user (*Configuration -> Web Services Access*) on the remote XiVO.

personal Back-end name: `personal`

Purpose: search directory entries among users' personal contacts

You should only have one source of type `personal`, because only one will be used to list personal contacts. The `personal` backend needs a working Consul installation. This backend works with the `personal` service, which allows users to add personal contacts.

The complete list of fields is in *Personal contacts*.

Configuration Example (a file inside `source_config_dir`):

```
1 type: personal
2 name: personal
3 first_matched_columns:
4   - number
5 format_columns:
6   firstname: "{firstname}"
7   lastname: "{lastname}"
8   number: "{number}"
```

`unique_column` is not configurable, its value is always `id`.

xivo Back-end name: `xivo`

Purpose: add users from a XiVO (may be remote) as directory entries

Configuration Example (a file inside `source_config_dir`):

```
1 type: xivo
2 name: my_xivo
3 confd_config:
4   https: True
5   host: xivo.example.com
6   port: 9486
7   version: 1.1
8   username: admin
9   password: password
10  timeout: 3
11 unique_column: id
12 first_matched_columns:
13   - exten
14 searched_columns:
15   - firstname
16   - lastname
17 format_columns:
```

```
18     number: "{exten}"
19     mobile: "{mobile_phone_number}"
```

confd_config:host the hostname of the XiVO (more precisely, of the xivo-confd service)

confd_config:port the port of the xivo-confd service (usually 9486)

confd_config:version the version of the xivo-confd API (should be 1.1)

Integration of XiVO dird with the rest of XiVO

Configuration values

Views In the directory displays (also in the *main configuration file* of xivo-dird, in the `views` section), the following keys are interpreted and displayed in xlet people of the XiVO Client:

title The `title` will be shown as a header for the column

type

- **agent**: the field value will be ignored and replaced by an icon showing the status of the agent assigned to the contact (e.g. green icon for logged agent, red icon for unlogged agent, ...)
- **callable**: a dropdown action on the `number` field will be added to call the field value.
- **email**: a dropdown action on the `number` field will be added to send an email to the field value.
- **favorite**: the boolean field value will be replaced by an icon showing if the status is favorite (yellow star filled) or not (yellow star empty).
- **name**: a decoration will be added to the field value (typically a color dot) showing the presence status of the contact (e.g. Disconnected, Available, Away, ...)
- **number**: only one number type can be defined per profile. The field value will be:
 - added a decoration (typically a color dot) showing the status of the phone of the contact (e.g. Offline, Ringing, Talking, ...)
 - replaced with a button to call the contact with your phone when using the mouse
- **personal**: the boolean field value will be used to show a deletion action for the contact
- **voicemail**: the voicemail number of the contact

See *People Xlet features Upgrade Notes* for an example with screenshots.

Personal contacts Here are the list of available attributes of a personal contact:

- `id`
- `company`
- `email`
- `fax`
- `firstname`
- `lastname`
- `mobile`
- `number`

To be able to edit and delete personal contacts, you need a column of type *personal* in your display.

Adding the *personal* column to your display In the web interface under *Services* → *CTI Server* → *Directories* → *Display filters*.

1. Edit the filter on which you which to enable favorites.
2. Add a column with the type *personal* and display format *personal*.

Favorites Enabling favorites in the XiVO client.

- Add a *unique_column* to your sources.
- Add a *favorite* column to your display

Adding a *unique_column* to your sources The web interface does not allow the administrator to specify the *unique_column* and *unique_column_format*. To add these configuration options, add a file to */etc/xivo-dird/sources.d* containing the *same name* than the directory definition and all missing fields.

Example:

Given an *ldap* directory source using Active Directory named *myactivedirectory*:

Fieldname	Value
location	{st}
name	{cn}
number	{telephoneNumber}
directory	Active Directory

Add a file */etc/xivo-dird/sources.d/myactivedirectory.yml* with the following content to enable favorites on this source.

```
name: myactivedirectory # the same name than the directory definition
unique_column: objectGUID
unique_column_format: binary_uuid
```

Adding the *favorite* column to your display In the web interface under *Services* → *CTI Server* → *Directories* → *Display filters*.

1. Edit the filter on which you which to enable favorites.
2. Add a column with the type *favorite* and display format *favorite*.

Customizing sources Some configuration options are not available in the web interface. To add configuration to a source that is configured in the web interface, create a file in */etc/xivo-dird/sources.d/* with the key *name* matching your web interface configuration and add all missing fields.

Example:

adding a timeout configuration to a CSV web service source

```
name: my_csv_web_service
timeout: 16
```

Launching xivo-dird

```
usage: xivo-dird [-h] [-c CONFIG_FILE] [-d] [-f] [-l LOG_LEVEL] [-u USER]

optional arguments:
  -h, --help            show this help message and exit
  -c CONFIG_FILE, --config-file CONFIG_FILE
                        The path where is the config file. Default: /etc/xivo-dird/config.yml
  -d, --debug           Log debug messages. Overrides log_level. Default:
                        False
  -f, --foreground      Foreground, don't daemonize. Default: False
  -l LOG_LEVEL, --log-level LOG_LEVEL
                        Logs messages with LOG_LEVEL details. Must be one of:
                        critical, error, warning, info, debug. Default: info
  -u USER, --user USER The owner of the process.
```

Terminology

Back-end A back-end is a connector to query a specific type of directory, e.g. one back-end to query LDAP servers, another back-end to query CSV files, etc.

Source A source is an instance of a back-end. One backend may be used multiples times to query multiple directories of the same type. For example, I could have the customer-csv and the employee-csv sources, each using the CSV back-end, but reading a different file.

Plugins A plugin is an extension point in xivo-dird. It is a way to add or modify the functionality of xivo-dird. There are currently three types of plugins:

- Back-ends to query different types of directories (LDAP, CSV, etc.)
- Services to provide different directory actions (lookup, reverse lookup, etc.)
- Views to expose directory results in different formats (JSON, XML, etc.)

API See <http://api.xivo.io>, section XiVO Dird.

XiVO dird phoned

xivo-dird-phoned is an interface to use directory service with phone. It offers a simple REST interface to authenticate a phone and search result from *XiVO dird*.

Usage xivo-dird-phoned is used through HTTP requests, using HTTP and HTTPS. Its default port is 9498 and 9499. As a user, the common operation is to search through directory from a phone. The phone need to send 2 informations:

- *xivo_user_uuid*: The XiVO user uuid that the phone is associated. It's used to search through personal contacts (see *personal*).
- *profile*: The profile that the user is associated. It's used to format results as configured.

Note: Since most phones don't support HTTPS, a small protection is to configure *authorized_subnets* in *Configuration Files* or in *Services* → *General settings* → *Phonebook* → *Hosts*

Launching xivo-dird-phoned On command line, type `xivo-dird-phoned -h` to see how to use it.

Purge Logs

Keeping records of personal communications for long periods may be subject to local legislation, to avoid personal data retention. Also, keeping too many records may become resource intensive for the server. To ease the removal of such records, `xivo-purge-db` is a process that removes old log entries from the database. This allows keeping records for a maximum period and deleting older ones.

By default, `xivo-purge-db` removes all logs older than a year (365 days). `xivo-purge-db` is run nightly.

Note: Please check the laws applicable to your country and modify `days_to_keep` (see below) in the configuration file accordingly.

Tables Purged The following features are impacted by `xivo-purge-db`:

- *Call Logs*
- *Call center statistics*
- *Switchboard Statistics*

More technically, the tables purged by `xivo-purge-db` are:

- `call_log`
- `cel`
- `queue_log`
- `stat_agent_periodic`
- `stat_call_on_queue`
- `stat_queue_periodic`
- `stat_switchboard_queue`

Configuration File We recommend to override the setting `days_to_keep` from `/etc/xivo-purge-db/config.yml` in a new file in `/etc/xivo-purge-db/conf.d/`.

Warning: Setting `days_to_keep` to 0 will NOT disable `xivo-purge-db`, and will remove ALL logs from your system.

See *Configuration priority* and `/etc/xivo-purge-db/config.yml` for more details.

Manual Purge It is possible to purge logs manually. To do so, log on to the target XiVO server and run:

```
xivo-purge-db
```

You can specify the number of days of logs to keep. For example, to purge entries older than 365 days:

```
xivo-purge-db -d 365
```

Usage of `xivo-purge-db`:

```
usage: xivo-purge-db [-h] [-d DAYS_TO_KEEP]

optional arguments:
  -h, --help            show this help message and exit
  -d DAYS_TO_KEEP, --days_to_keep DAYS_TO_KEEP
                        Number of days data will be kept in tables
```

Maintenance After an execution of `xivo-purge-db`, postgresql's [Autovacuum Daemon](#) should perform a `VACUUM ANALYZE` automatically (after 1 minute). This command marks memory as reusable but does not actually free disk space, which is fine if your disk is not getting full. In the case when `xivo-purge-db` hasn't run for a long time (e.g. upgrading to 15.11 or when `days_to_keep` is decreased), some administrator may want to perform a `VACUUM FULL` to recover disk space.

Warning: `VACUUM FULL` will require a service interruption. This may take several hours depending on the size of purged database.

You need to:

```
$ xivo-service stop
$ sudo -u postgres psql asterisk -c "VACUUM (FULL) "
$ xivo-service start
```

Archive Plugins In the case you want to keep archives of the logs removed by `xivo-purge-db`, you may install plugins to `xivo-purge-db` that will be run before the purge.

XiVO does not provide any archive plugin. You will need to develop plugins for your own need. If you want to share your plugins, please open a [pull request](#).

Archive Plugins (for Developers) Each plugin is a Python callable (function or class constructor), that takes a dictionary of configuration as argument. The keys of this dictionary are the keys taken from the configuration file. This allows you to add plugin-specific configuration in `/etc/xivo-purge-db/conf.d/`.

There is an example plugin in the [xivo-purge-db git repo](#).

Example Archive name: sample

Purpose: demonstrate how to create your own archive plugin.

Activate Plugin Each plugin needs to be explicitly enabled in the configuration of `xivo-purge-db`. Here is an example of file added in `/etc/xivo-purge-db/conf.d/`:

```
1 enabled_plugins:
2     archives:
3         - sample
```

sample.py The following example will be save a file in `/tmp/xivo_purge_db.sample` with the following content:

Save tables before purge. 365 days to keep!

```
1 sample_file = '/tmp/xivo_purge_db.sample'
2
3 def sample_plugin(config):
4     with open(sample_file, 'w') as output:
5         output.write('Save tables before purge. {0} days to keep!'.format(config['days_to_keep']))
```

Install sample plugin The following `setup.py` shows an example of a python library that adds a plugin to `xivo-purge-db`:

```
1 #!/usr/bin/env python
2 # -*- coding: utf-8 -*-
3
4 from setuptools import setup
5 from setuptools import find_packages
```

```

6
7
8  setup(
9      name='xivo-purge-db-sample-plugin',
10     version='0.0.1',
11
12     description='An example program',
13     packages=find_packages(),
14     entry_points={
15         'xivo_purge_db.archives': [
16             'sample = xivo_purge_db_sample.sample:sample_plugin',
17         ],
18     }
19 )

```

XiVO service

XiVO has many running services. To restart the whole stack, the *xivo-service* command can be used to make sure the service is restarted in the right order.

Usage Show all services status:

```
xivo-service status
```

Stop XiVO services:

```
xivo-service stop
```

Start XiVO services:

```
xivo-service start
```

Restart XiVO services:

```
xivo-service restart
```

The commands above will only act upon XiVO services. Appending an argument *all* will also act upon *nginx* and *postgresql*. Example:

```
xivo-service restart all
```

UDP port 5060 will be closed while services are restarting.

XiVO sysconfd

xivo-sysconfd is the system configuration server for XiVO. It does quite a few different things; here's a non exhaustive list:

- configuring network (interfaces, hostname, DNS)
- configuring high availability
- starting/stopping/restarting services
- reloading asterisk configuration
- sending some events to components (*xivo-agentd*, *xivo-agid* and *xivo-ctid*)

Configuration File Default location: `/etc/xivo/sysconfd.conf`. Format: INI.

The default location may be overwritten by the command line options.

Here's an example of the configuration file:

```
[general]
xivo_config_path = /etc/xivo
templates_path = /usr/share/xivo-sysconfd/templates
custom_templates_path = /etc/xivo/sysconfd/custom-templates
backup_path = /var/backups/xivo-sysconfd

[resolveconf]
hostname_file = /etc/hostname
hostname_update_cmd = /etc/init.d/hostname.sh start
hosts_file = /etc/hosts
resolveconf_file = /etc/resolve.conf

[network]
interfaces_file = /etc/network/interfaces

[wizard]
templates_path = /usr/share/xivo-config/templates
custom_templates_path = /etc/xivo/custom-templates

[commonconf]
commonconf_file = /etc/xivo/common.conf
commonconf_cmd = /usr/sbin/xivo-update-config
commonconf_monit = /usr/sbin/xivo-monitoring-update

[openssl]
certsdir = /var/lib/xivo/certificates

[monit]
monit_checks_dir = /usr/share/xivo-monitoring/checks
monit_conf_dir = /etc/monit/conf.d

[request_handlers]
synchronous = true

[bus]
username = guest
password = guest
host = localhost
port = 5672
exchange_name = xivo
exchange_type = topic
exchange_durable = true
```

request_handlers section

synchronous If this option is true, when xivo-sysconfd receives a request to reload the dialplan for example, it will wait for the dialplan reload to complete before replying to the request.

When this option is false, xivo-sysconfd reply to the request immediately.

Setting this option to false will speed up some operation (for example, editing a user from the web interface or from xivo-confd), but this means that there will be a small delay (up to a few seconds in the worst case) between the time you create your user and the time you can dial successfully its extension.

Ecosystem

Devices

In XiVO, there is two kind of devices:

Officially Supported Devices The officially supported devices will be supported across upgrades and phone features are guaranteed to be supported on the latest version.

xivo-provd plugins for these devices can be installed from the *“officially supported devices” repository*.

Aastra Aastra has been acquired by Mitel in 2014. In XiVO, the 6700 series and 6800 series phones are still referenced as Aastra phones, for historical and compatibility reasons.

	6731i	6735i	6737i	6739i	6755i	6757i
Provisioning	Y	Y	Y	Y	Y	Y
H-A	Y	Y	Y	Y	Y	Y
Directory XIVO	Y	Y	Y	Y	Y	Y
Funckeys	8	26	30	55	26	30
Supported programmable keys						
User with supervision function	Y	Y	Y	Y	Y	Y
Group	Y	Y	Y	Y	Y	Y
Queue	Y	Y	Y	Y	Y	Y
Conference Room with supervision function	Y	Y	Y	Y	Y	Y
General Functions						
Online call recording	N	N	N	N	N	N
Phone status	Y	Y	Y	Y	Y	Y
Sound recording	Y	Y	Y	Y	Y	Y
Call recording	Y	Y	Y	Y	Y	Y
Incoming call filtering	Y	Y	Y	Y	Y	Y
Do not disturb	Y	Y	Y	Y	Y	Y
Group interception	Y	Y	Y	Y	Y	Y
Listen to online calls	Y	Y	Y	Y	Y	Y
Directory access	Y	Y	Y	Y	Y	Y
Filtering Boss - Secretary	Y	Y	Y	Y	Y	Y
Transfers Functions						
Blind transfer	HK	Y	Y	HK	Y	Y
Indirect transfer	HK	Y	Y	HK	Y	Y
Forwards Functions						
Disable all forwarding	Y	Y	Y	Y	Y	Y
Enable/Disable forwarding on no answer	Y	Y	Y	Y	Y	Y
Enable/Disable forwarding on busy	Y	Y	Y	Y	Y	Y
Enable/Disable forwarding unconditional	Y	Y	Y	Y	Y	Y
Voicemail Functions						
Enable voicemail with supervision function	Y	Y	Y	Y	Y	Y
Reach the voicemail	Y	Y	Y	HK	Y	Y
Delete messages from voicemail	Y	Y	Y	Y	Y	Y
Agent Functions						
Connect/Disconnect a static agent	Y	Y	Y	Y	Y	Y
Connect a static agent	Y	Y	Y	Y	Y	Y
Disconnect a static agent	Y	Y	Y	Y	Y	Y
Parking Functions						
Parking	Y	Y	Y	Y	Y	Y
Continued on next page						

Table 1.2 – continued from previous page

	6731i	6735i	6737i	6739i	6755i	6757i
Parking position	Y	Y	Y	Y	Y	Y
Paging Functions						
Paging	Y	Y	Y	Y	Y	Y

6700i series Supported expansion modules:

- Aastra® M670i (for Aastra® 35i/37i/39i/53i/55i/57i)
- Aastra® M675i (for Aastra® 35i/37i/39i/55i/57i)

	6863i	6865i	6867i	6869i
Provisioning	Y	Y	Y	NT
H-A	Y	Y	Y	Y
Directory XIVO	Y	Y	Y	Y
Funckey	0	8	38	68
Supported programmable keys				
User with supervision function	N	Y	Y	Y
Group	N	Y	Y	Y
Queue	N	Y	Y	Y
Conference Room with supervision function	N	Y	Y	Y
General Functions				
Online call recording	N	Y	Y	Y
Phone status	N	Y	Y	Y
Sound recording	N	Y	Y	Y
Call recording	N	Y	Y	Y
Incoming call filtering	N	Y	Y	Y
Do not disturb	N	Y	Y	Y
Group interception	N	Y	Y	Y
Listen to online calls	N	Y	Y	Y
Directory access	N	Y	Y	Y
Filtering Boss - Secretary	N	Y	Y	Y
Transfers Functions				
Blind transfer	HK	HK	HK	HK
Indirect transfer	HK	HK	HK	HK
Forwards Functions				
Disable all forwarding	N	Y	Y	Y
Enable/Disable forwarding on no answer	N	Y	Y	Y
Enable/Disable forwarding on busy	N	Y	Y	Y
Enable/Disable forwarding unconditional	N	Y	Y	Y
Voicemail Functions				
Enable voicemail with supervision function	N	Y	Y	Y
Reach the voicemail	N	Y	Y	Y
Delete messages from voicemail	N	Y	Y	Y
Agent Functions				
Connect/Disconnect a static agent	N	Y	Y	Y
Connect a static agent	N	Y	Y	Y
Disconnect a static agent	N	Y	Y	Y
Parking Functions				
Parking	N	Y	Y	Y
Parking position	N	Y	Y	Y
Paging Functions				
Paging	N	Y	Y	Y

6800i series Supported expansion modules:

- Aastra® M680 (for Aastra® 6865i/6867i/6869i)
- Aastra® M685 (for Aastra® 6865i/6867i/6869i)

DECT Infrastructure

	RFP35	RFP36
Provisioning	N	N
H-A	N	N
Directory XIVO	N	N
Funkeys	0	0

Cisco

ATAs

	SPA122	SPA3102	SPA8000
Provisioning	Y	Y	Y
H-A	N	N	N
Directory XIVO	N	N	N
Funkeys	0	0	0

For best results, activate *DHCP Integration* on your XiVO.

These devices can be used to connect faxes. For better success with faxes some parameters must be changed. You can read the *Using analog gateways* section.

Note: If you want to manually resynchronize the configuration from the ATA device you should use the following url:

```
http://ATA_IP/admin/resync?http://XIVO_IP:8667/CONF_FILE
```

where :

- *ATA_IP* is the IP address of the ATA,
- *XIVO_IP* is the IP address of your XiVO,
- *CONF_FILE* is one of *spa3102.cfg*, *spa8000.cfg*

	7905G	7906G	7911G	7912G	7920	7921G	7940G	7941
Provisioning	Y	Y	Y	Y	Y	Y	Y	Y
H-A	Y	Y	Y	Y	NT	NT	Y	Y
Directory XIVO	FK	FK	FK	FK	N	N	FK	FK
Funkeys	0	0	0	0	0	0	1	1
					Supported programmable keys			
User with supervision function	N	N	N	N	N	N	N	N
Group	N	N	N	N	N	N	N	N
Queue	N	N	N	N	N	N	N	N
Conference Room with supervision function	N	N	N	N	N	N	N	N
General Functions								
Online call recording	N	N	N	N	N	N	N	N
Phone status	N	N	N	N	N	N	N	N
Sound recording	N	N	N	N	N	N	N	N
Call recording	N	N	N	N	N	N	N	N
Incoming call filtering	N	N	N	N	N	N	N	N
Do not disturb	SK	SK	SK	SK	N	N	SK	SK
Group interception	N	N	N	N	N	N	N	N

Table 1.4 – continued from previous page

	7905G	7906G	7911G	7912G	7920	7921G	7940G	7941G
Listen to online calls	N	N	N	N	N	N	N	N
Directory access	Y	Y	Y	Y	N	N	Y	Y
Filtering Boss - Secretary	N	N	N	N	N	N	N	N
Transfers Functions								
Blind transfer	N	N	N	N	N	N	N	N
Indirect transfer	SK	SK	SK	SK	SK	SK	SK	SK
Forwards Functions								
Disable all forwarding	N	N	N	N	N	N	N	N
Enable/Disable forwarding on no answer	N	N	N	N	N	N	N	N
Enable/Disable forwarding on busy	N	N	N	N	N	N	N	N
Enable/Disable forwarding unconditional	N	N	N	N	N	N	N	N
Voicemail Functions								
Enable voicemail with supervision function	N	N	N	N	N	N	N	N
Reach the voicemail	SK	SK	SK	SK	N	N	HK	HK
Delete messages from voicemail	N	N	N	N	N	N	N	N
Agent Functions								
Connect/Disconnect a static agent	N	N	N	N	N	N	N	N
Connect a static agent	N	N	N	N	N	N	N	N
Disconnect a static agent	N	N	N	N	N	N	N	N
Parking Functions								
Parking	N	N	N	N	N	N	N	N
Parking position	N	N	N	N	N	N	N	N
Paging Functions								
Paging	N	N	N	N	N	N	N	N

Cisco 7900 Series

Warning: These phones can only be used in SCCP mode. They are limited to the *features supported in XiVO's SCCP implementation*.

To install firmware for xivo-cisco-sccp plugins, you need to manually download the firmware files from the Cisco website and save them in the `/var/lib/xivo-provd/plugins/$plugin-name/var/cache` directory.

This directory is created by XiVO when you install the plugin (i.e. `xivo-cisco-sccp-legacy`). If you create the directory manually, the installation will fail.

Warning: Access to Cisco firmware updates requires a Cisco account with sufficient privileges. The account requires paying for the service and remains under the responsibility of the client or partner. Avencall is not responsible for these firmwares and does not offer any updates.

For example, if you have installed the `xivo-cisco-sccp-legacy` plugin and you want to install the `7940-7960-fw, networklocale` and `userlocale_fr_FR` package, you must:

- Go to <http://www.cisco.com>
- Click on “Log In” in the top right corner of the page, and then log in
- Click on the “Support” menu
- Click on the “Downloads” tab, then on “Voice & Unified Communications”
- Select “IP Telephony”, then “Unified Communications Endpoints”, then the model of your phone (in this example, the 7940G)
- Click on “Skinny Client Control Protocol (SCCP) software”
- Choose the same version as the one shown in the plugin
- Download the file with an extension ending in “.zip”, which is usually the last file in the list
- In the XiVO web interface, you’ll then be able to click on the “install” button for the firmware

The procedure is similar for the network locale and the user locale package, but:

- Instead of clicking on “Skinny Client Control Protocol (SCCP) software”, click on “Unified Communications Manager Endpoints Locale Installer”
- Click on “Linux”
- Choose the same version of the one shown in the plugin
- For the network locale, download the file named “po-locale-combined-network.cop.sgn”
- For the user locale, download the file named “po-locale-\$locale-name.cop.sgn, for example “po-locale-fr_FR.cop.sgn” for the “fr_FR” locale
- Both files must be placed in `/var/lib/xivo-provd/plugins/$plugin-name/var/cache` directory. Then install them in the XiVO Web Interface.

Note: Currently user and network locale 11.5.1 should be used for plugins xivo-sccp-legacy and xivo-cisco-sccp-9.4

	D40	D50	D70
Provisioning	Y	NYT	Y
H-A	Y	NYT	Y
Directory XIVO	N	NYT	N
Funckeys	2	14	106
Supported programmable keys			
User with supervision function	N	NYT	N
Group	Y	NYT	Y
Queue	Y	NYT	Y
Conference Room with supervision function	Y	NYT	Y
General Functions			
Online call recording	N	NYT	N
Phone status	Y	NYT	Y
Sound recording	Y	NYT	Y
Call recording	Y	NYT	Y
Incoming call filtering	Y	NYT	Y
Do not disturb	HK	NYT	HK
Group interception	Y	NYT	Y
Listen to online calls	N	NYT	N
Directory access	N	NYT	N
Filtering Boss - Secretary	Y	NYT	Y
Transfers Functions			
Blind transfer	HK	NYT	HK
Indirect transfer	HK	NYT	HK
Forwards Functions			
Disable all forwarding	Y	NYT	Y
Enable/Disable forwarding on no answer	Y	NYT	Y
Enable/Disable forwarding on busy	Y	NYT	Y
Enable/Disable forwarding unconditional	Y	NYT	Y
Voicemail Functions			
Enable voicemail with supervision function	Y	NYT	Y
Reach the voicemail	HK	NYT	HK
Delete messages from voicemail	Y	NYT	Y
Agent Functions			
Connect/Disconnect a static agent	Y	NYT	Y
Connect a static agent	Y	NYT	Y
Continued on next page			

Table 1.5 – continued from previous page

	D40	D50	D70
Disconnect a static agent	Y	NYT	Y
Parking Functions			
Parking	N	NYT	N
Parking position	N	NYT	N
Paging Functions			
Paging	Y	NYT	Y

Digium

Note: Some function keys are shared with line keys

Particularities:

- For best results, activate *DHCP Integration* on your XiVO.
- Impossible to do directed pickup using a BLF function key.
- Only supports DTMF in RFC2833 mode.
- Does not work reliably with Cisco ESW520 PoE switch. When connected to such a switch, the D40 tends to reboot randomly, and the D70 does not boot at all.
- It's important to not edit the phone configuration via the phones' web interface when using these phones with XiVO.
- Paging doesn't work.

Mitel The Mitel 6700 Series and 6800 Series SIP Phones are supported in XiVO. See the *Aastra* section.

Patton The following analog VoIP gateways are supported:

	SN4112	SN4114	SN4116	SN4118	SN4316	SN4324	SN4332
Provisioning	Y	Y	Y	Y	Y	Y	Y
H-A	Y	Y	Y	Y	Y	Y	Y

XiVO only supports configuring the FXS ports of these gateways. It does not support configuring the FXO ports. If you have a gateway on which you would like to configure the FXO ports, you'll need to write the FXO ports configuration manually by creating a *custom template* for your gateway.

It's only possible to enter a provisioning code on the first FXS port of a gateway. For example, if you have a gateway with 8 FXS ports, the first port can be configured by dialing a provisioning code from it, but ports 2 to 7 can only be configured via the XiVO web interface. Also, if you dial the *“reset to autoprov” extension* from any port, the configuration of all the ports will be reset, not just the port on which the extension was dialed. These limitations might go away in the future.

These gateways are configured with a few regional parameters (France by default). These parameters are easy to change by writing a *custom template*.

Telnet access and web access are enabled by default. You should change the default password by setting an administrator password via a XiVO “template device”.

By downloading and installing the Patton firmwares, you agree to the *Patton Electronics Company conditions*.

To provision a gateway that was previously configured manually, use the following commands on your gateway (configure mode), replacing XIVO_IP by the IP address of your XiVO server:

```
profile provisioning PF_PROVISIONING_CONFIG
destination configuration
location 1 http://XIVO_IP:8667/$(system.mac).cfg
activation reload graceful
exit
provisioning execute PF_PROVISIONING_CONFIG
```

	SoundPoint IP					SoundStation IP	
	SPIP331	SPIP335	SPIP450	SPIP550	SPIP560	SPIP650	SPIP5000
Provisioning ¹	NT ¹	Y	Y	Y	NT ¹	NT ¹	NT ¹
H-A	N	Y	N	Y	N	N	N
Directory XIVO	N	N	N	FK	N	N	N
Funkeys	N	0	2	3	3	47	0
						Supported programmable keys	
User with supervision function	NYT	N	NYT	Y	NYT	NYT	NYT
Group	NYT	N	NYT	Y	NYT	NYT	NYT
Queue	NYT	N	NYT	Y	NYT	NYT	NYT
Conference Room with supervision function	NYT	N	NYT	Y	NYT	NYT	NYT
General Functions							
Online call recording	NYT	N	NYT	N	NYT	NYT	NYT
Phone status	NYT	N	NYT	Y	NYT	NYT	NYT
Sound recording	NYT	N	NYT	Y	NYT	NYT	NYT
Call recording	NYT	N	NYT	Y	NYT	NYT	NYT
Incoming call filtering	NYT	N	NYT	Y	NYT	NYT	NYT
Do not disturb	NYT	SK	NYT	HK	NYT	NYT	NYT
Group interception	NYT	N	NYT	Y	NYT	NYT	NYT
Listen to online calls	NYT	N	NYT	Y	NYT	NYT	NYT
Directory access	NYT	N	NYT	Y	NYT	NYT	NYT
Filtering Boss - Secretary	NYT	N	NYT	Y	NYT	NYT	NYT
Transfers Functions							
Blind transfer	NYT	SK	NYT	N	NYT	NYT	NYT
Indirect transfer	NYT	SK	NYT	HK	NYT	NYT	NYT
Forwards Functions							
Disable all forwarding	NYT	N	NYT	Y	NYT	NYT	NYT
Enable/Disable forwarding on no answer	NYT	SK	NYT	Y	NYT	NYT	NYT
Enable/Disable forwarding on busy	NYT	SK	NYT	Y	NYT	NYT	NYT
Enable/Disable forwarding unconditional	NYT	SK	NYT	Y	NYT	NYT	NYT
Voicemail Functions							
Enable voicemail with supervision function	NYT	N	NYT	Y	NYT	NYT	NYT
Reach the voicemail	NYT	SK	NYT	HK	NYT	NYT	NYT
Delete messages from voicemail	NYT	N	NYT	Y	NYT	NYT	NYT
Agent Functions							
Connect/Disconnect a static agent	NYT	N	NYT	Y	NYT	NYT	NYT
Connect a static agent	NYT	N	NYT	Y	NYT	NYT	NYT
Disconnect a static agent	NYT	N	NYT	Y	NYT	NYT	NYT
Parking Functions							
Parking	NYT	N	NYT	N	NYT	NYT	NYT
Parking position	NYT	N	NYT	N	NYT	NYT	NYT
Paging Functions							
Paging	NYT	N	NYT	Y	NYT	NYT	NYT

Polycom Particularities:

- The latest Polycom firmwares can take a lot of time to download and install due to their size (~650 MiB). For this reason, these files are explicitly excluded from the XiVO backups.
- For directed call pickup to work via the BLF function keys, you need to make sure that the option *Set caller-id in dialog-info+xml notify* is enabled on your XiVO. This option is located on the *Services* → *IPBX* → *General settings* → *SIP Protocol* page, in the *Signaling* tab.

Also, directed call pickup via a BLF function key will not work if the extension number of the supervised user is different from its caller ID number.

¹These devices are marked as *Not Tested* because other similar models using the same firmware have been tested instead. If these devices ever present any bugs, they will be troubleshooted by the XiVO support team.

- Default password is **9486** (i.e. the word “xivo” on a telephone keypad).
- On the VVX101 and VVX201, to have the two line keys mapped to the same SIP line, create a *custom template* with the following content:

```
{% extends 'base.tpl' -%}

{% block remote_phonebook -%}
{% endblock -%}

{% block model_specific_parameters -%}
reg.1.lineKeys="2"
{% endblock -%}
```

This is especially useful on the VVX101 since it supports a maximum of 1 SIP line and does not support function keys.

Note: (XiVO HA cluster) BLF function key saved on the master node are not available.

Supported expansion modules:

- Polycom® VVX Color Expansion (for Polycom® VVX 300/310/400/410/500/600)
- Polycom® VVX Paper Expansion (for Polycom® VVX 300/310/400/410/500/600)
- Polycom® SoundPoint IP Backlit (for Polycom® SoundPoint 650)

Warning: Polycom® VVX® Camera are not supported.

	370	710	715	720	D725	D745	760	D765	821	870
Provisioning	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
H-A	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Directory XIVO	HK	SK	SK	HK	HK	HK	HK	HK	HK	HK
Funckeys	12	5	5	18	18	32	16	16	12	15
Supported programmable keys										
User with supervision function	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Group	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Queue	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Conference Room with supervision function	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
General Functions										
Online call recording	N	N	N	N	N	N	N	N	N	N
Phone status	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Sound recording	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Call recording	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Incoming call filtering	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Do not disturb	HK	SK	SK	HK	HK	HK	HK	HK	HK	HK
Group interception	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Listen to online calls	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Directory access	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Filtering Boss - Secretary	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Transfers Functions										
Blind transfer	Y	SK	SK	HK	HK	HK	HK	HK	HK	HK
Indirect transfer	Y	SK	SK	HK	HK	HK	HK	HK	HK	HK
Forwards Functions										
Disable all forwarding	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Enable/Disable forwarding on no answer	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y

Continued on next page

Table 1.7 – continued from previous page

	370	710	715	720	D725	D745	760	D765	821	870
Enable/Disable forwarding on busy	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Enable/Disable forwarding unconditional	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Voicemail Functions										
Enable voicemail with supervision function	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Reach the voicemail	HK	HK	HK	HK	HK	HK	HK	HK	HK	HK
Delete messages from voicemail	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Agent Functions										
Connect/Disconnect a static agent	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Connect a static agent	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Disconnect a static agent	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Parking Functions										
Parking	Y	N	N	N	N	N	N	N	Y	Y
Parking position	Y	N	N	N	N	N	N	N	Y	Y
Paging Functions										
Paging	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y

Snom Supported expansion modules:

- Snom® Vision (for Snom® 7xx series and Snom® 8xx series)
- Snom® D7 (for Snom® 7xx series)

Note: For some models, function keys are shared with line keys

There's the following known limitations/issues with the provisioning of Snom phones in XiVO:

- If you are using Snom phones with [HA](#), you should not assign multiple lines to the same device.
- The Snom D745 has limited space for function key labels: long labels might be split in a suboptimal way.
- When using a D7 expansion module, the function key label will not be shown on the first reboot or resynchronization. You'll need to reboot or resynchronize the phone a second time for the label to be shown properly.
- After a factory reset of a phone, if no language and timezone are set for the “default config device” in XiVO → *Configuration* → *Provisioning* → *Template device*, you will be forced to select a default language and timezone on the phone UI.

	T19P	T19P E2	T20P	T21P	T21P E2	T22P	T26P	T28P
Provisioning	Y	Y	Y	Y	Y	Y	Y	Y
H-A	Y	Y	Y	Y	Y	Y	Y	Y
Directory XIVO	N	Y	N	N	Y	N	N	N
Funkeys	0	0	2	2	2	3	13	16
Supported programmable keys								
User with supervision function	N	N	Y	Y	Y	Y	Y	Y
Group	N	N	Y	Y	Y	Y	Y	Y
Queue	N	N	Y	Y	Y	Y	Y	Y
Conference Room with supervision function	N	N	Y	Y	Y	Y	Y	Y
General Functions								
Online call recording	N	N	N	N	N	N	N	N
Phone status	N	N	Y	Y	Y	Y	Y	Y
Sound recording	N	N	Y	Y	Y	Y	Y	Y
Call recording	N	N	Y	Y	Y	Y	Y	Y
Incoming call filtering	N	N	Y	Y	Y	Y	Y	Y

Table 1.8 – continued from previous page

	T19P	T19P E2	T20P	T21P	T21P E2	T22P	T26P	T28P
Do not disturb	N	N	Y	SK	SK	SK	SK	SK
Group interception	N	N	Y	Y	Y	Y	Y	Y
Listen to online calls	N	N	Y	Y	Y	Y	Y	Y
Directory access	N	N	Y	Y	Y	Y	Y	Y
Filtering Boss - Secretary	N	N	Y	Y	Y	Y	Y	Y
Transfers Functions								
Blind transfer	SK	SK	HK	HK	HK	HK	HK	HK
Indirect transfer	SK	SK	HK	HK	HK	HK	HK	HK
Forwards Functions								
Disable all forwarding	N	N	Y	Y	Y	Y	Y	Y
Enable/Disable forwarding on no answer	N	N	Y	Y	Y	Y	Y	Y
Enable/Disable forwarding on busy	N	N	Y	Y	Y	Y	Y	Y
Enable/Disable forwarding unconditional	N	N	Y	Y	Y	Y	Y	Y
Voicemail Functions								
Enable voicemail with supervision function	N	N	Y	Y	Y	Y	Y	Y
Reach the voicemail	N	N	HK	HK	HK	HK	HK	HK
Delete messages from voicemail	N	N	Y	Y	Y	Y	Y	Y
Agent Functions								
Connect/Disconnect a static agent	N	N	Y	Y	Y	Y	Y	Y
Connect a static agent	N	N	Y	Y	Y	Y	Y	Y
Disconnect a static agent	N	N	Y	Y	Y	Y	Y	Y
Parking Functions								
Parking	N	N	Y	Y	Y	Y	Y	Y
Parking position	N	N	Y	Y	Y	Y	Y	Y
Paging Functions								
Paging	N	N	Y	Y	Y	Y	Y	Y

Yealink See also the list of [community supported Yealink models](#).

Regarding the W52P (DECT), there is firmware for both the base station and the handset. The base and the handset are [probably going to work if they are not using the same firmware version](#), although this does not seem to be officially recommended. By default, a base station will try to upgrade the firmware of an handset over the air (OTA) if the following conditions are met:

- Handset with firmware 26.40.0.15 or later
- Base station with firmware 25.40.0.15 or later
- Handset with hardware 26.0.0.6 or later

Otherwise, you'll have to manually upgrade the handset firmware via USB.

In all cases, you should consult the Yealink documentation on [Upgrading W52x Handset Firmware](#).

Note: Some function keys are shared with line keys

Supported expansion modules:

- Yealink® EXP38 (for Yealink® T26P/T28P)
- Yealink® EXP39 (for Yealink® T26P/T28P)
- Yealink® EXP40 (for Yealink® T46G/T48G)

Caption :

Community Supported Devices The community supported devices are only supported by the community. In other words, maintenance, bug, corrections and features are developed by members of the XiVO community. XiVO does not officially endorse support for these devices.

xivo-provd plugins for these devices can be installed from the “*community supported devices*” repository.

Aastra 6700i and 9000i series:

Model	Tested ¹	Fkeys ²	XiVO HA ³
6730i	No	8	Yes
6753i	Yes	6	Yes
6757i	Yes	30	Yes
9143i	Yes	7	Yes
9480i	No	6	Yes
9480CT	No	6	Yes

Alcatel-Lucent IP Touch series:

Model	Tested ¹	Fkeys ²	XiVO HA ³
4008 Extended Edition	Yes	4	No
4018 Extended Edition	Yes	4	No

Note that you *must not* download the firmware for these phones unless you agree to the fact it comes from a non-official source.

For the plugin to work fully, you need these additional packages:

```
apt-get install p7zip python-pexpect telnet
```

Avaya 1200 series IP Deskphones (previously known as Nortel IP Phones):

Model	Tested ¹	Fkeys ²	XiVO HA ³
1220 IP	Yes	0	No
1230 IP	No	0	No

Cisco Cisco Small Business SPA300 series:

Model	Tested ¹	Fkeys ²	XiVO HA ³
SPA301	No	1	No
SPA303	No	3	No

Note: Function keys are shared with line keys for all SPA phones

Cisco Small Business SPA500 series:

Model	Tested ¹	Fkeys ²	XiVO HA ³
SPA501G	Yes	8	No
SPA502G	No	1	No
SPA504G	Yes	4	No
SPA508G	Yes	8	No
SPA509G	No	12	No
SPA512G	No	1	No
SPA514G	No	4	No
SPA525G	Yes	5	No
SPA525G2	No	5	No

¹Tested means the device has been tested by the XiVO development team and that the developers have access to this device.

²Fkeys is the number of programmable function keys that you can configure from the XiVO web interface. It is not necessarily the same as the number of physical function keys the device has (for example, a 6757i has 12 physical keys but you can configure 30 function keys because of the page system).

³XiVO HA means the device is confirmed to work with *XiVO HA*.

The SPA500 expansion module is supported.

Cisco Small Business IP Phones (previously known as Linksys IP Phones)

Model	Tested ¹	Fkeys ²	XiVO HA ³
SPA901	No	1	No
SPA921	No	1	No
SPA922	No	1	No
SPA941	No	4	No
SPA942	Yes	4	No
SPA962	Yes	6	No

Note: You must install the firmware of each SPA9xx phones you are using since they reboot in loop when they can't find their firmware.

The SPA932 expansion module is supported.

ATAs:

Model	Tested ¹	Fkeys ²	XiVO HA ³
PAP2	No	0	No
SPA2102	No	0	No
SPA8800	No	0	No
SPA112	No	0	No

For best results, activate *DHCP Integration* on your XiVO.

Note: These devices can be used to connect Faxes. For better success with faxes some parameters must be changed. You can read the *Using analog gateways* section.

Note: If you want to manually resynchronize the configuration from the ATA device you should use the following url:

`http://ATA_IP/admin/resync?http://XIVO_IP:8667/CONF_FILE`

where :

- *ATA_IP* is the IP address of the ATA,
 - *XIVO_IP* is the IP address of your XiVO,
 - *CONF_FILE* is one of `spa2102.cfg`, `spa8000.cfg`
-

Fanvil	Model	Tested ¹	Fkeys ²	XiVO HA ³
	C62P	Yes	5	Yes

Gigaset Also known as Siemens.

Model	Tested ¹	Fkeys ²	XiVO HA ³
C470 IP	No	0	No
C475 IP	No	0	No
C590 IP	No	0	No
C595 IP	No	0	No
C610 IP	No	0	No
C610A IP	No	0	No
S675 IP	No	0	No
S685 IP	No	0	No
N300 IP	No	0	No
N300A IP	No	0	No
N510 IP PRO	No	0	No

Jitsi

Model	Tested ¹	Fkeys ²	XiVO HA ³
Jitsi	Yes	—	No

Panasonic Panasonic KX-HTXXX series:

Model	Tested ¹	Fkeys ²	XiVO HA ³
KX-HT113	No	—	No
KX-HT123	No	—	No
KX-HT133	No	—	No
KX-HT136	No	—	No

Note: This phone is for testing for the moment

Polycom

Model	Tested ¹	Fkeys ²	XiVO HA ³
SPIP320	No	0	No
SPIP321	No	0	No
SPIP330	No	0	No
SPIP430	No	0	No
SPIP501	Yes	0	No
SPIP600	No	0	No
SPIP601	No	0	No
SPIP670	No	47	No

SoundStation IP:

Model	Tested ¹	Fkeys ²	XiVO HA ³
SPIP4000	No	0	No

Others:

Model	Tested ¹	Fkeys ²	XiVO HA ³
VVX1500	No	0	No

Snom

Model	Tested ¹	Fkeys ²	XiVO HA ³
300	No	6	Yes
320	Yes	12	Yes
360	No	—	Yes
820	Yes	4	Yes
MP	No	—	Yes
PA1	No	0	Yes

Note: For some models, function keys are shared with line keys

Warning: If you are using Snom phones with HA, you should not assign multiple lines to the same device.

There's a known issue with the provisioning of Snom phones in XiVO:

- After a factory reset of a phone, if no language and timezone are set for the “default config device” in *XiVO* → *Configuration* → *Provisioning* → *Template device*, you will be forced to select a default language and timezone on the phone UI.

Technicolor Previously known as Thomson:

Model	Tested ¹	Fkeys ²	XiVO HA ³
ST2022	No	—	—
ST2030	Yes	10	Yes

Note: Function keys are shared with line keys

	Model	Tested ¹	Fkeys ²	XiVO HA ³	Plugin
Yealink	CP860	No	0	—	xivo-yealink-v72
	T23P	No	3	—	xivo-yealink-v80
	T23G	Yes	3	Yes	xivo-yealink-v80
	T27P	Yes	21	Yes	xivo-yealink-v80
	T29G	No	27	—	xivo-yealink-v80
	T49G	Yes	29	Yes	xivo-yealink-v80

Note: Some function keys are shared with line keys

	Model	Tested ¹	Fkeys ²	XiVO HA ³
Zenitel	IP station	Yes	1	No

The officially supported devices will be supported across upgrades and phone features are guaranteed to be supported on the latest version.

The community supported devices are only supported by the community. In other words, maintenance, bug, corrections and features are developed by members of the XiVO community. XiVO does not officially endorse support for these devices.

The next topics lists the officially and community supported devices. For each vendor, a table shows the various features supported by XiVO. Here's an example:

	Model X	Model Y	Model Z
Provisioning	Y	Y	Y
H-A	Y	Y	Y
Directory XiVO	N	Y	Y
Funckeys	0	2	8
	Supported programmable keys		
User with supervision function	Y	Y	Y

The rows have the following meaning:

Provisioning Is the device supported by the *auto-provisioning* system ?

H-A Is the device supported by the *high availability* system ?

Directory XiVO Is the device supported by the *remote directory* ? In other word, is it possible to consult the XiVO's remote directory from the device ?

Funckeys How many function keys can be configured on the device from the XiVO web interface ?

The number of function keys that can be configured on a device is not necessarily the same as the number of physical function keys the device has. For example, an Aastra 6757i has 12 physical keys but you can configure 30 function keys because of the page system.

Inside a table, the following legend is used:

- Y = Yes / Supported
- N = No / Not supported
- NT = Not tested
- NYT = Not yet tested

Each table also contains a section about the supported function keys. In that section, the following legend can also be used:

- FK = Funckey
- SK = SoftKey
- HK = HardKey
- MN = Menu

Function keys work using the extensions in *Services* → *Extensions*. It is important to enable the function keys you want to use. Also, the enable transfer option in the user configuration services tab must be enabled to use transfer function keys.

Administration

Advanced Configuration

This section describes the advanced system configuration.

XiVO General Settings XiVO offers the possibility to configure the general settings via the *Configuration* → *Management* → *General* page.

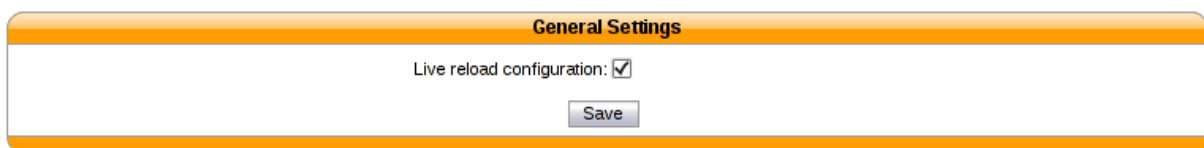


Fig. 1.33: Configure XiVO General Settings

Live reload configuration permit to reload its configuration on command received from WEBI (this option is enabled by default).

Telephony certificates XiVO offers the possibility to create and manage X.509 certificates via the *Configuration* → *Management* → *Certificates* page.

These certificates can be used for:

- enabling SIP TLS
- enabling encryption between the CTI server and the XiVO clients

For the certificate used for HTTPS, see [HTTPS certificate](#).

Fig. 1.34: Adding a certificate

Creating certificates You can add a certificate by clicking on the add button at the top right of the page. You'll then be shown this page:

You should look at the [examples](#) if you don't know which attributes to set when creating your certificates.

Removing certificates When removing a certificate, you should remove all the files related to that certificates.

Warning: If you remove a certificate that is used somewhere in XiVO, then you need to manually reconfigure that portion of XiVO.

For example, if you remove the certificate files used for SIP TLS, then you need to manually disable SIP TLS or asterisk will look for certificate file but it won't be able to find them.

Examples In the following examples, if a field is not specified than you should leave it at its default value.

Creating certificates for SIP TLS You need to create both a CA certificate and a server certificate.

CA certificate:

- *Name* : phones-CA
- *Certification authority (checkbox)* : checked
- *Autosigned* : checked
- *Valid end date* : at least one month in the future
- *Common name* : the FQDN (Fully Qualified Domain Name) of your XiVO
- *Organization* : your organization's name, or blank
- *Email* : your email or organization's email

Server certificate:

- *Name* : phones
- *Certification authority (select)* : phones-CA
- *Valid end date* : at least one month in the future

- *Common name* : the FQDN of your XiVO
- *Organization* : your organization's name, or blank
- *Email* : your email or organization's email

Creating certificate for CTI server

- *Name* : xivo-ctid
- *Autosigned* : checked
- *Valid end date* : at least one month in the future
- *Common name* : the FQDN of your XiVO
- *Organization* : your organization's name, or blank
- *Email* : your email or organization's email

Warning: You must *not* set a password for the certificate. If the certificate is password protected, the CTI server will not be able to use it.

LDAP XiVO offers the possibility to integrate LDAP servers. Once configured properly, you'll be able to search your LDAP servers from your XiVO client and from your phones (if they support this feature).

Note: This page describes how to add LDAP servers as sources of contacts. For other sources of contacts, see [Directories](#).

Add a LDAP Server You can add a LDAP server by clicking on the add button at the top right corner of the *Configuration* → *Management* → *LDAP Servers* page. You'll then be shown this page:

LDAP Servers > Add

Name:

Host:

Port:

Security layer:

Protocol version:

Description:

Fig. 1.35: Adding a LDAP server

Enter the following information:

- Name: the server's display name
- Host: the hostname or IP address
- Port: the port number (default: 389)
- Security layer: select SSL if it is activated on your server and you want to use it (default: disabled)
 - SSL means TLS/SSL (doesn't mean StartTLS) and port 636 should then be used
- Protocol version: the LDAP protocol version (default: 3)

Warning: When editing an LDAP server, you'll have to restart the CTI server for the changes to be taken into account.

Notes on SSL/TLS usage If you are using SSL with an LDAP server that is using a CA certificate from an unknown certificate authority, you'll have to put the certificate file as a single file ending with `.crt` into `/usr/local/share/ca-certificates` and run `update-ca-certificates`.

You also need to make sure that the `/etc/ldap/ldap.conf` file contains a line `TLS_CACERT /etc/ssl/certs/ca-certificates.crt`.

After that, restart `spawn-fcgi` with `service spawn-fcgi restart`.

Also, make sure to use the FQDN of the server in the host field when using SSL. The host field must match exactly what's in the CN attribute of the server certificate.

Add a LDAP Filter Next thing to do after adding a LDAP server is to create a LDAP filter via the *Services* → *IPBX configuration* → *LDAP Filters* page.

You can add a LDAP filter by clicking on the add button at the top right of the page. You'll then be shown this page:

The screenshot shows a web interface titled "LDAP filters > Edit". It contains several input fields: "Name" with the value "test", "LDAP Server" with a dropdown menu showing "debian-ldap (192.168.32.194)", "User" with "uid=alice,ou=people", "Password" with "foobar", "Base DN" with "ou=people,dc=example,dc=org", "Filter" (empty), and "Description" (empty text area). A "Save" button is located at the bottom of the form.

Fig. 1.36: Adding a LDAP Filter

Enter the following information:

- Name: the filter's display name
- LDAP server: the LDAP server this filter applies to
- User: the dn of the user used to do search requests
- Password: the password of the given user

- Base DN: the base dn of search requests
- Filter: if specified, *it replace the default filter*

Use a Custom Filter In some cases, you might have to use a custom filter for your search requests instead of the default filter.

In custom filters, occurrence of the pattern %Q is replaced by what the user entered on its phone.

Here's some examples of custom filters:

- `cn=%Q*`
- `& (cn=%Q*) (mail=*@example.org)`
- `| (cn=%Q*) (displayName=%Q*)`

Add a Directory Definition The next step is to add a directory definition for the LDAP filter you just created. See the [directories](#) section for more information.

Here's an example of an LDAP directory definition:

Update directories

Name:

URI:

Delimiter:

Direct match:

Match reverse directories:

Mapped fields:

Fieldname	Value
<input type="text" value="display_name"/>	<input type="text" value="{cn}"/>
<input type="text" value="phone"/>	<input type="text" value="{telephoneNumber}"/>
<input type="text" value="firstname"/>	<input type="text" value="{givenName}"/>
<input type="text" value="lastname"/>	<input type="text" value="{sn}"/>

Description

You need to restart the Dird server to apply changes.

Fig. 1.37: *Services → IPBX → IPBX configuration → LDAP filters*

If a custom filter is defined in the LDAP filter configuration, the fields in *direct match* will be added to that filter using an `&`. To only use the *filter* field of your LDAP filter configuration, do not add any *direct match* fields in your directory definition.

Example:

- Given an LDAP filter with *filter* `st=Canada`
- Given a directory definition with a *direct match* `cn,o`
- Then the resulting filter when doing a search will be `& (st=Canada) (| (cn=%Q*) (o=%Q*))`

Boss Secretary Filter

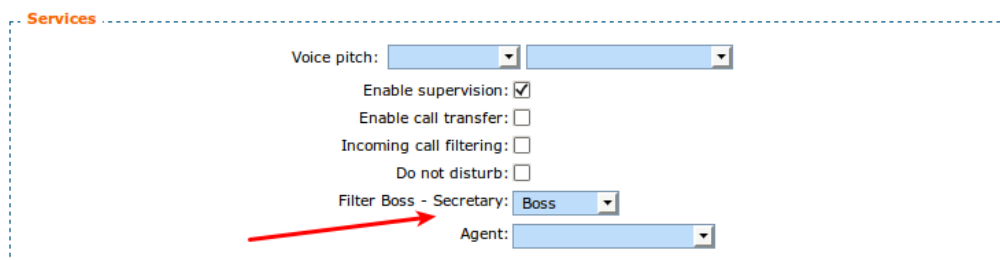
The boss secretary filter allow to set a secretary or a boss role to a user. Filters can then be created to filter calls directed to a boss using different strategies.

Quick Summary

In order to be able to use the boss secretary filter you have to :

- Select a boss role for one the users
- Select a secretary role for one of the users
- Create a filter to set a strategy for this boss secretary filter
- Add a function key for the user boss and the user secretary

Defining a Role The secretary or boss role can be set in the user's configuration page under the service tab. To use this feature, at least one boss and one secretary must be defined.



The screenshot shows a configuration page titled 'Services'. It contains several settings: 'Voice pitch' with two dropdown menus, 'Enable supervision' (checked), 'Enable call transfer' (unchecked), 'Incoming call filtering' (unchecked), 'Do not disturb' (unchecked), and 'Filter Boss - Secretary' (set to 'Boss'). A red arrow points to the 'Agent' dropdown menu below the 'Filter Boss - Secretary' setting.

Creating a Filter The filter is used to associate a boss to one or many secretaries and to set a ring strategy. The call filter is added in the *Services* → *IPBX* → *Call management* → *Call filters* page.

Different ringing strategies can be applied :

- Boss rings first then all secretaries one by one
- Boss rings first then secretaries are all ringing simultaneously
- Secretaries ring one by one
- Secretaries are all ringing simultaneously
- Boss and secretaries are ringing simultaneously
- Change the caller id if the secretary wants to know which boss was initially called

When one of serial strategies is used, the first secretary called is the last in the list. The order can be modified by drag and drop in the list.

Usage The call filter function can be activated and deactivated by the boss or the secretary using the *37 extension. The extension is defined in *IPBX services* > *Extensions*.

The call filter has to be activated for each secretary if more than one is defined for a given boss.

The extension to use is *37<callfilter member id>.

In this example, you would set 2 Func Keys *373 and *374 on the Boss.

On the secretary Jina LaPlante you would set *373.

On the secretary Ptit Nouveau you would set *374.

7

Name: Bos secretary

Context: Default (default)

Call from: All

Mode: Secretaries simultaneously

CallerID mode : Secretaries in serial

Identity: Bruce Willis

Ringing time: Unlimited

Items selected	Remove all		Add all
		Jean-Yves LEBLEU	+

Secretary

2 items selected	Remove all		Add all
↑ Jina LaPlante (*373)	-	John Smith	+
↑ Ptit Nouveau (*374)	-		

Function Keys A more convenient way to active the boss secretary filter is to assign a function key on the boss' phone or the secretary's phone. In the user's configuration under `Func Keys`. A function key can be added for each secretaries of a boss.

If supervision is activated, the key will light up when filter is activated for this secretary. If a secretary also has a function key on the same boss/secretary combination the function key's BLF will be in sync between each phones.

Warning: With SCCP phones, you must configure a custom `Func Keys`.

Call Completion

The call completion feature (or CCSS, for Call Completion Supplementary Services) in XiVO allows for a caller to be automatically called back when a called party has become available.

1. To illustrate, let's say Alice attempts to call Bob.
2. Bob is currently on a phone call with Carol, though, so Bob rejects the call from Alice
3. Alice then dials *40 to request call completion.
4. Once Bob has finished his phone call, Alice will be automatically called back by the system.
5. When she answers, Bob will be called on her behalf.

This feature has been introduced in XiVO in version 14.17.

Description Call completion can be used in two scenarios:

- when the called party is busy (Call Completion on Busy Subscriber)
- when the called party doesn't answer (Call Completion on No Response)

We have already discussed the busy scenario in the introduction section.

Let's now illustrate the no answer scenario:

1. Alice attempts to call Bob.
2. Bob doesn't answer the phone. Alternatively, Alice hangs up before Bob has the time to answer the call.
3. Alice then dial *40 to request call completion.
4. When Bob's phone becomes busy and then is no longer busy, Alice is automatically called back.
5. When she answers, Bob will be called on her behalf.

The important thing to note here is step #4. Bob's phone needs to become busy and then no longer busy for Alice to be called back. This means that if Bob was away when Alice called him, but when he came back he did not received nor placed any call, then Alice will not be called back.

In fact, in all scenarios, after call completion has been requested by the caller, the called phone needs to transition from busy to no longer busy for the caller to be called back. This means that in the following scenario:

1. Alice attempts to call Bob.
2. Bob is currently on a phone call, so he doesn't answer the call from Alice.
3. Bob finish his call a few seconds later.
4. Alice then dials *40 to request call completion (Bob is not busy anymore).

Then, for Alice to be called back, Bob needs to become busy and then not busy.

If Alice is busy when Bob becomes not busy, then the call completion callback will only happen after both Alice and Bob are not busy.

When call completion is active, it can be cancelled by dialing the *40 extension.

Some timers governs the use of call completion. These are:

- offer timer: the time the caller has to request call completion. Defaults to 30 (seconds).
- busy available timer: when call completion on busy subscriber is requested, if this timer expires before the called party becomes available, then the call completion attempt will be cancelled. Defaults to 900 (seconds).
- no response available timer: similar to the “busy available timer”, but when call completion on no response is requested. Defaults to 900 (seconds).
- recall timer: when the caller who requested call completion is called back, how long the original caller’s phone rings before giving up. Defaults to 30 (seconds).

It’s currently impossible to modify the value of these timers in XiVO.

Special Scenarios There are four special scenarios:

- the call completion will not activate
- the call completion will activate and call back for the original called party
- the call completion will activate and call back for the rerouted called party
- the call completion will activate and call back for the original called party but fail to join him

Call completion will not activate It is not possible to activate call completion in the following two scenarios.

First scenario: Alice tries to call Bob, but Bob has currently reached its “simultaneous calls” limit. When activating call completion, Alice hears that the call completion can not be activated.

Note: The “simultaneous calls” option is configured per user via the XiVO web interface.

Second scenario: Alice tries to call Bob, but the call is redirected to Charlie.

This occurs when Bob redirects/rejects the call with any of the following:

- Unconditional call forwarding towards Charlie
- Closed schedule towards Charlie
- Call permission forbidding Alice to call Bob
- Preprocess subroutine forwarding the call towards Charlie

Call completion will activate and call back for the original called party Scenario: Alice tries to call Bob, but the call is redirected to Charlie. When activating call completion, Alice hears that the call completion is activated and eventually Alice is called back to speak with Bob.

This occurs when Bob redirects/rejects the call with any of the following:

- No-answer call forwarding towards Charlie
- Busy call forwarding towards Charlie

Call completion will activate and call back for the rerouted called party Scenario: Alice tries to call Bob, but the call is redirected to Charlie. When activating call completion, Alice hears that the call completion is activated and eventually Alice is called back to speak with Charlie.

This occurs when Bob redirects the call with any of the following:

- Boss-Secretary filter to the secretary Charlie

Call completion will activate and call back for the original called party but fail to join him Scenario: Alice tries to call Bob, but the call is redirected to Charlie. When activating call completion, Alice hears that the call completion is activated and eventually Alice is called back to speak with Bob. But when Alice answers, Bob is not called. If Alice activates call completion again, she will hear that the call completion was cancelled.

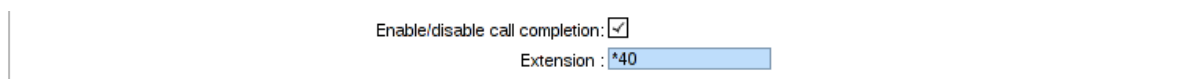
This occurs when Bob redirects/rejects the call with any of the following:

- Do Not Disturb mode
- a new call forwarding rule that was applied after Alice activated call completion:
 - Unconditional call forwarding towards Charlie
 - Closed schedule towards Charlie
 - Call permission forbidding Alice to call Bob
 - Preprocess subroutine forwarding the call towards Charlie

Limitations

- Call completion can only be used with SIP lines. It can't be used with SCCP lines.
- It can't be used with outgoing calls and incoming calls, except if these calls are passing through a customized trunk of type Local.
- It can't be used with groups or queues.
- The call completion feature can't be enabled only for a few users; either all users have access to it, or none.

Configuration The call completion extension is enabled via the *Services → IPBX → IPBX services → Extensions* page, in the *General* tab.



The screenshot shows a configuration interface with two elements: a checkbox labeled 'Enable/disable call completion:' which is checked, and a text input field labeled 'Extension :' containing the value '*40'.

Fig. 1.38: Call Completion Extension

If your XiVO has been installed in version 14.16 or earlier, then this extension is by default disabled. Otherwise, this extension is by default enabled.

Call Permissions

You can manage call permissions via the *Services → IPBX → Call management → Call permissions* page.

Call permissions can be used for:

- denying a user from calling a specific extension
- denying a user of a group from calling a specific extension
- denying a specific extension on a specific outgoing call from being called
- denying an incoming call coming from a specific extension from calling you

More than one extension can match a given call permission, either by specifying more than one extension for that permission or by using extension patterns.

You can also create permissions that allow a specific extension to be called instead of being denied. This make it possible to create a general “deny all” permission and then an “allow for some” one.

Finally, instead of unconditionally denying calling a specific extension, call permissions can instead challenge the user for a password to be able to call that extension.

As you can see, you can do a lot of things with XiVO’s call permissions. They can be used to create fairly complex rules. That said, it is probably *not* a good idea to so because it’s pretty sure you’ll get it somehow wrong.

Examples Note that when creating or editing a call permission, you must at least:

- fill the *Name* field
- have one extension / extension pattern in the *Extensions* field

Denying a user from calling a specific extension

- Add the extension in the extensions list
- In the *Users* tab, select the user

Note: User's *Rightcall Code* (*Services* -> *IPBX* -> *IPBX Settings* -> *Users* under *Services* tab) overwrite all password call permissions for the user.

Warning: The extension can be anything but it will only work if it's the extension of a user or an extension that pass through an outgoing call. It does *not* work, for example, if the extension is the number of a conference room.

Denying a user of a group from calling a specific extension First, you must create a group and add the user to this group. Note that groups aren't required to have a number.

Then,

- Add the extension in the extensions list
- In the *Groups* tab, select the group

Denying users from calling a specific extension on a specific outgoing call

- Add the extension in the extensions list
- In the *Outgoing calls* tab, select the outgoing call

Note that selecting both a user and an outgoing call for the same call permission doesn't mean the call permission applies only to that user. In fact, it means that the user can't call that extension and that the extension can't be called on the specific outgoing call. This is redundant and you will get the same result by not selecting the user.

Denying an incoming call coming from a specific extension from calling you Call permissions on incoming calls are semantically different from the other scenarios since the extension that you add to the permission will match the extension of the caller (i.e. the caller number) and *not* the extension that the caller dialed (i.e. the callee number).

- Add the extension in the extensions list.
- In the *Incoming calls* tab, select the incoming call

Call Logs

Call logs are pre-generated from CEL entries. The generation is done automatically by xivo-call-logd. xivo-call-logd is also run nightly to generate call logs from CEL that were missed by xivo-call-logd.

Note: The oldest call logs are periodically removed. See [Purge Logs](#) for more details.

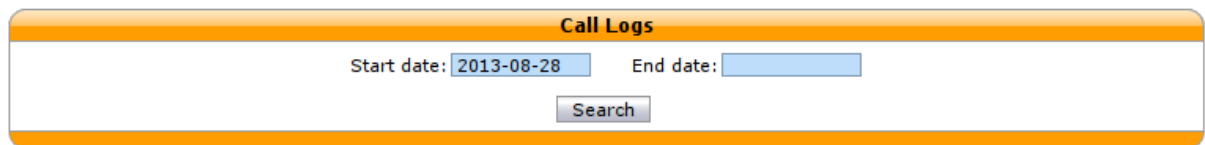


Fig. 1.39: Calls Records Dashboard

Search Dashboard Call logs can be accessed using the menu *Services* → *IPBX* → *Call management* → *Call Logs* page.

Specifying no start date returns all available call logs. Specifying a start date and no end date returns all call logs from start date until now.

Call logs are presented in a CSV format. Here's an example:

```
Call Date,Caller,Called,Period,user Field
2015-01-02T00:00:00,Alice (1001),1002,2,userfield
```

The CSV format has the following specifications:

- field names are listed on the first line
- fields are separated by commas: ,
- if there is a comma in a field value, the value is surrounded by double quotes: "
- the UTF-8 character encoding is used

REST API Call logs are also available from *xivo-confd REST API*.

Manual generation Call logs can also be generated manually. To do so, log on to the target XiVO server and run:

```
xivo-call-logs
```

To avoid running for too long in one time, the call logs generation is limited to the N last unprocessed CEL entries (default 20,000). This means that successive calls to `xivo-call-logs` will process N more CELs, making about N/10 more calls available in call logs, going further back in history, while processing new calls as well.

You can specify the number of CEL entries to consider. For example, to generate calls using the 100,000 last unprocessed CEL entries:

```
xivo-call-logs -c 100000
```

CLI Tools

XiVO comes with a collection of console (CLI) tools to help administer the server.

xivo-dist `xivo-dist` is the xivo repository sources manager. It is used to switch between distributions (production, development, release candidate, archived version). Example use cases :

- switch to production repository : `xivo-dist xivo-five`
- switch to development repository : `xivo-dist xivo-dev`
- switch to release candidate repository : `xivo-dist xivo-rc`
- switch to an archived version's repository (here 14.18) : `xivo-dist xivo-14.18`

Conference Room

Adding a conference room In this example, we'll add a conference room with number 1010.

First, you need to define a conference room number range for the "default" context via the "Services / IPBX / IPBX configuration / Contexts" page.

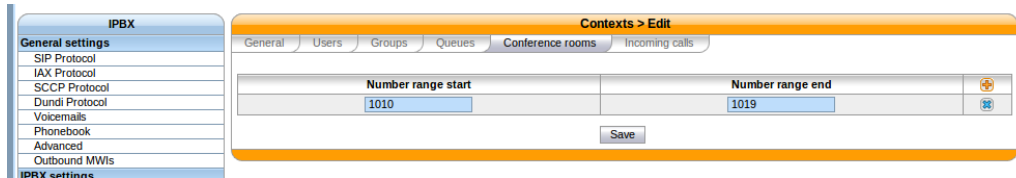


Fig. 1.40: Adding a conference room number range to the default context

You can then create a conference room via the "Services / IPBX / IPBX settings / Conference rooms" page.

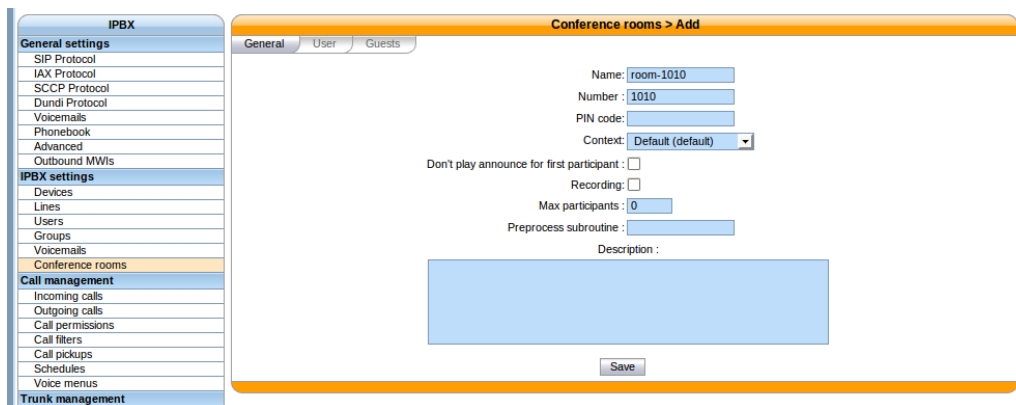


Fig. 1.41: Creating conference room 1010

In this example, we have only filled the "Name" and "Number" fields, the others have been left to their default value.

As you can see, there's quite a few options when adding / editing a conference room. Here's a description of the most common one:

General / PIN code Protects your conference room with a PIN number. People trying to join the room will be asked for the PIN code.

General / Don't play announce for first participant Don't play the "you are currently the only person in this conference" for the first participant.

General / Max participants Limits the number of participants in the conference room. A value of 0 means unlimited.

CTI Server

The CTI server configuration options can be found in the web-interface under the services tab.

General Options The general options allow the administrator to manage network connections between the CTI server and the clients.

The section named `STARTTLS options` allows the administrator to enable encrypted communications between the clients and xivo-ctid and specify the certificate and private keys to use.

If no certificate and private key is configured, xivo-ctid will use the ones located in `/usr/share/xivo-certs`.

StartTLS options

StartTLS: ☒

Certificate:

Private Key:

Parting options are used to isolate XiVO users from each other. These options should be used when using the same XiVO for different enterprises.

Context separation is based on the user's line context. A user with no line is not the member of any context and will not be able to do anything with the CTI client.

Note: xivo-dird must be restarted to take into account this parameter.

Contexts Separation: ☒

Authentication xivo-ctid uses xivo-auth to authenticate users. The default authentication backend is *xivo_user*. To change the authentication backend, add a configuration file in `/etc/xivo-ctid/conf.d` with the following content:

```
auth:
  backend: backend_name
```

where *backend* name is the name of an enabled *xivo-auth Backends Plugins*.

Presence Option In the *Status* menu, under *Presences*, you can edit presences group. The default presence group is *francais*. When editing a group, you will see a list of presences and their descriptions.

Presence Name	Description	Action
<input type="checkbox"/> > available	Available	
<input type="checkbox"/> > away	Away	
<input type="checkbox"/> > berightback	Be right back	
<input type="checkbox"/> > disconnected	Disconnected	
<input type="checkbox"/> > donotdisturb	Do not disturb	
<input type="checkbox"/> > outtolunch	Out to lunch	

To use another presence group, you can edit the CTI profile you are using and select the appropriate presence group for that profile.

Available configuration

- *Presence name* is the name of the presence
- *Display name* is the human readable representation of this presence
- *Color status* is the color associated to this presence
- *Other reachable statuses* is the list of presence that can be switched from this presence state
- *Actions* are post selection actions that are triggered by selecting this presence

Edit CTI profile

General | Xlets | Preferences

Name:

Status

Presence: ←

Phonehints:

Services

5 items selected	Remove all		Add all
↑ Enable DND	-	Enable voicemail	+
↑ Unconditional transfer to a number	-	Call record	+
↑ Transfer on busy	-	Incall record	+
↑ Transfer on no-answer	-		
↑ Call filter	-		

Edit presence

Presence name :

Display name :
The human readable name to be displayed

Color status :
Color of icon status

Other reachable statuses from this mode

Déconnecté

➤

➤

Disponible

Sorti

Parti Manger

Bientôt de retour

Action	Params
<input type="text" value="Activate DND mode"/>	<input type="text" value="true"/>
<input type="text" value="Activate pause to all queue"/>	<input type="text"/>

Actions	action	param
	Enable DND	{ 'true','false' }
	Pause agent in all queues	
	Unpause agent in all queues	
	Agent logoff	

Enable encryption To enable encryption of CTI communications between server and clients, you have to enable STARTTLS in *CTI Server* → *General settings* → *General*

Custom certificates can be added in *Configuration* → *Certificates* and used in *CTI Server* → *General settings* → *General*

In your XiVO Client, in the menu *XiVO Client* → *Configure* → *Connection*, click on the lock icon.

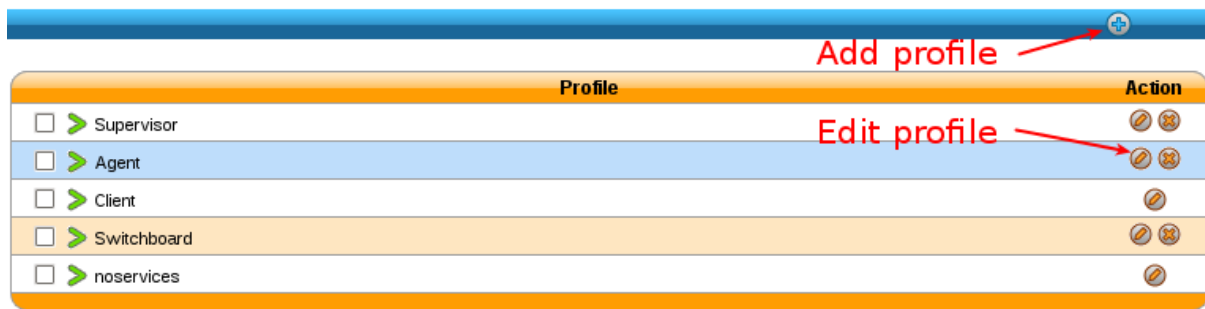
Note: A client which chooses to use encryption will not be able to connect to a server that does not have STARTTLS enabled.

Warning: For now, there is no mechanism for strong authentication of the server. The connection is encrypted, but the identity of the server is not verified.

CTI profiles The CTI profiles define which features are made available to a user. You can configure which profile will be used by a user in the menu *IPBX* → *PBX Settings* → *Users*:

The screenshot shows the 'Users > Edit' interface for a user named 'Alice Wonderland'. The interface has several tabs: General, Lines, No answer, Services, Voicemail, Groups, and Func Keys. The 'General' tab is selected. On the left, there is a placeholder for a user picture. On the right, there are various configuration fields: First name (Alice), Last name (Wonderland), User picture (with a 'Browse...' button), Mobile phone number, Schedules (dropdown), Ringing time (30 seconds), Simultaneous calls (5), On-Hold Music (default), Language (dropdown), Timezone (dropdown), Caller ID ('Alice Wonderland'), Outgoing Caller ID (Default), Preprocess subroutine, and User field. Below these fields is a section titled 'XiVO Client' enclosed in a dashed blue box. Inside this section, there is a checkbox for 'Enable XiVO Client' which is checked, followed by 'Login' (alice), 'Password' (alice), and 'Profile' (Switchboard). The 'Profile' dropdown is circled in red. At the bottom of the 'XiVO Client' section is a 'Description' field with a large text area. A 'Save' button is located at the bottom of the entire form.

You can also customize the default profiles or add new profiles in the menu *CTI Server* → *Profiles*:



Xlets To choose which features are available to users using a profile, you have to select which *Xlets* will be available.

The Xlets are detailed in [Xlets](#).

The *Position* attribute determines how the Xlets will be laid out:

- *dock* will display a Xlet in its own frame. This frame can have some options:
 - *Floating* means that the frame can be detached from the main window of the CTI Client.
 - *Closable* means that the Xlet can be hidden
 - *Movable* means that the Xlet can be moved (either inside the main window or outside)
 - *Scroll* means that the Xlet will display a scroll bar if the Xlet is too large.
- *grid* will display a Xlet inside the main window, and it will not be movable. Multiple *grid* Xlets will be laid out vertically (the second below the first).
- *tab* will display a Xlet inside a tab of the Xlet *Tabber*. Thus the Xlet *Tabber* is required and can't be in a *tab* position.

The *Number* attribute gives the order of the Xlets, beginning with 0. The order applies only to Xlets having the same *Position* attribute.

Display customer informations

Sheet Configuration Sheets can be defined under *Services* → *CTI Server* → *Models* in the web interface. Once a sheet is defined, it has to be assigned to an event in the *Services* → *CTI Server* → *Events* menu.

Model The model contains the content of the displayed sheet.

Event Events are actions that trigger the defined sheet. A sheet can be assigned to many events. In that case, the sheet will be raised for each event.

CTI Server	Model	Description	Action
General settings			
General			
Profiles			
Status			
Presences			
Phone hints			
Directories			
Definitions			
Reverse directories			
Direct directories			
Display filters			
Sheets			
Models			
Events			
Control			
Restart CTI server			

Model	Description	Action
<input type="checkbox"/> > custom1	sheet_action_custom1	
<input type="checkbox"/> > dial	sheet_action_dial	
<input type="checkbox"/> > Demo	Demo sheet	
<input type="checkbox"/> > queue	sheet_action_queue	
<input type="checkbox"/> > XIVO	Modèle de fiche de base.	

Update model

General settings Sheet Systray Actions

Name :

Focus: ☐

Description

General settings You must give a name to your sheet to be able to select it later.

The `Focus` checkbox makes the XiVO Client pop up when the sheet is displayed, if the XiVO Client was hidden.

Sheets There are two different ways to configure the contents of the sheet:

- creating a custom sheet from the Qt designer. This gives you a total control on the layout of the information and allows you to save and process data entered during or after a call.
- listing the different fields and their content. The information will be automatically laid out in a linear fashion and will be read-only.

Custom sheet

Update model

General settings Sheet Systray Actions

Disabled: ☐

Qt interface:

	Field title	Field type	Default value	Display value	
<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text" value="form"/>	<input type="text"/>	<input type="text" value="qtui"/>	<input type="button" value="+"/>

Configuring the sheet The `Qt interface` field is the path to the UI file created by the Qt Designer. The path can either be a local file on your XiVO starting with `file://`, or a HTTP URL.

You must add a field with type `form` and display value `qtui` for the form to be displayed.

Create a custom sheet with Qt Designer The Qt Designer is part of the Qt development kit and is also available in the Qt Creator. They are available on the [Qt project website](#).

Here is an example of a small form created with Qt Designer.

The Qt Designer screenshot.

Warning: In Qt Designer, one must set 'vertical layout' on the top widget (right click on the top widget > Lay out > Vertical layout).

You can download the file generated by this example from Qt Designer: `example-form.ui`

Text fields (`QLineEdit`, `QLabel`, `QPlainTextEdit`) can contain variables that will be substituted. See the [variable list](#) for more information.

Contacts Fiches Fax Historique Répertoire Services Répertoire personnel Conférences

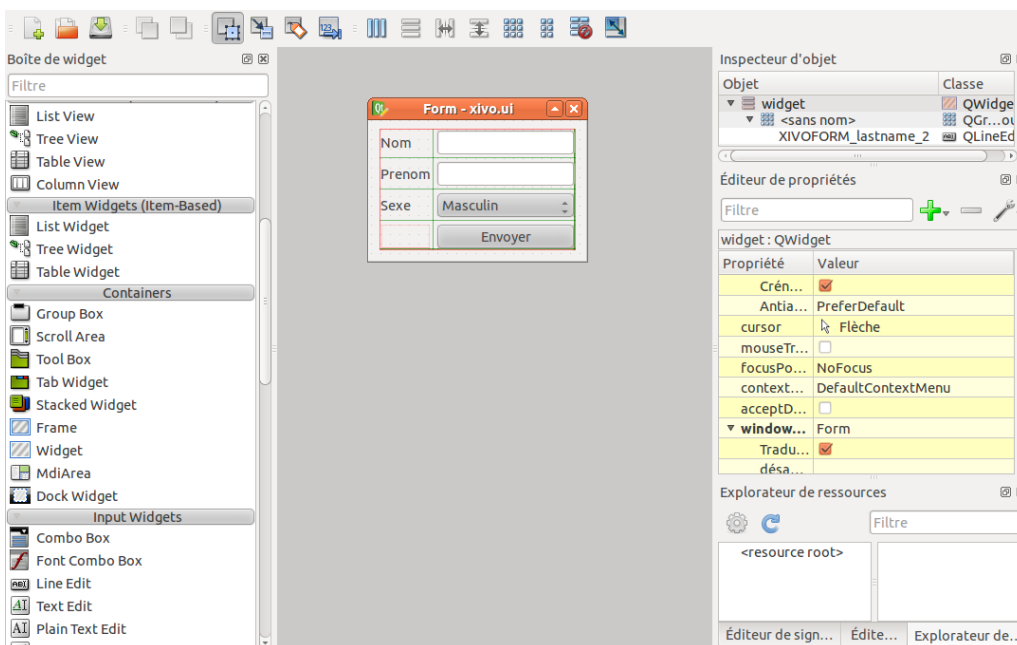
20:39:47

Nom

Prenom

Sexe Masculin

Envoyer



Update model

General settings Sheet Systray Actions

Disabled: ☒

Qt interface:

	Field title	Field type	Default value	Display value	
<input checked="" type="checkbox"/>	Nom	title		{xivo-calleridname}	
<input checked="" type="checkbox"/>	Numéro	text		{xivo-calleridnum}	
<input checked="" type="checkbox"/>	Origine	text		{xivo-origin}	

Save

List of fields Default XiVO sheet example :

Example showing all kinds of fields:

	Field title	Field type	Default value	Display value
<input checked="" type="checkbox"/>	Phone	phone	Unknown	{xivo-calleridnum}
<input checked="" type="checkbox"/>	Titre	title	test	test titre
<input checked="" type="checkbox"/>	picture	picture	picture	{xivo-callerpicture}
<input checked="" type="checkbox"/>	Test	text	text	{dp-test}
<input checked="" type="checkbox"/>		form		qtui
<input checked="" type="checkbox"/>	Uniq	text		{xivo-uniqueid}
<input checked="" type="checkbox"/>	url	url		http://git.xivo.fr
<input checked="" type="checkbox"/>	urlx	urlx		http://duckduckgo.c

Save

Each field is represented by the following parameters :

- Field title : name of your line used as label on the sheet.
- Field type : define the type of field displayed on the sheet. Supported field types :
 - title : to create a title on your sheet
 - text : show a text
 - url : a simple url link, open your default browser.
 - urlx : an url button
 - phone : create a tel: link, you can click to call on your sheet.
 - form : show the form from an ui predefined. It's an xml ui. You need to define qtui in display format.
- Default value : if given, this value will be used when all substitutions in the display value field fail.
- Display value : you can define text, variables or both. See the [variable list](#) for more information.

Variables Three kinds of variables are available :

- *xivo-* prefix is reserved and set inside the CTI server:
 - *xivo-where* for sheet events, event triggering the sheet
 - *xivo-origin* place from where the lookup is requested (did, internal, forcelookup)
 - *xivo-direction* incoming or internal
 - *xivo-did* DID number
 - *xivo-calleridnum*
 - *xivo-calleridname*
 - *xivo-calleridrdnis* contains information whether there was a transfer
 - *xivo-calleridton* Type Of Network (national, international)
 - *xivo-calledidnum*
 - *xivo-calledidname*
 - *xivo-ipbxid* (*xivo-astid* in 1.1)

- *xivo-directory* : for directory requests, it is the directory database the item has been found
- *xivo-queue* *name* queue called
- *xivo-agent* *number* agent number called
- *xivo-date* formatted date string
- *xivo-time* formatted time string, when the sheet was triggered
- *xivo-channel* asterisk channel value (for advanced users)
- *xivo-uniqueid* asterisk uniqueid value (for advanced users)
- *db-* prefixed variables are defined when the reverse lookup returns a result.

For example if you want to access to the reverse lookup full name, you need to define a field `fullname` in the directory definition, mapping to the full name field in your directory. The `{db-fullname}` will be replaced by the caller full name. Every field of the directory is accessible this way.

- *dp-* prefixed ones are the variables set through the dialplan (through UserEvent application)

For example if you want to access from the dialplan to a variable `dp-test` you need to add in your dialplan this line (in a subroutine):

```
UserEvent(dialplan2cti,UNIQUEID: ${UNIQUEID},CHANNEL: ${CHANNEL},VARIABLE: test,VALUE: "Salut
```

The `{dp-test}` displays Salut.

Sending informations during/after a call After showing a sheet, the XiVO Client can also send back information to XiVO for post-processing or archiving.

Here are the requirements:

- The sheet must contain a button named `save` to submit information
- Supported widgets:
 - `QCalendarWidget`
 - `QCheckBox`
 - `QComboBox`
 - `QDateEdit`
 - `QDateTime`
 - `QDateTimeEdit`
 - `QDoubleSpinBox`
 - `QLabel`
 - `QLineEdit`
 - `QList`
 - `QPlainTextEdit`
 - `QRadioButton`
 - `QSpinBox`
 - `QTimeEdit`
- Fields must have their name starting with `XIVOFORM_`

If you want to send information that is not visible, you can make the widget invisible on the sheet:

- change the `maximumWidth` or `maximumHeight` property to 0

- edit the `.ui` file and add the following property to the widget:

```
<property name="visible">
  <bool>false</bool>
</property>
```

When a CTI client submits a custom sheet, a `call_form_result` event is published on the event bus.

Systray Mostly the same syntax as the sheet with less field types available (title, body). A Systray popup will display a single title (the last one added to the list of fields) and zero, one or more fields of type 'body'.

Field title	Field type	Default value	Display value
Nom	title		{xivo-calledidname}
Numéro	body		{xivo-calleridnum}
Origine	body		{xivo-origin}

Save

Warning: The popup message on MacOSX works with Growl <http://growl.info>. We could get simple sheet popup to work using the free Growl Fork <http://www.macupdate.com/app/mac/41038/growl-fork> Note that this is not officially supported.

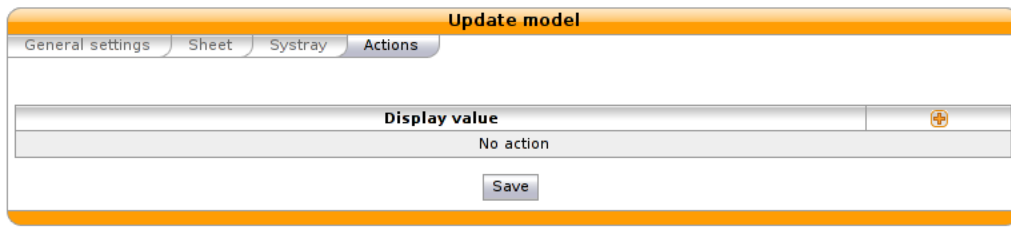
Actions The action is for the xivo client, so if you configure an action, please be sure you understand it's executed *by the client*. You need to allow this action in the client configuration too (menu *XiVO Client -> Configure*, tab *Functions*, tick option *Customer Info* and in sub-tab *Customer Info* tick the option *Allow the Automatic Opening of URL*).

The field in this tab receives the URL that will be displayed in your browser. You can also use variable substitution in this field.

- `http://example.org/foo` opens the URL on the default browser
- `http://example.org/{xivo-did}` opens the URL on the default browser, after substituting the `{xivo-did}` variable. If the substitution fails, the URL will remain `http://example.org/{xivo-did}`, i.e. the curly brackets will still be present.
- `http://example.org/{xivo-did}?origin={xivo-origin}` opens the URL on the default browser, after substituting the variables. If at least one of the substitution is successful, the failing substitutions will be replaced by an empty string. For example, if `{xivo-origin}` is replaced by 'outcall' but `{xivo-did}` is not substituted, the resulting URL will be `http://example.org/?origin=outcall`
- `tcp://x.y.z.co.fr:4545/?var1=a1&var2=a2` connects to TCP port 4545 on x.y.z.co.fr, sends the string `var1=a1&var2=a2`, then closes
- `udp://x.y.z.co.fr:4545/?var1=a1&var2=a2` connects to UDP port 4545 on x.y.z.co.fr, sends the string `var1=a1&var2=a2`, then closes

Note: any string that would not be understood as an URL will be handled like and URL it is a process to launch and will be executed as it is written

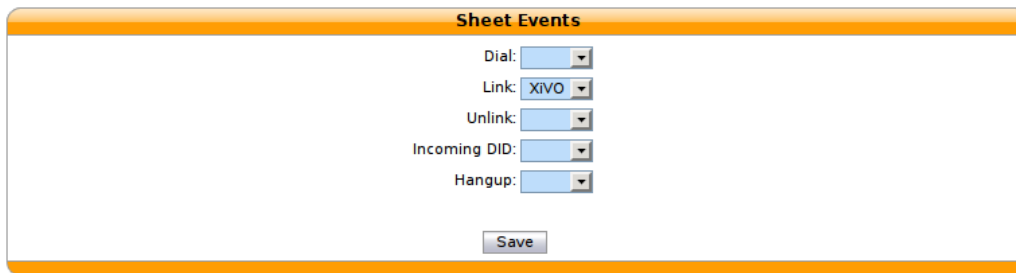
For `tcp://` and `udp://`, it is a requirement that the string between `/` and `?` is empty. An extension could be to define other serialization methods, if needed.



Event configuration You can configure a sheet when a specific event is called. For example if you want to receive a sheet when an agent answers to a call, you can choose a sheet model for the Link event.

The following events are available :

- Dial: When the member's phone starts ringing for calls on a group or queue or when the user receives a call
- Link: When a user or agent answers a call
- Unlink: When a user or agent hangup a call received from a queue
- Incoming DID: Received a call in a DID
- Hangup: Hangup the call



The informations about a call are displayed via the XiVO Client on forms called sheets.

Example: Display a Web page when an agent answers a call The first step is to assign the URL to a dialplan variable. Go in the *Services* → *IPBX* → *Configuration files* and create a new file called `setsheeturl.conf`. In this file, put the following:

```
[setsheeturl]
exten = s,1,NoOp(Starting Set Sheet URL)
same  = n,Set(SHEET_URL_CTI=http://documentation.xivo.solutions)
same  = n,UserEvent(dialplan2cti,UNIQUEID: ${UNIQUEID},CHANNEL: ${CHANNEL},VARIABLE: mysheeturl,V
same  = n,Return()
```

You can replace `documentation.xivo.solutions` by the URL you want.

The second step is to set the URL when the call is queued. To do that, we will use a preprocessing subroutine. This is configured in the queue configuration : go to *Services* → *Call center* → *Queues* and edit the queue. Set the field *Preprocessing subroutine* to `setsheeturl` (the same as above).

The third step is to configure the sheet to open the wanted URL. Go to *Services* → *CTI Server* → *Sheets* → *Models* and create a new sheet. Keep the default for everything except the Action tab, add a field and set it to `{dp-mysheeturl}` (the same as above).

The fourth and final step is to trigger the sheet when the agent answers the queued call. Go to *Services* → *CTI Server* → *Sheets* → *Events* and link the event *Agent* linked to the sheet you just created.

That's it, you can assign agents to your queue, log the agents and make them answer calls with the XiVO Client opened, and your browser should open the specified URL.

Devices

Synchronize a device First you have to display the list of devices.



Fig. 1.42: Click on the synchronize button for a device.





	MAC	IP	Vendor	Model	Plugin	Action
<input type="checkbox"/>	00:14:7f:e1:37:62	10.97.5.100	Technicolor	ST2030	xivo-technicolor-ST2030-2.74	
<input type="checkbox"/>	00:08:5d:13:ca:05	10.97.5.102	Aastra	6739i	xivo-aastra-3.2.2.56	
<input type="checkbox"/>	00:0e:08:dd:64:2e	10.97.5.103	Cisco	SPA962	xivo-cisco-spa-legacy	
<input type="checkbox"/>	00:14:7f:e1:42:b3	10.97.5.104	Technicolor	ST2030	xivo-technicolor-ST2030-2.74	

Fig. 1.43: List devices

You will see a pop-up to confirm synchronization Click on the <ok> button.

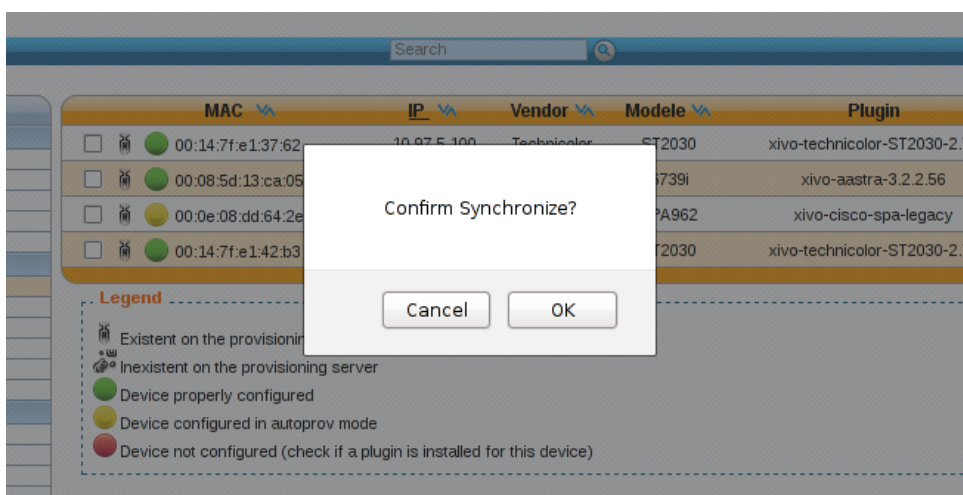


Fig. 1.44: Alert confirm synchronize

You must wait until the full synchronization process has completed to determine the state returned back from the device. This can take several seconds. It is important to wait and do nothing during this time.

If synchronization is successful, a green information balloon notifies you of success.

If synchronization fails, a red information balloon warns you of failure.

Synchronize multiple devices

Warning: When using multiple synchronization, the individual return states will not be displayed.

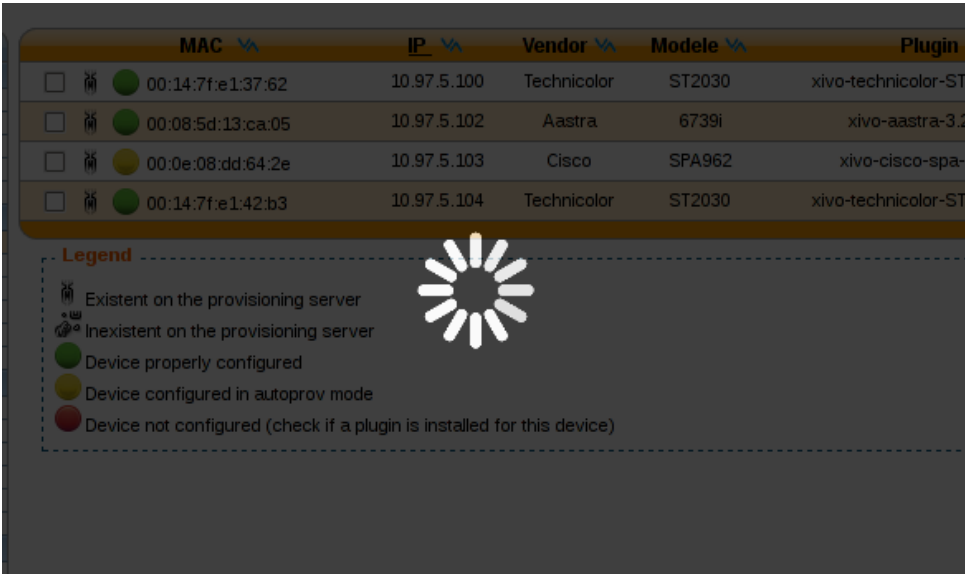
Select the devices you want to synchronize by checking the boxes.

A pop-up will appear requesting confirmation.

If mass synchronization was successfully sent to the devices, a green information balloon notifies you of success.

Directories

This page documents how to add and configure directories from custom sources. Directories added from custom sources can be used for lookup via the *XiVO Client*, directory feature of phones or for *reverse lookup* on incoming calls.



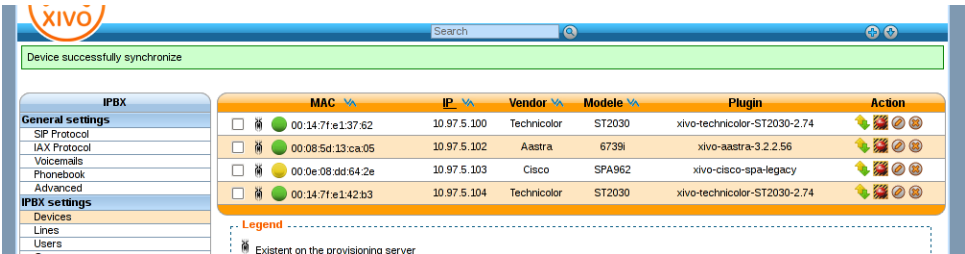
The interface shows a table with columns: MAC, IP, Vendor, Modele, and Plugin. Below the table is a legend with four items: 'Existent on the provisioning server' (green circle), 'Inexistent on the provisioning server' (yellow circle), 'Device properly configured' (green circle), and 'Device configured in autoprov mode' (yellow circle). A large white loading spinner is centered on the legend.

	MAC	IP	Vendor	Modele	Plugin
<input type="checkbox"/>	00:14:7f:e1:37:62	10.97.5.100	Technicolor	ST2030	xivo-technicolor-ST
<input type="checkbox"/>	00:08:5d:13:ca:05	10.97.5.102	Aastra	6739i	xivo-aastra-3.2
<input type="checkbox"/>	00:0e:08:dd:64:2e	10.97.5.103	Cisco	SPA962	xivo-cisco-spa-
<input type="checkbox"/>	00:14:7f:e1:42:b3	10.97.5.104	Technicolor	ST2030	xivo-technicolor-ST

Legend

- Existent on the provisioning server
- Inexistent on the provisioning server
- Device properly configured
- Device configured in autoprov mode
- Device not configured (check if a plugin is installed for this device)

Fig. 1.45: Request synchronization processing



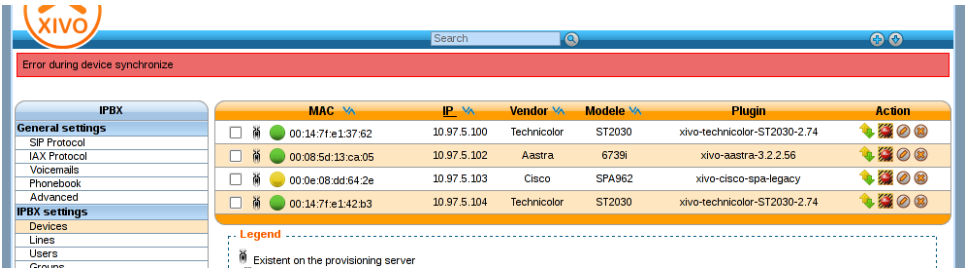
The interface shows a green banner at the top stating 'Device successfully synchronize'. Below it is a table with columns: MAC, IP, Vendor, Modele, Plugin, and Action. The table contains four rows of device information. A legend below the table indicates 'Existent on the provisioning server' with a green circle.

	MAC	IP	Vendor	Modele	Plugin	Action
<input type="checkbox"/>	00:14:7f:e1:37:62	10.97.5.100	Technicolor	ST2030	xivo-technicolor-ST2030-2.74	
<input type="checkbox"/>	00:08:5d:13:ca:05	10.97.5.102	Aastra	6739i	xivo-aastra-3.2.2.56	
<input type="checkbox"/>	00:0e:08:dd:64:2e	10.97.5.103	Cisco	SPA962	xivo-cisco-spa-legacy	
<input type="checkbox"/>	00:14:7f:e1:42:b3	10.97.5.104	Technicolor	ST2030	xivo-technicolor-ST2030-2.74	

Legend

- Existent on the provisioning server

Fig. 1.46: Device successfully synchronized



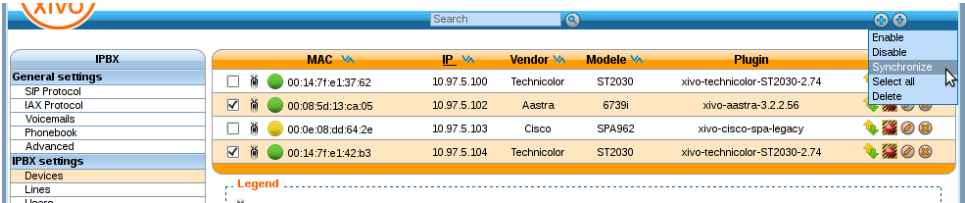
The interface shows a red banner at the top stating 'Error during device synchronize'. Below it is a table with columns: MAC, IP, Vendor, Modele, Plugin, and Action. The table contains four rows of device information. A legend below the table indicates 'Existent on the provisioning server' with a green circle.

	MAC	IP	Vendor	Modele	Plugin	Action
<input type="checkbox"/>	00:14:7f:e1:37:62	10.97.5.100	Technicolor	ST2030	xivo-technicolor-ST2030-2.74	
<input type="checkbox"/>	00:08:5d:13:ca:05	10.97.5.102	Aastra	6739i	xivo-aastra-3.2.2.56	
<input type="checkbox"/>	00:0e:08:dd:64:2e	10.97.5.103	Cisco	SPA962	xivo-cisco-spa-legacy	
<input type="checkbox"/>	00:14:7f:e1:42:b3	10.97.5.104	Technicolor	ST2030	xivo-technicolor-ST2030-2.74	

Legend

- Existent on the provisioning server

Fig. 1.47: Error during device synchronization



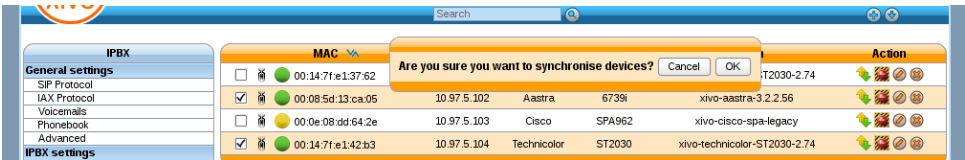
The interface shows a table with columns: MAC, IP, Vendor, Modele, Plugin, and Action. The table contains four rows of device information. A context menu is open over the 'Action' column, showing options: 'Enable', 'Disable', 'Synchronize', 'Select all', and 'Delete'. The 'Synchronize' option is highlighted.

	MAC	IP	Vendor	Modele	Plugin	Action
<input type="checkbox"/>	00:14:7f:e1:37:62	10.97.5.100	Technicolor	ST2030	xivo-technicolor-ST2030-2.74	
<input checked="" type="checkbox"/>	00:08:5d:13:ca:05	10.97.5.102	Aastra	6739i	xivo-aastra-3.2.2.56	
<input type="checkbox"/>	00:0e:08:dd:64:2e	10.97.5.103	Cisco	SPA962	xivo-cisco-spa-legacy	
<input checked="" type="checkbox"/>	00:14:7f:e1:42:b3	10.97.5.104	Technicolor	ST2030	xivo-technicolor-ST2030-2.74	

Legend

- Existent on the provisioning server

Fig. 1.48: Synchronize selected devices



The interface shows a table with columns: MAC, IP, Vendor, Modele, Plugin, and Action. The table contains four rows of device information. A confirmation dialog box is open, asking 'Are you sure you want to synchronise devices?' with 'Cancel' and 'OK' buttons. The 'Synchronize' option is highlighted in the context menu.

	MAC	IP	Vendor	Modele	Plugin	Action
<input type="checkbox"/>	00:14:7f:e1:37:62	10.97.5.100	Technicolor	ST2030	xivo-technicolor-ST2030-2.74	
<input checked="" type="checkbox"/>	00:08:5d:13:ca:05	10.97.5.102	Aastra	6739i	xivo-aastra-3.2.2.56	
<input type="checkbox"/>	00:0e:08:dd:64:2e	10.97.5.103	Cisco	SPA962	xivo-cisco-spa-legacy	
<input checked="" type="checkbox"/>	00:14:7f:e1:42:b3	10.97.5.104	Technicolor	ST2030	xivo-technicolor-ST2030-2.74	

Legend

- Existent on the provisioning server

Fig. 1.49: Synchronize selected devices confirmation

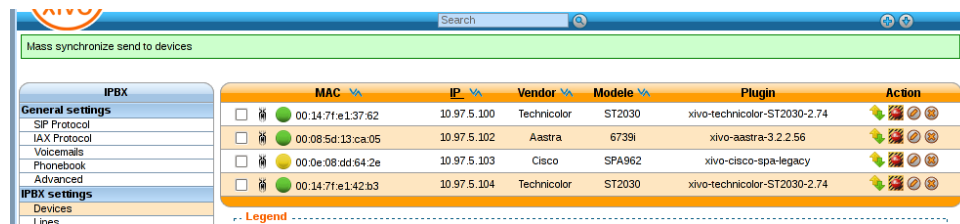


Fig. 1.50: Mass synchronization request sent successfully

An example of *adding a source* and *configuring source access* is made for each type of source:

XiVO directories This type of directory is used to query the users of a XiVO. On a fresh install, the local XiVO is already configured. The URI field for this type of directory should be the base URL of a *xivo-confd* server.

This directory type matches the *xivo* backend in *xivo-dird*.

Available fields

- id
- agent_id
- line_id
- firstname
- lastname
- email
- exten
- context
- mobile_phone_number
- userfield
- description
- voicemail_number

Example

Adding a source

Configuring source access Here is an example of a configuration where the userfield was used as a free field to store the DID number of the user and the description to store its location.

CSV File directories The source file of the directory must be in CSV format. You will be able to choose the headers and the separator in the next steps. For example, the file will look like:

```
title|firstname|lastname|displayname|society|mobilenumber|email
mr|Emmett|Brown|Brown Emmett|DMC|5555551234|emmet.brown@dmc.example.com
```

This directory type matches the *csv* backend in *xivo-dird*.

For file directories, the *Direct match* and the *Match reverse directories* must be filled with the name of the column used to match entries.

Directories Servers > Edit

Directory name:

Type:

URI:

XiVO directory

Username:

Password:

Verify certificate:

Custom CA certificate:

Description

XiVO internal users

Fig. 1.51: Configuration → Management → Directories

Update directories

Name:

URI:

Delimiter:

Direct match:

Match reverse directories:

Mapped fields:

Fieldname	Value
directory	Répertoire XIVO Interne
display_name	{firstname} {lastname}
firstname	{firstname}
lastname	{lastname}
name	{firstname} {lastname}
phone	{exten}
mobile	{mobile_phone_number}
did	{userfield}
location	{description}

Description

Répertoire XIVO Interne

You need to restart the Dird server to apply changes.

Fig. 1.52: Services → CTI Server → Directories → Definitions

Available fields Available fields are the one's contained in the CSV file.

Example csv-phonebook.csv:

```
title|firstname|lastname|displayname|society|phone|email
mr|Emmett|Brown|Brown Emmett|DMC|5555551234|emmet.brown@dmc.example.com
ms|Alice|Wonderland|Wonderland Alice|DMC|5555551235|alice.wonderland@dmc.example.com
```

Directories Servers > Add

Directory name:

Type:

URI:

Description

Fig. 1.53: *Configuration → Management → Directories*

Adding a source

Update directories

Name:

URI:

Delimiter:

Direct match:

Match reverse directories:

Mapped fields:

Fieldname	Value
directory	DMC directory
display_name	{title} {displayname}
email	{email}
firstname	{firstname}
lastname	{lastname}
phone	{phone}
society	{society}
title	{title}

Description

You need to restart the Dird server to apply changes.

Fig. 1.54: *Services → CTI Server → Directories → Definitions*

Configuring source access

CSV Web service directories The data returned by the Web service must have the same format than the file directory. In the same way, you will be able to choose the headers and the separator in the next step.

This directory type matches the *CSV web service* backend in *xivo-dird*.

For web service directories, the *Direct match* and the *Match reverse directories* must be filled with the name of the HTTP query parameter that will be used when doing the HTTP requests.

Note that the CSV returned by the Web service is not further processed.

Manual configuration needs to be done to use a secure (SSL) connection. See *CSV web service* for more details.

Available fields Available fields are the ones contained in the CSV result.

Example `http://example.org:8000/ws-phonebook` return csv:

```
title|firstname|lastname|displayname|society|phone|email
mr|Emmett|Brown|Brown Emmett|DMC|5555551234|emmet.brown@dmc.example.com
ms|Alice|Wonderland|Wonderland Alice|DMC|5555551235|alice.wonderland@dmc.example.com
```

Fig. 1.55: Configuration → Management → Directories

Adding a source

Configuring source access Given you have the following directory definition:

- *Direct match* : search
- *Match reverse directories* : phone

When a direct lookup for “Alice” is performed, then the following HTTP request:

```
GET /ws-phonebook?search=Alice HTTP/1.1
```

is emitted. When a reverse lookup for “5555551234” is performed, then the following HTTP request:

```
GET /ws-phonebook?phone=5555551234 HTTP/1.1
```

is emitted. On the reverse lookup, a filtering is performed on the result. In this example, it should have phone as column.

Phonebook directories This type of directory source is the internal phonebook of a XiVO. The *URI* field is the one used to query the phonebook.

This directory type matches the *phonebook* backend in *xivo-dird*.

Available fields

Add directory

Name:

URI:

Delimiter:

Direct match:

Match reverse directories:

Mapped fields:

Fieldname	Value
<input type="text" value="directory"/>	<input type="text" value="CSV web service example"/>
<input type="text" value="firstname"/>	<input style="background-color: #e0e0e0;" type="text" value="{firstname}"/>
<input type="text" value="lastname"/>	<input style="background-color: #e0e0e0;" type="text" value="{lastname}"/>
<input type="text" value="display_name"/>	<input style="background-color: #e0e0e0;" type="text" value="{title} {displayname}"/>
<input type="text" value="phone"/>	<input style="background-color: #e0e0e0;" type="text" value="{phone}"/>
<input type="text" value="email"/>	<input style="background-color: #e0e0e0;" type="text" value="{email}"/>
<input type="text" value="society"/>	<input style="background-color: #e0e0e0;" type="text" value="{society}"/>

Description

You need to restart the Dird server to apply changes.

Fig. 1.56: *Services → CTI Server → Directories → Definitions*

General phone book section These fields are set in the General tab of the phone book.

- phonebook.description
- phonebook.displayname
- phonebook.email
- phonebook.firstname
- phonebook.fullname (this value is automatically generated as “<firstname> <lastname>”, e.g. “John Doe”)
- phonebook.lastname
- phonebook.society
- phonebook.title
- phonebook.url

Phone numbers These are the different phone numbers that are available

- phonebooknumber.fax.number
- phonebooknumber.home.number
- phonebooknumber.mobile.number
- phonebooknumber.office.number
- phonebooknumber.other.number

Addresses Each configured address can be accessed

Address uses the following syntax *phonebookaddress.[location].[field]*, e.g. *phonebookaddress.office.zipcode*.

Locations

- home
- office
- other

Fields

- address1
- address2
- city
- country
- state
- zipcode

Example

Directories Servers > Edit

Directory name:

Type:

URI:

Description

Fig. 1.57: Configuration → Management → Directories

URI: http://localhost/service/ipbx/json.php/private/pbx_services/phonebook

Adding a source

Configuring source access Default phonebook is set in *Directories -> Definitions -> xivodir*.

XiVO dird internal phonebook This type of directory source is the internal phonebook of XiVO dird. The URI field is used to connect to the xivo-dird database.

This directory type matches the *dird_phonebook* backend in *xivo-dird*.

Example

Adding a source

Configuring available fields

Update directories

Name:

URI:

Delimiter:

Direct match:

Match reverse directories:

Mapped fields:

Fieldname	Value
<input type="text" value="company"/>	<input type="text" value="{phonebook.society}"/>
<input type="text" value="directory"/>	<input type="text" value="Répertoire XIVO Externe"/>
<input type="text" value="display_name"/>	<input type="text" value="{phonebook.displayname}"/>
<input type="text" value="firstname"/>	<input type="text" value="{phonebook.firstname}"/>
<input type="text" value="fullname"/>	<input type="text" value="{phonebook.fullname}"/>
<input type="text" value="lastname"/>	<input type="text" value="{phonebook.lastname}"/>
<input type="text" value="mail"/>	<input type="text" value="{phonebook.email}"/>
<input type="text" value="nom"/>	<input type="text" value="{phonebook.firstname} {phonebook.last"/>
<input type="text" value="phone"/>	<input type="text" value="{phonebooknumber.office.number}"/>
<input type="text" value="phone_home"/>	<input type="text" value="{phonebooknumber.home.number}"/>
<input type="text" value="phone_mobile"/>	<input type="text" value="{phonebooknumber.mobile.number}"/>
<input type="text" value="phone_other"/>	<input type="text" value="{phonebooknumber.other.number}"/>
<input type="text" value="reverse"/>	<input type="text" value="{phonebook.fullname}"/>

Description

Répertoire XIVO Externe

You need to restart the Dird server to apply changes.

Fig. 1.58: *Services → CTI Server → Directories → Definitions*

Directories Servers > Edit

Directory name:

Type:

URI:

Tenant: ?

Phonebook: ?

Description

Fig. 1.59: *Configuration → Management → Directories*
URI : The URI to connect to the xivo-dird database
Tenant : Name of the tenant, the entity is used in the default configuration
Phonebook : Name of the phonebook to use

Update directories

Name:

URI:

Delimiter:

Direct match:

Match reverse directories:

Mapped fields:

Fieldname	Value
<input type="text" value="name"/>	<input type="text" value="{firstname} {lastname}"/>
<input type="text" value="phone"/>	<input type="text" value="{number_office}"/>

Description

You need to restart the Dird server to apply changes.

Fig. 1.60: *Services → CTI Server → Directories → Definitions**Name* : Name of this source*Direct match* : Fields to match when doing a lookup*Match reverse directories* : Fields to match when doing a reverse lookup*Mapped fields* : Add fields to be compatible with a configured display**Adding the source to a profile****Note:** Phone IP should be in the authorized subnet to access the directories. See [Remote directory](#).**Adding a source****Note:** See [LDAP](#) for adding this source.You can add new data sources via the *Configuration → Management → Directories* page.

- *Directory name*: the name of the directory
- *Type*: there are 4 types of directory:
 - *XiVO*
 - *CSV File*
 - *CSV Web service*
 - *Phonebook*
- *URI*: the data source
- *Description*: (optional) a description of the directory

Configuring source access Go in *Services → CTI Server → Directories → Definitions* and add a new directory definition.

- *Name*: the name of the directory definition
- *URI*: the data source
- *Delimiter*: (optional) the field delimiter in the data source
- *Direct match*: the list used to match entries for direct lookup (comma separated)

Edit CTI context

Name:

Display filter:

Directories

3 items selected	Remove all	<input type="text"/>	Add all
↕ xivodir	—		
↕ internal	—		
↕ main	—		

Description

Contexte par défaut

You need to restart the Dird server to apply changes.

Fig. 1.61: *Services → CTI Server → Directories → Direct directories*
Directories : Add the new directory source to the profile

- *Match reverse directories*: (optional) the list used to match entries for reverse lookup (comma separated)
- *Mapped fields*: used to add or modify columns in this directory source
 - *Fieldname*: the identifier for this new field
 - *Value*: a python format string that can be used to modify the data returned from a data source

Reverse lookup It's possible to do reverse lookups on incoming calls to show a better caller ID name when the caller is in one of our directories.

Reverse lookup will only be tried if at least one of the following conditions is true:

- The caller ID name is the same as the caller ID number
- The caller ID name is “unknown”

Also, reverse lookup is performed after *caller ID number normalization* (since XiVO 13.11).

To enable reverse lookup, you need to add an entry in *Mapped fields*:

- *Fieldname*: `reverse`
- *Value*: the header of your data source that you want to see as the caller ID on your phone on incoming calls

Example

- *Match reverse directories*: `phonebooknumber.office.number,phonebooknumber.mobile.number,phonebooknumber.home.number`
- *Fieldname*: `reverse`
- *Value*: `phonebook.society`

This configuration will show the contact's company name on the caller ID name, when the incoming call will match office, mobile or home number.

Update directories

Name:

URI:

Delimiter:

Direct match:

Match reverse directories: ←

Mapped fields:

Fieldname	Value
company	{phonebook.society}
directory	Répertoire XiVO Externe
display_name	{phonebook.displayname}
firstname	{phonebook.firstname}
fullname	{phonebook.fullname}
lastname	{phonebook.lastname}
mail	{phonebook.email}
nom	{phonebook.firstname} {phonebook.last
phone	{phonebooknumber.office.number}
phone_home	{phonebooknumber.home.number}
phone_mobile	{phonebooknumber.mobile.number}
phone_other	{phonebooknumber.other.number}
← reverse	→ {phonebook.society}

Fig. 1.62: *Services → CTI Server → Directories → Definitions*

Phone directory Phone directory takes 2 *Fieldname* by default:

- `display_name`: the displayed name on the phone
- `phone`: the number to call

Examples: You will find below some useful configurations of *Mapped fields*.

Adding a name field from firstname and lastname Given a configuration where the directory source returns results with fields `firstname` and `lastname`. To add a *name* column to a directory, the administrator would add the following *Mapped fields*:

- *Fieldname*: `name`
- *Value*: `{firstname} {lastname}`

Prefixing a field Given a directory source that need a prefix to be called, a new field can be created from an existing one. To add a prefix 9 to the numbers returned from a source, the administrator would add the following *Mapped fields*:

- *Fieldname*: `number`
- *Value*: `9{number}`

Adding a static field Sometimes, it can be useful to add a field to the search results. A string can be added without any formatting. To add a *directory* field to the *xivodir* directory, the administrator would add the following *Mapped fields*:

- *Fieldname*: `directory`
- *Value*: `XiVO internal directory`

Configuring source display

XiVO Client Edit the default display filter or create your own in *Services* → *CTI Server* → *Directories* → *Display filters*.

Field title	Field type	Default value	Field name
Name	name		name
Number	number		number
Favorite	favorite		favorite
Personal	personal		
Source			directory
Mobile	callable		mobile
SDA	callable		sda
Location			location

Description

You need to restart the Dird server to apply changes.

Save

Fig. 1.63: *Services* → *CTI Server* → *Directories* → *Display filters*

Each line in the display filter will result in a header in your XiVO Client.

- *Field title*: text displayed in the header.
- *Field type*: type of the column, this information is used by the XiVO Client. (see [type description](#))
- *Default value*: value that will be used if this field is empty for one of the configured sources.
- *Field name*: name of the field in the directory definitions. The specified names should be available in the configured sources. To add new column name to a directory definition see above.

Phone The only way to configure display phone directory is through [XiVO dird configuration](#).

Adding a directory To include a directory in direct directory definition:

1. Go to *Services* → *CTI Server* → *Directories* → *Direct directories*.
2. Edit your context.
3. Select your display filter.
4. Add the directories in the *Directories* section.

To include a directory in reverse directory definition:

1. Go to *Services* → *CTI Server* → *Directories* → *Reverse directories*.
2. Add the directories to include to reverse lookups in the *Related directories* section.

Applying changes To reload the directory configuration for XiVO Client, phone lookups and reverse lookups, use *one* of these methods:

- *Services* → *IPBX* → *Control* → *Restart Dird server*
- `console service xivo-dird restart`

Directed Pickup

Directed pickup allows a user to intercept calls made to another user.

For example, if a user with number 1001 is ringing, you can dial *81001 from your phone and it will intercept (i.e. pickup) the call to this user.

The extension prefix used to pickup calls can be changed via the *Services* → *IPBX* → *IPBX services* → *Extensions* page.

Custom Line Limitation There is a case where directed pickup does not work, which is the following:

```
Given you have a user U with a line of type "customized"
Given this custom line is using DAHDI technology
Given this user is a member of group G
When a call is made to group G
Then you won't be able to intercept the call made to U by pressing *8<line number of U>
```

If you find yourself in this situation, you'll need to write a bit of dialplan.

For example, if you have the following:

- a user with a custom line with number 1001 in context default
- a custom line with interface DAHDI/g1/5551234

Then add the following, or similar:

```
[custom_lines]
exten = line1001,1,NoOp()
same = n,Set(__PICKUPMARK=1001%default)
same = n,Dial(DAHDI/g1/5551234)
same = n,Hangup()
```

And do a `dialplan reload` in the asterisk CLI.

Then, edit the line of the user and change the interface value to `Local/line1001@custom_lines`

Note that you'll need to update your dialplan if you update the number of the line or the context.

Entities

Purpose In some cases, as the telephony provider, you want different independent organisations to have their telephony served by your XiVO, e.g. different departments using the same telephony infrastructure, but you do not want each organisation to see or edit the configuration of other organisations.

Configuration In *Configuration* → *Entities*, you can create entities, one for each independant organisation.

In *Configuration* → *Users*, you can select an entity for each administrator.

Note: Once an entity is linked with an administrator, it can not be deleted. You have to unlink the entity from all administrator to be able to delete it.

For the new entity to be useful, you need to create contexts in this entity. You may need:

- an Internal context for users, groups, queues, etc.
- an Incall context for incoming calls
- an Outcall context for outgoing calls, which should be included in the Internal context for the users to be able to call external numbers

Limitations

Global Fields Some fields are globally unique and will collide when the same value is used in different entities:

- User CTI login
- Agent number
- Queue name
- Context name

An error message will appear when creating resources with colliding parameters, saying the resource already exists, even if the entity-linked administrator can not see them.

Affected Lists Only the following lists may be filtered by entity:

- Lines
- Users
- Devices
- Groups
- Voicemails
- Conference Rooms
- Incoming calls
- Call filters
- Call pickups
- Schedules
- Agents
- Queues

For the devices:

- The filtering only applies to the devices associated with a line.
- The devices in autoprov mode or not configured mode are visible by every administrator.

REST API The REST API does not have the notion of entity. When creating a resource without context via REST API, the resource will be associated to an arbitrary entity. Affected resources are:

- Contexts
- Call filters
- Group pickups
- Schedules
- Users

Fax

Fax transmission It's possible to send faxes from XiVO using the fax Xlet in the XiVO client.

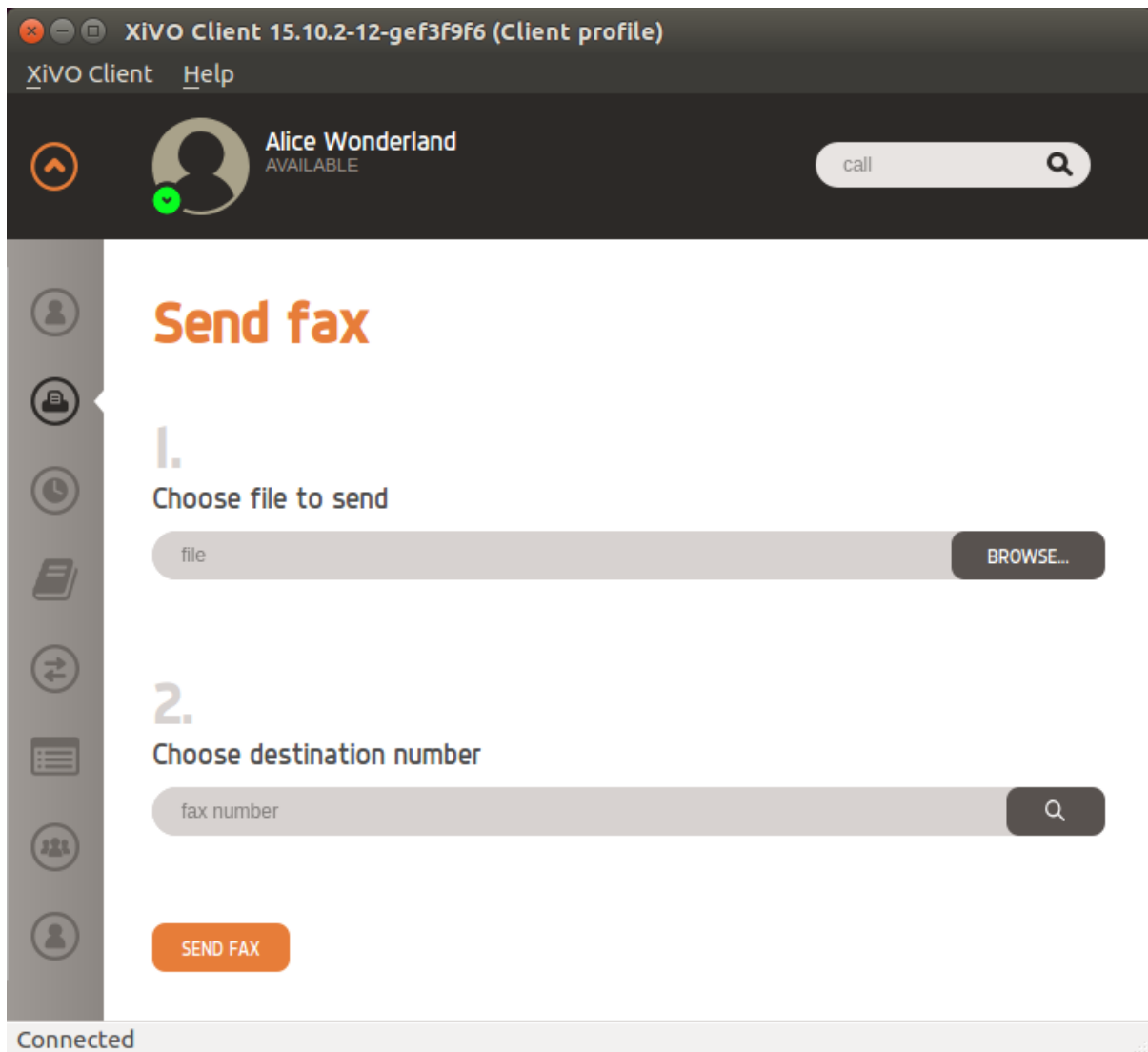


Fig. 1.64: The fax Xlet in the XiVO Client

The file to send must be in PDF format.

Fax reception

Adding a fax reception DID If you want to receive faxes from XiVO, you need to add incoming calls definition with the *Application* destination and the *FaxToMail* application for every DID you want to receive faxes from.

This applies even if you want the action to be different from sending an email, like putting it on a FTP server. You'll still need to enter an email address in these cases even though it won't be used.

Note that, as usual when adding incoming call definitions, you must first define the incoming call range in the used context.

Changing the email body You can change the body of the email sent upon fax reception by editing `/etc/xivo/mail.txt`.

The following variable can be included in the mail body:

- `%(dstnum)s`: the DID that received the fax

If you want to include a regular percent character, i.e. `%`, you must write it as `%%` in `mail.txt` or an error will occur when trying to do the variables substitution.

The `agid` service must be restarted to apply changes:

```
service xivo-agid restart
```

Changing the email subject You can change the subject of the email sent upon fax reception by editing `/etc/xivo/asterisk/xivo_fax.conf`.

Look for the `[mail]` section, and in this section, modify the value of the `subject` option.

The available variable substitution are the same as for the email body.

The `agid` service must be restarted to apply changes:

```
service xivo-agid restart
```

Changing the email from You can change the from of the email sent upon fax reception by editing `/etc/xivo/asterisk/xivo_fax.conf`.

Look for the `[mail]` section, and in this section, modify the value of the `email_from` option.

The `agid` service must be restarted to apply changes:

```
service xivo-agid restart
```

Changing the email realname You can change the realname of the email sent upon fax reception by editing `/etc/xivo/asterisk/xivo_fax.conf`.

Look for the `[mail]` section, and in this section, modify the value of the `email_realname` option.

The `agid` service must be restarted to apply changes:

```
service xivo-agid restart
```

Using the advanced features The following features are only available via the `/etc/xivo/asterisk/xivo_fax.conf` configuration file. They are not available from the web-interface.

The way it works is the following:

- you first declare some backends, i.e. actions to be taken when a fax is received. A backend name looks like `mail`, `ftp_example_org` or `printer_office`.
- once your backends are defined, you can use them in your destination numbers. For example, when someone calls the DID 100, you might want the `ftp_example_org` and `mail` backend to be run, but otherwise, you only want the `mail` backend to be run.

Here's an example of a valid `/etc/xivo/asterisk/xivo_fax.conf` configuration file:

```
[general]
tiff2pdf = /usr/bin/tiff2pdf
mutt = /usr/bin/mutt
lp = /usr/bin/lp

[mail]
subject = FAX reception to %(dstnum)s
content_file = /etc/xivo/mail.txt
email_from = no-reply+fax@xivo.solutions
email_realname = Service Fax

[ftp_example_org]
host = example.org
username = foo
password = bar
directory = /foobar

[dstnum_default]
dest = mail

[dstnum_100]
dest = mail, ftp_example_org
```

The section named `dstnum_default` will be used only if no DID-specific actions are defined.

After editing `/etc/xivo/asterisk/xivo_fax.conf`, you need to restart the `agid` server for the changes to be applied:

```
service xivo-agid restart
```

Using the FTP backend The FTP backend is used to send a PDF version of the received fax to an FTP server.

An FTP backend is always defined in a section beginning with the `ftp` prefix. Here's an example for a backend named `ftp_example_org`:

```
[ftp_example_org]
host = example.org
port = 2121
username = foo
password = bar
directory = /foobar
convert_to_pdf = 0
```

The `port` option is optional and defaults to 21.

The `directory` option is optional and if not specified, the document will be put in the user's root directory.

The `convert_to_pdf` option is optional and defaults to 1. If it is set to 0, the TIFF file will not be converted to PDF before being sent to the FTP server.

The uploaded file are named like `${XIVO_SRCNUM}-${EPOCH}.pdf`.

Using the printer backend To use the printer backend, you must have the `cups-client` package installed on your XiVO:

```
$ apt-get install cups-client
```

The printer backend uses the `lp` command to print faxes.

A printer backend is always defined in a section beginning with the `printer` prefix. Here's an example for a backend named `printer_office`:

```
[printer_office]
name = office
convert_to_pdf = 1
```

When a fax will be received, the system command `lp -d office <faxfile>` will be executed.

The `convert_to_pdf` option is optional and defaults to 1. If it is set to 0, the TIFF file will not be converted to PDF before being printed.

Warning: You need a CUPS server set up somewhere on your network.

Using the mail backend By default, a mail backend named `mail` is defined. You can define more mail backends if you want. Just look what the default mail backend looks like.

Fax detection XiVO does not currently support Fax Detection. A workaround is described in the [Fax detection](#) section.

Using analog gateways XiVO is able to provision Cisco SPA122 and Linksys SPA2102, SPA3102 and SPA8000 analog gateways which can be used to connect fax equipments. This section describes the creation of custom template *for SPA3102* which modifies several parameters.

Note: With SPA ATA plugins \geq v0.8, you **should not need** to follow this section anymore since all of these parameters are now set in the base templates of all, except for `Echo_Canc_Adapt_Enable`, `Echo_Supp_Enable`, `Echo_Canc_Enable`.

Note: Be aware that most of the parameters are or could be country specific, i.e. :

- Preferred Codec,
 - FAX Passthru Codec,
 - RTP Packet Size,
 - RTP-Start-Loopback Codec,
 - Ring Waveform,
 - Ring Frequency,
 - Ring Voltage,
 - FXS Port Impedance
-

1. Create a custom template for the SPA3102 base template:

```
cd /var/lib/xivo-provd/plugins/xivo-cisco-spa3102-5.1.10/var/templates/
cp ../../templates/base.tpl .
```

2. Add the following content before the `</flat-profile>` tag:

```

<!-- CUSTOM TPL - for faxes - START -->

{% for line_no, line in sip_lines.iteritems() %}
<!-- Dial Plan: L{{ line_no }} -->
<Dial_Plan_{{ line_no }}_ ua="na">([x*#].)</Dial_Plan_{{ line_no }}_>

<Call_Waiting_Serv_{{ line_no }}_ ua="na">No</Call_Waiting_Serv_{{ line_no }}_>
<Three_Way_Call_Serv_{{ line_no }}_ ua="na">No</Three_Way_Call_Serv_{{ line_no }}_>

<Preferred_Codec_{{ line_no }}_ ua="na">G711a</Preferred_Codec_{{ line_no }}_>
<Silence_Supp_Enable_{{ line_no }}_ ua="na">No</Silence_Supp_Enable_{{ line_no }}_>
<Echo_Canc_Adapt_Enable_{{ line_no }}_ ua="na">No</Echo_Canc_Adapt_Enable_{{ line_no }}_>
<Echo_Supp_Enable_{{ line_no }}_ ua="na">No</Echo_Supp_Enable_{{ line_no }}_>
<Echo_Canc_Enable_{{ line_no }}_ ua="na">No</Echo_Canc_Enable_{{ line_no }}_>
<Use_Pref_Codec_Only_{{ line_no }}_ ua="na">yes</Use_Pref_Codec_Only_{{ line_no }}_>
<DTMF_Tx_Mode_{{ line_no }}_ ua="na">Normal</DTMF_Tx_Mode_{{ line_no }}_>

<FAX_Enable_T38_{{ line_no }}_ ua="na">Yes</FAX_Enable_T38_{{ line_no }}_>
<FAX_T38_Redundancy_{{ line_no }}_ ua="na">1</FAX_T38_Redundancy_{{ line_no }}_>
<FAX_Passthru_Method_{{ line_no }}_ ua="na">ReINVITE</FAX_Passthru_Method_{{ line_no }}_>
<FAX_Passthru_Codec_{{ line_no }}_ ua="na">G711a</FAX_Passthru_Codec_{{ line_no }}_>
<FAX_Disable_ECAN_{{ line_no }}_ ua="na">yes</FAX_Disable_ECAN_{{ line_no }}_>
<FAX_Tone_Detect_Mode_{{ line_no }}_ ua="na">caller or callee</FAX_Tone_Detect_Mode_{{ line_no }}_>

<Network_Jitter_Level_{{ line_no }}_ ua="na">very high</Network_Jitter_Level_{{ line_no }}_>
<Jitter_Buffer_Adjustment_{{ line_no }}_ ua="na">disable</Jitter_Buffer_Adjustment_{{ line_no }}_>
{% endfor %}

<!-- SIP Parameters -->
<RTP_Packet_Size ua="na">0.020</RTP_Packet_Size>
<RTP-Start-Loopback_Codec ua="na">G711a</RTP-Start-Loopback_Codec>

<!-- Regional parameters -->
<Ring_Waveform ua="rw">Sinusoid</Ring_Waveform> <!-- options: Sinusoid/Trapezoid -->
<Ring_Frequency ua="rw">50</Ring_Frequency>
<Ring_Voltage ua="rw">85</Ring_Voltage>

<FXS_Port_Impedance ua="na">600+2.16uF</FXS_Port_Impedance>
<Caller_ID_Method ua="na">Bellcore (N.Amer,China)</Caller_ID_Method>
<Caller_ID_FSK_Standard ua="na">bell 202</Caller_ID_FSK_Standard>

<!-- CUSTOM TPL - for faxes - END -->

```

3. Reconfigure the devices with:

```
xivo-provd-cli -c 'devices.using_plugin("xivo-cisco-spa3102-5.1.10").reconfigure()'
```

4. Then reboot the devices:

```
xivo-provd-cli -c 'devices.using_plugin("xivo-cisco-spa3102-5.1.10").synchronize()'
```

Most of this template can be copy/pasted for a SPA2102 or SPA8000.

Using a SIP Trunk Fax transmission, to be successful, *MUST* use G.711 codec. Fax streams cannot be encoded with lossy compression codecs (like G.729a).

That said, you may want to establish a SIP trunk using G.729a for all other communications to save bandwidth. Here's a way to be able to receive a fax in this configuration.

Note: There are some prerequisites:

- your SIP Trunk must offer both G.729a and G.711 codecs

- your fax users must have a customized outgoing calleridnum (for the codec change is based on this variable)

1. We assume that outgoing call rules and fax users with their DID are created
2. Create the file `/etc/asterisk/extensions_extra.d/fax.conf` with the following content:

```
;; For faxes :
; The following subroutine forces inbound and outbound codec to alaw.
; For outbound codec selection we must set the variable with inheritance.
; Must be set on each Fax DID
[pre-incall-fax]
exten = s,1,NoOp(### Force alaw codec on both inbound (operator side) and outbound (analog gw)
exten = s,n,Set(SIP_CODEC_INBOUND=alaw)
exten = s,n,Set(__SIP_CODEC_OUTBOUND=alaw)
exten = s,n,Return()

; The following subroutine forces outbound codec to alaw based on outgoing callerid number
; For outbound codec selection we must set the variable with inheritance.
; Must be set on each outgoing call rule
[pre-outcall-fax]
exten = s,1,NoOp(### Force alaw codec if caller is a Fax ###)
exten = s,n,GotoIf("${CALLERID(num)}" = "0112697845"?alaw:)
exten = s,n,GotoIf("${CALLERID(num)}" = "0112697846"?alaw:end)
exten = s,n(alaw),Set(__SIP_CODEC_OUTBOUND=alaw)
exten = s,n(end),Return()
```

3. For each Fax users' DID add the following string in the Preprocess subroutine field:

```
pre-incall-fax
```

4. For each Outgoing call rule add the the following string in the Preprocess subroutine field:

```
pre-outcall-fax
```

Graphics

The Services/Graphics section gives a historical overview of a XiVO system's activity based on snapshots recorded every 5 minutes. Graphics are available for the following resources :

- CPU
- Entropy
- Interruptions
- IRQ Stats
- System Load
- Memory Usage
- Open Files
- Open Inodes
- Swap Usage

Each section is presented as a series of 4 graphics : daily, weekly, monthly and yearly history. Each graphic can be clicked on to zoom. All information presented is read only.

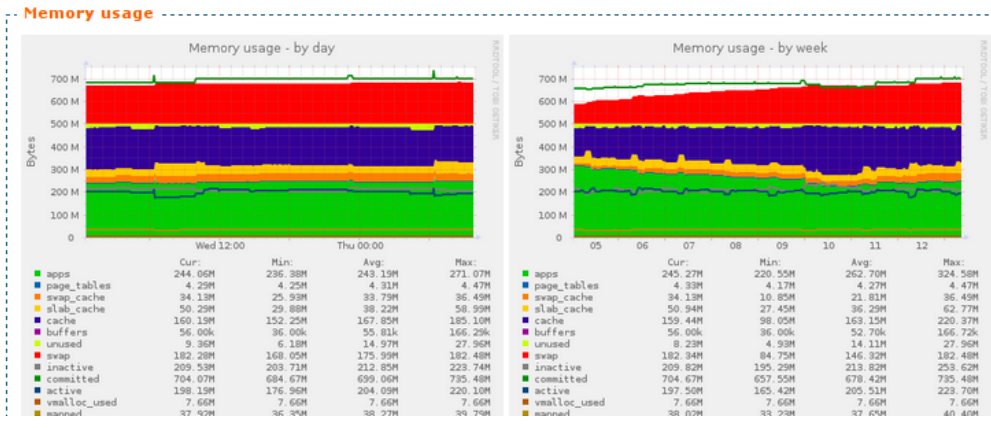
Groups

Groups are used to be able to call a set or users.

Group name cannot be general reserved in asterisk configuration.

Local timer interrupts	349.73	215.98	348.42	412.76	Local timer interrupts	350.25	99.43	363.09	7.12k
Spurious interrupts	0.00	0.00	0.00	0.00	Spurious interrupts	0.00	0.00	0.00	0.00
Performance monitoring interrupts	0.00	0.00	0.00	0.00	Performance monitoring interrupts	0.00	0.00	0.00	0.00
IRQ work interrupts	0.00	0.00	0.00	0.00	IRQ work interrupts	0.00	0.00	0.00	0.00
Rescheduling interrupts	0.00	0.00	0.00	0.00	Rescheduling interrupts	0.00	0.00	0.00	0.00
Function call interrupts	0.00	0.00	0.00	0.00	Function call interrupts	0.00	0.00	0.00	0.00
TLB shutdowns	0.00	0.00	0.00	0.00	TLB shutdowns	0.00	0.00	0.00	0.00
Thermal event interrupts	0.00	0.00	0.00	0.00	Thermal event interrupts	0.00	0.00	0.00	0.00
Threshold APIC interrupts	0.00	0.00	0.00	0.00	Threshold APIC interrupts	0.00	0.00	0.00	0.00
Machine check exceptions	0.00	0.00	0.00	0.00	Machine check exceptions	0.00	0.00	0.00	0.00
Machine check polls	3.33m	3.22m	3.33m	3.45m	Machine check polls	3.33m	2.88m	3.33m	6.67m
ERR	0.00	0.00	0.00	0.00	ERR	0.00	0.00	0.00	0.00
HIS	0.00	0.00	0.00	0.00	HIS	0.00	0.00	0.00	0.00

Runin 2.0.6-4+deb7u2
Last update: Thu Mar 13 09:45:01 2014



Group Pickup

Pickup groups allow users to intercept calls directed towards other users of the group. This is done either by dialing a special extension or by pressing a function key.

Quick Summary

In order to be able to use group pickup you have to:

- Create a pickup group
- Enable an extension to intercept calls
- Add a function key to interceptors

Creating a Pickup Group Pickup groups can be created in the *Services* → *IPBX* → *Call management* → *Call pickups* page.

In the *general* tab, you can define a name and a description for the pickup group. In the *Interceptors* tab, you can define a list of users, groups or queues that can intercept calls. In the *Intercepted* tab, you can define a list of users, groups or queues that can be intercepted.

Pickup groups > Edit test

General Interceptors Intercepted

Groups

0 items selected Remove all Add all

huge (3000@pcm-dev) +

Queues

Create queue

Users

3 items selected Remove all Add all

↕ Père Noël	—	User 0500	+
↕ Linda	—	User 0501	+
↕ Fernando L'Igüane	—	User 0502	+
		User 0503	+
		User 0504	+
		User 0505	+
		User 0506	+

Save

Enabling an Interception Extension The pickup extension can be defined in the *Services* → *IPBX* → *IPBX services* → *Extensions* page.

The extension used by group pickup is called *Group interception* it's default value is *8.

Warning: The extension must be enabled even if a function key is used.

Adding a Function Key to an Interceptor To assign a function to an interceptor, go to *Services* → *IPBX* → *IPBX settings* → *Users*, edit an interceptor and go to the *Func Keys* tab.

Add a new function key of type *Group Interception* and save.

Key	Type	Destination	Label	Supervision
1	Filtering Boss - Secretary	fernando / Fernando L'Iguane	Linda	Enabled
2	Group Interception		Interception	Disabled

Save

Server/Hardware

This section describes how to configure the telephony hardware on a XiVO server.

Note: Currently XiVO supports only Digium Telephony Interface cards

The configuration process is the following :

Load the correct DAHDI modules For your Digium card to work properly you must load the appropriate DAHDI kernel module. This is done via the file `/etc/dahdi/modules` and this page will guide you through its configuration.

Know which card is in your server You can see which cards are detected by issuing the `dahdi_hardware` command:

```
dahdi_hardware
pci:0000:05:0d.0      wcb4xxp-      d161:b410 Digium Wildcard B410P
pci:0000:05:0e.0      wct4xxp-      d161:0205 Wildcard TE205P (4th Gen)
```

This command gives the card name detected and, more importantly, the DAHDI kernel module needed for this card. In the above example you can see that two cards are detected in the system:

- a Digium B410P *which needs* the `wcb4xxp` module
- and a Digium TE205P *which needs* the `wct4xxp` module

Create the configuration file Now that we know the modules we need, we can create our configuration file:

1. Create the file `/etc/dahdi/modules`:

```
touch /etc/dahdi/modules
```

2. Fill it with the modules name you found with the `dahdi_hardware` command (one module name per line). In our example, your `/etc/dahdi/modules` file should contain the following lines:

```
wcb4xxp
wct4xxp
```

Note: In the `/usr/share/dahdi/modules.sample` file you can find all the modules supported in your XiVO version.

Apply the configuration To apply the configuration, restart the services:

```
xivo-service restart
```

Next step Now that you have loaded the correct module for your card you must:

1. check if you need to follow one of the *Specific configuration* sections below,
2. and continue with the next configuration step which is to *configure the echo canceller*.

Specific configuration This section lists some specific configuration. You should not follow them unless you have a specific need.

TE13x, TE23x, TE43x: E1/T1 selection With E1/T1 cards you must select the correct *line mode* between:

- E1 : the European standard,
- and T1 : North American standard

For old generation cards (TE12x, TE20x, TE40x series) the *line mode* is selected via a physical jumper.

For new generation cards like TE13x, TE23x, TE43x series the *line mode* is selected by configuration.

If you're configuring one of these **TE13x, T23x, T43x** cards then you **MUST** create a configuration file to set the line mode to E1:

1. Create the file `/etc/modprobe.d/xivo-wcte-linemode.conf`:

```
touch /etc/modprobe.d/xivo-wcte-linemode.conf
```

2. Fill it with the following lines replacing DAHDI_MODULE_NAME by the correct module name (`wcte13xp`, `wcte43x` ...):

```
# set the card in E1/T1 mode
options DAHDI_MODULE_NAME default_linemode=e1
```

3. Then, restart the services:

```
xivo-service restart
```

Hardware Echo-cancellation It is *recommended* to use telephony cards with an hardware echo-canceller module.

Warning: with **TE13x, TE23x and TE43x** cards, you **MUST** install the echo-canceller firmware. Otherwise the card won't work properly.

Know which firmware you need If you have an hardware echo-canceller module you **have to** install its firmware.

You first need to know which firmware you have to install. The simplest way is to restart dahdi and then to lookup in the `dmesg` which firmware does DAHDI request at startup:

```
xivo-service restart
dmesg |grep firmware
[5461540.738209] wct4xxp 0000:01:0e.0: firmware: agent aborted loading dahdi-fw-oct6114-064.bin (
[5461540.738310] wct4xxp 0000:01:0e.0: VPM450: firmware dahdi-fw-oct6114-064.bin not available fr
```

In the example above you can see that the module `wct4xxp` requested the `dahdi-fw-oct6114-064.bin` firmware file but did not found it. But you now know that you need the `dahdi-fw-oct6114-064.bin` firmware.

Install the firmware When you know which firmware you need you can install it with `xivo-fetchfw` utility.

1. Use `xivo-fetchfw` to find the name of the package. You can search for `digium` occurrences in the available packages:

```
xivo-fetchfw search digium
```

2. Find the package name which matches the firmware file you need. In our example, we need the `dahdi-fw-oct6114-064.bin` file which is supplied by the package named `digium-oct6114-064`:

```
xivo-fetchfw install digium-oct6114-064
```

Activate the Hardware Echo-cancellation Now that you installed hardware echo-canceller firmware you must activate it in `/etc/asterisk/chan_dahdi.conf` file:

```
echocancel = 1
```

Apply the configuration To apply the configuration, restart the services:

```
xivo-service restart
```

Next step Now that you have loaded the correct module for your card you must:

1. check if you need to follow one of the [Specific configuration](#) sections below,
2. and continue with the next configuration step which is to [configure your card](#) according to the operator links.

Specific configuration This section describes some specific configuration. You should not follow them unless you have a specific need.

Use the Hardware Echo-canceller for DTMF detection If you have an hardware echo-canceller you *may* want to use it to detect the DTMF signal (instead of asterisk).

1. Create the file `/etc/modprobe.d/xivo-hwec-dtmf.conf`:

```
touch /etc/modprobe.d/xivo-hwec-dtmf.conf
```

2. Fill it with the following lines replacing `DAHDI_MODULE_NAME` by the correct module name (`wctel3xp`, `wct4xxp` ...):

```
options DAHDI_MODULE_NAME vpmdtmfsupport=1
```

3. Then, restart the services:

```
xivo-service restart
```

Card configuration Now that you have *loaded the correct DAHDI modules* and *configured the echo canceller* you can proceed with the card configuration. Follow one of the appropriate link below :

BRI card configuration

Verifications Verify that the `wcb4xxp` module is uncommented in `/etc/dahdi/modules`.

If it wasn't, do again the step [Load the correct DAHDI modules](#).

Generate DAHDI configuration Issue the command:

```
dahdi_genconf
```

Warning: it will erase all existing configuration in `/etc/dahdi/system.conf` and `/etc/asterisk/dahdi-channels.conf` files !

Configure

DAHDI system.conf configuration First step is to check `/etc/dahdi/system.conf` file:

- check the span numbering,
- if needed change the clock source,

See detailed explanations of this file in the [/etc/dahdi/system.conf](#) section.

Below is **an example** for a typical french BRI line span:

```
# Span 1: B4/0/1 "B4XXP (PCI) Card 0 Span 1" (MASTER) RED
span=1,1,0,ccs,ami
# termtype: te
bchan=1-2
hardhdlc=3
echocanceller=mg2,1-2
```

Asterisk dahdi-channels.conf configuration Then you have to modify the `/etc/asterisk/dahdi-channels.conf` file:

- remove the unused lines like:

```
context = default
group = 63
```

- change the context lines if needed,
- the signalling should be one of:
 - `bri_net`
 - `bri_cpe`
 - `bri_net_ptmp`
 - `bri_cpe_ptmp`

See some explanations of this file in the [/etc/asterisk/dahdi-channels.conf](#) section.

Below is **an example** for a typical french BRI line span:

```
; Span 1: B4/0/1 "B4XXP (PCI) Card 0 Span 1" (MASTER) RED
group = 0,11 ; belongs to group 0 and 11
context = from-extern ; incoming call to this span will be sent in 'from-extern' context
switchtype = euroisdn
signalling = bri_cpe ; use 'bri_cpe' signalling
channel => 1-2 ; the above configuration applies to channels 1 and 2
```

Next step Now that you have configured your BRI card:

1. you must check if you need to follow one of the [Specific configuration](#) sections below,
2. then, if you have another type of card to configure, you can go back to the [configure your card](#) section,
3. if you have configured all your card you have to configure the [DAHDI interconnections](#) in the web interface.

Specific configuration You will find below 3 configurations that we recommend for BRI lines. These configurations were tested on different type of french BRI lines with success.

Note: The pre-requisites are:

- XiVO >= 14.12,
- Use per-port dahdi interconnection (see the *DAHDI interconnections* section)

If you don't know which one to configure we recommend that you try each one after the other in this order:

1. *PTMP without layer1/layer2 persistence*
2. *PTMP with layer1/layer2 persistence*
3. *PTP with layer1/layer2 persistence*

PTMP without layer1/layer2 persistence In this mode we will configure asterisk and DAHDI:

- to use Point-to-Multipoint (PTMP) signalling,
- and to leave Layer1 and Layer2 DOWN

Follow theses steps to configure:

1. **Before** the line `#include dahdi-channels.conf` add, in file `/etc/asterisk/chan_dahdi.conf`, the following lines:

```
layer1_presence = ignore
layer2_persistence = leave_down
```

2. In the file `/etc/asterisk/dahdi-channels.conf` use `bri_cpe_ptmp` signalling:

```
signalling = bri_cpe_ptmp
```

3. Create the file `/etc/modprobe.d/xivo-wcb4xxp.conf` to deactivate the layer1 persistence:

```
touch /etc/modprobe.d/xivo-wcb4xxp.conf
```

4. Fill it with the following content:

```
options wcb4xxp persistentlayer1=0
```

5. Then, apply the configuration by restarting the services:

```
xivo-service restart
```

Note: Expected behavior:

- The `dahdi show status` command should show the BRI spans in *RED* status if there is no call,
- For outgoing calls the layer1/layer2 should be brought back up by the XiVO (i.e. asterisk/chan_dahdi),
- For incoming calls the layer1/layer2 should be brought back up by the operator,
- You can consider that there is *a problem* only if incoming or outgoing calls are rejected.

PTMP with layer1/layer2 persistence In this mode we will configure asterisk and DAHDI:

- to use Point-to-Multipoint (PTMP) signalling,
- and to keep Layer1 and Layer2 UP

Follow theses steps to configure:

1. **Before** the line `#include dahdi-channels.conf` add, in file `/etc/asterisk/chan_dahdi.conf`, the following lines:

```
layer1_presence = required
layer2_persistence = keep_up
```

2. In the file `/etc/asterisk/dahdi-channels.conf` use `bri_cpe_ptmp` signalling:

```
signalling = bri_cpe_ptmp
```

3. If it exists, delete the file `/etc/modprobe.d/xivo-wcb4xxp.conf`:

```
rm /etc/modprobe.d/xivo-wcb4xxp.conf
```

4. Then, apply the configuration by restarting the services:

```
xivo-service restart
```

Note: Expected behavior:

- The `dahdi show status` command should show the BRI spans in **OK** status even if there is no call,
 - In asterisk CLI you may see the spans going Up/Down/Up : it is *a problem* only if incoming or outgoing calls are rejected.
-

PTP with layer1/layer2 persistence In this mode we will configure asterisk and DAHDI:

- to use Point-to-Point (PTP) signalling,
- and use default behavior for Layer1 and Layer2.

Follow theses steps to configure:

1. In file `/etc/asterisk/chan_dahdi.conf` remove all occurrences of `layer1_presence` and `layer2_persistence` options.
2. In the file `/etc/asterisk/dahdi-channels.conf` use `bri_cpe` signalling:

```
signalling = bri_cpe
```

3. If it exists, delete the file `/etc/modprobe.d/xivo-wcb4xxp.conf`:

```
rm /etc/modprobe.d/xivo-wcb4xxp.conf
```

4. Then, apply the configuration by restarting the services:

```
xivo-service restart
```

Note: Expected behavior:

- The `dahdi show status` command should show the BRI spans in **OK** status even if there is no call,
 - In asterisk CLI you should not see the spans going Up and Down : if it happens, it is *a problem* only if incoming or outgoing calls are rejected.
-

PRI card configuration

Verifications Verify that the correct module is configured in `/etc/dahdi/modules` depending on the card you installed in your server.

If it wasn't, do again the step *Load the correct DAHDI modules*

Warning: *TE13x, TE23x, TE43x* cards :

- these cards need a specific dahdi module configuration. See *TE13x, TE23x, TE43x: E1/T1 selection* paragraph,
- you **MUST** install the correct echo-canceller firmware to be able to use these cards. See *Hardware Echo-cancellation* paragraph.

Generate DAHDI configuration Issue the command:

```
dahdi_genconf
```

Warning: it will erase all existing configuration in `/etc/dahdi/system.conf` and `/etc/asterisk/dahdi-channels.conf` files !

Configure

DAHDI system.conf configuration First step is to check `/etc/dahdi/system.conf` file:

- check the span numbering,
- if needed change the clock source,
- usually (at least in France) you should remove the `crc4`

See detailed explanations of this file in the */etc/dahdi/system.conf* section.

Below is **an example** for a typical french PRI line span:

```
# Span 1: TE2/0/1 "T2XXP (PCI) Card 0 Span 1" CCS/HDB3/CRC4 RED
span=1,1,0,ccs,hdb3
# termtype: te
bchan=1-15,17-31
dchan=16
echocanceller=mg2,1-15,17-31
```

Asterisk dahdi-channels.conf configuration Then you have to modify the `/etc/asterisk/dahdi-channels.conf` file:

- remove the unused lines like:

```
context = default
group = 63
```

- change the context lines if needed,
- the signalling should be one of:
 - `pri_net`
 - `pri_cpe`

Below is **an example** for a typical french PRI line span:

```
; Span 1: TE2/0/1 "T2XXP (PCI) Card 0 Span 1" CCS/HDB3/CRC4 RED
group = 0,11 ; belongs to group 0 and 11
context = from-extern ; incoming call to this span will be sent in 'from-extern' context
switchtype = euroisdn
```

```
signalling = pri_cpe      ; use 'pri_cpe' signalling
channel => 1-15,17-31    ; the above configuration applies to channels 1 to 15 and 17 to 31
```

Next step Now that you have configured your PRI card:

1. you must check if you need to follow one of the *Specific configuration* sections below,
2. then, if you have another type of card to configure, you can go back to the *configure your card* section,
3. if you have configured all your card you have to configure the *DAHDI interconnections* in the web interface.

Specific configuration

Multiple PRI cards and sync cable If you have several PRI cards in your server you should link them with a synchronization cable to share the exact same clock.

To do this, you need to:

- use the coding wheel on the Digium cards to give them an order of recognition in DAHDI/Asterisk (see [Digium_telephony_cards_support](#)),
- daisy-chain the cards with a sync cable (see [Digium_telephony_cards_support](#)),
- load the DAHDI module with the `timingcable=1` option.

Create `/etc/modprobe.d/xivo-timingcable.conf` file and insert the line:

```
options DAHDI_MODULE_NAME timingcable=1
```

Where `DAHDI_MODULE_NAME` is the DAHDI module name of your card (e.g. `wct4xxp` for a TE205P).

Analog card configuration

Limitations

- XiVO does not support hardware echocanceller on the TDM400 card. Users of TDM400 card willing to setup an echocanceller will have to use a software echocanceller like OSLEC.

Verifications Verify that one of the `{wctdm,wctdm24xxp}` module is uncommented in `/etc/dahdi/modules` depending on the card you installed in your server.

If it wasn't, do again the step *Load the correct DAHDI modules*

Note: Analog cards work with card module. You must add the appropriate card module to your analog card. Either:

- an FXS module (for analog equipment - phones, ...),
 - an FXO module (for analog line)
-

Generate DAHDI configuration Issue the command:

```
dahdi_genconf
```

Warning: it will erase all existing configuration in `/etc/dahdi/system.conf` and `/etc/asterisk/dahdi-channels.conf` files !

Configure

DAHDI system.conf configuration First step is to check `/etc/dahdi/system.conf` file:

- check the span numbering,

See detailed explanations of this file in the `/etc/dahdi/system.conf` section.

Below is **an example** for a typical FXS analog line span:

```
# Span 2: WCTDM/4 "Wildcard TDM400P REV I Board 5"
fxoks=32
echocanceller=mg2,32
```

Asterisk dahdi-channels.conf configuration Then you have to modify the `/etc/asterisk/dahdi-channels.conf` file:

- remove the unused lines like:

```
context = default
group = 63
```

- change the context and callerid lines if needed,
- the signalling should be one of:
 - `fxo_ks` for **FXS** lines -yes it is the reverse
 - `fxs_ks` for **FXO** lines - yes it is the reverse

Below is **an example** for a typical french PRI line span:

```
; Span 2: WCTDM/4 "Wildcard TDM400P REV I Board 5"
signalling=fxo_ks
callerid="Channel 32" <4032>
mailbox=4032
group=5
context=default
channel => 32
```

Next step Now that you have configured your PRI card:

1. you must check if you need to follow one of the *Specific configuration* sections below,
2. then, if you have another type of card to configure, you can go back to the *configure your card* section,
3. if you have configured all your card you have to configure the *DAHDI interconnections* in the web interface.

Specific configuration

FXS modules If you use **FXS** modules you should create the file `/etc/modprobe.d/xivo-tdm` and insert the line:

```
options DAHDI_MODULE_NAME fastringer=1 booststringer=1
```

Where `DAHDI_MODULE_NAME` is the DAHDI module name of your card (e.g. `wctdm` for a TDM400P).

FXO modules If you use **FXO** modules you should create file `/etc/modprobe.d/xivo-tdm`:

```
options DAHDI_MODULE_NAME opermode=FRANCE
```

Where `DAHDI_MODULE_NAME` is the DAHDI module name of your card (e.g. `wctdm` for a TDM400P).

Voice Compression Card configuration

Verifications Verify that the `wctc4xxp` module is uncommented in `/etc/dahdi/modules`.

If it wasn't, do again the step *Load the correct DAHDI modules*.

Configure To configure the card you have to:

1. Install the card firmware:

```
xivo-fetchfw install digium-tc400m
```

2. Comment out the following line in `/etc/asterisk/modules.conf`:

```
noload = codec_dahdi.so
```

3. Restart asterisk:

```
service asterisk restart
```

Next step Now that you have configured your Voice Compression card:

1. you must check if you need to follow one of the *Specific configuration* sections below,
2. then, if you have another type of card to configure, you can go back to the *configure your card* section.

Specific configuration

Select the transcoding mode The Digium TC400 card can be used to transcode:

- 120 G.729a channels,
- 92 G.723.1 channels,
- or 92 G.729a/G.723.1 channels.

Depending on the codec you want to transcode, you can modify the `mode` parameter which can take the following value:

- `mode = mixed` : this the default value which activates transcoding for 92 channels in G.729a or G.723.1 (5.3 Kbit and 6.3 Kbit)
- `mode = g729` : this option activates transcoding for 120 channels in G.729a
- `mode = g723` : this option activates transcoding for 92 channels in G.723.1 (5.3 Kbit et 6.3 Kbit)

1. Create the file `/etc/modprobe.d/xivo-transcode.conf`:

```
touch /etc/modprobe.d/xivo-transcode.conf
```

2. And insert the following lines:

```
options wctc4xxp mode=g729
```

3. Apply the configuration by restarting the services:

```
xivo-service restart
```

4. Verify that the card is correctly seen by asterisk with the `transcoder show` CLI command - this command should show the encoders/decoders registered by the TC400 card:

```
*CLI> transcoder show
0/0 encoders/decoders of 120 channels are in use.
```

Apply configuration If you didn't do it already, you have to restart the services to apply the configuration:

```
xivo-service restart
```

At the end of this page you will also find some general notes and DAHDI.

Notes on configuration files

/etc/dahdi/system.conf A *span* is created for each card port. Below is an example of a standard E1 port:

```
span=1,1,0,ccs,hdb3
dchan=16
bchan=1-15,17-31
echocanceller=mg2,1-15,17-31
```

Each span has to be declared with the following information:

```
span=<spannum>,<timing>,<LBO>,<framing>,<coding>[,crc4]
```

- **spannum**: corresponds to the span number. It starts to 1 and has to be incremented by 1 at each new span. This number **MUST** be unique.
- **timing**: describes the how this span will be considered regarding the synchronization :
 - 0 : do not use this span as a synchronization source,
 - 1 : use this span as the primary synchronization source,
 - 2 : use this span as the secondary synchronization source etc.
- **LBO**: 0 (not used)
- **framing**: correct values are `ccs` or `cas`. For ISDN lines, `ccs` is used.
- **coding**: correct values are `hdb3` or `ami`. For example, `hdb3` is used for an E1 (PRI) link, whereas `ami` is used for T0 (french BRI) link.
- **crc4**: this is a framing option for PRI lines. For example it is rarely use in France.

Note that the `dahdi_genconf` command should usually give you the correct parameters (if you correctly set the cards jumper). All these information should be checked with your operator.

/etc/asterisk/chan_dahdi.conf This file contains the general parameters of the DAHDI channel. It is not generated via the `dahdi_genconf` command.

/etc/asterisk/dahdi-channels.conf This file contains the parameters of each channel. It is generated via the `dahdi_genconf` command.

Below is an example of span definition:

```
group=0,11
context=from-extern
switchtype = euroisdn
signalling = pri_cpe
channel => 1-15,17-31
```

Note that parameters are read from top to bottom in a last match fashion and are applied to the given channels when it reads a line `channel =>`.

Here the channels 1 to 15 and 17 to 31 (it is a typical E1) are set:

- in groups 0 and 11 (see [DAHDI interconnections](#))
- in context `from-extern`: all calls received on these channels will be sent in the context `from-extern`
- and configured with `switchtype euroisdn` and `signalling pri_cpe`

Debug

Check IRQ misses It's always useful to verify if there isn't any *missed IRQ* problem with the cards.

Check:

```
cat /proc/dahdi/<span number>
```

If the *IRQ misses* counter increments, it's not good:

```
cat /proc/dahdi/1
Span 1: WCTDM/0 "Wildcard TDM800P Board 1" (MASTER)
IRQ misses: 1762187
 1 WCTDM/0/0 FXOKS (In use)
 2 WCTDM/0/1 FXOKS (In use)
 3 WCTDM/0/2 FXOKS (In use)
 4 WCTDM/0/3 FXOKS (In use)
```

Digium gives some hints in their *Knowledge Base* here : <http://kb.digium.com/entry/1/63/>

PRI Digium cards needs 1000 interruption per seconds. If the system cannot supply them, it increment the IRQ missed counter.

As indicated in Digium *KB* you should avoid shared IRQ with other equipments (like HD or NIC interfaces).

Incall

General Configuration You can configure incoming calls settings in *Services* → *IPBX* → *Call Management* → *Incoming calls*.

DID (Direct Inward Dialing) Configuration When a “+” character is prepended a called DID, the “+” character is discarded.

Example:

Bob has a DID with number 1000. Alice can call Bob by dialing either 1000 or +1000, without configuring another DID.

Interconnections

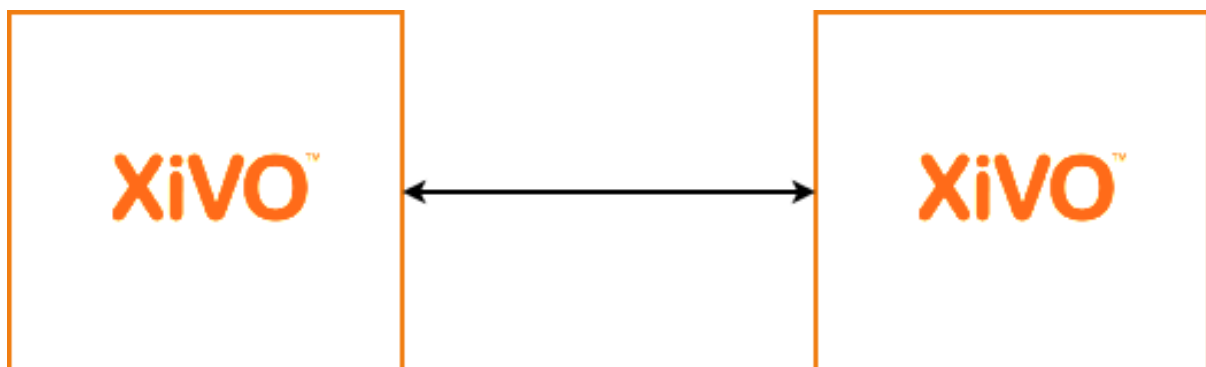


Fig. 1.65: Situation diagram

Interconnect two XiVO directly Interconnecting two XiVO will allow you to send and receive calls between the users configured on both sides.

The steps to configure the interconnections are:

- Establish the trunk between the two XiVO, that is the SIP connection between the two servers
- Configure outgoing calls on the server(s) used to emit calls
- Configure incoming calls on the server(s) used to receive calls

For now, only SIP interconnections have been tested.

Establish the trunk The settings below allow a trunk to be used in both directions, so it doesn't matter which server is A and which is B.

Consider XiVO A wants to establish a trunk with XiVO B.

On XiVO B, go on page *Services* → *IPBX* → *Trunk management* → *SIP Protocol*, and create a SIP trunk:

```
Name : xivo-trunk
Username: xivo-trunk
Password: pass
Connection type: Friend
IP addressing type: Dynamic
Context: <see below>
```

Note: For the moment, Name and Username need to be the same string.

The `Context` field will determine which extensions will be reachable by the other side of the trunk:

- If `Context` is set to `default`, then every user, group, conf room, queue, etc. that have an extension if the `default` context will be reachable directly by the other end of the trunk. This setting can ease configuration if you manage both ends of the trunk.
- If you are establishing a trunk with a provider, you probably don't want everything to be available to everyone else, so you can set the `Context` field to `Incalls`. By default, there is no extension available in this context, so we will be able to configure which extension are reachable by the other end. This is the role of the incoming calls: making bridges from the `Incalls` context to other contexts.

On XiVO A, create the other end of the SIP trunk on the *Services* → *IPBX* → *Trunk management* → *SIP Protocol*:

```
Name: xivo-trunk
Username: xivo-trunk
Password: pass
Identified by: Friend
Connection type: Static
Address: <XiVO B IP address or hostname>
Context: Incalls
```

Register tab:

```
Register: checked
Transport: udp
Username: xivo-trunk
Password: pass
Remote server: <XiVO B IP address or hostname>
```

On both XiVO, activate some codecs, *Services* → *IPBX* → *General Settings* → *SIP protocol*, tab *Signaling*:

```
Enabled codecs: at least GSM (audio)
```

At that point, the Asterisk command `sip show registry` on XiVO B should print a line showing that XiVO A is registered, meaning your trunk is established.

Set the outgoing calls The outgoing calls configuration will allow XiVO to know which extensions will be called through the trunk.

On the call emitting server(s), go on the page *Services* → *IPBX* → *Call management* → *Outgoing calls* and add an outgoing call.

Tab General:

```
Trunks: xivo-trunk
```

Tab Exten:

```
Exten: **99. (note the period at the end)
Stripnum: 4
```

This will tell XiVO: if any extension begins with `**99`, then try to dial it on the trunk `xivo-trunk`, after removing the 4 first characters (the `**99` prefix).

The most useful special characters to match extensions are:

```
. (period): will match one or more characters
X: will match only one character
```

You can find more details about pattern matching in Asterisk (hence in XiVO) on [the Asterisk wiki](#).

Set the incoming calls Now that we have calls going out from a XiVO, we need to route incoming calls on the XiVO destination.

Note: This step is only necessary if the trunk is linked to an Incoming calls context.

To route an incoming call to the right destination in the right context, we will create an incoming call in *Services* → *IPBX* → *Call management* → *Incoming calls*.

Tab General:

```
DID: 101
Context: Incalls
Destination: User
Redirect to: someone
```

This will tell XiVO: if you receive an incoming call to the extension `101` in the context `Incalls`, then route it to the user `someone`. The destination context will be found automatically, depending on the context of the line of the given user.

So, with the outgoing call set earlier on XiVO A, and with the incoming call above set on XiVO B, a user on XiVO A will dial `**99101`, and the user `someone` will ring on XiVO B.

Interconnect a XiVO to a VoIP provider When you want to send and receive calls to the global telephony network, one option is to subscribe to a VoIP provider. To receive calls, your XiVO needs to tell your provider that it is ready and to which IP the calls must be sent. To send calls, your XiVO needs to authenticate itself, so that the provider knows that your XiVO is authorized to send calls and whose account must be credited with the call fare.

The steps to configure the interconnections are:

- Establish the trunk between the two XiVO, that is the SIP connection between the two servers
- Configure outgoing calls on the server(s) used to emit calls
- Configure incoming calls on the server(s) used to receive calls

Establish the trunk You need the following information from your provider:

- a username
- a password
- the name of the provider VoIP server

- a public phone number

On your XiVO, go on page *Services* → *IPBX* → *Trunk management* → *SIP Protocol*, and create a SIP/IAX trunk:

```
Name : provider_username
Username: provider_username
Password: provider_password
Connection type: Peer
IP addressing type: voip.provider.example.com
Context: Incalls (or another incoming call context)
```

Register tab:

```
Register: checked
Transport: udp
Name: provider_username
Username: provider_username
Password: provider_password
Remote server: voip.provider.example.com
```

Note: For the moment, Name and Username need to be the same value.

If your XiVO is behind a NAT device or a firewall, you should set the following:

```
Monitoring: Yes
```

This option will make Asterisk send a signal to the VoIP provider server every 60 seconds (default settings), so that NATs and firewall know the connection is still alive. If you want to change the value of this cycle period, you have to select the appropriate value of the following parameter:

```
Qualify Frequency:
```

At that point, the Asterisk command `sip show registry` should print a line showing that you are registered, meaning your trunk is established.

Set the outgoing calls The outgoing calls configuration will allow XiVO to know which extensions will be called through the trunk.

Go on the page *Services* → *IPBX* → *Call management* → *Outgoing calls* and add an outgoing call.

Tab General:

```
Trunks: provider_username
```

Tab Exten:

```
Exten: 418. (note the period at the end)
```

This will tell XiVO: if an internal user dials a number beginning with 418, then try to dial it on the trunk `provider_username`.

The most useful special characters to match extensions are:

```
. (period): will match one or more characters
X: will match only one character
```

You can find more details about pattern matching in Asterisk (hence in XiVO) on [the Asterisk wiki](#).

Set the incoming calls Now that we have calls going out, we need to route incoming calls.

To route an incoming call to the right destination in the right context, we will create an incoming call in *Services* → *IPBX* → *Call management* → *Incoming calls*.

Tab General:

```
DID: your_public_phone_number
Context: Incalls (the same than configured in the trunk)
Destination: User
Redirect to: the_front_desk_guy
```

This will tell XiVO: if you receive an incoming call to the public phone number in the context `Incalls`, then route it to the user `the_front_desk_guy`. The destination context will be found automatically, depending on the context of the line of the given user.

Interconnect a XiVO to a PBX via an ISDN link The goal of this architecture can be one of:

- start a smooth migration between an old telephony system towards IP telephony with XiVO
- bring new features to the PBX like voicemail, conference, IVR etc.

First, XiVO is to be integrated transparently between the operator and the PBX. Then users or features are to be migrated from the PBX to the XiVO.

Warning: It requires a special call routing configuration on both the XiVO and the PBX.

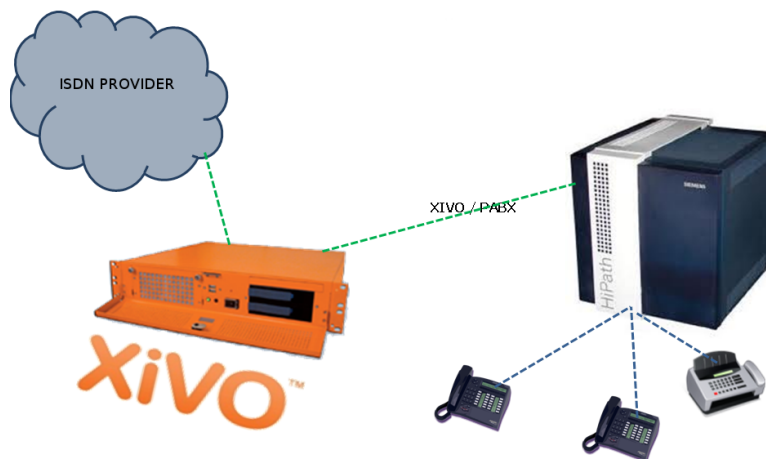


Fig. 1.66: Interconnect a XiVO to a PBX

Hardware

General uses You must have an ISDN card able to support both the provider and PBX ISDN links.

Example : If you have two provider links towards the PBX, XiVO should have a 4 spans card : two towards the provider, and two towards the PBX.

If you use two cards If you use two cards, you have to :

- Use a cable for clock synchronization between the cards
- Configure the *wheel* to define the cards order in the system.

Please refer to the section [Sync cable](#)

Configuration You have now to configure two files :

1. `/etc/dahdi/system.conf`
2. `/etc/asterisk/dahdi-channels.conf`

system.conf You mainly need to configure the `timing` parameter on each *span*. As a general rule :

- **Provider *span*** - XiVO will get the clock from the provider : the `timing` value is to be different from 0 (see `/etc/dahdi/system.conf` section)
- **PBX *span*** - XiVO will provide the clock to the PBX : the `timing` value is to be set to 0 (see `/etc/dahdi/system.conf` section)

Below is an example with two provider links and two PBX links:

```
# Span 1: TE4/0/1 "TE4XXP (PCI) Card 0 Span 1" (MASTER)
span=1,1,0,ccs,hdb3          # Span towards Provider
bchan=1-15,17-31
dchan=16
echocanceller=mg2,1-15,17-31

# Span 2: TE4/0/2 "TE4XXP (PCI) Card 0 Span 2"
span=2,2,0,ccs,hdb3          # Span towards Provider
bchan=32-46,48-62
dchan=47
echocanceller=mg2,32-46,48-62

# Span 3: TE4/0/3 "TE4XXP (PCI) Card 0 Span 3"
span=3,0,0,ccs,hdb3          # Span towards PBX
bchan=63-77,79-93
dchan=78
echocanceller=mg2,63-77,79-93

# Span 4: TE4/0/4 "TE4XXP (PCI) Card 0 Span 4"
span=4,0,0,ccs,hdb3          # Span towards PBX
bchan=94-108,110-124
dchan=109
echocanceller=mg2,94-108,110-124
```

dahdi-channels.conf In the file `/etc/asterisk/dahdi-channels.conf` you need to adjust, for each *span* :

- **group** : the group number (e.g. 0 for provider links, 2 for PBX links),
- **context** : the context (e.g. `from-extern` for provider links, `from-pabx` for PBX links)
- **signalling** : `pri_cpe` for provider links, `pri_net` for PBX side

Warning: most of the PBX uses overlap dialing for some destination (digits are sent one by one instead of by block). In this case, the `overlapdial` parameter has to be activated on the PBX spans:

```
overlapdial = incoming
```

Below an example of `/etc/asterisk/dahdi-channels.conf`:

```
; Span 1: TE4/0/1 "TE4XXP (PCI) Card 0 Span 1" (MASTER)
group=0,11
context=from-extern
switchtype = euroisdn
signalling = pri_cpe
channel => 1-15,17-31
```

```
; Span 2: TE4/0/2 "TE4XXP (PCI) Card 0 Span 2"
group=0,12
context=from-extern
switchtype = euroisdn
signalling = pri_cpe
channel => 32-46,48-62

; PBX link #1
; Span 3: TE4/0/3 "TE2XXP (PCI) Card 0 Span 3"
group=2,13
context=from-pabx      ; special context for PBX incoming calls
overlapdial=incoming  ; overlapdial activation
switchtype = euroisdn
signalling = pri_net   ; behave as the NET termination
channel => 63-77,79-93

; PBX link #2
; Span 4: TE4/0/4 "T4XXP (PCI) Card 0 Span 4"
group=2,14
context=from-pabx      ; special context for PBX incoming calls
overlapdial=incoming  ; overlapdial activation
switchtype = euroisdn
signalling = pri_net   ; behave as the NET termination
channel => 94-108,110-124
```

Passthru function

Route PBX incoming calls We first need to create a route for calls coming from the PBX

Create a file named `pbx.conf` in the directory `/etc/asterisk/extensions_extra.d/`, # Add the following lines in the file:

```
[from-pabx]
exten = _X.,1,NoOp(### Call from PBX ${CARLLERID(num)} towards ${EXTEN} ###)
exten = _X.,n,Goto(default,${EXTEN},1)
```

This dialplan routes incoming calls from the PBX in the default context of XiVO. It enables call from the PBX : * towards a SIP phone (in default context) * towards outgoing destination (via the `to-extern` context included in default context)

Create the to-pabx context In the webi, create a context named `to-pabx`:

- Name : `to-pabx`
- Display Name : `TO PBX`
- Context type : `Outcall`
- Include sub-contexts : `No context inclusion`

This context will permit to route incoming calls from the XiVO to the PBX.

Route incoming calls to PBX In our example, incoming calls on spans 1 and 2 (spans plugged to the provider) are routed by `from-extern` context. We are going to create a default route to redirect incoming calls to the PBX.

Create an incoming call as below :

- DID : `XXXX` (according to the number of digits sent by the provider)
- Context : `Incoming calls`
- Destination : `Customized`

Contexts > Edit

General Users Groups Queues Conference rooms Incoming calls

Name:

Displayed name:

Entity:

Context type:

Include sub-contexts

0 items selected	Remove all	<input type="text"/>	Add all
		Appels entrants (from-extern)	+
		Appels internes (default)	+
		Appels sortants (to-extern)	+

Description:

Save

- Command : Goto(to-pabx,\${XIVO_DSTNUM},1)

Incoming calls > Add

General Call permissions Schedules

DID:

Context:

Destination :

Command:

CallerID mode :

Preprocess subroutine :

Description :

Save

Create the interconnections You have to create two interconnections :

- provider side : dahdi/g0
- PBX side : dahdi/g2

In the menu *Services* → *IPBX* → *Trunk management* → *Customized* page :

- Name : t2-operateur
- Interface : dahdi/g0
- Context : to-extern

The second interconnection :

- Name : t2-pabx
- Interface : dahdi/g2
- Context : to-pabx

Customized trunk > Add

Name:

Interface:

Interface suffix:

Context:

Description :

Customized trunk > Add

Name:

Interface:

Interface suffix:

Context:

Description :

Create outgoing calls You must create two rules of outgoing calls in the menu *Services* → *IPBX* → *Call management* → *Outgoing calls* page :

1. Redirect calls to the PBX :
 - Name : fsc-pabx
 - Context : to-pabx
 - Trunks : choose the *t2-pabx* interconnection

Outgoing calls > Edit vers-pabx

General | Exten | Call permissions | Schedules

Name:

Context:

Use ENUM: ☐

Internal: ☐

Preprocess subroutine:

Ringing time before hangup:

Trunks:

1 items selected	Remove all		Add all
✱ t2-pabx (dahdi/g2)	—	idefisk-maq2 (SIP)	+
		jocelyn (SIP)	+
		loadtester (SIP)	+
		redirection (local)	+
		t2colt (dahdi/g0)	+
		test-audiocodes (SIP)	+
		toulouse (SIP)	+

In the extensions tab :

- Exten : XXXX
2. Create a rule “fsc-operateur”:
 - Name : fsc-operateur
 - Context : to-extern
 - Trunks : choose the “t2-operateur” interconnection

	Extern prefix	Prefix	Exten	Stripnum	Callerid
1			XXXX	0	

Save

In the extensions tab:

```
exten = X.
```

Specific VoIP providers

Simon Telephonics The following configuration is based on the example found [here](#)

- username: GV18005551212
- password: password
- exten: 18005551212
- host: gvgw.simonics.com

General SIP configuration Under *Services* → *IPBX* → *General settings* → *SIP Protocol*.

- General
 - Match users with ‘username’ field: *checked*

SIP Protocol properties

General Network Security Signaling T38 Jitter Buffer Default Real time Internals Auth. credentials

Port: 5060

UDP address: 0.0.0.0

Allow TCP connections: ☐

TCP address:

Automatic user creation: Persist

Accept unauthenticated calls: ☐

Allow subscribe: ☒

Allow overlap: ☒

Match users with 'username' field: ☒

Support 302 redirections: ☐

Automatic domain: ☐

Domain:

Use domain as realm: ☐

Trunk settings Under *Services* → *IPBX* → *Trunk management* → *SIP Protocol*.

- General
 - Name: GV18005551212
 - Authentication username: GV18005551212
 - Password: password
 - Caller ID: 18005551212
 - Connection type: Friend
 - IP Addressing type: static
 - * gvgw.simonics.com
 - Context: <your incoming call context>
- Register

- Register: checked
- Transport: UDP
- Name: GV18005551212
- Password: password
- Remote Server: GV18005551212
- Contact: 18005551212

- Signaling
 - Monitoring: Yes

Outgoing calls See the *Set the outgoing calls* section.

Incoming calls See the *Set the incoming calls* section.

Create an interconnection There are three types of interconnections :

- Customized
- SIP
- IAX

Customized interconnections Customized interconnections are mainly used for interconnections using DAHDI or Local channels:

- *Name* : it is the name which will appear in the outcall interconnections list,
- *Interface* : this is the channel name (for DAHDI see [DAHDI interconnections](#))
- *Interface suffix* (optional) : a suffix added after the dialed number (in fact the Dial command will dial:

```
<Interface>/<EXTEN><Interface suffix>
```

- *Context* : currently not relevant

DAHDI interconnections To use your DAHDI links you must create a customized interconnection.

Name : the name of the interconnection like **e1_span1** or **bri_port1**

Interface : must be of the form dahdi/[group order][group number] where :

- group order is one of :
 - g : pick the first available channel in group, searching from lowest to highest,
 - G : pick the first available channel in group, searching from highest to lowest,
 - r : pick the first available channel in group, going in round-robin fashion (and remembering where it last left off), searching from lowest to highest,
 - R : pick the first available channel in group, going in round-robin fashion (and remembering where it last left off), searching from highest to lowest.
- group number is the group number to which belongs the span as defined in the [/etc/asterisk/dahdi-channels.conf](#).

Warning: if you use a BRI card you MUST use per-port dahdi groups. You should not use a group like g0 which spans over several spans.

For example, add an interconnection to the menu *Services* → *IPBX* → *Trunk management* → *Customized*

Name : interconnection name
Interface : dahdi/g0

Customized trunk > Edit t2colt (dahdi/g0)

Name:

Interface:

Interface suffix:

Context:

Description :

Save

Debug Interesting Asterisk commands:

```
sip show peers
sip show registry
sip set debug on
```

Caller ID When setting up an interconnection with the public network or another PBX, it is possible to set a caller ID in different places. Each way to configure a caller ID has it's own use case.

The format for a caller ID is the following "My Name" <9999> If you don't set the number part of the caller ID, the dialplan's number will be used instead. This might not be a good option in most cases.

Outgoing call caller ID When you create an outgoing call, it's possible to set the it to internal, using the check box in the outgoing call configuration menu. When this option is activated, the caller's caller ID will be forwarded to the trunk. This option is use full when the other side of the trunk can reach the user with it's caller ID number.

When the caller's caller ID is not usable to the called party, the outgoing call's caller id can be fixed to a given value that is more use full to the outside world. Giving the public number here might be a good idea.

	Extern prefix	Prefix	Exten	Stripnum	Callerid
1			99X.	2	XiVO <5555555555

A user can also have a forced caller ID for outgoing calls. This can be use full for someone who has his own public number. This option can be set in the user's configuration page. The Outgoing Caller ID id option must be set to Customize. The user can also set his outgoing caller ID to anonymous.

The order of precedence when setting the caller ID in multiple place is the following.

1. Internal

Users > Edit

User1

General

Lines

No answer

Services

Voicemail

Groups

Func Keys

First name:

User1

Last name:

User picture:

Browse...

Mobile phone number:

Schedules:

Ringing time:

30 seconds

Simultaneous calls:

5

On-Hold Music:

default

Language:

fr_FR

Timezone:

Caller ID:

"User1"

Outgoing Caller ID:

Customize

"Bob" <5555551234:

Subroutine preprocess:

User field :

XIVO Client

Enable XIVO Client:

☒

Login:

pascal

Password:

pascal

Profile:

Client

Description:

Save

2. User's outgoing caller ID
3. Outgoing call
4. Default caller ID

Interactive Voice Response

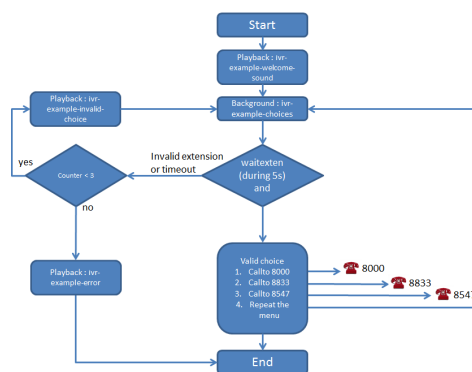
Introduction

Interactive voice response (IVR) is a technology that allows a computer to interact with humans through the use of voice and DTMF tones input via keypad. In telecommunications, IVR allows customers to interact with a company's host system via a telephone keypad or by speech recognition, after which they can service their own inquiries by following the IVR dialogue.

—Wikipedia

The IVR function is not yet available in graphic mode in XiVO. This functionality is currently supported using scripts, also named dialplan.

Use Case: Minimal IVR



Flowchart

Configuration File and Dialplan First step, you need to create a configuration file, that contain an asterisk context and your IVR dialplan. In our example, both (file and context) are named dp-ivr-example.

Copy all these lines in the newly created configuration file (in our case, dp-ivr-example) :

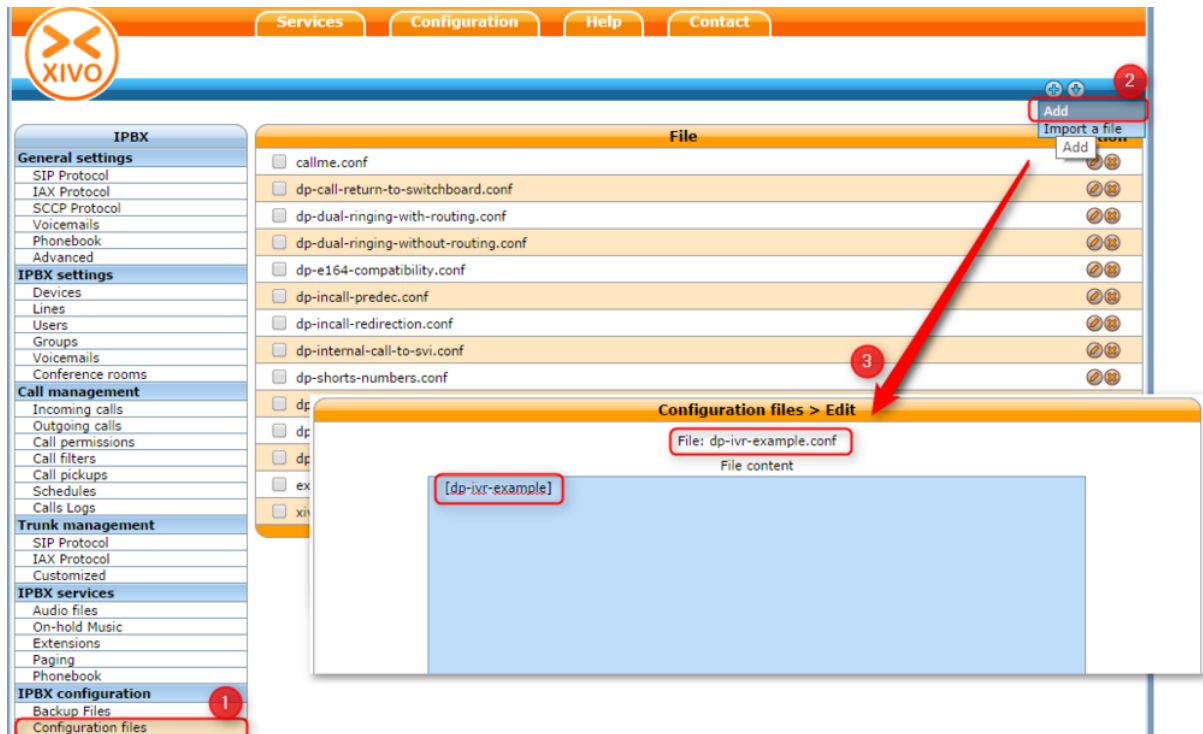
```
[dp-ivr-example]

exten = s,1,NoOp(### dp-ivr-example.conf ###)
same = n,NoOp(Set the context containing your ivr destinations.)
same = n,Set(IVR_DESTINATION_CONTEXT=my-ivr-destination-context)
same = n,NoOp(Set the directory containing your ivr sounds.)
same = n,Set(GV_DIRECTORY_SOUNDS=/var/lib/xivo/sounds/ivr-sounds)
same = n,NoOp(the system answers the call and waits for 1 second before continuing)
same = n,Answer(1000)

same = n,NoOp(the system plays the first part of the audio file "welcome to ...")
same = n(first),Playback(${GV_DIRECTORY_SOUNDS}/ivr-example-welcome-sound)

same = n,NoOp(variable "counter" is set to 0)
same = n(beginning),Set(counter=0)

same = n,NoOp(variable "counter" is incremented and the label "start" is defined)
same = n(start),Set(counter=${counter} + 1)
```



```

same = n,NoOp(counter variable is now = ${counter})
same = n,NoOp(waiting for 1 second before reading the message that indicate all choices)
same = n,Wait(1)
same = n,NoOp(play the message ivr-example-choices that contain all choices)
same = n,Background(${GV_DIRECTORY_SOUNDS}/ivr-example-choices)
same = n,NoOp(waiting for DTMF during 5s)
same = n,Waitexten(5)

;##### CHOICE 1 #####
exten = 1,1,NoOp(pressed digit is 1, redirect to 8000 in ${IVR_DESTINATION_CONTEXT} context)
exten = 1,n,Goto(${IVR_DESTINATION_CONTEXT},8000,1)

;##### CHOICE 2 #####
exten = 2,1,NoOp(pressed digit is 2, redirect to 8833 in ${IVR_DESTINATION_CONTEXT} context)
exten = 2,n,Goto(${IVR_DESTINATION_CONTEXT},8833,1)

;##### CHOICE 3 #####
exten = 3,1,NoOp(pressed digit is 3, redirect to 8547 in ${IVR_DESTINATION_CONTEXT} context)
exten = 3,n,Goto(${IVR_DESTINATION_CONTEXT},8547,1)

;##### CHOICE 4 #####
exten = 4,1,NoOp(pressed digit is 4, redirect to start label in this context)
exten = 4,n,Goto(s,start)

;##### TIMEOUT #####
exten = t,1,NoOp(no digit pressed for 5s, process it like an error)
exten = t,n,Goto(i,1)

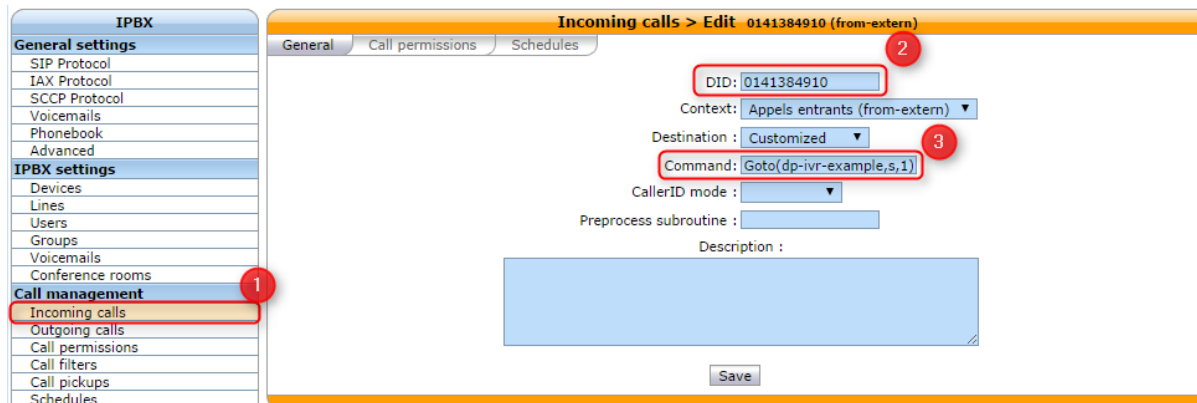
;##### INVALID CHOICE #####
exten = i,1,NoOp(if counter variable is 3 or more, then goto label "error")
exten = i,n,GotoIf(${ ${counter} }>=3,error)
exten = i,n,NoOp(pressed digit is invalid and less than 3 errors: the guide ivr-exemple-invalid-choices)
exten = i,n,Playback(${GV_DIRECTORY_SOUNDS}/ivr-example-invalid-choice)
exten = i,n,Goto(s,start)
exten = i,n(error),Playback(${GV_DIRECTORY_SOUNDS}/ivr-example-error)

```

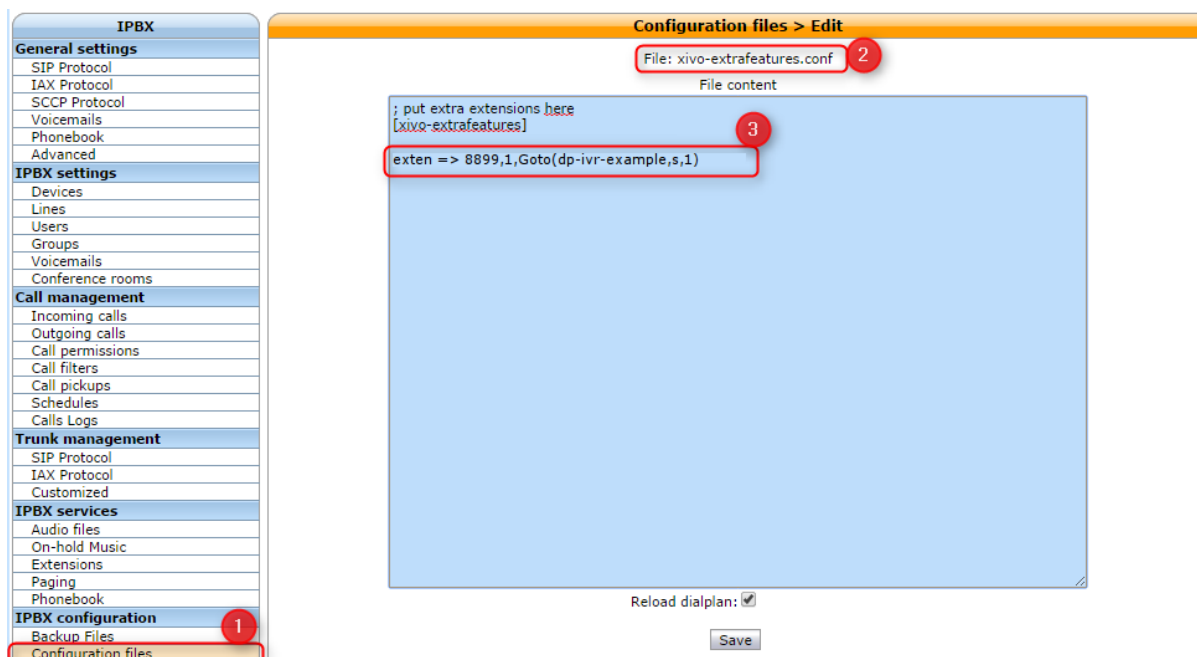
```
exten = i,n, Hangup()
```

IVR external dial To call the script dp-ivr-example from an external phone, you must create an incoming call and redirect the call to the script dp-ivr-example with the command :

```
Goto(dp-ivr-example,s,1)
```



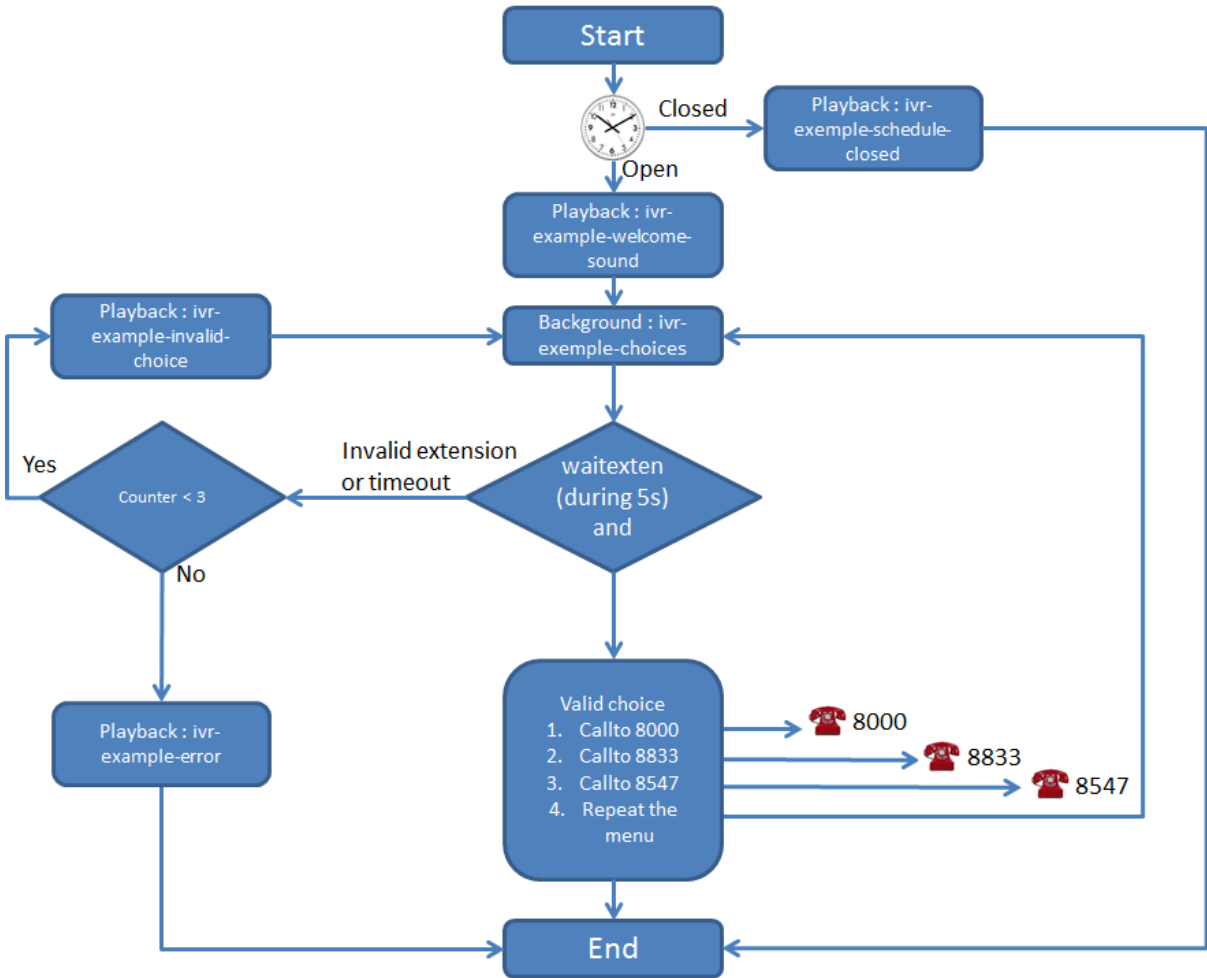
IVR internal dial To call the script dp-ivr-example from an internal phone you must create an entry in the default context (xivo-extrafeatures is included in default). The best way is to add the extension in the file xivo-extrafeatures.conf.



```
exten => 8899,1,Goto(dp-ivr-example,s,1)
```

Use Case: IVR with a schedule In many cases, you need to associate your IVR to a schedule to indicate when your company is closed.

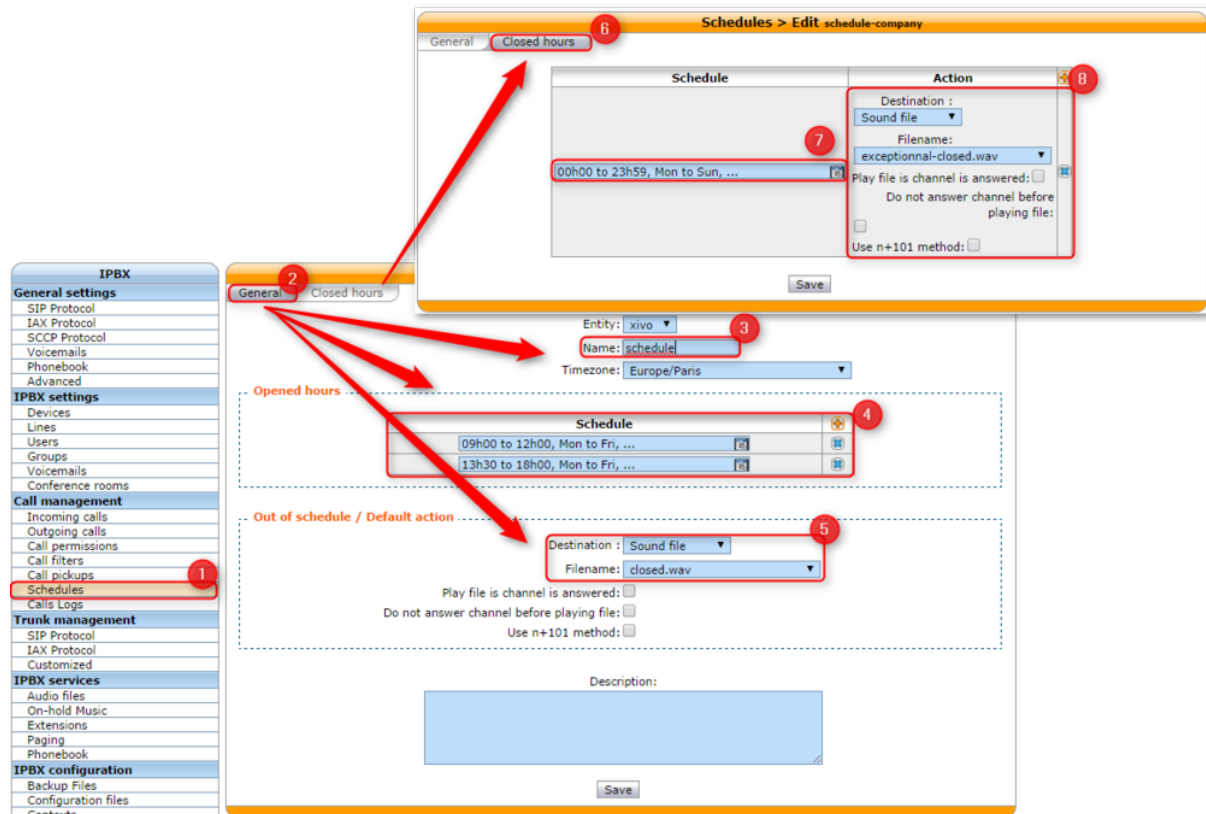
Flowchart



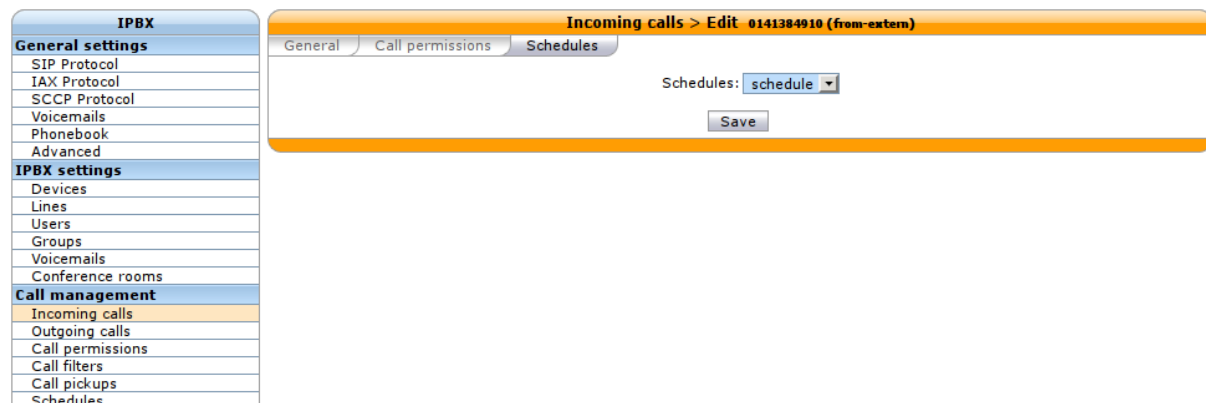
Create Schedule First step, create your schedule (1) from the menu *Call management* → *Schedules*. In the General tab, give a name (3) to your schedule and configure the open hours (4) and select the sound which is played when the company is closed.

In the Closed hours tab (6), configure all special closed days (7) and select the sound that indicate to the caller that the company is exceptionally closed.

The IVR script is now only available during workdays.

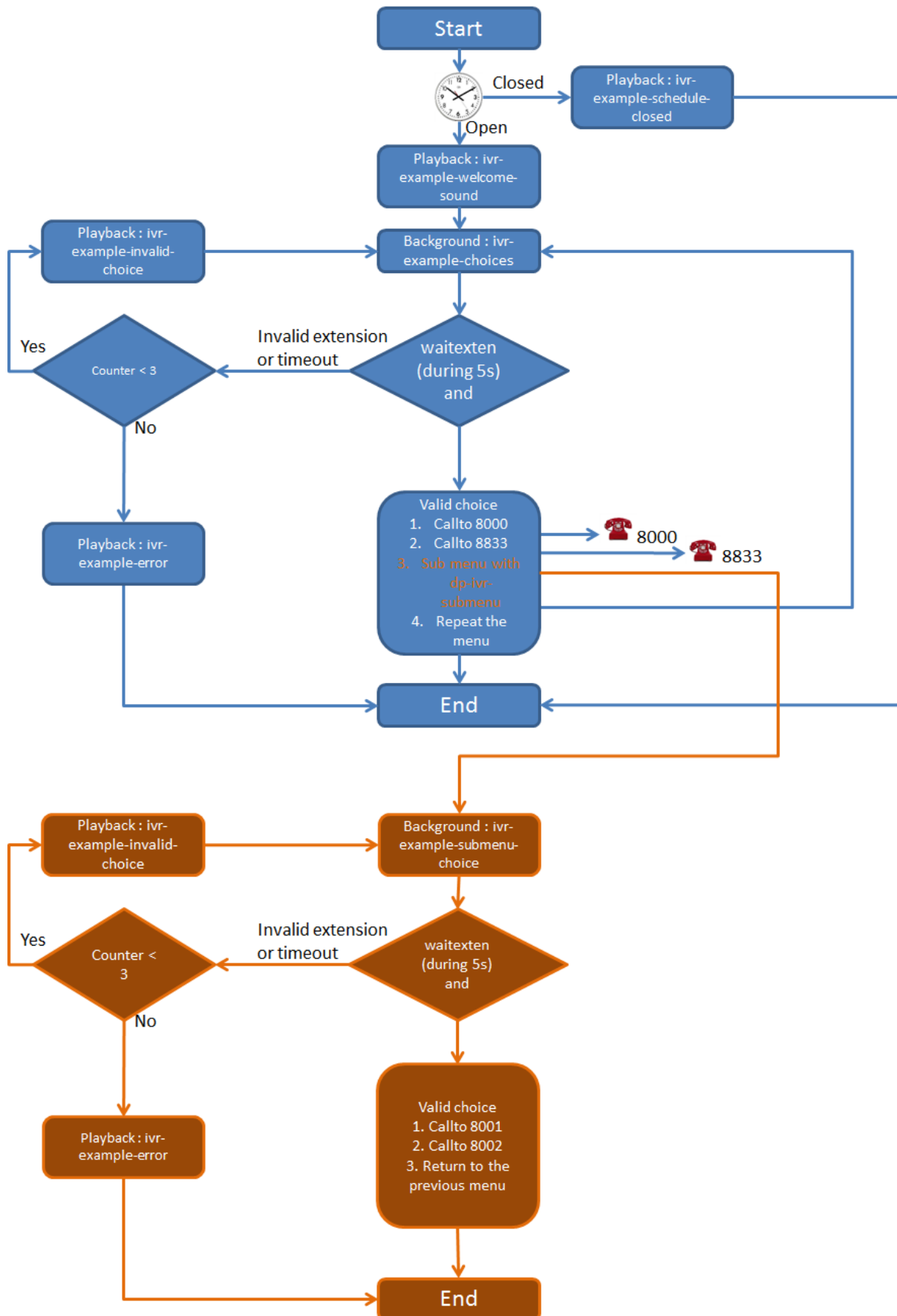


Assign Schedule to Incall Return editing your Incall (*Call management* → *Incoming calls*) and assign the newly created schedule in the “Schedules” tab



Use Case: IVR with submenu

Flowchart



Configuration File and Dialplan Copy all these lines (2 contexts) in a configuration file on your XiVO server :

```
[dp-ivr-example]

exten = s,1,NoOp(### dp-ivr-example.conf ###)
same = n,NoOp(Set the context containing your ivr destinations.)
same = n,Set(IVR_DESTINATION_CONTEXT=my-ivr-destination-context)
same = n,NoOp(Set the directory containing your ivr sounds.)
same = n,Set(GV_DIRECTORY_SOUNDS=/var/lib/xivo/sounds/ivr-sounds)
same = n,NoOp(the system answers the call and waits for 1 second before continuing)
same = n,Answer(1000)

same = n,NoOp(the system plays the first part of the audio file "welcome to ...")
same = n(first),Playback(${GV_DIRECTORY_SOUNDS}/ivr-example-welcome-sound)

same = n,NoOp(variable "counter" is set to 0)
same = n(beginning),Set(counter=0)

same = n,NoOp(variable "counter" is incremented and the label "start" is defined)
same = n(start),Set(counter=${counter} + 1)

same = n,NoOp(counter variable is now = ${counter})
same = n,NoOp(waiting for 1 second before reading the message that indicate all choices)
same = n,Wait(1)
same = n,NoOp(play the message ivr-example-choices that contain all choices)
same = n,Background(${GV_DIRECTORY_SOUNDS}/ivr-example-choices)
same = n,NoOp(waiting for DTMF during 5s)
same = n,Waitexten(5)

;##### CHOICE 1 #####
exten = 1,1,NoOp(pressed digit is 1, redirect to 8000 in ${IVR_DESTINATION_CONTEXT} context)
exten = 1,n,Goto(${IVR_DESTINATION_CONTEXT},8000,1)

;##### CHOICE 2 #####
exten = 2,1,NoOp(pressed digit is 2, redirect to 8833 in ${IVR_DESTINATION_CONTEXT} context)
exten = 2,n,Goto(${IVR_DESTINATION_CONTEXT},8833,1)

;##### CHOICE 3 #####
exten = 3,1,NoOp(pressed digit is 3, redirect to the submenu dp-ivr-submenu)
exten = 3,n,Goto(dp-ivr-submenu,s,1)

;##### CHOICE 4 #####
exten = 4,1,NoOp(pressed digit is 4, redirect to start label in this context)
exten = 4,n,Goto(s,start)

;##### TIMEOUT #####
exten = t,1,NoOp(no digit pressed for 5s, process it like an error)
exten = t,n,Goto(i,1)

;##### INVALID CHOICE #####
exten = i,1,NoOp(if counter variable is 3 or more, then goto label "error")
exten = i,n,GotoIf(${counter}>=3?error)
exten = i,n,NoOp(pressed digit is invalid and less than 3 errors: the guide ivr-example-invalid-c
exten = i,n,Playback(${GV_DIRECTORY_SOUNDS}/ivr-example-invalid-choice)
exten = i,n,Goto(s,start)
exten = i,n(error),Playback(${GV_DIRECTORY_SOUNDS}/ivr-example-error)
exten = i,n,Hangup()

[dp-ivr-submenu]

exten = s,1,NoOp(### dp-ivr-submenu ###)
```



```

same = n,NoOp(the system answers the call and waits for 1 second before continuing)
same = n,Answer(1000)

same = n,NoOp(variable "counter" is set to 0)
same = n(beginning),Set(counter=0)

same = n,NoOp(variable "counter" is incremented and the label "start" is defined)
same = n(start),Set(counter=${counter} + 1)

same = n,NoOp(counter variable is now = ${counter})
same = n,NoOp(waiting for 1 second before reading the message that indicate all choices)
same = n,Wait(1)
same = n,NoOp(play the message ivr-example-choices that contain all choices)
same = n,Background(${GV_DIRECTORY_SOUNDS}/ivr-example-submenu-choices)
same = n,NoOp(waiting for DTMF during 5s)
same = n,Waitexten(5)

;##### CHOICE 1 #####
exten = 1,1,NoOp(pressed digit is 1, redirect to 8000 in ${IVR_DESTINATION_CONTEXT} context)
exten = 1,n,Goto(${IVR_DESTINATION_CONTEXT},8000,1)

;##### CHOICE 2 #####
exten = 2,1,NoOp(pressed digit is 2, redirect to 8001 in ${IVR_DESTINATION_CONTEXT} context)
exten = 2,n,Goto(${IVR_DESTINATION_CONTEXT},8001,1)

;##### CHOICE 3 #####
exten = 3,1,NoOp(pressed digit is 3, redirect to the previous menu dp-ivr-example)
exten = 3,n,Goto(dp-ivr-example,s,beginning)

;##### TIMEOUT #####
exten = t,1,NoOp(no digit pressed for 5s, process it like an error)
exten = t,n,Goto(i,1)

;##### INVALID CHOICE #####
exten = i,1,NoOp(if counter variable is 3 or more, then goto label "error")
exten = i,n,GotoIf(${counter}>=3?error)
exten = i,n,NoOp(pressed digit is invalid and less than 3 errors: the guide ivr-example-invalid-choice)
exten = i,n,Playback(${GV_DIRECTORY_SOUNDS}/ivr-example-invalid-choice)
exten = i,n,Goto(s,start)
exten = i,n(error),Playback(${GV_DIRECTORY_SOUNDS}/ivr-example-error)
exten = i,n,Hangup()

```

Monitoring














































The Monitoring section gives an overview of a XiVO system's status and of all monitored processes. It is divided into 6 sections :

- *System*
- *Device*
- *CPU*
- *Network*
- *Memory*
- *Other Services*

System				
Name	o0-git			
Operating system	Linux			
Kernel version	3.2.0-4-686-pae			
IP address	10.33.255.1			
DNS address	10.33.255.1			
Uptime	1 day(s) 05:44:09			
Load average	0.65 0.73 0.51			

Device				
Partition	Percent	Free	Used	Total
data-system	61.70 %	0	3792.1	6138.0
data-var	58.90 %	0	1683.5	2854.0

Memory						
type	Percent	Free	Used	Buffers	Cached	Total
Physical memory	59.54 %	9.70 MiB	218.25 MiB	31.25 KiB	138.56 MiB	366.54 MiB
Swap partition	8.80 %	872.83 MiB	84.20 MiB	-	-	957.03 MiB

Other services						
Process	Status	Uptime	CPU	Memory		Action
asterisk	Running	1 day(s) 02:41:40	0.20 %	3.85 %	14.10 MiB	  
data-system	Accessible	-	-	-	-	  
data-var	Accessible	-	-	-	-	  
isc-dhcp-server	Running	1 day(s) 05:42:38	0.00 %	0.32 %	1.16 MiB	  
ntpd	Running	1 day(s) 05:42:32	0.00 %	0.33 %	1.19 MiB	  
rabbitmq	Running	1 day(s) 02:41:49	0.00 %	2.17 %	7.95 MiB	  
xivo-agent	Running	1 day(s) 02:41:30	0.00 %	3.87 %	14.18 MiB	  
xivo-agid	Running	1 day(s) 02:41:36	0.00 %	1.93 %	7.06 MiB	  
xivo-ami	Running	1 day(s) 02:41:33	0.00 %	1.27 %	4.65 MiB	  
xivo-call-logd	Running	1 day(s) 02:41:32	0.00 %	2.82 %	10.33 MiB	  
xivo-confgend	Running	1 day(s) 02:41:43	0.00 %	3.19 %	11.71 MiB	  
xivo-ctid	Running	1 day(s) 02:41:27	0.00 %	6.07 %	22.24 MiB	  
xivo-provd	Running	1 day(s) 02:41:41	0.00 %	2.36 %	8.66 MiB	  
xivo-restapid	Unmonitored	-	-	-	-	  
xivo-sysconfd	Running	1 day(s) 02:41:45	0.00 %	1.61 %	5.89 MiB	  

System Displays generic information about the operating system, network addresses, uptime and load average. Read only.

Device Displays free/used space on physical storage partitions. Read only.

CPU Monitors the CPU usage. Read only.

Network Displays network interfaces and corresponding network traffic. Read only.

Memory Displays Physical and swap memory usage. Read only.

Other Services Lists Xivo related processes (most of which are daemons) with their corresponding status, uptime, resource usage and controls to Restart service (blue button), stop service (red button) and stop monitoring service (grey button).

Music on Hold

The menu *Services* → *IPBX* → *IPBX services* → *On-hold Music* leads to the list of available on-hold musics.

Categories Available categories are:

- files: play sound files. Formats supported:

Format Name	Filename Extension
G.719	.g719
G.723	.g723 .g723sf
G.726	.g726-40 .g726-32 .g726-24 .g726-16
G.729	.g729
GSM	.gsm
iLBC	.ilbc
Ogg Vorbis	.ogg (only mono files sampled at 8000 Hz)
G.711 A-law	.alaw .al .alw
G.711 μ -law	.pcm .ulaw .ul .mu .ulw
G.722	.g722
Au	.au
Siren7	.siren7
Siren14	.siren14
SLN	.raw .sln .sln12 .sln16 .sln24 .sln32 .sln44 .sln48 .sln96 .sln192
VOX	.vox
WAV	.wav .wav16
WAV GSM	.WAV .wav49

Only 1 audio channel must be present per file, i.e. files must be in mono.

If your music on hold files don't seem to work, you should look for errors in the asterisk logs.

The on-hold music will always play from the start.

- mp3: play MP3 files.

The on-hold music will play from an arbitrary position on the track, it will not play from the start.

- custom: do not play sound files. Instead, run an external process. That process must send on stdout the same binary format than WAV files.

Example process: `/usr/bin/mpg123 -s --mono -y -f 8192 -r 8000 http://streaming.example.com/stream.mp3`

Note: Processes run by custom categories are started as soon as the category is created and will only stop when the category is deleted. This means that on-hold music fed from online streaming will constantly be receiving network traffic, even when there are no calls.

Paging

With XiVO, you can define paging (i.e. intercom) extensions to page a group of users. When calling a paging extension, the phones of the specified users will auto-answer, if they support it.

You can manage your paging extensions via the *Services* → *IPBX* → *Paging* page.

When adding a new paging extension, the number can be any numeric value; to call it, you just need to prefix the paging number with *11.

Parking

With XiVO it is possible to park calls, the same way you may park your car in a car parking. If you define supervised keys on a phone set for all the users of a system, when a call is parked, all the users are able to see that some one is waiting for an answer, push the phone key and get the call back to the phone.

There is a default parking number, 700, which is already configured when you install XiVO, but you may change the default configuration by editing the parking extension in menu *Service* → *IPBX* → *IPBX Services* → *Extensions* → *Advanced* → *Parking*

Paging > Edit

General
Users


Number:

Full duplex audio: ☐

Ignore attempts to forward the call: ☐

Record the page into a file: ☐

Quiet, do not play beep to caller: ☐

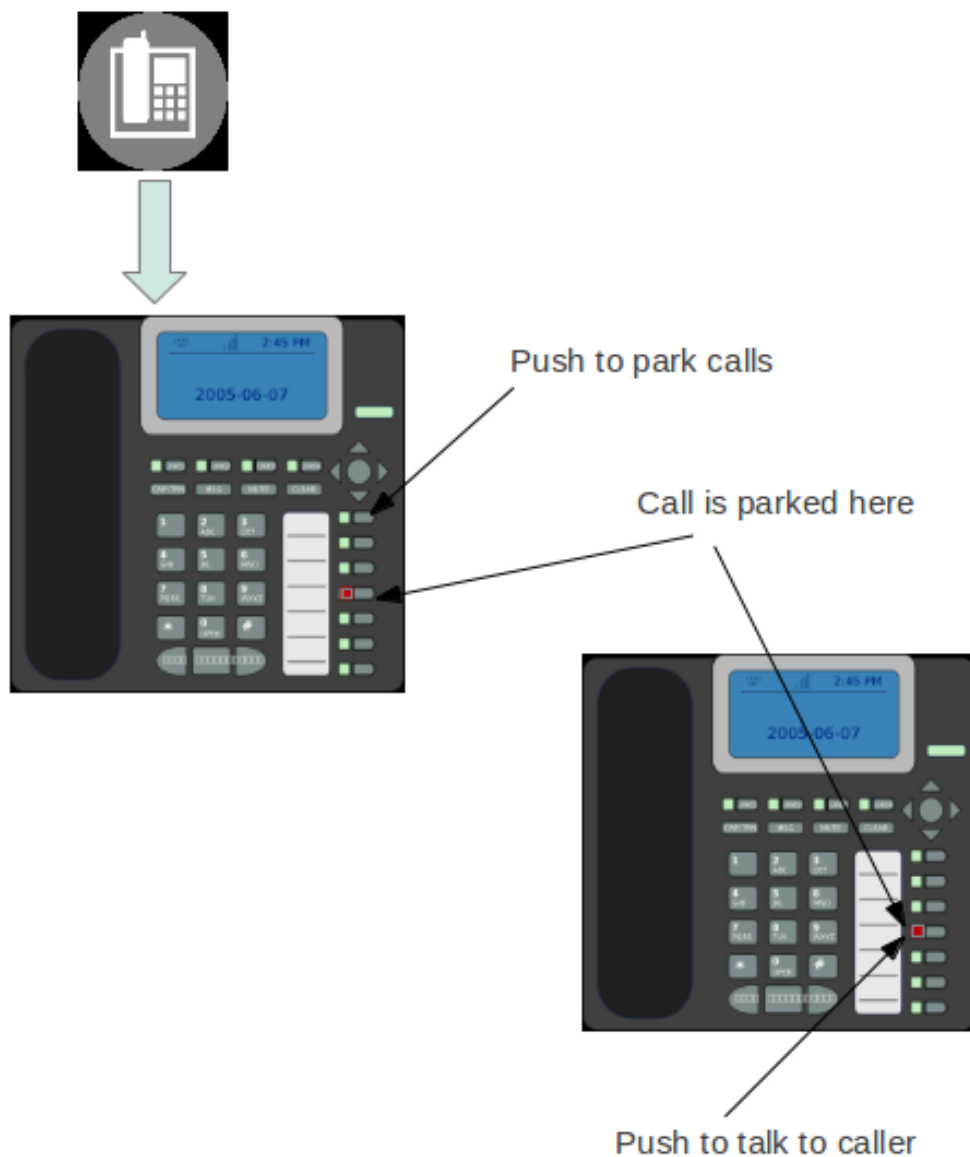
Timeout: 

Do not play simultaneous announcement to caller: ☐

Play simultaneous announcement to called users: ☐

The announcement to playback in all devices:

Description:



eral Voicemail Agents **Advanced**

Extension: 900
 Context: parkedcalls
 Wait delay: 30 seconds
 Extension to parked calls: 901-910
 Look for the next call: ☐
 Parkings hints: ☒
 Allow dynamically created parkinglots: ☐
 On parkedcall timeout: Send parked call to the dialplan
 Who to play courtesy tone when picking up parked call: Caller
 Allow DTMF based transfers when picking up parked call: None
 Allow DTMF based parking when picking up parked call: None
 Allow DTMF based hangups when picking up parked call: None
 Allow DTMF based one-touch recording when picking up parked call: None
 MOH class to play to parked calls: default
 Use ADSI announces: ☐
 Save

Using this extension, you may define the parking number used to park call, the parking lots, whether the system is rotating over the parking lots to park the calls, enable parking hint if you want to be able to supervise the parking using phone keys and other system default parameters.

You have two options in case of parking timeout :

- Callback the peer that parked this call

In this case the call is sent back to the user who parked the call.

- Send park call to the dialplan

In case you don't want to call back the user who parked the call, you have the option to send the call to any other extension or application. If the parking times out, the call is sent back to the dialplan in context [parkedcallstimeout]. You can define this context in a dialplan configuration file *Service* → *IPBX* → *Configuration Files* where you may define this context with dialplan commands.

Example:

```
[parkedcallstimeout]
exten = s,1,Noop('park call time out')
same = n,Playback(hello-world)
same = n,Hangup()
```

It is also usual to define supervised phone keys to be able to park and unpark calls as in the example below.

1	User	Jean-Yves LEBLEU	Enabled	
4	Parking	900	Parking	Disabled
5	Parking position	901	701	Enabled
6	Parking position	902	702	Enabled
7	Parking position	903	703	Enabled
8	Parking position	904	704	Enabled

Save

Phonebook

A global phone book can be defined in *Services → IPBX → IPBX Services → Phonebook*. The phone book can be used from the XiVO Client, from the phones directory look key if the phone is compatible and are used to set the Caller ID for incoming calls.

You can add entries one by one or you can mass-import from a CSV file.

Note: To configure phonebook, see [Directories](#).

Mass-import contacts Go in the *Services → IPBX → IPBX Services → Phonebook* section and move your mouse cursor on the + button in the upper right corner. Select *Import a file*.

The file to be imported must be a CSV file, with a pipe character | as field delimiter. The file must be encoded in UTF-8 (without an initial BOM).

Mandatory headers are :

- title (possible values : “mr”, “mrs”, “ms”)
- displayname

Optional headers are :

- firstname
- lastname
- society
- mobilenumbers¹
- email
- url
- description
- officenumbers¹
- faxnumbers¹
- officeaddress1
- officeaddress2
- officecity
- officestate
- officezipcode
- officecountry²
- homenumbers¹
- homeaddress1
- homeaddress2
- homecity
- homestate
- homezipcode
- homecountry²

¹ These fields must contain only numeric characters, no space, point, etc.

² These fields must contain ISO country codes. The complete list is described [here](#).

- othernumber ¹
- otheraddress1
- otheraddress2
- othercity
- otherstate
- otherzipcode
- othercountry ²

Provisioning

XiVO supports the auto-provisioning of a large number of telephony *Devices*, including SIP phones, SIP ATAs, and even softphones.

Introduction The auto-provisioning feature found in XiVO make it possible to provision, i.e. configure, a lots of telephony devices in an efficient and effortless way.

How it works Here's a simplified view of how auto-provisioning is supported on a typical SIP hardphone:

1. The phone is powered on
2. During its boot process, the phone sends a DHCP request to obtain its network configuration
3. A DHCP server replies with the phone network configuration + an HTTP URL
4. The phone use the provided URL to retrieve a common configuration file, a MAC-specific configuration file, a firmware image and some language files.

Building on this, configuring one of the supported phone on XiVO is as simple as:

1. *Configuring the DHCP Server*
2. *Installing the required provd plugin*
3. Powering on the phone
4. Dialing the user's provisioning code from the phone

And *voila*, once the phone has rebooted, your user is ready to make and receive calls. No manual editing of configuration files nor fiddling in the phone's web interface.

Limitations

- Device synchronisation does not work in the situation where multiple devices are connected from behind a NAPT network equipment. The devices must be resynchronised manually.

External links

- [Introduction to provd plugin model](#)
- [HTTP/TFTP requests processing in provd - part 1](#)
- [HTTP/TFTP requests processing in provd - part 2](#)

Basic Configuration You have two options to get your phone to be provisioned:

- Set up a DHCP server
- Tell manually each phone where to get the provisioning informations

You may want to manually configure the phones if you are only trying XiVO or if your network configuration does not allow the phones to access the XiVO DHCP server.

You may want to set up a DHCP server if you have a significant number of phones to connect, as no manual intervention will be required on each phone.

Configuring the DHCP Server XiVO includes a DHCP server that facilitate the auto-provisioning of telephony devices. It is *not* activated by default.

There's a few things to know about the peculiarities of the included DHCP server:

- it only answers to DHCP requests from *supported devices*.
- it only answers to DHCP requests coming from the VoIP subnet (see *network configuration*).

This means that if your phones are on the same broadcast domain than your computers, and you would like the DHCP server on your XiVO to handle both your phones and your computers, that won't do it.

The DHCP server is configured via the *Configuration* → *Network* → *DHCP* page:

Fig. 1.67: *Configuration* → *Network* → *DHCP*

Active Activate/desactivate the DHCP server.

Pool start The lower IP address which will be assigned dynamically. This address should be in the VoIP subnet.
Example: 10.0.0.10.

Pool end The higher IP address which will be assigned dynamically. This address should be in the VoIP subnet.
Example: 10.0.0.99.

Extra network interfaces A list of space-separated network interface name. Example: eth0.

Useful if you have done some custom configuration in the `/etc/dhcp/dhcpd_extra.conf` file. You need to explicitly specify the additional interfaces the DHCP server should listen on.

After saving your modifications, you need to click on *Apply system configuration* for them to be applied.

Installing provd Plugins The installation and management of provd plugins is done via the *Configuration* → *Provisioning* → *Plugin* page:

The page shows the list of both the installed and installable plugins. You can see if a plugin is installed or not by looking at the *Action* column.

Here's the list of other things that can be done from this page:

- update the list of installable plugins, by clicking on the top right icon. On a fresh XiVO installation, this is the first thing to do.
- install a new plugin
- upgrade an installed plugin
- uninstall an installed plugin

Name	Description	Version	Size	Action
null	Plugin that offers no configuration service and rejects TFTP/HTTP.	1.0-a	1.05 kb	
xivo-aastra-2.6.0.2019	Greffon pour Aastra 6730i, 6731i, 6751i, 6753i, 6755i, 6757i, 675...	0.3	8.29 kb	
xivo-aastra-3.2.2.1136	Greffon pour Aastra 6730i, 6731i, 6739i, 6753i, 6755i, 6757i, 675...	0.1.16 / 0.4	8.74 kb	
xivo-aastra-3.2.2.6268	Greffon pour Aastra 6735i et 6737i en version 3.2.2.6268.	0.3	7.5 kb	
xivo-aastra-switchboard	Greffon pour Aastra 6731i, 6755i, 6757i en version 3.2.2.1136.	0.1	7.99 kb	
xivo-alcatel-2.01.10	Greffon pour Alcatel IP Touch 4008 and 4018 "extended edition" en...	0.1.1	6.07 kb	
xivo-avaya-4.1.13	Greffon pour Avaya (auparavant connu comme Nortel) 1220 IP et 123...	0.1.1	4.65 kb	
xivo-cisco-pap2t-5.1.6	Greffon pour Cisco PAP2T en version 5.1.6.	0.2	10.92 kb	
xivo-cisco-sccp-9.0.3	Greffon pour Cisco 7911G, 7941G, 7941G-GE and 7961G en version 9...	0.2 / 0.3	6.9 kb	
xivo-cisco-sccp-legacy	Greffon pour Cisco 7912G, 7940G et 7960G en version 8.1.2 du logi...	0.2 / 0.3	7.05 kb	
xivo-cisco-spa-7.4.8	Greffon pour Cisco SMB SPA301, 303, 501G, 502G, 504G, 508G, 509G,...	0.3	11.99 kb	
xivo-cisco-spa-legacy	Greffon pour Cisco (auparavant connu comme Linksys) SPA901, 921, ...	0.3	12.02 kb	
xivo-cisco-spa2102-5.2.12	Greffon pour Cisco SPA2102 en version 5.2.12.	0.3	10.89 kb	
xivo-cisco-spa3102-5.1.10	Greffon pour Cisco SPA3102 en version 5.1.10.	0.2 / 0.3	10.92 kb	
xivo-cisco-spa8000-6.1.3	Greffon pour Cisco SPA8000 en version 6.1.3.	0.3	10.91 kb	
xivo-cisco-spa8800-6.1.7	Greffon pour Cisco SPA8800 en version 6.1.7.	0.3	10.9 kb	
xivo-digium-1.1.0.0	Greffon pour Digium D40, D50 et D70 en version 1.1.0.0.	0.3	4.09 kb	
xivo-gigaset-C470	Greffon pour Gigaset (aussi connu comme Siemens) C470 IP, C475 IP...	0.1.0	4.79 kb	
xivo-gigaset-C590	Greffon pour Gigaset (aussi connu comme Siemens) C590 IP, C595 IP...	0.1.0	5.2 kb	
xivo-jitsi-1	Greffon pour Jitsi en version 1.0.	0.1.1	3.04 kb	

Fig. 1.68: Configuration → Provisioning → Plugin

- edit an installed plugin, i.e. install/uninstall optional files that are specific to each plugin, like firmware or language files

After installing a new plugin, you are automatically redirected to its edit page. You can then download and install optional files specific to the plugin. You are strongly advised to install firmware and language files for the phones you'll use although it's often not a strict requirement for the phones to work correctly.

Warning: If you uninstall a plugin that is used by some of your devices, they will be left in an unconfigured state and won't be associated to another plugin automatically.

The search box at the top comes in handy when you want to find which plugin to install for your device. For example, if you have a Cisco SPA508G, enter "508" in the search box and you should see there's 1 plugin compatible with it.

Note: If your device has a number in its model name, you should use only the number as the search keyword since this is what usually gives the best results.

It's possible there will be more than 1 plugin compatible with a given device. In these cases, the difference between the two plugins is usually just the firmware version the plugins target. If you are unsure about which version you should install, you should look for more information on the vendor website.

It's good practice to only install the plugins you need and no more.

Alternative plugins repository By default, the list of plugins available for installation are the stable plugins for the officially supported devices.

This can be changed in the *Configuration* → *Provisioning* → *General* page, by setting the *URL* field to one of the following value:

- <http://provd.xivo.solutions/plugins/1/stable/> – *officially supported devices* “stable” repository (*default*)
- <http://provd.xivo.solutions/plugins/1/testing/> – *officially supported devices* “testing” repository

- `http://provd.xivo.solutions/plugins/1/archive/` – officially supported devices “archive” repository
- `http://provd.xivo.solutions/plugins/1/addons/stable/` – *community supported devices* “stable” repository
- `http://provd.xivo.solutions/plugins/1/addons/testing/` – community supported devices “testing” repository

The difference between the stable and testing repositories is that the latter might contain plugins that are not working properly or are still in development.

The archive repository contains plugins that were once in the stable repository.

After setting a new URL, you must refresh the list of installable plugins by clicking the update icon of the *Configuration* → *Provisioning* → *Plugin* page.

How to manually tell the phones to get their configuration If you have set up a DHCP server on XiVO and the phones can access it, you can skip this section.

The according provisioning plugins must be installed.

Aastra On the web interface of your phone, go to *Advanced settings* → *Configuration server*, and enter the following settings:

Download Protocol



HTTP Server

HTTP Path

HTTP Port

HTTP

<XIVO IP address>

Aastra

8667

Polycom On the phone, go to *Menu* → *Settings* → *Advanced* → *Admin Settings* → *Network configuration* → *Server Menu* and enter the following settings:

- Server type: HTTP
- Server address: `http://<XIVO IP address>:8667/000000000000.cfg`

Then save and reboot the phone.

Snom On the web interface of your phone, go to *Setup* → *Advanced* → *Update* and enter the following settings:

Update:

Update Policy: ?

Setting URL: ?

Settings refresh timer: ?

PnP Config: ☐ on ☒ off ?

Yealink On the web interface of your phone, go to *Settings* → *Auto Provision*, and enter the following settings:

- Server URL: `http://<XiVO IP address>:8667`

Yealink | T46G

Auto Provision

PNP Active ☒ On ☐ Off ?

DHCP Active ☒ On ☐ Off ?

Custom Option(128~254) ?

DHCP Option Value ?

Server URL ?

Save the changes by clicking on the *Confirm* button and then click on the *Autoprovision Now* button.

Autoprovisioning a Device Once you have installed the proper provd plugins for your devices and setup correctly your DHCP server, you can then connect your devices to your network.

But first, go to *Services* → *IPBX* → *Devices* page. You will then see that no devices are currently known by your XiVO:

XiVO

IPBX

General settings

- SIP Protocol
- IAX Protocol
- SCCP Protocol
- Dundi Protocol
- Voicemails
- Phonebook
- Advanced
- Outbound MWIs

IPBX settings

- Devices**
- Lines
- Users
- Groups
- Voicemails
- Conference rooms

Call management

- Incoming calls
- Outgoing calls
- Call permissions
- Call filters
- Call pickups
- Schedules

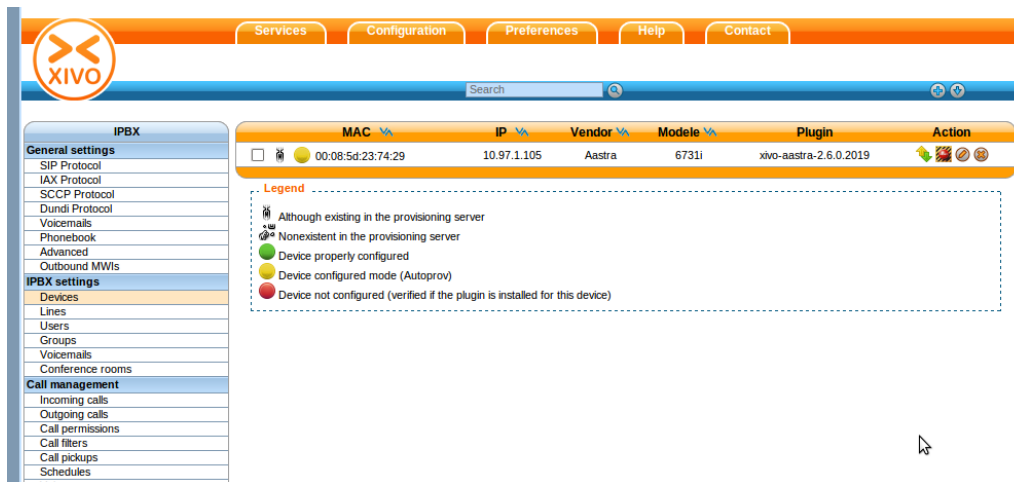
MAC IP Vendor Model Plugin Action

No device found

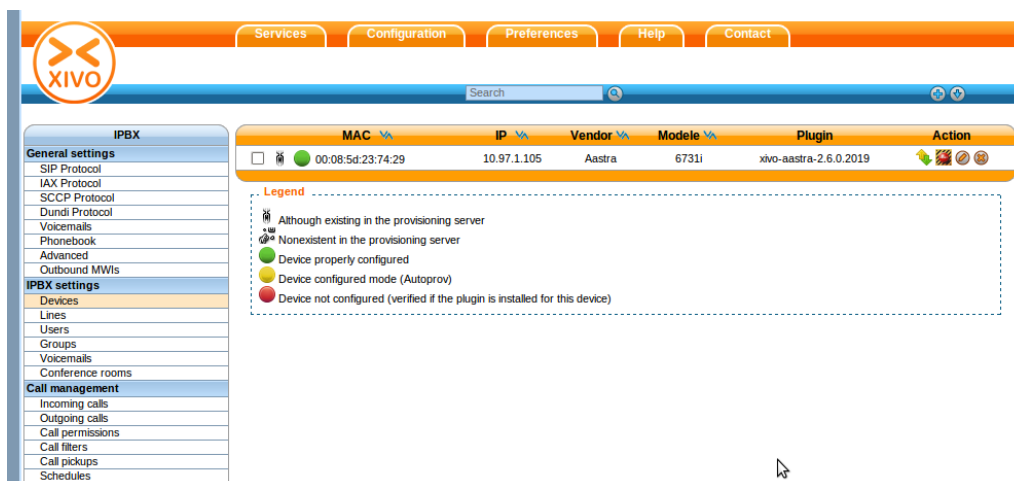
Legend

- Although existing in the provisioning server
- Nonexistent in the provisioning server
- Device properly configured
- Device configured mode (Autoprovision)
- Device not configured (verified if the plugin is installed for this device)

You can then power on your devices on your LAN. For example, after you power on an Aastra 6731i and give it the time to boot and maybe upgrade its firmware, you should then see the phone having its first line configured as 'autoprov', and if you refresh the devices page, you should see that your XiVO now knows about your 6731i:



You can then dial from your Aastra 6731i the provisioning code associated to a line of one of your user. You will hear a prompt thanking you and your device should then reboot in the next few seconds. Once the device has rebooted, it will then be properly configured for your user to use it. And also, if you update the device page, you'll see that the icon next to your device has now passed to green:



Resetting a Device

From the Device List in the Webi To remove a phone from XiVO or enable a device to be used for another user there are two different possibilities :

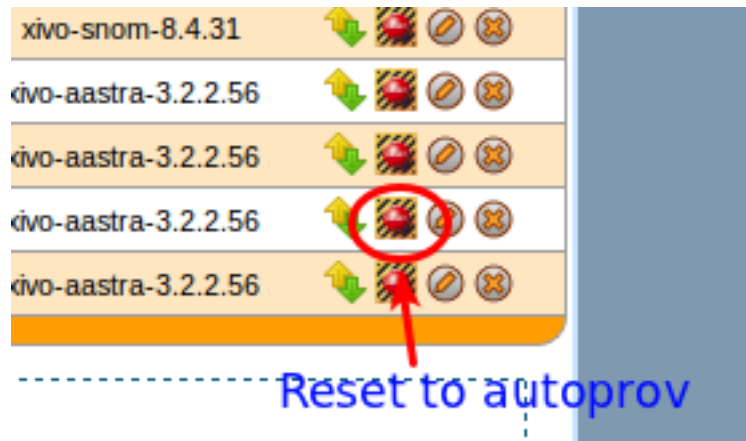
- click on the `reset to autoprov` button on the web interface

The phone will restarts and display autoprov, ready to be used for another user.

From the User Form in the Webi

Device With one User Only Associated Edit the user associated to the device and put the device field to null.

- click on the `Save` button on the web interface



The phone doesn't restart and the phone is in autoprov mode in the device list.

You can synchronize the device to reboot it.

Device with Several Users Associated Edit the primary user associated to the terminal (one with the line 1) and put the device field to null.

- click on the *Save* button on the web interface

The primary line of the phone has been removed, so the device will lose its funkeys associated to primary user but there others lines associated to the device will stay provisionned.

The phone doesn't restart and the phone is in autoprov mode in the device list.

You can synchronize the device for reboot it.

From a Device

- Dial ***guest** (*48378) on the phone dialpad followed by **xivo** (9486) as a password

The phone restarts and display autoprov, ready to be used for another user.

Advanced Configuration

DHCP Integration If your phones are getting their network configuration from your XiVO's DHCP server, it's possible to activate the DHCP integration on the *Configuration* → *Provisioning* → *General* page.

What DHCP integration does is that, on every DHCP request made by one of your phones, the DHCP server sends information about the request to `provd`, which can then use this information to update its device database.

This feature is useful for phones which lack information in their TFTP/HTTP requests. For example, without DHCP integration, it's impossible to extract model information for phones from the Cisco 7900 series. Without the model information extracted, there's chance your device won't be automatically associated to the best plugin.

This feature can also be useful if your phones are not always getting the same IP addresses, for one reason or another. Again, this is useful only for some phones, like the Cisco 7900; it has no effect for Aastra 6700.

Creating Custom Templates Custom templates comes in handy when you have some really specific configuration to make on your telephony devices.

Templates are handled on a per plugin basis. It's not possible for a template to be shared by more than one plugin since it's a design limitation of the plugin system of `provd`.

Note: When you install a new plugin, templates are not migrated automatically, so you must manually copy them from the old plugin directory to the new one. This does not apply for a plugin upgrade.

Let's suppose we have installed the `xivo-aastra-3.3.1-SP2` plugin and want to write some custom templates for it.

First thing to do is to go into the directory where the plugin is installed:

```
cd /var/lib/xivo-provd/plugins/xivo-aastra-3.3.1-SP2
```

Once you are there, you can see there's quite a few files and directories:

```
tree
.
+-- common.py
+-- entry.py
+-- pkgs
|   +-- pkgs.db
+-- plugin-info
+-- README
+-- templates
|   +-- 6730i.tpl
|   +-- 6731i.tpl
|   +-- 6739i.tpl
|   +-- 6753i.tpl
|   +-- 6755i.tpl
|   +-- 6757i.tpl
|   +-- 9143i.tpl
|   +-- 9480i.tpl
|   +-- base.tpl
+-- var
    +-- cache
    +-- installed
    +-- templates
    +-- tftpboot
        +-- Aastra
            +-- aastra.cfg
```

The interesting directories are:

templates This is where the original templates lies. You *should not* edit these files directly but instead copy the one you want to modify in the `var/templates` directory.

var/templates This is the directory where you put and edit your custom templates.

var/tftpboot This is where the configuration files lies once they have been generated from the templates. You should look at them to confirm that your custom templates are giving you the result you are expecting.

Warning: When you uninstall a plugin, the plugin directory is removed altogether, including all the custom templates.

A few things to know before writing your first custom template:

- templates use the [Jinja2 template engine](#).
- when doing an `include` or an `extend` from a template, the file is first looked up in the `var/templates` directory and then in the `templates` directory.
- device in autoprov mode are affected by templates, because from the point of view of `provd`, there's no difference between a device in autoprov mode or fully configured. This means there's usually no need to modify static files in `var/tftpboot`. And this is a bad idea since a plugin upgrade will override these files.

Custom template for every devices

```
cp templates/base.tpl var/templates
vi var/templates/base.tpl
xivo-provd-cli -c 'devices.using_plugin("xivo-aastra-3.3.1-SP2").reconfigure()'
```

Once this is done, if you want to synchronize all the affected devices, use the following command:

```
xivo-provd-cli -c 'devices.using_plugin("xivo-aastra-3.3.1-SP2").synchronize()'
```

Custom template for a specific model Let's suppose we want to customize the template for our 6739i:

```
cp templates/6739i.tpl var/templates
vi var/templates/6739i.tpl
xivo-provd-cli -c 'devices.using_plugin("xivo-aastra-3.3.1-SP2").reconfigure()'
```

Custom template for a specific device To create a custom template for a specific device you have to create a device-specific template named `<device_specific_file_with_extension>.tpl` in the `var/templates/` directory :

- for an Aastra phone, if you want to customize the file `00085D2EECFB.cfg` you will have to create a template file named `00085D2EECFB.cfg.tpl`,
- for a Snom phone, if you want to customize the file `000413470411.xml` you will have to create a template file named `000413470411.xml.tpl`,
- for a Polycom phone, if you want to customize the file `0004f2211c8b-user.cfg` you will have to create a template file named `0004f2211c8b-user.cfg.tpl`,
- and so on.

Here, we want to customize the content of a device-specific file named `00085D2EECFB.cfg`, we need to create a template named `00085D2EECFB.cfg.tpl`:

```
cp templates/6739i.tpl var/templates/00085D2EECFB.cfg.tpl
vi var/templates/00085D2EECFB.cfg.tpl
xivo-provd-cli -c 'devices.using_mac("00085D2EECFB").reconfigure()'
```

Note: The choice to use this syntax comes from the fact that `provd` supports devices that do not have MAC addresses, namely softphones.

Also, some devices have more than one file (like Snom), so this way make it possible to customize more than 1 file.

The template to use as the base for a device specific template will vary depending on the need. Typically, the model template will be a good choice, but it might not always be the case.

Changing the Plugin Used by a Device From time to time, new firmwares are released by the devices manufacturer. This sometimes translate to a new plugin being available for these devices.

When this happens, it almost always means the new plugin obsoletes the older one. The older plugin is then considered “end-of-life”, and won’t receive any new updates nor be available for new installation.

Let’s suppose we have the old `xivo-aastra-3.2.2.1136` plugin installed on our xivo and want to use the newer `xivo-aastra-3.3.1-SP2` plugin.

Both these plugins can be installed at the same time, and you can manually change the plugin used by a phone by editing it via the *Services* → *IPBX* → *Devices* page.

If you are using custom templates in your old plugin, you should copy them to the new plugin and make sure that they are still compatible.

Once you take the decision to migrate all your phones to the new plugin, you can use the following command:

```
xivo-provd-cli -c 'helpers.mass_update_devices_plugin("xivo-aastra-3.2.2.1136", "xivo-aastra-3.3.1")'
```

Or, if you also want to synchronize (i.e. reboot) them at the same time:

```
xivo-provd-cli -c 'helpers.mass_update_devices_plugin("xivo-aastra-3.2.2.1136", "xivo-aastra-3.3.1", true)'
```

You can check that all went well by looking at the *Services* → *IPBX* → *Devices* page.

NAT The provisioning server has partial support for environment where the telephony devices are behind a NAT equipment.

By default, each time the provisioning server receives an HTTP/TFTP request from a device, it makes sure that only one device has the source IP address of the request. This is not a desirable behaviour when the provisioning server is used in a NAT environment, since in this case, it's normal that more than 1 devices have the same source IP address (from the point of view of the server).

If *all* your devices used on your XiVO are behind a NAT, you should disable this behaviour by setting the NAT option to 1 via the *Configuration* → *Provisioning* → *General* page.

Enabling the NAT option will also improve the performance of the provisioning server in this scenario.

If you have many devices behind a NAT equipment, you should also check the [security](#) section to make sure the IP address of your NAT equipment doesn't get banned unintentionally.

Limitations

- You must only have phones of the following brands:
 - Aastra
 - Cisco SPA
 - Yealink
- All your devices must be behind a NAT equipment (the devices may be grouped behind different NAT equipments, not necessarily the same one)
- You must provision the devices via the Web interface, i.e. associate the devices from the user form. Using the 6-digit provisioning code on the phone will produce unexpected results (i.e. the wrong device will be provisioned)

For technical information about why other devices are not supported, you can look at [this issue](#) on the XiVO bug tracker.

Security By design, the auto-provisioning process is vulnerable to:

- Leakage of sensitive information: some files that are served by the provisioning server contains sensitive information, e.g. SIP credentials that are used by SIP phones to make calls. Depending on your network configuration and the amount of information an attacker has on your telephony ecosystem (phone vendor, MAC address, etc.), he could retrieve the content of some files containing sensitive information.
- Denial-of-service attack: in its default configuration, each time the provisioning server identify a request coming from a new device, it creates a new device object in its database. An attacker could spoof requests to the provisioning server to create a huge amount of devices, creating a denial-of-service condition.

That said, starting from XiVO 16.08, XiVO adds [Fail2ban](#) support to the provisioning server to drastically lower the likelihood of such attacks. Every time a request for a file potentially containing sensitive information is requested, a log line is appended to the `/var/log/xivo-provd-fail2ban.log` file, which is monitored by fail2ban. The same thing happens when a new device is automatically created by the provisioning server.

The fail2ban configuration for the provisioning server is located at `/etc/fail2ban/jail.d/xivo.conf`. You may want to adjust the `findtime` / `maxretry` value if you have special requirements. In particular, if you have many phones behind a NAT equipment, you'll probably have to adjust these values, since every request

coming from your phones behind your NAT will appear to the provisioning server as coming from the same source IP address, and this IP address will then be more likely to get banned promptly if you, for example, reboot all your phones at the same time. Another solution would be to add your IP address to the list of ignored IP address of fail2ban. See the fail2ban(1) man page for more information.

System Requirements XiVO 16.08 or later is required. You also need to use compatible xivo-provd plugins. Here's the list of official plugins which are compatible:

Plugin family	Version
xivo-aastra	>= 1.6
xivo-cisco-sccp	>= 1.1
xivo-cisco-spa	>= 1.0
xivo-digium	>= 1.0
xivo-polycom	>= 1.7
xivo-snom	>= 1.6
xivo-yealink	>= 1.26

Remote directory If you have a phone provisioned with XiVO and its one of the supported ones, you'll be able to search in your XiVO directory and place call directly from your phone.

See the list of *supported devices* to know if a model supports the XiVO directory or not.

Configuration For the remote directory to work on your phones, the first thing to do is to go to the *Services* → *IPBX* → (*General settings*) *Phonebook* page.

You then have to add the range of IP addresses that will be allowed to access the directory. So if you know that your phone's IP addresses are all in the 192.168.1.0/24 subnet, just click on the small "+" icon and enter "192.168.1.0/24", then save.

Once this is done, on your phone, just click on the "remote directory" function key and you'll be able to do a search in the XiVO directory from it.

Jitsi Jitsi (<http://jitsi.org/>) is an opensource softphone (previously SIP Communicator).

XiVO now support Jitsi sofphones provisioning. Here are the steps to follow :

Requirements This how to needs :

1. Jitsi installed,
2. SIP line created

Add Jitsi plugin on XiVO Open XiVO Web interface, and go to Configuration tab, Then chose *Provisioning* → *Plugins menu*, Install the Jitsi plugin you want to use : e.g.:

```
xivo-jitsi-1
```

You can now launch your Jitsi softphone

Configuring Jitsi

1. Launch Jitsi,
2. If you don't have any accounts configured Jitsi will launch a windows and you can click
3. Use online provisioning. Otherwise go to Tools -> Options -> Advanced -> Provisioing, Click on Enable provisioning
4. Select Manually specify a provisioning URI,

5. Enter the following URI where <provd_ip> is the VoIP interface IP address of your XiVO and <provd_port> is the provd port (default : 8667)

```
http://<provd_ip>:<provd_port>/jitsi?uuid=${uuid}
```

6. When done, quit Jitsi,
7. Launch Jitsi again,
 - You should now be connected with in autoprov mode,
 - You could see a new device in the devices list,
8. You can now provision the phones by typing the provisioning code (you get it in the Lines list),
9. Quit Jitsi again (configuration syncing is not available with the Jitsi plugin)
10. And launch Jitsi again : you should now be connected with you phone account

SCCP Configuration

Provisioning To be able to provision SCCP phones you should :

- activate the *DHCP Server*,
- activate the *DHCP Integration*,

Then install a plugin for SCCP Phone: *Configuration → Provisioning → Plugins*

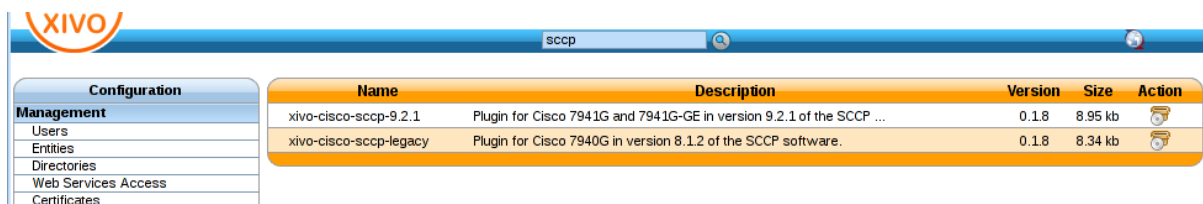


Fig. 1.69: Installing xivo cisco-sccp plugin

At this point you should have a fully functional DHCP server that provides IP address to your phones. Depending on what type of CISCO phone you have, you need to install the plugin sccp-legacy, sccp-9.4 or both.

Note: Please refer to the *Provisioning page* for more information on how to install CISCO firmwares.

Once your plugin is installed, you'll be able to edit which firmwares and locales you need. If you are unsure, you can choose all without any problem.

Now if you connect your first SCCP phone, you should be able to see it in the device list.

Listing the detected devices: *Services → IPBX → IPBX settings → Devices*

When connecting a second SCCP phone, the device will be automatically detected as well.

SCCP General Settings

Review SCCP general settings: *Services → IPBX → IPBX settings → SCCP general settings*

User creation The last step is to create a user with a **SCCP line**.

Creating a user with a SCCP line: *Services → IPBX → IPBX settings → Users*

Before saving the newly configured user, you need to select the *Lines* menu and add a SCCP line. Now, you can save your new user.

Congratulations ! Your SCCP phone is now ready to be called !

Configuration

Management

- Users
- Entities
- General
- Directories
- Web Services Access
- Certificates
- High Availability
- LDAP Servers

Network

- Interfaces
- Resolver
- Mail
- DHCP

Support

- XIVO
- Alerts

Provisioning

- General
- Template line
- Template device
- Plugins

Control System

- Apply network configuration
- Apply system configuration

Edit plugin xivo-cisco-sccp-legacy v. 0.3

Description

Plugin for Cisco 7912G, 7940G and 7960G in version 8.1.2 of the SCCP software.
Please see the documentation if you want to install Cisco firmwares.

Name	Description	Size	Version	Action
7912-fw	Firmware for Cisco 7912G	331.06 kb	8.0.4	
userlocale_es_ES	es_ES user locale	4.11 mb	9.0.2	
userlocale_de_DE	de_DE user locale	3.4 mb	9.0.2	
7940-7960-fw	Firmware for Cisco 7940G and 7960G	684.95 kb	8.1.2	
networklocale	Network locale	8.92 mb	9.0.2	
userlocale_fr_FR	fr_FR user locale	4.19 mb	9.0.2	

Fig. 1.70: Editing the xivo-cisco-sccp-legacy plugin

IPBX

General settings

- SIP Protocol
- IAX Protocol
- Voicemails
- Phonebook
- Advanced

IPBX settings

- Devices
- Lines
- Users
- Groups
- Voicemails

	MAC	Phone number	IP	Vendor	Modele	Plugin	Action
<input type="checkbox"/>	00:1a:a2:7a:bb:fc	-	10.97.5.103	Cisco	7912G	xivo-cisco-sccp-legacy	

Legend

- Existent on the provisioning server
- Inexistent on the provisioning server
- Device properly configured
- Device configured in autoprov mode
- Device not configured (check if a plugin is installed for this device)

Fig. 1.71: Device list

IPBX

General settings

- SIP Protocol
- IAX Protocol
- Voicemails
- Phonebook
- Advanced

IPBX settings

- Devices
- Lines
- Users
- Groups
- Voicemails
- Conference rooms

Call management

	MAC	Phone number	IP	Vendor	Modele	Plugin	Action
<input type="checkbox"/>	00:17:5a:4a:a3:6d	-	10.97.5.102	Cisco	7941G	xivo-cisco-sccp-legacy	
<input type="checkbox"/>	00:1a:a2:7a:bb:fc	-	10.97.5.103	Cisco	7912G	xivo-cisco-sccp-legacy	

Legend

- Existent on the provisioning server
- Inexistent on the provisioning server
- Device properly configured
- Device configured in autoprov mode
- Device not configured (check if a plugin is installed for this device)

Fig. 1.72: Device list

SCCP protocol properties

Enable direct media: ☒ ?

Dial timeout: ?

Default language: ?

Codecs

Customize codecs: ☒

Disabled codecs: ?

1 items selected	Remove all	Add all
G.729A (Audio)	-	G.711 u-law (Audio)
		G.711 A-law (Audio)

Fig. 1.73: SCCP general settings

Users

Search

Full name	Provisioning	Phone number	Nb Lines
No user found			

Fig. 1.74: Add a new user

General settings

- SIP Protocol
- IAX Protocol
- Voicemails
- Phonebook
- Advanced

IPBX settings

- Devices
- Lines
- Users**
- Groups
- Voicemails
- Conference rooms

First name:

Last name:

User picture:

Mobile phone number:

Create a schedules ☐

Ringing time:

Fig. 1.75: Edit user informations

Users > Add

Entity: ?

	Protocol	Name	Context	Number	Site	Device	Line (N°)
1	SCCP		Default	1001	local	00:17:5a:4a:a3:6d	1

Fig. 1.76: Add a line to a user

Function keys With SCCP phones, the only function keys that can be configured are:

- *Key*: Only the order is important, not the number
- *Type*: Customized; Any other type doesn't work
- *Destination*: Any valid extension
- *Label*: Any label
- *Supervision*: Enabled or Disabled

Direct Media

SCCP Phones support directmedia (direct RTP). In order for SCCP phones to use directmedia, one must enable the directmedia

Services → IPBX → IPBX settings → SCCP general settings

Features

Features	Supported
Receive call	Yes
Initiate call	Yes
Hangup call	Yes
Transfer call	Yes
Congestion Signal	Yes
Autoanswer (custom dialplan)	Yes
Call forward	Yes
Multi-instance per line	Yes
Message waiting indication	Yes
Music on hold	Yes
Context per line	Yes
Paging	Yes
Direct RTP	Yes
Redial	Yes
Speed dial	Yes
BLF (Supervision)	Yes
Resync device configuration	Yes
Do not disturb (DND)	Yes
Group listen	Yes
Caller ID	Yes
Connected line ID	Yes
Group pickup	Yes
Auto-provisioning	Not yet
Multi line	Not yet
Codec selection	Yes
NAT traversal	Not yet
Type of Service (TOS)	Manual

Telephone	Device type	Supported	Firmware version	Timezone aware
	7905	Yes	8.0.3	No
	7906	Yes	SCCP11.9-4-2SR1-1	Yes
	7911	Yes	SCCP11.9-4-2SR1-1	Yes
	7912	Yes	8.0.4(080108A)	No
	7920	Yes	3.0.2	No
	7921	Yes	1.4.5.3	Yes
	7931	Yes	SCCP31.9-4-2SR1-1	Yes
	7937	Testing		
	7940	Yes	8.1(SR.2)	No
	7941	Yes	SCCP41.9-4-2SR1-1	Yes
	7941GE	Yes	SCCP41.9-4-2SR1-1	Yes
	7942	Yes	SCCP42.9-4-2SR1-1	Yes
	7945	Testing		
	7960	Yes	8.1(SR.2)	No
	7961	Yes	SCCP41.9-4-2SR1-1	Yes
	7962	Yes	SCCP42.9-4-2SR1-1	Yes
	7965	Testing		
	7970	Testing		
	7975	Testing		
	CIPC	Yes	2.1.2	Yes

Models not listed in the table above won't be able to connect to Asterisk at all. Models listed as "Testing" are not yet officially supported in XiVO: use them at your own risk.

The "Timezone aware" column indicates if the device supports the timezone tag in its configuration file, i.e. in the file that the device request to the provisioning server when it boots. If you have devices that don't support the timezone tag and these devices are in a different timezone than the one of the XiVO, you can look at [the issue #5161](#) for a potential solution.

Schedules

Schedules are specific time frames that can be defined to open or close a service. Within schedules you may specify opening days and hours or close days and hours.

A default destination as user, group ... can be defined when the schedule is in closed state.

Schedules can be applied to :

- Users
- Groups
- Inbound calls
- Outbound calls
- Queues

Creating Schedules A schedule is composed of a name, a timezone, one or more opening hours or days that you may setup using a calendar widget, a destination to be used when the schedule state is closed.

With the calendar widget you may select months, days of month, days of week and opening time.

You may also optionally select closed hours and destination to be applied when period is inside the main schedule. For example, your main schedule is opened between 08h00 and 18h00, but you are closed between 12h00 and 14h00.

Using Schedule on Users When you have a schedule associated to a user, if this user is called during a closed period, the caller will first hear a prompt saying the call is being transferred before being actually redirected to the closed action of the schedule.

Schedules > Add

General

Closed hours

Name: workinghours

Timezone: America/New_York

Opened hours

Schedule

09h00 to 18h00, Mon to Fri, ...

Out of schedule / Default action

Destination : None

Description:

Save

Fig. 1.77: Creating a schedule

Schedule

09h00 to 18h00, Mon to Fri, ...

Months

Jan

Feb

Mar

Apr

May

Jun

Jul

Aug

Sep

Oct

Nov

Dec

None

Days of month

1

2

3

4

5

6

7

8

9

10

11

12

13

14

15

16

17

18

19

20

21

22

23

24

25

26

27

28

29

30

31

None

Days of week

Mon

Tue

Wed

Thu

Fri

Sat

Sun

All

Hours

18:00

09:00

Fig. 1.78: Schedule calendar widget

Closed hours

Schedule	Action
09h00 to 18h00, Mon to Fri, ...	Action : End call
Schedule	Choice: Hangup

Months

Jan Feb Mar Apr May Jun
Jul Aug Sep Oct Nov Dec
None

Days of month

1	2	3	4	5	6
7	8	9	10	11	12
13	14	15	16	17	18
19	20	21	22	23	24
25	26	27	28	29	30
31					None

Days of week

Mon	Tue	Wed	Thu	Fri	Sat
Sun					All

Hours

18:00
09:00

Fig. 1.79: Schedule closed hours

If you don't want this prompt to be played, you can change the behaviour by:

1. editing the `/etc/xivo/asterisk/xivo_globals.conf` file and setting the `XIVO_FWD_SCHEDULE_OUT_ISDA` to 1
2. reloading the asterisk dialplan with an asterisk `-rx "dialplan reload"`.

Sound Files

Add Sounds Files On a fresh install, only `en_US` and `fr_FR` sounds are installed. Canadian French and German are available too.

To install Canadian French sounds you have to execute the following command:

```
apt-get install asterisk-sounds-wav-fr-ca xivo-sounds-fr-ca
```

To install German sounds you have to execute the following command:

```
apt-get install asterisk-sounds-wav-de-de xivo-sounds-de-de
```

Now you may select the newly installed language for your users.

Convert Your Wav File Asterisk will read natively WAV files encoded in wav 8kHz, 16 bits, mono.

The following command will return the encoding format of the <file>

```
$ file <file>
RIFF (little-endian) data, WAVE audio, Microsoft PCM, 16 bit, mono 8000 Hz
```

The following command will re-encode the <input file> with the correct parameters for asterisk and write into the <output file>:


```
sox <input file> -b 16 -c 1 -t wav <output file> rate -I 8000
```

Switchboard

This page describes the configuration needed to have a switchboard on your XiVO.

Overview Switchboard functionality is available in the XiVO client. The goal of this page is to explain how to configure your switchboard and how to use it.

The switchboard xlet and profile allow an operator to view incoming calls, answer them, put calls on hold, view the calls on hold and pick up the calls on hold.

Limitations

Note: The shortcut keys of the switchboard do not work on the Mac version of the XiVO client.

Note: The enter shortcut to answer a call will not work if the focus is currently on a widget that will consume the key press. ie: a text field, a drop down

Note: Attended transfers to the switchboard cannot be managed with the switchboard xlets depending on the moment at which the call was completed.

Table of Contents

Switchboard Configuration Be sure to read the [limitations](#) before configuring a switchboard.

Server configuration

Quick Summary In order to configure a switchboard on your XiVO, you need to:

- Create a queue for your switchboard
- Create a queue for your switchboard's calls on hold
- Create the users that will be operators
- Activate the switchboard option for your phone
- Create an agent for your user
- Assign the incoming calls to the switchboard queue
- For each operator, add a function key for logging in or logging out from the switchboard queue.
- Set "no answer" destinations on the switchboard queue

Supported Devices The supported phones for the switchboard are:

Brand	Model	XiVO version	Plugin version
Aastra	6755i	>= 14.07	>= xivo-aastra-3.3.1-SP2, v1.0
Aastra	6757i	>= 14.07	>= xivo-aastra-3.3.1-SP2, v1.0
Aastra	6735i	>= 14.07	>= xivo-aastra-3.3.1-SP2, v1.2
Aastra	6737i	>= 14.07	>= xivo-aastra-3.3.1-SP2, v1.2
Polycom	VVX 400	>= 15.11	>= xivo-polycom-5.3.0, v1.3
Polycom	VVX 410	>= 15.11	>= xivo-polycom-5.3.0, v1.3
Snom	720	>= 14.14	>= xivo-snom-8.7.3.25.5, v1.0
Snom	D725	>= 14.14	>= xivo-snom-8.7.5.17, v1.4
Yealink	T46G	>= 15.01	>= xivo-yealink-72.0, v1.22.1

Create a Queue for Your Switchboard All calls to the switchboard will first be distributed to a switchboard queue.

To create this queue, go to *Services* → *Call center* → *Queues* and click the add button.

The following configuration is mandatory

- The *General* → *Name* field has to be `__switchboard`
- The *General* → *Ring strategy* field has to be *Ring all*
- The *General* → *Preprocess subroutine* field has to be `xivo_subr_switchboard`
- The *Application* → *Allow caller to hang up call* option has to be *enabled*
- The *Application* → *Allow callee to transfer the call* option has to be *enabled*
- The *Advanced* → *Member reachability timeout* option has to be *disabled*
- The *Advanced* → *Time before retrying a call to a member* option has to be *1 second*
- The *Advanced* → *Delay before reassigning a call* option has to be *disabled*
- The *Advanced* → *Call a member already on* option has to be *disabled*
- The *Advanced* → *Autopause agents* option has to be *No*

Other important fields

- The *General* → *Display name* field is the name displayed in the XiVO client xlets and in the statistics
- The *General* → *Number* field is the number that will be used to reach the switchboard internally (typically 9)

Create a Queue for Your Switchboard on Hold The switchboard uses a queue to track its calls on hold.

To create this queue, go to *Services* → *Call center* → *Queues* and click the add button.

The following configuration is mandatory

- The *General* → *Name* field has to be `__switchboard_hold`
- The *General* → *Number* field has to be a valid number in a context reachable by the switchboard

Other important fields

- The *General* → *Display name* field is the name displayed in the XiVO client xlets and in the statistics

Warning: This queue MUST have **NO** members

Create the Users that Will be Operators Each operator needs to have a user configured with a line. The XiVO client profile has to be set to *Switchboard*.

The following configuration is mandatory for switchboard users

- The *General* → *First name* field has to be set
- The *General* → *Enable XiVO Client* option has to be *enabled*
- The *General* → *Login* field has to be set
- The *General* → *Password* field has to be set
- The *General* → *Profile* field has to be set to *Switchboard*
- The *Lines* → *Number* field has to have a valid extension
- The *Lines* → *Device* field has to be a *supported device*
- The *Services* → *Enable call transfer* option has to be *enabled*
- The *Services* → *Enable supervision* option has to be *enabled*

Activate the Switchboard Option for your Phone The switchboard option must be activated on the phone. It's possible to activate this option only on *supported phones* and plugins.

- Edit device associated to your user in *Services* → *Devices*
- Check the switchboard checkbox and save
- Synchronize your phone to apply the changes

Polycom Phones To be able to use a Polycom phone for the switchboard, the XiVO must be able to do HTTP requests to the phone. This might be problematic if there's a NAT between your XiVO and your phone.

It's possible to configure the Polycom switchboard via the *configuration files* of xivo-ctid. The following options are available:

```
switchboard_polycom:
  username: xivo_switchboard
  password: xivo_switchboard
  answer_delay: 0.5
```

You will also need to change the XML API username/password by creating a *custom template* for your phone.

Snom Phones When using a Snom switchboard, you must not configure a function key on position 1.

To be able to use a Snom phone for the switchboard, the XiVO must be able to do HTTP requests to the phone. This might be problematic if there's a NAT between your XiVO and your phone. The following command should work from your XiVO's bash command line `wget http://guest:guest@<phone IP address>/command.htm?key=SPEAKER`. If this command does not activate the phone's speaker, your network configuration will have to be *fixed* before you can use the Snom switchboard.

It's possible to configure the Snom switchboard via the *configuration files* of xivo-ctid. The following options are available:

Users > Add

General

Lines

No answer

Services

Voicemail

Groups

Func Keys

First name: Bob

Last name:

User picture:

Parcourir...

 Aucun fichier sélectionné.

Mobile phone number:

Schedules:

Ringing time: 30 seconds

Simultaneous calls: 5

On-Hold Music: default

Language:

Timezone:

Caller ID: Bob

Outgoing Caller ID: Default

Preprocess subroutine:

User field :

XiVO Client

Enable XiVO Client: ☒

Login: switchboard

Password: password

Profile: Switchboard

Description:

Save

IP: 10.34.1.163

MAC: 00:08:5d:33:e5:76

Plugin: xivo-aastra-3.3.1-SP2

Device config template: Default config device

Switchboard: ☒

```
switchboard_snom:
  username: guest
  password: guest
  answer_delay: 0.5
```

You have to change the username and password option if you have changed the administrator username or administrator password for your phone in *Configuration* → *Provisioning* → *Template Device*.

Yealink Phones When using a Yealink switchboard, you must not configure a function key on position 1.

Create an Agent for the Operator Each operator needs to have an associated agent.

Warning: Each agent MUST ONLY be a member of the Switchboard queue

To create an agent:

- Go to *Services* → *Call center* → *Agents*
- Click on the group *default*
- Click on the *Add* button

Agents > Add an agent

General Users Queues Advanced

First name:

Last name:

Number:

Password:

Context:

Language:

Group:

- Associate the user to the agent in the *Users* tab

Agents > Add an agent

General Users Queues Advanced

1 items selected		Remove all	<input type="text"/>	Add all
↑	Bob	—	Abraham Maharba	+
			Alice Wonderland	+
			Charlie Chaplin	+
			Voice Mail	+

- Assign the Agent to the *Switchboard Queue* (**and ONLY to the Switchboard queue**)

Send Incoming Calls to the Switchboard Queue Incoming calls must be sent to the *Switchboard* queue to be distributed to the operators. To do this, we have to change the destination of our incoming call for the switchboard queue.

Name	Penalty
_switchboard	0

In this example, we associate our incoming call (DID 444) to our *Switchboard* queue:

Set “No Answer” Destinations on the *Switchboard* Queue When there are no operators available to answer a call, “No Answer” destinations should be used to redirect calls towards another destination.

You also need to set the timeout of the Switchboard queue to know when calls will be redirected.

The reachability timeout must not be disabled nor be too short.

The time before retrying a call to a member should be as low as possible (1 second).

In this example we redirect “No Answer”, “Busy” and “Congestion” calls to the *everyone* group and “Fail” calls to the *guardian* user.

Queues > Edit __switchboard (9@default)

General Announces Members Application No answer **Advanced** Schedules Diversions

Exit context: ?

Service level: ?

Member reachability timeout: 30 seconds ?

Time before retrying a call to a member: 1 second ?

Weight: 0 ?

Delay before reassigning a call: Disabled ?

Maximum number of people allowed to wait: 0 ?

You can also choose to redirect all the calls to another user or a voice mail.

Queues > Edit __switchboard (9@default)

General Announces Members Application **No answer** Advanced Schedules Diversions

No answer

Destination: Group

Redirect to: everyone (200@default)

Ring time:

Busy

Destination: Group

Redirect to: everyone (200@default)

Ring time:

Congestion

Destination: Group

Redirect to: everyone (200@default)

Ring time:

Fail

Destination: User

Redirect to: Guardian

Ring time:

Save

XIVO Client configuration

Directory xlet The transfer destination is chosen in the Directory xlet. You **must** follow the *Directory Xlet* section to be able to use it.

Configuration for multiple switchboards The above documentation can be used for multiple switchboards on the same XIVO by replacing the `__switchboard` and `__switchboard_hold` queues name and configuring the operators XIVO client accordingly in the *XIVO Client* → *Configure* → *Functions* → *Switchboard* window.

All switchboard queues should be added to the `xivo-ctid` configuration. New queues can be added by adding a file in `/etc/xivo-ctid/conf.d`. For example, the following content should be used for a new switchboard queue names `__switchboard_two` and an hold queue names `__switchboard_hold_two`.

The screenshot shows a configuration window with tabs: Connection, Account, GUI Settings, Functions (selected), and Advanced. Under Functions, there are checkboxes for 'Presence reporting' (checked) and 'Customer Info' (unchecked). Below these are sub-tabs: 'Presence reporting', 'Customer Info', 'Dialer', and 'Switchboard' (selected). The 'Switchboard' sub-tab contains two text input fields: 'Switchboard queue name' with the value '_switchboard' and 'Switchboard call on hold queue name' with the value '_switchboard_hold'. At the bottom right are 'OK' and 'Cancel' buttons.

```
{
  "switchboard_queues": {
    "__switchboard_two": true
  },
  "switchboard_hold_queues": {
    "__switchboard_hold_two": true
  }
}
```

Switchboard Usage

Warning: The *switchboard configuration* must be completed before using the switchboard. This includes:

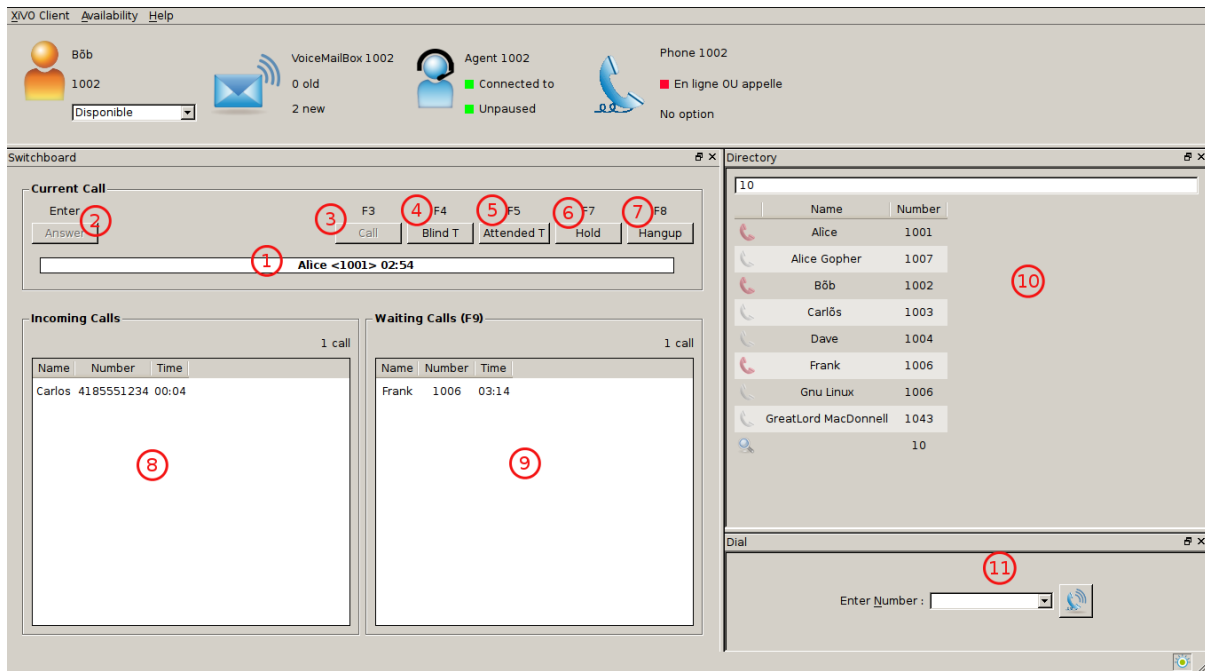
- Device, User, Agent and Queues configuration (see above),
- Directory xlet configuration (see *Directory Xlet*)

If it's not the case, the user must disconnect his XiVO client and reconnect.

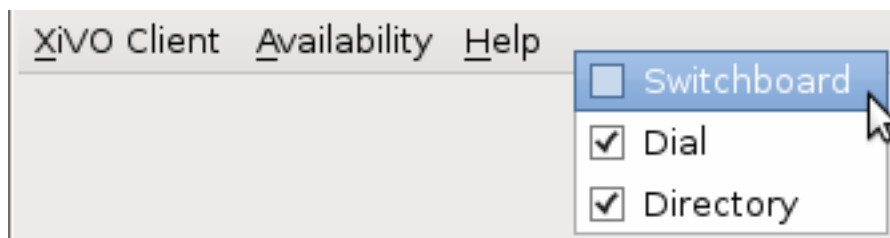
Be sure to read the *limitations* before using the switchboard.

The XiVO Client Switchboard Profile When the user connects with his XiVO Client, he gets the Switchboard profile.

1. *Current Call* frame
2. *Answer* button
3. *Call* button
4. *Blind transfer* button
5. *Attended transfer* button
6. *Hold* button
7. *Hangup* button
8. *Incoming Calls* list
9. *Waiting Calls* list
10. *Directory Xlet*
11. *Dial Xlet*



Note: If you don't see the Switchboard Xlet, right-click on the grey bar at the right of the *Help* menu and check *Switchboard*:



The operator can login his agent using a function key or an extension to start receiving calls.

Call flow

Answering an incoming call When the switchboard receives a call, the new call is added to the *Incoming Calls* list on the left and the phone starts ringing. The user can answer this call by:

- clicking on any call in the list
- clicking the *Answer* button
- pressing the *Enter* key

Note: The XIVO Client must be the active window for the keyboard shortcuts to be handled

The operator can select which call to answer by:

- clicking directly on the incoming call
- pressing *F6* to select the incoming calls frame and pressing the up and down arrow keys

Selecting a call to answer while talking will not answer the call.

Once the call has been answered, it is removed from the incoming calls list and displayed in the *Current Call* frame.

Making a Call The switchboard operator can do the following operations:

- Press the *Call* button or press *F3*
- Search for the call destination in the directory xlet
- Press to confirm the selection and start the call

Hanging Up a Call The switchboard operator can hang up its current call by either:

- Clicking the *Hangup* button
- Pressing the *F8* key

If the operator has placed a new call via the *Directory* or *Dial* xlet and that call has not yet been answered, he can cancel it in the same way.

Distributing a call Once the call has been answered and placed in the current call frame, the operator has 3 choices:

- transfer the call to another user
 - using the *Blind transfer* button or the *F4* key.
 - using the *Attended transfer* button or the *F5* key
- put the call on hold using the *Hold* button or the *F7* key
- end the call using the *Hangup* button or the *F8* key.

Transferring a call Transfer buttons allow the operator to select towards which destination he wishes to transfer the call. This is made through the *Directory* xlet. For details about the xlet *Directory* usage and configuration see [Directory Xlet](#).

Once the destination name has been entered, press *Enter*. If multiple destinations are displayed, you can choose by:

- double-clicking on the destination
- using *Up/Down* arrows then:
 - pressing *Enter*
 - pressing the transfer button again

Blind transfers are straightforward: once the call is transferred, the operator is free to manage other calls.

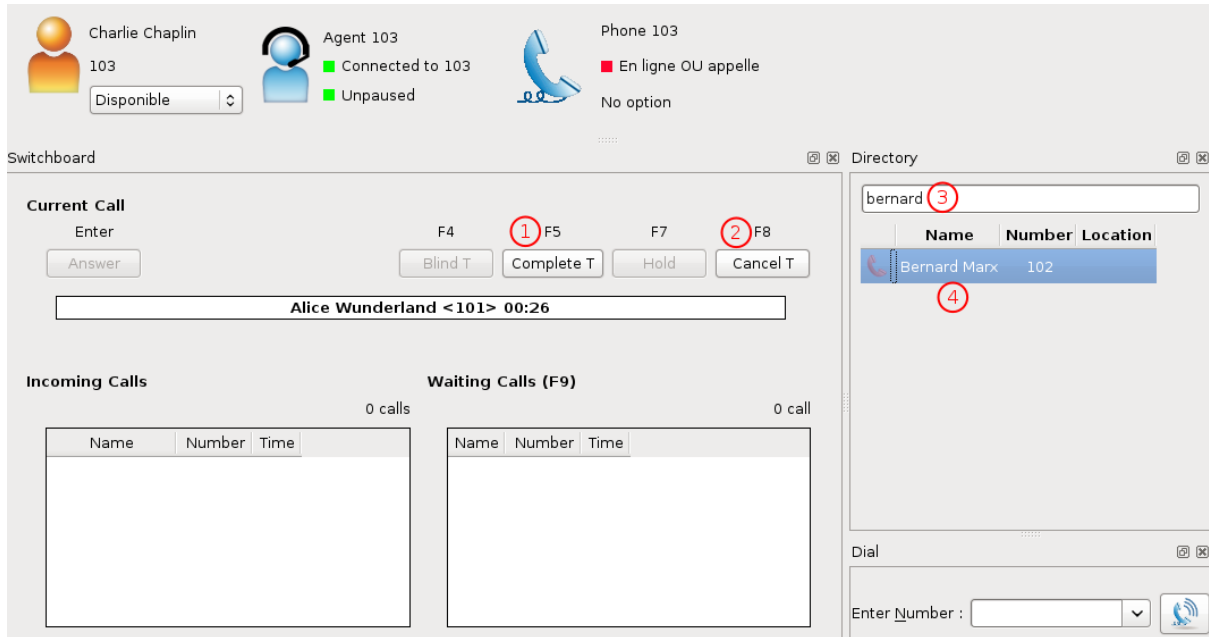
Attended transfers are a bit more complicated: the operator needs to wait for the transfer destination to answer before completing the transfer.

In this example, the operator is currently asking *Bernard Marx* if he can transfer *Alice Wonderland* to him.

1. *Complete transfer* button
2. *Cancel transfer* button
3. Transfer destination filtering field (xlet *Directory*)
4. Transfer destination list (xlet *Directory*)

Once the destination has answered, you can:

- cancel the transfer with *F8* key
- complete the transfer with *F5* key



Note: The operator can not complete an attended transfer while the transfer destination is ringing. In this case, the operator must cancel the attended transfer and use the *Blind transfer* action.

Putting a call on hold If the user places the call on hold, it will be removed from the *Current call* frame and displayed in the *Waiting calls* list. The time counter shows how long the call has been waiting, thus it will be reset each time the call returns in the *Waiting calls* list. The calls are ordered from the oldest to the newest.

Retrieving a call on hold Once a call has been placed on hold, the operator will most certainly want to retrieve that call later to distribute it to another destination.

To retrieve a call on hold:

- click the desired call in the *Waiting calls* list
- with the keyboard:
 - move the focus to the *Waiting calls* list (F9 key)
 - choose the desired call with the arrow keys
 - press the *Enter* key.

Once a call has been retrieved from the *Waiting calls* list, it is moved back into the *Current Call* frame, ready to be distributed.

Switchboard Statistics

Limitations

Note: Statistics are produced by *xivo-ctid*. If a call is received when *xivo-ctid* is stopped, no statistics will be produced for that call.

Note: Statistics are only generated for calls answered in XiVO 16.03.

Note: Statistics are only available for existing switchboard queues, i.e. deleting a queue will also delete the associated statistics.

Daily Statistics Switchboard statistics can be retrieved in CSV format via the web interface in *Services* → *Statistics* → *Switchboard* → *Statistics*.

- Start date: when empty, the result will contain statistics from the beginning
- End date: when empty, the result will contain statistics until the current time

Note: Switchboard statistics older than a year are automatically removed. See [Purge Logs](#) for more details.

Report The generated CSV report includes the following columns:

- date: The date at which the calls were received
- entered: The number of calls to the switchboard for the given date excluding calls when the switchboard was closed (e.g. with a [schedule](#))
- answered: The number of calls that have been answered by the operator and then transferred or completed by the operator
- transferred: The number of calls that have been transferred by the switchboard operator to another destination
- abandoned: The number of calls that have been abandoned in the switchboard queue or while waiting in the hold queue
- forwarded: The number of calls that have been forwarded to another destination:
 - a call reaching a full queue
 - a call waiting until the max ring time is reached
 - a call forwarded because of a diversion rule
 - a call forwarded because of a leave empty condition
- waiting_time_average: The average time spent in the switchboard and hold queue for all calls that entered the switchboard

Events Switchboard statistic events are published on the bus to be consumed by collectd.

Configuration In order to process these events, you need:

- collectd installed on your XiVO: `apt-get install collectd`
- In `/etc/collectd/collectd.conf.d/amqp.conf`, configure collectd to read events from the bus (RabbitMQ):

```
LoadPlugin amqp

<Plugin "amqp">
  <Subscribe "xivo">
    Host "127.0.0.1"
```

```

Port "5672"
VHost "/"
User "guest"
Password "guest"
Exchange "collectd"
ExchangeType "topic"
RoutingKey "collectd.#"
</Subscribe>
</Plugin>

```

- another service receiving events from collectd, e.g. logstash, graphite, another collectd.

Event description The collectd events have the following attributes:

- **host**: the UUID of the XiVO.
- **plugin**: switchboard
- **plugin_instance**: the name (not the display name) of the queue for incoming calls of the switchboard.
- **type**: counter or gauge.
- **type_instance**: the following values.

entered This event is produced when a call enters the switchboard on an open schedule. Calls that did not enter the queue, if the queue was full for example, will also generate an entered event.

abandoned This event is produced when the called hangs up while waiting in the incoming queue or in the hold queue.

transferred This event is produced when a call is transferred from the switchboard by the operator. For attended transfers, the event is sent when the transfer is completed.

forwarded This event is produced when a call is redirected to another destination under certain conditions. This include:

- When the queue is full
- When the queue timeout is reached
- When no agent are logged with a join empty configuration
- When a diversion occurred

completed This event is produced when a call was answered by the operator without being transferred to another destination.

wait_time This event is produced when a call is completed, its value is the sum of all times spent in the hold queue and the time spent in the incoming queue before being answered.

Users

Users Configuration.

User Import and Export

CSV Import Users can be imported and associated to other resources by use of a CSV file. CSV Importation can be used in situations where you need to modify many users at the same in an efficient manner, or for migrating users from one system to another. A CSV file can be created and edited by spreadsheet tools such as Excel, LibreOffice/OpenOffice Calc, etc.

CSV file The first line of a CSV file contains a list of field names (also sometimes called “columns”). Each new line afterwards are users to import. CSV data must respect the following conditions:

- Files must be encoded in UTF-8
- Fields must be separated with a ,
- Fields can be optionally quoted with a "
- Double-quotes can be escaped by writing them twice (e.g. Robert "Bob" Jenkins)
- Empty fields or headers that are not defined will be considered null.
- Fields of type *bool* must be either 0 for false, or 1 for true.
- Fields of type *int* must be a positive number

In the following tables, columns have been grouped according to their resource. Each resource is created and associated to its user when all required fields for that resource are present.

	Field	Type	Re-quired	Values	Description
User	entity_id	int	Yes		Entity ID (Defined in menu <i>Configuration</i> → <i>Management</i> → <i>Entities</i>)
	firstname	string	Yes		User's firstname
	lastname	string			User's lastname
	email	string			User's email
	language	string		de_DE, en_US, es_ES, fr_FR, fr_CA	User's language
	mobile_phone_number	string			Mobile phone number
	outgoing_caller_id	string			Customize outgoing caller id for this user
	enabled	bool			Enable/Disable the user
	supervision_enabled	bool			Enable/Disable supervision
	call_transfer_enabled	bool			Enable/Disable call transfers by DTMF
	dtmf_hangup_enabled	bool			Enable/Disable hangup by DTMF
	simultaneous_calls	int			Number of calls a user can have on his phone simultaneously
	ring_seconds	int		Must be a multiple of 5	Number of seconds a call will ring before ending
	call_permission_password	string			Overwrite all passwords set in call permissions associated to the user

	Field	Type	Required	Values	Description
CTI Profile	cti_profile_enabled	bool	No		Activates the XiVO Client account for this user
	username	string	Yes, if profile enabled		XiVO Client username
	password	string	Yes, if profile enabled		XiVO Client password
	cti_profile_name	string	Yes, if profile enabled		XiVO Client profile (Defined in menu <i>Services</i> → <i>CTI server</i> → <i>Profiles</i>)

	Field	Type	Re-quired	Val-ues	Description
Phone	exten	string	Yes		Number for calling the user. The number must be inside the range of acceptable numbers defined for the context
	context	string	Yes		Context
	line_protocol	string	Yes	sip, sccp	Line protocol
	sip_username	string			SIP username
	sip_secret	string			SIP secret

	Field	Type	Re-quired	Val-ues	Description
Incoming call	in-call_exten	string	Yes		Number for calling the user from an incoming call (i.e outside of XiVO). The number must be inside the range of acceptable numbers defined for the context.
	in-call_context	string	Yes		context used for calls coming from outside of XiVO
	in-call_ring_seconds	int			Number of seconds a call will ring before ending

	Field	Type	Re-quired	Values	Description
Voicemail	voicemail_name	string	Yes		Voicemail name
	voicemail_number	string	Yes		Voicemail number
	voicemail_context	string	Yes		Voicemail context
	voicemail_password	string		A sequence of digits or #	Voicemail password
	voicemail_email	string			Email for sending notifications of new messages
	voice-mail_attach_audio	bool			Enable/Disable attaching audio files to email message
	voice-mail_delete_messages	bool			Enable/Disable deleting message after notification is sent
	voice-mail_ask_password	bool			Enable/Disable password checking

	Field	Type	Re-quired	Values	Description
Call permissions	call_permissions	string		list separated by semicolons (;)	Names of the call permissions to assign to the user

Importing a file Once your file is ready, you can import it via *Services* → *IPBX* → *IPBX settings* → *Users*. At the top of the page there is a plus button. A submenu will appear when the mouse is on top. Click on Import a file.



Fig. 1.80: Import Users

Examples The following example defines 3 users who each have a phone number. The first 2 users have a SIP line, where as the last one uses SCCP:

```
entity_id,firstname,lastname,exten,context,line_protocol
1,John,Doe,1000,default,sip
1,George,Clinton,1001,default,sip
1,Bill,Bush,1002,default,sccp
```

The following example imports a user with a phone number and a voicemail:

```
entity_id,firstname,lastname,exten,context,line_protocol,voicemail_name,voicemail_number,voicemail
1,John,Doe,1000,default,sip,Voicemail for John Doe,1000,default
```

The following exmple imports a user with both an internal and external phone number (e.g. incoming call):

```
entity_id,firstname,lastname,exten,context,line_protocol,incall_exten,incall_context
1,John,Doe,1000,default,sip,2050,from-extern
```

CSV Update The field list for an update is the same as for an import with the addition of the column uid, which is mandatory. For each line in the CSV file, the updater goes through the following steps:

1. Find the user, using the uid
2. For each resource (line, voicemail, extension, etc) find out if it already exists.
3. If an existing resource was found, associate it with the user. Otherwise, create it.
4. Update all remaining fields

The following restrictions must also be respected during update:

- Columns that are not included in the CSV header will not be updated.
- A field that is empty (i.e, "") will be converted to NULL, which will unset the value.
- A line's protocol cannot be changed (i.e you cannot go from "sip" to "sccp" or vice-versa).
- An incall cannot be updated if the user has more than one incall associated.

Updating is done through the same menu as importing (*Services* → *IPBX* → *IPBX settings* → *Users*). A submenu will appear when the mouse is on top. Click on *Update from file* in the submenu.

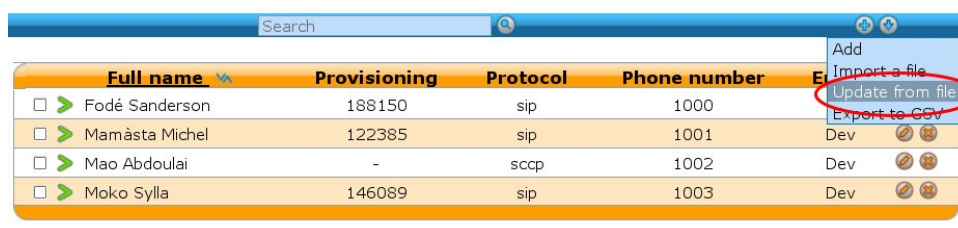


Fig. 1.81: *Services* → *IPBX* → *IPBX settings* → *Users* → *Update from file*

CSV Export CSV exports can be used as a scaffold for updating users, or as a means of importing users into another system. An export will generate a CSV file with the same list of columns as an import, with the addition of uid and provisioning_code.

Exports are done through the same menu as importing (*Services* → *IPBX* → *IPBX settings* → *Users*). Click on *Export to CSV* in the submenu. You will be asked to download a file.

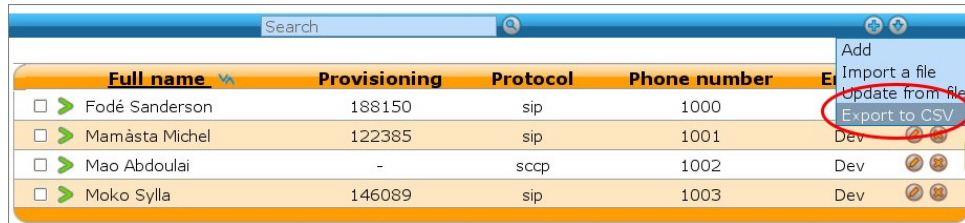


Fig. 1.82: Services → IPBX → IPBX settings → Users → Export to CSV

Function keys Function keys can be configured to customize the user's phone keys. Key types are pre-defined and can be browsed through the Type drop-down list. The Supervision field allows the key to be supervised. A supervised key will light up when enabled. In most cases, a user cannot add multiple times exactly the same function key (example : two user function keys pointing to the same user). Adding the same function key multiple times can lead to undefined behavior and generally will delete one of the two function keys.

Warning: SCCP device only supports type “Customized”.

Key	Type	Destination	Label	Supervision
1	Do not disturb			Enabled
2	Incoming call filtering			Enabled
3	Enable / Disable forwarding unconditional	102		Enabled
4	Enable / Disable forwarding on busy	102		Enabled
5	Enable / Disable forwarding on no answer	102		Enabled
6	Enable / Disable forwarding unconditional	103		Enabled

For User keys, start to key in the user name in destination, XiVO will try to complete with the corresponding user. If the forward unconditional function key is used with no destination the user will be prompted when the user presses the function key and the BLF will monitor *ALL* unconditional forward for this user.

Extensions

***3 (online call recording)** To enable online call recording, you must check the “Enable online call recording” box in the user form.

Services

Enable supervision: ☒

Enable call transfer: ☒

Enable online call recording: ☒

Call recording: ☐

Incoming call filtering: ☐

Do not disturb: ☐

Filter Boss - Secretary: No

Agent:

Fig. 1.83: Users Services

When this option is activated, the user can press *3 during a conversation to start/stop online call recording. The recorded file will be available in the `monitor` directory of the *Services* → *IPBX* → *Audio files* menu.

***26 (call recording)** You can enable/disable the recording of all calls for a user in 2 different way:

1. By checking the “Call recording” box of the user form.

Fig. 1.84: Users Services

2. By using the extension *26 from your phone (the “call recording” option must be activated in *Services* → *IPBX* → *Extensions*).

When this option is activated, all calls made to or made by the user will be recorded in the `monitor` directory of the *Services* → *IPBX* → *Audio files* menu.

Voicemail

Voicemail Configuration.

General Configuration The global voicemail configuration is located under *Services* → *IPBX* → *General Settings* → *Voicemails*.

Adding voicemails There are 2 ways to add a voicemail:

- Using *Services* → *IPBX* → *IPBX settings* → *Voicemails*
- Using the user’s configuration

Using *Services* → *IPBX* → *IPBX settings* → *Voicemails* New voicemails can be added using the + button.

Fig. 1.85: Add voicemails from voicemail menu

Once your voicemail is configured, you have to edit the user configuration and search the voicemail previously created and then associate it to your user.

Fig. 1.86: Search for a voicemail in the user's configuration

Using the user's configuration The other way is to add the voicemail from user's configuration in the 'voicemail' tab by

1. Clicking the + button
2. Filling the voicemail form
3. Saving

Fig. 1.87: Add a voicemail from the user's configuration

Note: The user's language *must* be set in the *general* tab

Disabling a voicemail You can disable a user's voicemail by un-checking the 'Enable voicemail' option on the Voicemail tab from user's configuration.

Deleting a voicemail Delete voicemail is done on *Services* → *IBX* → *IPBX settings* → *Voicemails* or from the user's *voicemail* tab.

Note:

- Deleting a voicemail is irreversible. It deletes all messages associated with that voicemail.

Fig. 1.88: Deactivate user's voicemail

- If the voicemail contains messages, the message waiting indication on the phone will not be deactivated until the next phone reboot.

Disable password checking Unchecking the option `Ask password` allows you to skip password checking for the voicemail only when it is consulted from an internal context.

- when calling the voicemail with *98
- when calling the voicemail with *99<voicemail number>

Warning: If the the *99 extension is enabled and a user does not have a password on its voicemail, anyone from the same context will be able to listen to its messages, change its password and greeting messages.

However, the password will be asked when the voicemail is consulted through an incoming call. For instance, let's consider the following incoming call:

With such a configuration, when calling this incoming call from the outside, we will be asked for:

- the voicemail number we want to consult
- the voicemail password, **even if the “Disable password checking option” is activated**

And then, we will be granted access to the voicemail.

Take note that the second “context” field contains the context of the voicemail. Voicemails of other contexts will not be accessible through this incoming call.

Warning: For security reasons, such an incoming call should be avoided if a voicemail in the given context has no password.

Advanced configuration

Remote *xivo-confd* If *xivo-confd* is on a remote host, *xivo-confd-client* configuration will be required to be able to change the voicemail passwords using a phone.

This configuration should be done:

```
mkdir -p /etc/systemd/system/asterisk.service.d
cat >/etc/systemd/system/asterisk.service.d/remote-confd-voicemail.conf <<EOF
[Service]
Environment=CONFD_HOST=localhost
Environment=CONFD_PORT=9486
Environment=CONFD_HTTPS=true
Environment=CONFD_USERNAME=<username>
Environment=CONFD_PASSWORD=<password>
EOF
systemctl daemon-reload
```

WebRTC

General notes

Note: added in version 2016.04

From version 2016.04 one can use WebRTC with *XiVO PBX* and *XiVO CC* in the following environment:

- LAN network (currently no support for WAN environment),
- with the:
 - *Web Assistant* with Chrome browser version 55 (tested on 55.0.2883.87 m 64-bit),
 - or *Desktop Assistant*

See [WebRTC Requirements](#).

Note: Current WebRTC implementation requires to create user with one line configured for WebRTC. To have user with both SIP and WebRTC line is not supported.

Configuration of user with WebRTC line

1. Create user
2. Add line to user without any device
3. Edit the line created and, in the *Advanced* tab, add *webrtc=yes* options:

Manual configuration of user with WebRTC line For the records

Lines > Edit

General Advanced IPBX Infos

Option	Value	
host	dynamic	
type	friend	
call-limit	10	
subscribermwi	no	
amaflags	default	
regseconds	0	
webrtc	yes	

Save

WebRTC manual configuration

Note: This is the manual way to configure a WebRTC line. It is here *for the record*. You should follow the *Configuration of user with WebRTC line* instead.

1. Create user
2. Optional: set codec to ulaw

Lines > Edit

General Advanced IPBX Infos

Username: 4raun97x
Password: 625gzwy
Context: Default (default)
Language:
Caller ID: "WebRTC WebRTC" <1099>
NAT:
DTMF:
Monitoring:

Codecs

Customize codecs: ☒
Codecs disallow: All

1 items selected Remove all

G.711 u-law (Audio)

Add all

G.723.1 (Audio)
GSM (Audio)
G.711 A-law (Audio)
ADPCM (Audio)
16 bit Signed Linear PCM (Audio)
LPC10 (Audio)
G.729A (Audio)

Save

3. Add line to user without any device
4. Configure Advanced Line options, so that it is usable with the softphone WebRTC

```
avpf = yes
call-limit = 1
dtlsenable = yes ; Tell Asterisk to enable DTLS for this peer
dtlsverify = no ; Tell Asterisk to not verify your DTLS certs
```

```
dtls-certfile=/etc/asterisk/keys/asterisk.pem ; Tell Asterisk where your DTLS cert file is
dtls-privatekey = /etc/asterisk/keys/asterisk.pem ; Tell Asterisk where your DTLS private key is
dtls-setup = actpass ; Tell Asterisk to use actpass SDP parameter when setting up DTLS
encryption = yes
force_avp = yes
icesupport = yes
transport = ws
```

Lines > Edit

General
Advanced
IPBX Infos

Option	Value	⛶
host	dynamic	⛶
type	friend	⛶
call-limit	1	⛶
subscribe-mwi	no	⛶
amaflags	default	⛶
transport	ws	⛶
regseconds	0	⛶
encryption	yes	⛶
avpf	yes	⛶
dtlsenable	yes	⛶
dtlsverify	no	⛶
dtls-certfile	/etc/asterisk/keys/asterisk.pem	⛶
dtls-privatekey	/etc/asterisk/keys/asterisk.pem	⛶
dtls-setup	actpass	⛶
force_avp	yes	⛶
icesupport	yes	⛶

Web Services Access

You may configure Web Services / REST API permissions in *Configuration* → *Management* → *Web Services Access*.

Web services access may have two different meanings:

- Who may access REST APIs of various XiVO daemons, and which resources in those REST APIs?
- Who may access PHP web services under `https://xivo.example.com/xivo/configuration/json.php/*?`

REST API access and permissions Those REST API interfaces are documented on <http://api.xivo.io>. They all require an authorization token, obtained by giving valid credentials to the REST API of xivo-auth. The relevant settings are:

- Login/Password: the xivo-auth credentials (for the xivo-auth *backend* xivo_service)
- ACL: The list of authorized REST API resources. See [REST API Permissions](#).

Unlike PHP web services, there is no host-based authorization, so the `Host` setting is not relevant.

A few REST API access are automatically generated during the installation of XiVO, so that XiVO services may authenticate each other.

You will probably only need to create such a REST API access when you want another non-XiVO service to communicate with XiVO via REST API.

PHP web services

Warning: DEPRECATED

Those web services are deprecated. There is no documentation about their usage, and the goal is to remove them. They are still protected with HTTP authentication, requiring a login and password. The relevant settings are:

- Login/Password: the HTTP authentication credentials
- Host: the authorized hosts that are allowed to make HTTP requests:
 - Empty value: HTTP authentication
 - Non-empty value: no HTTP authentication, all requests coming from this host will be accepted. Valid hosts may be: a hostname, an IP address, a CIDR block.

There is no fine-grained permissions: either the user has access to every PHP web services, or none.

xivo-confd **Warning: DEPRECATED**

There is also a special case for authentication with xivo-confd. See [REST API](#) for more details.

Contact Center

In XiVO, the contact center is implemented to fulfill the following objectives :

- Call routing
Includes basic call distribution using call queues and skills-based routing
- Agent and Supervisor workstation.
Provides the ability to execute contact center actions such as: agent login, agent logout and to receive real time statistics regarding contact center status
- Statistics reporting
Provides contact center management reporting on contact center activities
- Advanced functionalities
Call recording
- Screen Pop-up

Agents

Introduction

A call center agent is the person who handles incoming or outgoing customer calls for a business. A call center agent might handle account inquiries, customer complaints or support issues. Other names for a call center agent include customer service representative (CSR), telephone sales or service representative (TSR), attendant, associate, operator, account executive or team member.

—SearchCRM

In this respect, agents in XiVO have no fixed line and can login from any registered device.

Getting Started

- Create a user with a SIP line and a provisioned device.
- Create agents.
- Create a queue adding created agent as member of queue.

Creating agents

Service > Call center > Agents > General These settings are specific for a given agent.

Service > Call center > Agents > Users These settings are specific for a given agent.

Service > Call center > Agents > Queues These settings are specific for a given agent.

Service > Call center > Agents > Advanced These settings are specific for a given agent.

Service > IPBX > General settings > Advanced > Agent These settings are global for all agents.

Queues

Call queues are used to distribute calls to the agents subscribed to the queue. Queues are managed on the *Services* → *Call Center* → *Queues* page.

The screenshot shows the 'Queues > Add' configuration window. It features a series of tabs: General, Announces, Members, Application, No answer, Advanced, Schedules, and Diversions. The 'General' tab is active. The form contains the following fields and controls:

- Name:** A text input field.
- Display name:** A text input field.
- Number:** A text input field.
- Ring strategy:** A dropdown menu set to 'Ring all' with a help icon.
- Context:** A dropdown menu set to 'Default (default)' with a help icon.
- On-Hold Music:** A dropdown menu set to 'default' with a help icon.
- Announce when a member picks up the call:** A dropdown menu with a help icon.
- Customize the name of the caller:** A dropdown menu.
- Preprocess subroutine:** A text input field.
- Save:** A button at the bottom center.

Fig. 1.89: *Services* → *Call Center* → *Queues* → *Add*

A queue can be configured with the following options:

- **Name:** used as an unique id, cannot be `general`
- **Display name:** Displayed on the supervisor screen
- **On-Hold music:** The music the caller will hear. The music is played when waiting and when the call is on hold.

A ring strategy defines how queue members are called when a call enters the queue. A queue can use one of the following ring strategies:

- **Linear:** For each call, in the same order, starting from the same member
 - For agents: In login order
 - For static members: In definition order
- **Least recent:** call the member who least recently hung up a call
- **Fewest calls:** call the member with the fewest completed calls
- **Round robin memory:** call the “next” member after the one who answered last
- **Random:** call a member at random
- **Weight random:** same as random, but taking the member penalty into account

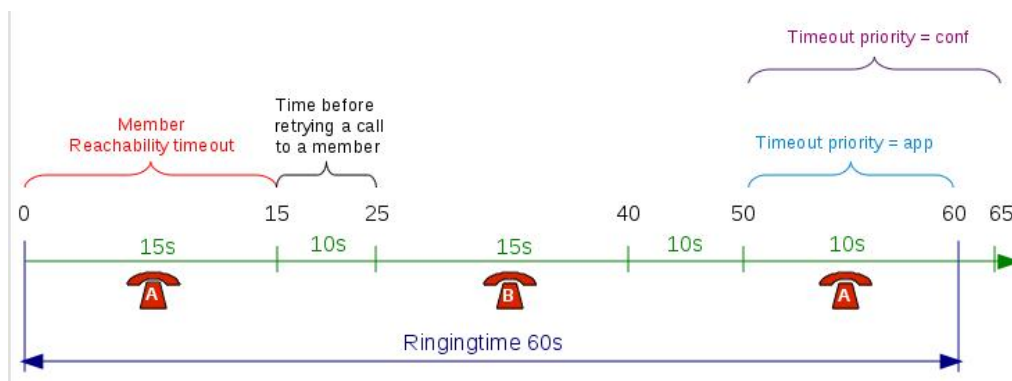
- Ring all: call all members at the same time

Warning: When editing a queue, you can't change the ring strategy to linear. This is due to an asterisk limitation. Unfortunately, if you want to change the ring strategy of a queue to linear, you'll have to delete it first and then create a new queue with the right strategy.

Note: When an agent is a member of many queues the order of call distribution between multiple queues is nondeterministic and cannot be configured.

Timers You may control how long a call will stay in a queue using different timers:

- Member reachability time out (Advanced tab): Maximum number of seconds a call will ring on an agent's phone. If a call is not answered within this time, the call will be forwarded to another agent.
- Time before retrying a call to a member (Advanced tab): Used once a call has reached the "Member reachability time out". The call will be put on hold for the number of seconds allotted before being redirected to another agent.
- Ringing time (Application tab): The total time the call will stay in the queue.
- Timeout priority (Application tab): Determines which timeout to use before ending a call. When set to "configuration", the call will use the "Member reachability time out". When set to "dialplan", the call will use the "Ringing time".



No Answer Calls can be diverted on no answer:

- No answer: The call reached the "Ringing time" in Application tab and no agent answered the call
- Congestion: The number of calls waiting has reached the "Maximum number of people allowed to wait" limit specified on the advanced tab
- Fail: No agent was available to answer the call when the call entered the queue ("Join an empty queue" condition on the advanced tab) or the call was queued and no agents were available to answer ("Remove callers if there are no agents" on the advanced tab)

Diversions Diversions can be used to redirect calls to another destination when a queue is very busy. Calls are redirected using one of the two following scenarios:

The diversion check is done only once per call, before the *preprocess subroutine* is executed and before the call enters the queue.

In the following sections, a waiting call is a call that has entered the queue but has not yet been answered by a queue member.

Queues > Edit blue (3000@default)

General Announces Members Application **No answer** Advanced Schedules Divisions

No answer

Destination : Queue

Redirect to : green (3006@default)

Ring time :

Busy

Destination : End call

Choice: Hangup

Congestion

Destination : User

Redirect to : Bill Johnson

Ring time :

Fail

Destination : Voicemail

Redirect to : 1456 (1456@default)

Play occupation message : ☐

Do not play introduction message : ☐

Do not play unavailable message : ☐

Use n+101 method : ☐

Save

Queues > Edit foobar (3005@default)

General Announces Members Application No answer Advanced Schedules **Divisions**

On estimated wait time overrun

Maximum estimated wait time: 5 minutes

Destination : End call

Choice: Busy

Delay before hangup:

On number of waiting calls per logged-in agent overrun

Maximum number of waiting calls per logged-in agent: 1

Destination : End call

Choice: Busy

Delay before hangup:

Save

Estimated Wait Time Overrun When this scenario is used, the administrator can set a destination for calls to be sent to when the estimated waiting time is over the threshold.

Note that if a new call arrives when there are no waiting calls in the queue, the call will **always** be allowed to enter the queue.

Note:

- this *estimated* waiting time is computed from the **actual hold time** of all **answered** calls in the queue (since last asterisk restart) according to an [exponential smoothing formula](#)
 - the estimated waiting time of a queue is updated only when a queue member answers a call.
-

Number of Waiting Calls per Logged-In Agent Overrun When this scenario is used, the administrator can set a destination for calls to be sent to when the number of waiting calls per logged-in agent is over the threshold.

The number of waiting calls includes the call for which the check is currently being performed.

The number of logged-in agents is the sum of user members and currently logged-in agent members. An agent only needs to be logged in and a member of the queue to participate towards the count of logged-in agents, regardless of whether he is available, on call, on pause or on wrapup.

The maximum number of waiting calls per logged-in agent can have a fractional part.

Here are a few examples:

```
Maximum number of waiting calls per logged-in agent: 1
Current number of waiting calls: 2
Current number of logged-in agents: 2
Number of waiting calls per logged-in agent when a new call arrives: 3 / 2 = 1.5
Call will be redirected

Maximum number of waiting calls per logged-in agent: 0.5
Number of waiting calls: 5
Number of logged-in agents: 12
Number of waiting calls per logged-in agent when a new call arrives: 6 / 12 = 0.5
Call will not be redirected
```

Note that if a new call arrives when there are no waiting calls in the queue, the call will **always** be allowed to enter the queue. For example, in the following scenario:

```
Maximum number of waiting calls per logged-in agent: 0.5
Current number of waiting calls: 0
Current number of logged-in agents: 1
Number of waiting calls per logged-in agent when a new call arrives: 1 / 1 = 1
```

Even if the number of waiting calls per logged-in agent (1) is greater than the maximum (0.5), the call will still be accepted since there are currently no waiting calls.

Supervision

Introduction Allows a contact center supervisor to monitor contact center activities such as:

- Monitoring real time information from call queues
- Agent activities per call queues
- Agent detailed activities

XiVO client as a Supervision Platform

Configuration A supervisor profile defined in *Service* → *CTI Server* → *Profiles* menu usually contains the following Xlets :

- Identity
- Queues
- Queue members
- Queues (entries detail)
- Agents (list)
- Agents (detail)

Note You may also see the *Agent Status Dashboard*

The screenshot displays the XIVO interface with several sections. The 'Queues' table at the top left lists queues with columns: Number, Queues, Waiting calls, EWT, Longest wait, Talking, Logged, and Availat. The 'Calls of a Queue' section shows details for 'Bakery (301) on xivo (default)'. The 'Queue Members' table lists agents with columns: Number, Firstname, Lastname, Logged, Paused, Answered calls, Last call, and Penalty. The 'Agent Details' section shows information for 'Bob Cat (102) on xivo (default)'. The 'Agents List (plain)' table lists agents with columns: Number, First name, Last name, Listen, Status since, Logged, Joined queues, Paused, and Paused queues. Arrows indicate the flow of information between these sections.

Supervision Panel

- Clicking on a queue's name in the queue list will display the agent list in the xlet *Queue Members* and show waiting calls in the *Calls of a Queue* xlet.
- Clicking on an agent's name in the agent list will display information on the agent in the *Agent Details* xlet
- Clicking on the + icon in the *Agent Details* xlet will display information about the selected queue in the *Calls of a Queue* and *Queue Members* xlets.

Queue List General information

The queue list is a dashboard displaying queue statistics and real-time counters for each queue configured on the XIVO.

Number	Queues	Waiting calls	EWT	Talking	Logged	Available	Received	Answered	Abandoned	Mean Waiting Time	Max Waiting Time	Efficiency	QOS
3947	UNIX	--	00:00	0	0	0	0	0	0	-	-	-	-
3256	tornato	--	00:00	0	0	0	0	0	0	-	-	-	-
3007	superqueue	--	00:00	0	N/A	0	0	0	0	-	-	-	-

Real-time Columns

The data of following columns display real-time information.

Queues queue name and number if configured to be displayed

Waiting calls The number of calls currently waiting for an agent in this queue. The background color can change depending of the configured thresholds

EWT Estimated waiting time

Longest wait The longest waiting time for currently waiting calls. The background color can change depending of the configured thresholds

Talking The number of agents currently in conversation in the queue. This column is set to 0 when the queue has just been created and no members have been added.

Logged The number of logged agents in the queue. This column is set to “N/A” when the queue has just been created and no members have been added.

Available The number of available agents ready to take a call in the queue. This column is set to N/A when the queue has just been created and no members have been added.

Last Period Columns

The data of following columns are based on statistics fetched from a fixed-width window of time, e.g. the last 60 minutes or the last 10 minutes. See below to configure the width of the window for each queue.

Received The number of calls received in this queue during the configured statistical window

Answered The number of calls answered in this queue during the configured statistical window

Abandoned The number of calls abandoned in this queue during the configured statistical window

Mean waiting time The mean wait time in the statistical time window, in mm:ss If no calls are received, “-” is displayed

Max waiting time The longest wait time in the statistical time window, in mm:ss If no calls are received, “-” is displayed

Efficiency Answered calls over received calls during the configured statistical window (unanswered calls that are still waiting are not taken into account). If no calls are received, “-” is displayed

QOS Percentage of calls taken within X seconds over answered calls during the configured statistical window. If no calls are received, “-” is displayed

Counter availability

When the XiVO client is started, “na” is displayed for counters that have not been initialized.

When the XiVO client is restarted, the counters are always displayed and calculated as if the application was not restarted. When the server is restarted, counters are reinitialized.

Enabling the xlet

The xlet can be added to any CTI profile from the web interface.

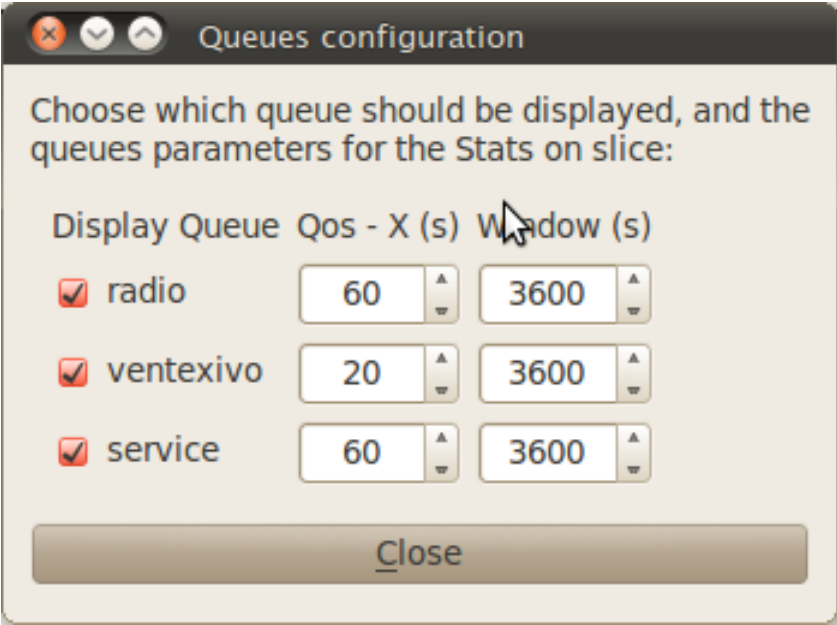
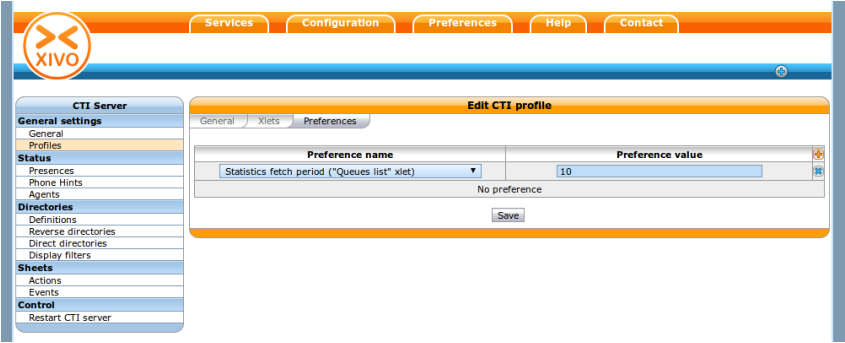


Configuration

Some values can be configured for the xlet. The statistic fetch timer can be set in the CTI profile preferences. This option is expressed in seconds and the default is 30 seconds.

The statistical period can be configured through the XiVO client once logged in by right-clicking on the Queue’s name in the *Queues* xlet. For each queue, you can configure the following information:

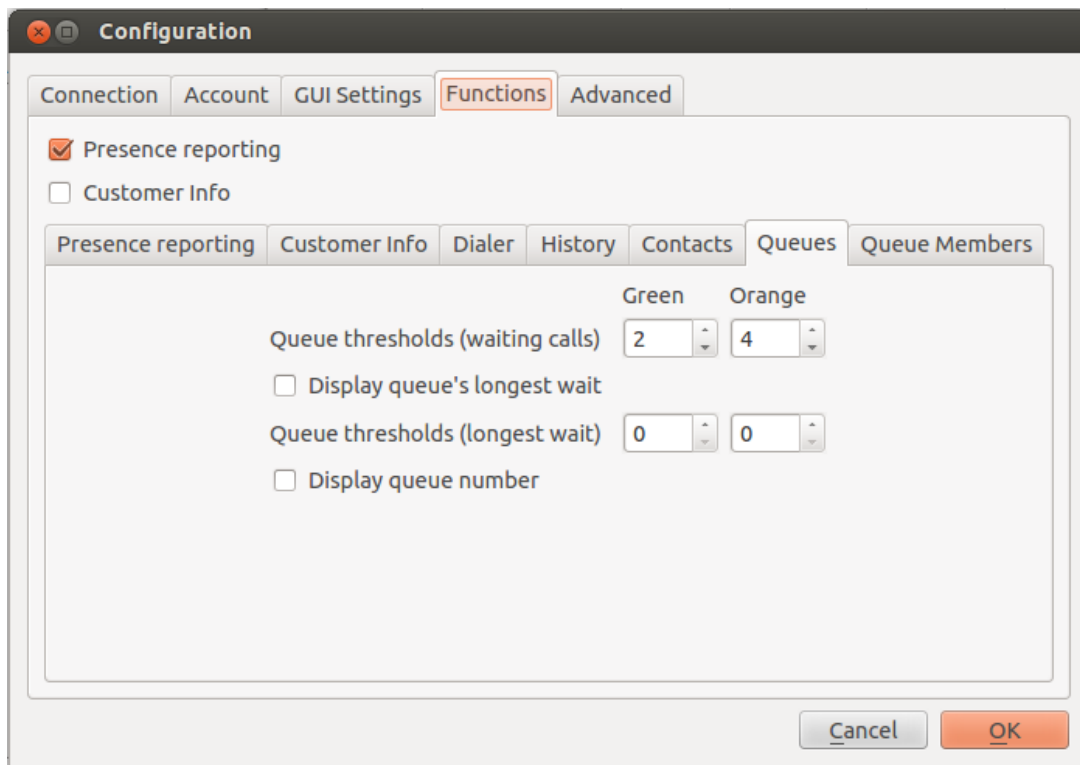
- Qos: maximum wait time for a call, in seconds.
- Window: period of time used for accumulating statistics, in seconds.



The data used to compute statistics on the XiVO server is only kept for a maximum of 3 hours. The window period cannot be configured to go beyond this limit.

Display options can also be set on the client side. A threshold can be configured to change the color of a column using the following parameters:

- Queue thresholds (waiting calls): number of waiting calls in the queue.
- Display queue's longest wait: Add a column displaying the number of seconds the longest call has waited.
- Queue thresholds (longest wait): number of seconds for the longest waiting call in the queue.
- Display queue number: Add a column displaying the queue's number.



Monitoring queues on high dimension screens

You may want to display the queue list on one big screen, visible by multiple people. However, the default font will not be large enough, so the information will not be readable.

You can change the font size of this Xlet by giving a configuration file when launching the XiVO Client:

```
> xivoclient -stylesheet big_fonts.qss # Windows and Mac
$ xivoclient -- -stylesheet big_fonts.qss # GNU/Linux
```

The `big_fonts.qss` file should contain:

```
QueuesView {font-size: 40px;}
QueuesView QHeaderView {font-size: 40px;}
```

Units of size that can be used are described on the [Qt documentation](#).

Agent List General information

The queue list is a dashboard displaying each agent configured on the XiVO.

Columns

Number The agent's number

First name & Last name The agent's first name and last name

Agents' List (plain)								
Number	First name	Last name	Listen	Status since	Logged	Joined queues	Paused	Paused queues
101	Alice	Wonderland	Listen	Not in use (10:58)	Logged	1	Unpaused	0
102	Bob	Cat	Listen	-	Unlogged	3	Unpaused	0
103	Charlie	Chaplin	Listen	In use (10:11)	Logged	1	Paused	1

Listen A *clickable cell* to listen to the agent's current call.

Clicking on the cell will make your phone ring. When you'll answer, you'll hear the conversation the agent is having.

You'll then be able to press the following digits on your phone to switch between the different "listen" modes:

- 4 - spy mode (default). No one hears you.
- 5 - whisper mode. Only the agent hears you.
- 6 - barge mode. Both the agent and the person he's talking to hear you.

Status since Shows the agent's status and the time spent in this status. An agent can have three statuses:

- *Not in use* when he is ready to answer an ACD call
- *Out of queue* when he called or answered a call not from the queue
- *In use* when he is either on call from a queue, on pause or on wrapup

Logged A *clickable cell* to log or unlog the agent

Joined queues The number of queues the agent will be receiving calls from

Paused A *clickable cell* to pause or unpause the agent

Paused queues The number of queues in which the agent is paused

Agent Details **General information**

Display advanced informations of an agent and enable to login/logoff, add/remove to a queue, and pause/unpause.

1. This is the status information of agent
2. Button to login/logoff agent
3. Supervision button of the Xlet "Calls of a queue"
4. Add/Remove agent for given queue
5. Pause/Unpause button for given queue

Queue members The queue members lists which agents or phones will receive calls from the selected queue and some of their attributes.

Columns

Number The agent number or the phone number of the queue member.

Firstname and Lastname First name and last name of the agent or the user to which the phone belongs.

Logged Whether the agent is logged or not. Blank for a phone.

Paused Whether the agent is paused or not. Blank for a phone.

Answered calls Number of calls answered by the member since last login (for an agent), or restart or configuration reload.

Last call Hangup time of the last answered calls.

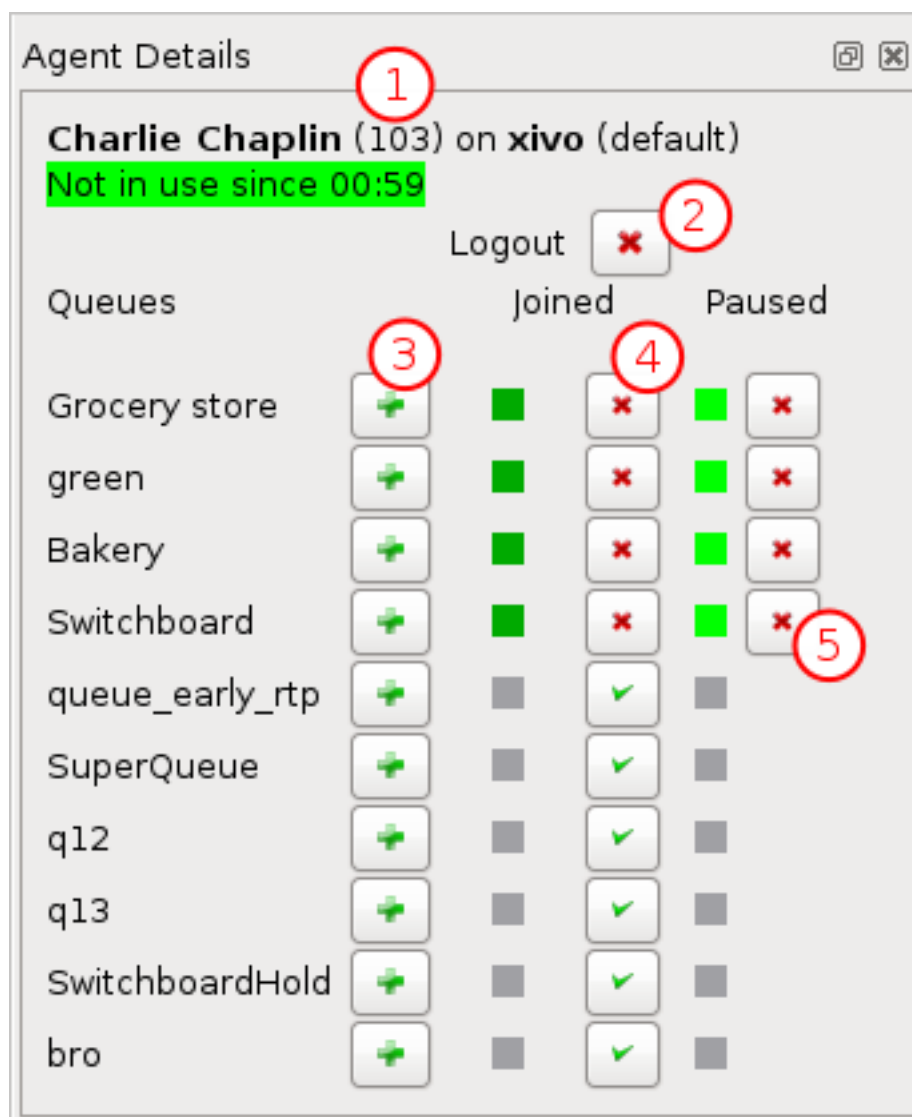


Fig. 1.90: Agent Details

Queue Members							
Épicerie (301@default) : 2 agent(s) and 2 phone(s)							
Number	Firstname	Lastname	Logged	Paused	Answered calls	Last call	Penalty
101	Alice	Wonderland	Logged in	Not paused	0		2
1151	Yoda	Kenobi			0		0
	Paul	Castagnette			0		0
102	Bob	Cat	Logged out	Not paused	0		6

Penalty Penalty of the queue member.

Link XiVO Client presence to agent presence You can configure XiVO to have the following scenario:

- The agent person leaves temporarily his office (lunch, break, ...)
- He sets his presence in the XiVO Client to the according state
- The agent will be automatically set in pause and his phone will not ring from queues
- He comes back to his office and set his presence to 'Available'
- The pause will be automatically cancelled

You can *configure the presence states* of CTI profiles and attach *Actions* to them, such as *Set in pause* or *Enable DND*.

You can then attach an action *Set in pause* for multiple presence states and attach an action *Cancel the pause* for the presence state *Available*.

For now, the actions attached to the mandatory presence *Disconnected* will not be taken into account.

Agent Status Dashboard

Overview The goal of the agent status dashboard xlet is to give contact center supervisors a better overview of agent status evolution in active queues.

Agent status dashboard

Queue 11002					Queue 11000			Queue 11007				
A. 9025 Not in use 00:24	A. 9056 Not in use 01:42	A. 9055 In use 04:16	A. 9002 In use 06:50	A. 9034 In use 00:52	A. 9019 In use 02:23	A. 9028 Not in use 02:40	A. 9054 In use 04:13	A. 9003 Not in use 03:59	A. 9120 Not in use 01:23	A. 9050 Not in use 01:44	A. 9038 In use 01:04	A. 9029 In use 01:33
A. 9001 In use 03:04	A. 9016 Not in use 02:13	A. 9043 In use 00:23	A. 9008 Not in use 00:19		0. 1 Not in use 18:29:09	A. 9037 In use 00:27	A. 9033 Not in use 00:20	A. 9063 Not in use 00:20	A. 9172 Not in use 02:26	A. 9095 Not in use 00:25	A. 9012 Not in use 03:18	A. 9047 In use 00:06
Queue 11001					A. 9187 In use 00:58	A. 9011 Not in use 00:32	A. 9046 In use 01:23	A. 9077 In use 00:37				
A. 9026 In use 00:16	A. 9057 In use 00:48	A. 9035 In use 02:17	A. 9001 In use 03:04	A. 9201 Not in use 00:46				Queue 11004				
A. 9017 In use 01:20	A. 9044 In use 01:48							A. 9023 Not in use 04:04	A. 9054 In use 04:13	A. 9192 Not in use 02:26	A. 9006 Not in use 00:11	A. 9032 In use 01:43
Ghost 11008					Queue 11005							
A. 9027 Not in use 01:10	A. 9006 Not in use 00:11	0. 1 Not in use 18:29:09	A. 9036 Not in use 00:35	A. 9001 In use 03:04	A. 9022 In use 00:12	A. 9049 In use 00:44	A. 9052 In use 01:13	A. 9067 Not in use 16:17	A. 9014 Not in use 06:19	A. 9041 Not in use 09:28	A. 9141 Not in use 11:05	
A. 9010 Not in use 01:45	A. 9018 In use 00:29	A. 9045 In use 01:01	A. 9000 In use 00:04		A. 9200 Not in use 02:11	A. 9002 In use 06:50	A. 9031 In use 02:12	Queue 11006				
					A. 9005 Not in use 01:02	A. 9040 Not in use 00:29						
								A. 9021 Not in use 01:59	A. 9057 In use 00:48	A. 9051 Not in use 00:27	A. 9030 Not in use 12:27	A. 9059 In use 01:39
								A. 9013 Not in use 17:43	A. 9048 Not in use 09:39	A. 9039 Not in use 15:52	A. 9004 Not in use 07:12	

Usage The xlet is *read-only* and presents a list of queues. For each queue, the xlet displays a status box for each logged in agent. Each status box gives the following information:

- Agent name
- Agent status: Shows the agent's status. An agent can have six statuses:
 - *Not in use* when he is ready to answer an ACD call
 - *Int. Incoming* when he answered an internal call not from a queue
 - *Int. Outgoing* when he emitted an internal call not from a queue
 - *Ext. Incoming* when he answered an external call not from a queue
 - *Ext. Outgoing* when he emitted an external call not from a queue
 - *In use* when he is either on call a from a queue, on pause or on wrapup

- Agent status since: Shows the time spent in the current status
- Background color:
 - green if *Not in use*
 - purple if *Int. Incoming* or *Int. Outgoing*
 - pink if *Ext. Incoming* or *Ext. Outgoing*
 - orange if *In use*

Note that the agent status will only change when the communication is established, not when phones are ringing.

Known bugs

1. If an agent emits a call via his XiVO Client, the status will change to *Int. Outgoing* or *Ext. Outgoing* when the destination phone rings, instead of when the destination answers.
2.
 - Given the agent is on an ACD call
 - When the agent logs out
 - When the agent hangs up the ACD call
 - When the agent logs back in via CTI Client
 - Then the agent may be seen as outgoing non-ACD communication, whether there is a non-ACD communication or not

To make the agent *Not in use* again, make a non-ACD call and hangup.

3.
 - Given the agent is on ACD call
 - When the agent calls someone else (e.g. his supervisor)
 - When the ACD call hangs up (while the agent talks to his supervisor)
 - Then the agent is seen as available, instead of in outgoing non-ACD communication.

This applies to all kinds of non-ACD calls.

Changing the disposition The disposition of the Xlet can be changed in two ways:

- Placement of queues
- Which queues are displayed

The disposition is saved whenever the XiVO Client is closed and restored when it is opened again.

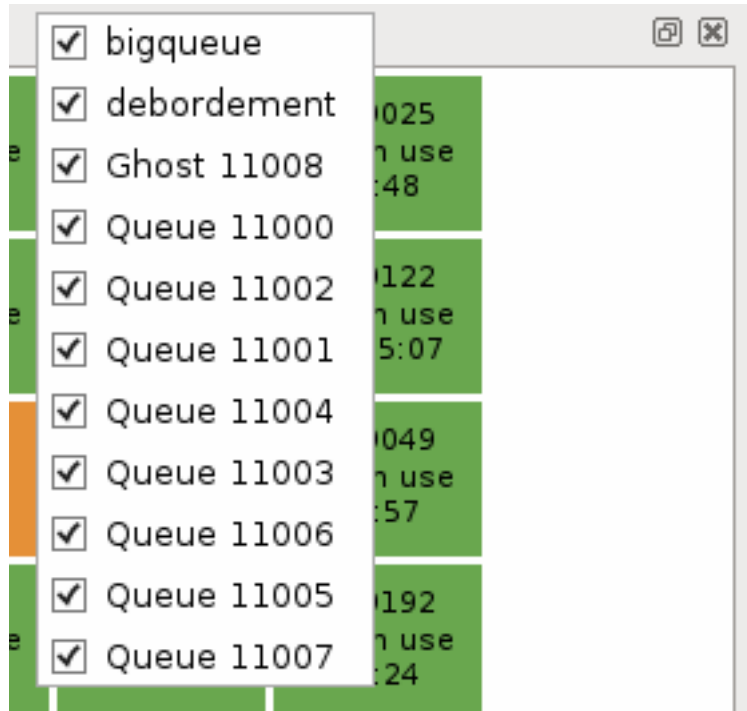
Changing the placement of queues The little windows containing each queue can be resized and moved around. That way, any layout can be achieved, according to the size and importance of each queue.

Choosing which queues are displayed There is a little contextual menu when right-clicking on the title bar of every queue window. Checking/unchecking the lines of this menu shows/hides the associated queue.

Known issues

- There is no profile containing this xlet. The profile must be created manually.
- There is no sorting on agents in a queue.
- An empty queue will display an empty box with no message specifying the queue has no logged agents.

Configuration No special configuration is necessary other than creating a CTI profile in which the Agent Status Dashboard is added.



Skills-Based Routing

Introduction

Skills-based routing (SBR), or Skills-based call routing, is a call-assignment strategy used in call centres to assign incoming calls to the most suitable agent, instead of simply choosing the next available agent. It is an enhancement to the Automatic Call Distributor (ACD) systems found in most call centres. The need for skills-based routing has arisen, as call centres have become larger and dealt with a wider variety of call types.

—Wikipedia

In this respect, skills-based routing is also based on call distribution to agents through waiting queues, but one or many skills can be assigned to each agent, and call can be distributed to the most suitable agent.

In skills-based routing, you will have to find a way to be able to tag the call for a specific skill need. This can be done for example by entering the call distribution system using different incoming call numbers, using an IVR to let the caller do his own choice, or by requesting to the information system database the customer profile.

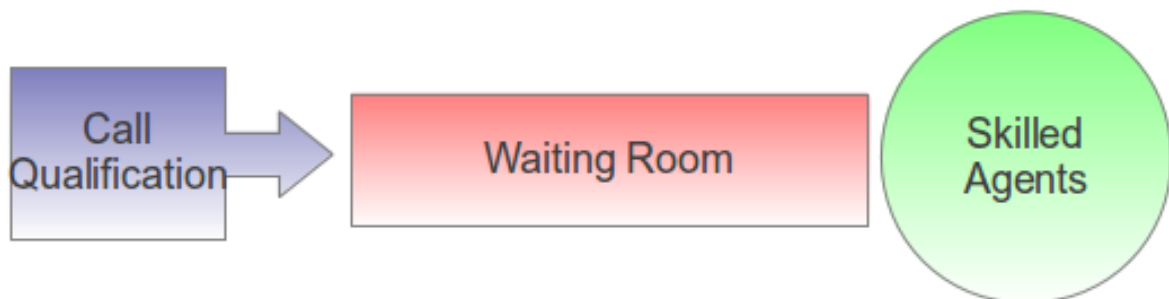


Fig. 1.91: Skills-Based Routing

Getting Started

- Create the skills

- Apply the skills to the agents
- Create the skill rule sets
- Assign the skill rule sets using a configuration file
- Apply the skill rule sets to call qualification, i.e. incoming calls by using the preprocess subroutine field

Note that you shouldn't use skill based routing on a queue with queue members of type user because the behaviour is not defined and might change in a future XiVO version.

Skills Skills are created using the menu *Services* → *Call center* → *Skills*. Each skill belongs to a category. First create the category, and in this category create different skills.

Note that a skill name can't contain upper case letters and must be globally unique (i.e. the same name can't be used in two different categories).

Skills > Edit

Category: language

Values:

Name	Description	
english		
french		

Save

Fig. 1.92: Skills Creation

Once all the skills are created you may apply them to agents. Agents may have one or more skills from different categories.

Agents > Add an agent

General Users Queues Queueskills Advanced

Skill	Weight	
english	75	
french	25	

Save

Fig. 1.93: Apply Skills to Agents

It is typical to use a value between 0 and 100 inclusively as the weight of a skill, although any integer is accepted.

Skill Rule Sets Once skills are created, rule sets can be defined.

A rule set is a list of rules. Here's an example of a rule set containing 2 rules:

1. WT < 60, english > 50
2. english > 0

The first rule of this rule set can be read as:

If the caller has been waiting for less than 60 seconds (WT < 60), only try to call agents which have the skill "english" set to a value higher than 50; otherwise, go to the next rule.

And the second rule can be read as:

Only try to call agents which have the skill “english” set to a value higher than 0.

Let’s examine some simple scenarios, because there’s actually some subtleties on how calls are distributed. We will suppose that we have a queue with the default settings and the following members:

- Agent A, with skill english set to 75
- Agent B, with skill english set to 25

Scenario 1 Given:

- Agent A is logged and not in use
- Agent B is logged and not in use
- There is no call in the queue

When a new call enters the queue, then it is distributed to Agent A. As long as Agent A is available and doesn’t answer the call, the call will never be distributed to Agent B, even after 60 seconds of waiting time.

When another call enters the queue, then after 60 seconds of waiting time, this call will be distributed to Agent B (and the first call will still be distributed only to Agent A).

The reason is that there’s a difference between a call that is being distributed (i.e. that is making agents ring) and a call that is waiting for being distributed. When a call is being distributed to a set of members, no other rule is tried as long as there’s at least 1 of these members available.

Scenario 2 Given:

- Agent A is not logged
- Agent B is logged and not in use
- There is no call in the queue

When a new call enters the queue, then it is *immediately* distributed to Agent B.

The reason is that when there’s no logged agent matching a rule, the next rule is immediately tried.

Rules Each rule set is composed of rules, and each rule has two parts, separated by a comma:

- the first part (optional) is the “*dynamic part*”
- the second part is the “*skill part*”

Each part contains an expression composed of operators, variables and integer constants.

Operators The following operators can be used inside rules:

Comparison operators:

- operand1 ! operand2 (is not equal)
- operand1 = operand2 (is equal)
- operand1 > operand2 (is greater than)
- operand1 < operand2 (is lesser than)

Logical operators:

- operand1 & operand2 (both are true)
- operand1 | operand2 (at least one of them are true)

‘!’ is the operator with the higher priority, and ‘|’ the one with the lower priority. You can use parentheses ‘()’ to change the priority of operations.

Dynamic Part The dynamic part can reference the following variables:

- WT
- EWT

The waiting time (WT) is the elapsed time since the call entered the queue. The time the call pass in an IVR or another queue is not taken into account.

The estimated waiting time (EWT) has never fully worked. It is mentioned here only for historical reason. You should not use it. It might be removed in a future XiVO version.

Examples

- WT < 60

Skill Part The skill part can reference any skills name as variables.

You can also use meta-variables, starting with a '\$', to substitute them with data set on the Queue() call. For example, if you call Queue() with the skill rule set argument equal to:

```
select_lang(lang=german)
```

Then every \$lang occurrence will be replaced by 'german'.

Fig. 1.94: Create Skill Rule Sets

Examples

- english > 50
- technic ! 0 & (\$os > 29 & \$lang > 39 | \$os > 39 & \$lang > 19)

Evaluation Note that the expression:

english | french

is equivalent to:

english ! 0 | french ! 0

Sometimes, a rule references a skill which is not defined for every agent. For example, given the following rule:

english > 0 | english < 1

Then, for an agent which has the skill english defined, the result of this expression is always true. For an agent which does not have the skill english defined, the result of this expression is always false.

Said differently, an agent without a skill X is not the same as an agent with the skill X set to the value 0.

Technically, this is what is happening when evaluating the rule “english > 0” for an agent without the skill english:


```

english > 0
=      <Substituting english with the agent value>
"undefined" > 0
=      <A comparison with "undefined" in at least one operand yields undefined>
"undefined"
=      <In a boolean context, "undefined" is equal to false>
false

```

This behaviour applies to every comparison operators.

Also, the syntax that is currently accepted for comparison is always of the form:

```
variable cmp_op constant
```

Where “variable” is a variable name, “cmp_op” is a comparison operator and “constant” is an integer constant. This means the following expressions are not accepted:

- 10 < english (but english > 10 is accepted)
- english < french (the second operand must be a constant)
- 10 < 11 (the first operand must be a variable name)

Apply Skill Rule Sets A skill rule set is attached to a call using a bit of dialplan. This dialplan is stored in a configuration file you may edit using menu *Services* → *IPBX* → *Configuration Files*.

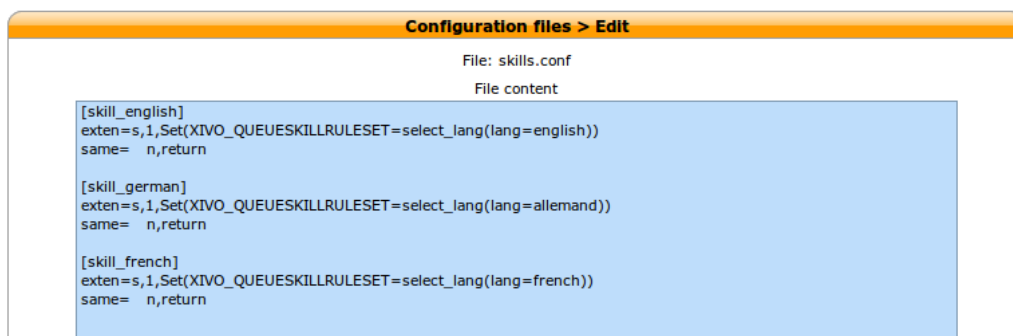


Fig. 1.95: Use Rule Set In Dialplan

In the figure above, 3 different languages are selected using three different subroutines.

Each of this different selections of subroutines can be applied to the call qualifying object. In the following example language selection is applied to incoming calls.

Example

Configuration file for simple skill selection :

```

[simple_skill_english]
exten = s,1,Set(XIVO_QUEUESKILLRULESET=english_rule_set)
same = n,Return()

[simple_skill_french]
exten = s,1,Set(XIVO_QUEUESKILLRULESET=french_rule_set)
same = n,Return()

```

Monitoring You may monitor your waiting calls with skills using the asterisk CLI and the command `queue show <queue_name>`:

The screenshot shows a web interface for editing an incoming call rule. The title bar is orange and reads 'Incoming calls > Edit 4160001210 (from-extern)'. Below it are three tabs: 'General' (selected), 'Call permissions', and 'Schedules'. The form contains several fields: 'DID' (4160001210), 'Context' (Incalls (from-extern)), 'Destination' (Queue), 'Redirect to' (bureautique (5000@default)), 'Ring time' (empty), 'CallerID mode' (empty), 'Preprocess subroutine' (skill_english, circled in red), and 'Description' (empty text area). A 'Save' button is at the bottom.

Fig. 1.96: Apply Rule Set to Incoming Call

```
xivo-jylebleu*CLI> queue show services
services has 1 calls (max unlimited) in 'ringall' strategy (0s holdtime, 2s talktime), W:0, C:1, L:1
Members:
  Agent/2000 (Not in use) (skills: agent-1) has taken no calls yet
  Agent/2001 (Unavailable) (skills: agent-4) has taken no calls yet
Virtual queue english:
Virtual queue french:
  1. SIP/jyl-dev-assur-00000017 (wait: 0:05, prio: 0)
Callers:
```

You may monitor your skills groups with the command `queue show skills groups <agent_name>`:

```
xivo-jylebleu*CLI> queue show skills groups <PRESS TAB>
agent-2 agent-3 agent-4 agent-48 agent-7 agent-1
xivo-jylebleu*CLI> queue show skills groups agent-1
Skill group 'agent-1':
- bank : 50
- english : 100
```

You may monitor your skills rules with the command `queue show skills rules <rule_name>`:

```
xivo-jylebleu*CLI> queue show skills rules <PRESS TAB>
english french select_lang
xivo-jylebleu*CLI> queue show skills rules english
Skill rules 'english':
=> english>90
```

Statistics

Overview The statistics page is used to monitor the efficiency of queues and agents. Statistics are automatically generated every six hours. They can also be generated manually.

Note: The contact center statistics do not apply to switchboard queues. See [Switchboard](#) for more details.

Note: The oldest statistics are periodically removed. See *Purge Logs* for more details.

Configuration In order to display call center statistics, you must create at least one configuration profile.

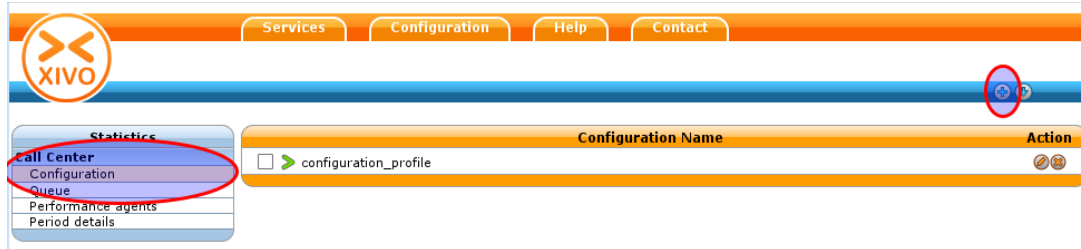


Fig. 1.97: Statistics Configuration

The configuration profile is used to generate reports from the cache. The cache is generated independently from the configuration so adding a new configuration does not require a new cache generation.

Configuration options

Field	Values	Description
name	string	Configuration name, useful for remembering what the configuration is used for
interval	enum [0-999] [day, week, month]	Default time interval used when displaying statistics. Examples: “-1” show statistics for yesterday “-3 weeks”: show statistics for the last 3 weeks
show on summary page		Display this configuration on the summary page
timezone	America/Montreal	Your time zone
Period cache		Maximum and minimum dates that can be used for displaying statistics
start	YYYY-MM	Start date
end	YYYY-MM	End date. If left to 0, use the servers’ current date
Working Hours		Work hours for agents
start	hh:mm	Beginning of working hours.
end	hh:mm	End of working hours
Periods		Number of calls answered for a time period
Period 1	number of seconds (Example: 20)	Show number of calls answered within 20 seconds in column “P1”
Period n	number of seconds (Example: 20)	Show number of calls answered within 20 seconds in column “Pn”

Note: Calls outside of working hours will not be in the cache. e.g. if working hours are from 8:00 AM to 16:00 PM, a call at 7:55 AM will not show up in the reports.

Note: Statistics are computed on the hour. e.g. If work hours are from 8:30 to 16:15, working hours should be set from 8:00 to 17:00.

Note: Period includes both bounds, if the same number is used for the higher bound and the lower bound of next period, some calls will be counted twice. i.e period 1 : 0-30 period 2 : 31-60 period 3 : 61

How to generate the cache The cache must be generated before using reports. By default, the cache is automatically generated every six hours.

However, you can safely generate it manually. The script to generate the cache is *xivo-stat fill_db*. When this script is run, statistics will be regenerated for the last 8 hours starting from the previous hour. e.g. If you run *xivo-stat* on 2012-08-04 11:47:00, statistics will be regenerated from 2012-08-04 03:00:00 to 2012-08-04 11:47:00

Note: *xivo-stat fill_db* can be a long operation when used for the first time or after a *xivo-stat clean_db*

Warning: The current events have an end date of the launch date of the script *xivo-stat* as the end date.

Clearing the cache If for some reason the cache generation fails or the cache becomes unusable, the administrator can safely clean the cache using *xivo-stat clean_db* and then regenerate it. This operation will only clear the cache and does *not* erase any other data.

Queue statistics Queue statistics can be viewed in *Services* → *Statistics* → *Queue*.

The first table displays a list of queues with all the counters for the period chosen from the Dashboard panel

					Dissuaded or Overflowed					
	Received	Answered	Abandoned	Closed	Saturated	NA	Blocking	AWT	Answered rate	QoS
STA1	1749	1521	84	21	0	0	123	00:00:13	88 %	0 %
STA2	1713	1454	73	38	0	0	148	00:00:11	86 %	0 %
STA3	1529	1367	76	23	0	0	63	00:00:10	90 %	0 %
STA4	2147	1776	115	17	0	0	239	00:00:16	83 %	0 %
STA5	1800	1594	93	28	0	0	85	00:00:13	89 %	0 %

By clicking on a queue name you may display detailed queue statistics

Statistics can be displayed :

By week

By month

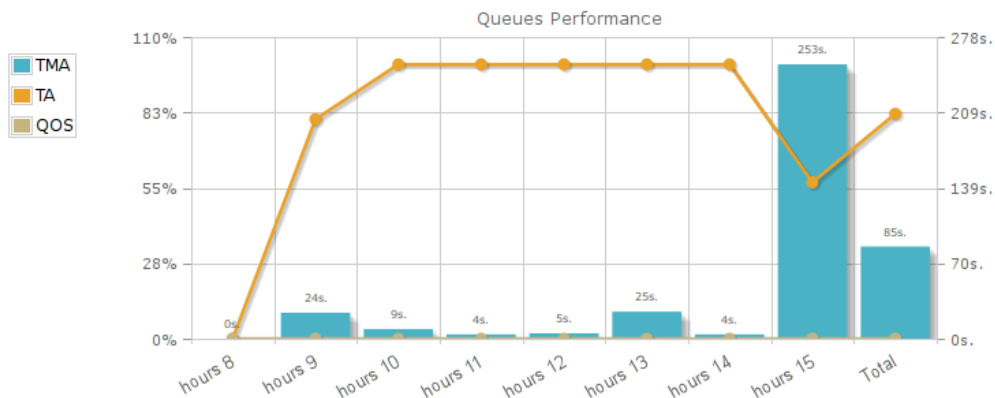
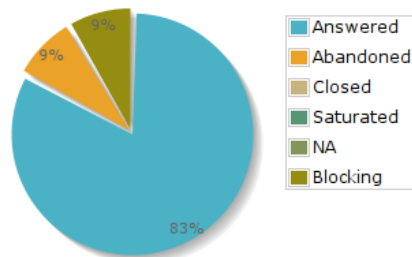
By year

Counters

- Received: Total number of received calls
- Answered: Calls answered by an agent
- Abandoned: Calls that were hung up while waiting for an answer
- Dissuaded or Overflowed:

				Dissuaded or Overflowed						
	Received	Answered	Abandoned	Closed	Saturated	NA	Blocking	AWT	Answered rate	QoS
8h-9h	0	0	0	0	0	0	0			
9h-10h	5	4	1	0	0	0	0	00:00:24	80 %	0 %
10h-11h	2	2	0	0	0	0	0	00:00:09	100 %	0 %
11h-12h	1	1	0	0	0	0	0	00:00:04	100 %	0 %
12h-13h	3	3	0	0	0	0	0	00:00:05	100 %	0 %
13h-14h	1	1	0	0	0	0	0	00:00:25	100 %	0 %
14h-15h	4	4	0	0	0	0	0	00:00:04	100 %	0 %
15h-16h	7	4	1	0	0	0	2	00:04:13	57 %	0 %
Total	23	19	2	0	0	0	2	00:01:25	82 %	0 %

Total call distributed



				Dissuaded or Overflowed						
	Received	Answered	Abandoned	Closed	Saturated	NA	Blocking	AWT	Answered rate	QoS
Monday 7	28	27	0	0	0	0	1	00:00:10	96 %	0 %
Tuesday 8	22	20	1	0	0	0	1	00:00:04	90 %	0 %
Wednesday 9	23	19	2	0	0	0	2	00:01:25	82 %	0 %
Thursday 10	21	21	0	0	0	0	0	00:00:07	100 %	0 %
Friday 11	34	30	1	0	0	0	3	00:00:08	88 %	0 %
Total	128	117	4	0	0	0	7	00:00:21	91 %	0 %

Total call distributed



				Dissuaded or Overflowed						
	Received	Answered	Abandoned	Closed	Saturated	NA	Blocking	AWT	Answered rate	QoS
1 week										
Tuesday	0	0	0	0	0	0	0			
Wednesday	10	0	0	0	0	0	10	00:00:00	0 %	
Thursday	13	12	1	0	0	0	0	00:00:23	92 %	0 %
Friday	19	17	2	0	0	0	0	00:00:25	89 %	0 %
2 week										
Monday	28	27	0	0	0	0	1	00:00:10	96 %	0 %
Tuesday	22	20	1	0	0	0	1	00:00:04	90 %	0 %
Wednesday	23	19	2	0	0	0	2	00:01:25	82 %	0 %
Thursday	21	21	0	0	0	0	0	00:00:07	100 %	0 %
Friday	34	30	1	0	0	0	3	00:00:08	88 %	0 %
3 week										
Monday	36	35	0	0	0	0	1	00:00:11	97 %	0 %
Tuesday	40	36	4	0	0	0	0	00:00:07	90 %	0 %
Wednesday	35	35	0	0	0	0	0	00:00:07	100 %	0 %
Thursday	51	51	0	0	0	0	0	00:00:05	100 %	0 %
Friday	16	16	0	0	0	0	0	00:00:04	100 %	0 %
4 week										
Monday	24	24	0	0	0	0	0	00:00:04	100 %	0 %
Tuesday	25	24	1	0	0	0	0	00:00:08	96 %	0 %
Wednesday	28	24	4	0	0	0	0	00:00:26	85 %	0 %
Thursday	40	37	0	3	0	0	0	00:00:09	100 %	0 %
Friday	39	37	2	0	0	0	0	00:00:06	94 %	0 %
5 week										
Monday	43	43	0	0	0	0	0	00:00:07	100 %	0 %
Tuesday	35	32	1	0	0	0	2	00:00:06	91 %	0 %
Wednesday	31	28	1	0	0	0	2	00:00:07	90 %	0 %
Thursday	27	15	1	0	0	0	11	00:00:49	55 %	0 %
Total	640	583	21	3	0	0	33	00:00:13	91 %	0 %

Total call distributed



	Dissuaded or Overflowed							AWT	Answered rate	QoS
	Received	Answered	Abandoned	Closed	Saturated	NA	Blocking			
January	0	0	0	0	0	0	0			
February	0	0	0	0	0	0	0			
March	0	0	0	0	0	0	0			
April	0	0	0	0	0	0	0			
May	0	0	0	0	0	0	0			
June	0	0	0	0	0	0	0			
July	119	98	7	13	0	0	1	00:00:08	92 %	1 %
August	95	72	6	3	0	0	14	00:00:10	78 %	0 %
September	181	153	9	0	0	0	19	00:00:11	84 %	0 %
October	214	159	9	0	0	0	46	00:00:13	74 %	0 %
November	177	169	6	1	0	0	1	00:00:06	96 %	0 %
December	194	180	5	1	0	0	8	00:00:07	93 %	0 %
Total	973	824	42	18	0	0	89	00:00:09	86 %	0 %

Total call distributed



- Closed: Calls received when the queue was closed
 - No answer (NA): Calls that reached the ring timeout delay
 - Saturated: Calls received when the queue was already full (“Maximum number of people allowed to wait:” limit of advanced tab) or when one of the diversion parameters were reached
 - Blocking : Calls received when no agents were available or when there were no agents to take the call, join an empty queue condition or remove callers if there are no agents condition is reached (advanced queue parameter tab).
- Average waiting time (AWT): The average waiting time of call on wait
 - Answered rate (HR): The ratio of answered calls over (received calls - closed calls)
 - Quality of service (QoS): Percentage of calls answered in less than x seconds over the number of answered calls, where x is defined in the configuration

Agent performance Agent performance statistics can be viewed in *Services* → *Statistics* → *Performance agents*.

Statistics		Performance agents					
Call Center							
Configuration							
Queue							
Performance agents							
Period details							

Dashboard							
test							
Bob (2002)							
Analysis Axis : Day							
Day: 2012-10-24							
Apply							
Cache Interval		2012-01 to date					
Time Slot		08:00 - 12:00					
Working days		MON TUE WED THU FRI SAT SUN					
Performance		215.55ms - 2.77 mb					

Performance agents					
	Answered	Nb. of calls	Login	Pause	Wrapup
8h-9h	11	00:02:07	00:52:16	00:00:00	00:01:50
9h-10h	3	00:00:25	00:01:42	00:00:00	00:00:30
10h-11h	0	00:00:00	00:00:00	00:00:00	00:00:00
11h-12h	2	00:00:12	00:03:56	00:00:09	00:00:40
Total	16	00:02:44	00:57:55	00:00:09	00:03:00

Note: The agent performance counters do not take into account transfer between agents: if agent A processes a call and transfers it to agent B, only the counters of agent A will be updated. Ignoring any info after the call transfer.







Counters

- Answered: Number of calls answered by the agent
- Conversation: Total time spent for calls answered during a given period
- Login: Total login time of an agent.
- Wrapup: Total time spent in wrapup by an agent.
- Pause: Total pause time of an agent

Warning: Data generated before XiVO 12.19 might have erroneous results for the Login time counter

Note: The Pause time counter only supports **PAUSEALL** and **UNPAUSEALL** command from cticlient. The agent must also be a member of a least 1 queue.

Note: Wrapup time events were added to XiVO in version 12.21

Performance agents					
	Nb. of calls	Total time			
	Answered	Conversation	Login	Pause	Wrapup
 Karine Boudoux (108)	87	09:59:57	30:58:51	10:53:16	00:21:30
 Fred Epric (100)	45	01:27:52	34:59:45	05:32:45	00:07:30
 Hipolyte Marroussou (102)	13	00:24:23	27:42:47	97:50:42	00:02:10
 Gérard Mensour (101)	0	00:00:00	00:00:00	00:00:00	00:00:00
 Irène Pourtox (106)	39	03:52:40	34:08:47	22:39:31	00:09:45
 Juliette Queriau (107)	78	09:03:51	34:06:24	91:23:52	00:19:30

Agent summary

	Nb. of calls	Total time			
	Answered	Conversation	Login	Pause	Wrapup
9h-10h	1	00:07:31	00:28:45	00:55:13	00:00:00
10h-11h	5	00:30:50	01:00:00	00:00:00	00:01:15
11h-12h	4	00:24:15	01:00:00	00:00:20	00:01:15
12h-13h	0	00:00:00	00:33:34	00:59:07	00:00:00
13h-14h	0	00:00:00	01:00:00	00:34:20	00:00:00
14h-15h	3	01:23:38	01:00:00	00:00:00	00:00:30
15h-16h	2	00:05:24	01:00:00	00:00:00	00:00:30
16h-17h	3	00:26:21	01:00:00	00:00:00	00:01:00
17h-18h	6	00:37:50	01:00:00	00:00:00	00:01:15
Total	24	03:35:49	08:02:20	02:29:01	00:05:45

Agent per day

	Nb. of calls	Total time			
	Answered	Conversation	Login	Pause	Wrapup
Monday 1	24	03:35:49	08:02:20	02:29:01	00:05:45
Tuesday 2	22	02:17:11	07:31:53	03:36:46	00:05:30
Wednesday 3	19	01:40:33	07:27:13	02:34:03	00:04:45
Thursday 4	22	02:26:24	07:57:25	02:13:23	00:05:30
Friday 5	0	00:00:00	00:00:00	00:00:00	00:00:00
Total	87	09:59:57	30:58:51	10:53:16	00:21:30

Agent per week

Agent per month

Agent per year

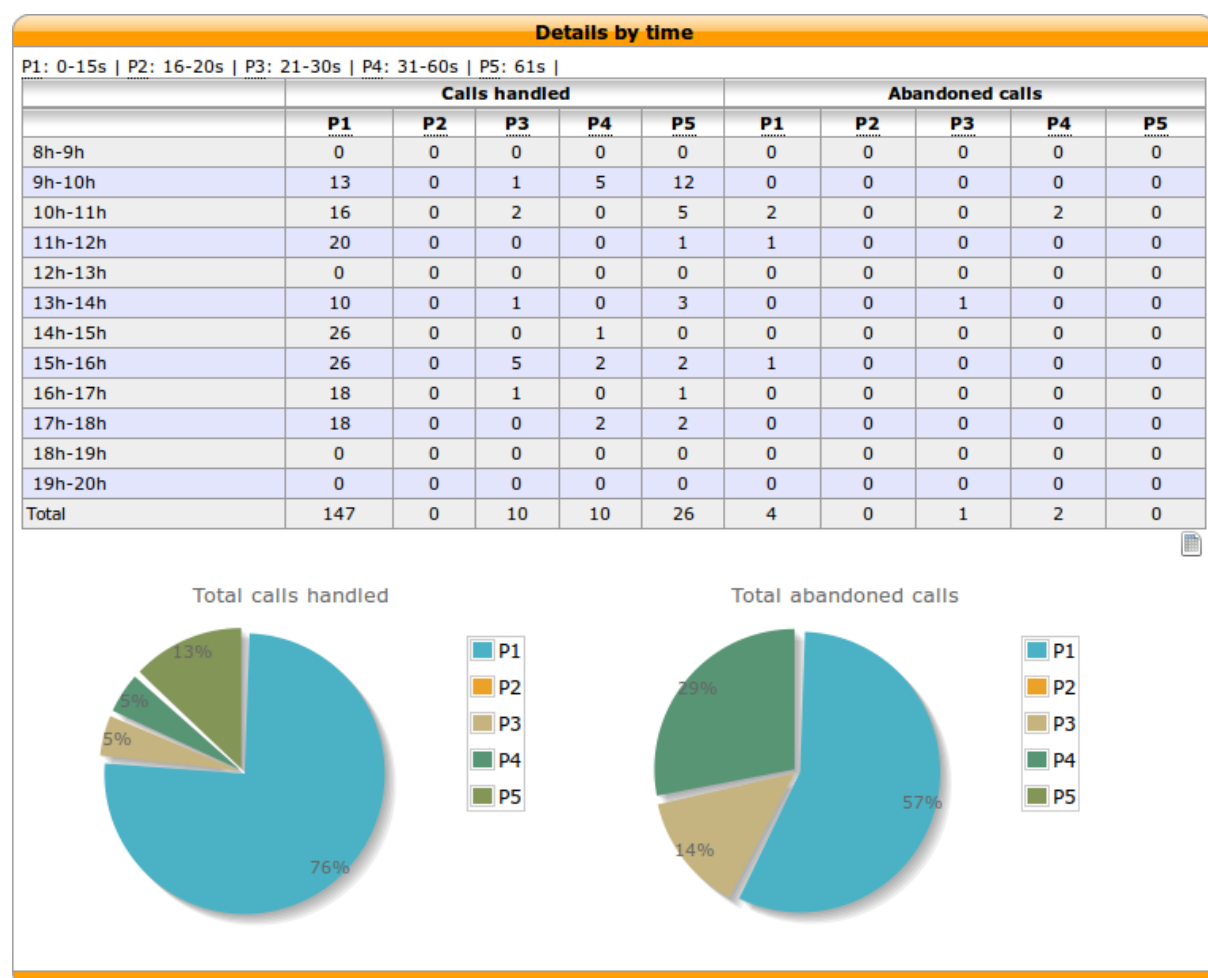
	Nb. of calls	Total time			
	Answered	Conversation	Login	Pause	Wrapup
23 week					
Monday	0	00:00:00	09:00:00	00:00:00	00:00:00
Tuesday	0	00:00:00	09:00:00	00:00:00	00:00:00
Wednesday	0	00:00:00	09:00:00	00:00:00	00:00:00
Thursday	0	00:00:00	09:00:00	00:00:00	00:00:00
Friday	0	00:00:00	09:00:00	00:00:00	00:00:00
24 week					
Monday	0	00:00:00	09:00:00	00:00:00	00:00:00
Tuesday	0	00:00:00	03:54:46	00:00:00	00:00:00
Wednesday	0	00:00:00	00:00:00	00:00:00	00:00:00
Thursday	0	00:00:00	04:35:05	04:56:29	00:00:00
Friday	10	01:23:58	08:24:59	03:11:31	00:01:35
25 week					
Monday	15	01:20:19	08:17:50	03:02:58	00:02:30
Tuesday	3	00:13:52	08:22:51	07:06:44	00:00:20
Wednesday	3	00:20:02	08:28:03	06:12:39	00:00:40
Thursday	0	00:00:00	08:24:06	08:24:01	00:00:00
Friday	1	00:09:22	08:27:27	05:27:33	00:00:15
26 week					
Monday	10	00:36:41	08:23:33	04:48:20	00:02:30
Tuesday	11	01:08:46	08:30:00	04:34:26	00:02:45
Wednesday	3	00:07:48	08:29:34	04:58:42	00:00:45
Thursday	5	00:40:10	04:00:58	07:26:52	00:01:15
Friday	9	01:12:33	08:19:18	04:27:04	00:02:15
Total	70	07:13:31	150:38:3	64:37:24	00:14:50

	Nb. of calls	Total time			
	Answered	Conversation	Login	Pause	Wrapup
January	0	00:00:00	00:00:00	00:00:00	00:00:00
February	0	00:00:00	00:00:00	00:00:00	00:00:00
March	0	00:00:00	00:00:00	00:00:00	00:00:00
April	0	00:00:00	00:00:00	00:00:00	00:00:00
May	1	00:00:34	97:17:07	00:00:00	00:00:05
June	89	09:01:48	159:03:1	69:17:25	00:19:35
July	39	03:52:40	34:08:47	22:39:31	00:09:45
August	0	00:00:00	00:00:00	00:00:00	00:00:00
September	0	00:00:00	00:00:00	00:00:00	00:00:00
October	0	00:00:00	00:00:00	00:00:00	00:00:00
November	0	00:00:00	00:00:00	00:00:00	00:00:00
December	0	00:00:00	00:00:00	00:00:00	00:00:00
Total	110	11:06:45	282:04:2	87:16:55	00:24:40

Period details Display by period defined in configuration, i.e. between 0 and 10s, 10s and 30s etc ... the number of handled calls and the number of abandoned calls

Details by time										
P1: 0-15s P2: 16-20s P3: 21-30s P4: 31-60s P5: 61s										
	Calls handled					Abandoned calls				
	P1	P2	P3	P4	P5	P1	P2	P3	P4	P5
blue	1992	25	140	152	250	38	3	6	15	26
green	0	0	0	0	0	0	0	0	0	0
red	0	0	0	0	0	0	0	0	0	0
yellow	640	2	18	7	2	10	1	0	2	1

You may click on a queue name to get more information for this queue



Period details by day

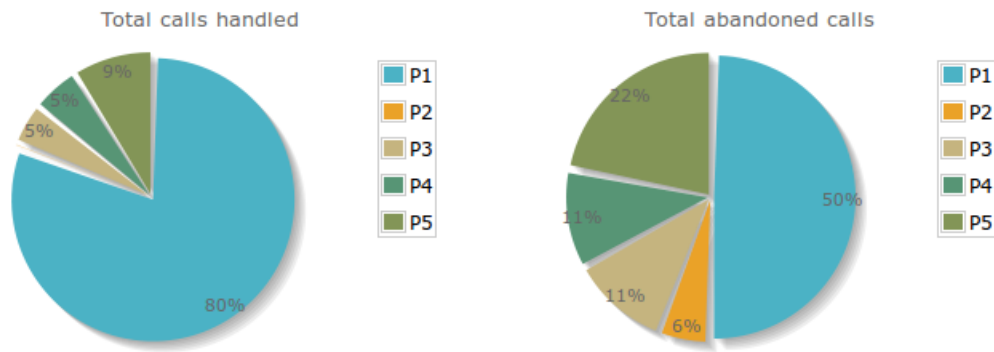
Period details by week

Period details by month

Period details by year

P1: 0-15s | P2: 16-20s | P3: 21-30s | P4: 31-60s | P5: 61s |

	Calls handled					Abandoned calls				
	P1	P2	P3	P4	P5	P1	P2	P3	P4	P5
Monday 1	147	0	10	10	26	4	0	1	2	0
Tuesday 2	145	2	8	8	6	2	0	0	0	1
Wednesday 3	128	0	8	7	7	1	0	1	0	2
Thursday 4	122	2	5	11	23	2	1	0	0	1
Friday 5	0	0	0	0	0	0	0	0	0	0
Total	542	4	31	36	62	9	1	2	2	4



Reporting

You may use your own reporting tools to be able to produce your own reports provided **you do not use the XiVO server original tables**, but copy the tables to your own data server. You may use the following procedure as a template :

- Allow remote database access on XiVO
- Create a postgresql account read only on asterisk database
- Create target tables in your database located on the data server
- Copy the statistic table content to your data server

General Architecture

1. The *queue_log* table of the *asterisk* database is filled by events from Asterisk and by custom dialplan events
2. *xivo-stat fill_db* is then used to read data from the *queue_log* table and generate the tables *stat_call_on_queue* and *stat_queue_periodic*
3. The web interface generate tables and graphics from the *stat_call_on_queue* and *stat_queue_periodic* tables depending on the selected configuration

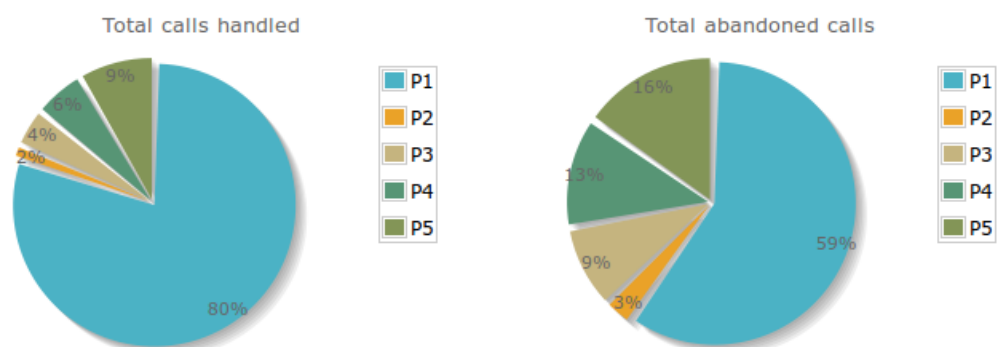
Statistic Data Table Content

stat_call_on_queue This table is used to store each call individually. Each call received on a queue generates a single entry in this table containing time related fields and a foreign key to the agent who answered the call and another on the queue on which the call was received.

It also contains the status of the call ie. answered, abandoned, full, etc.

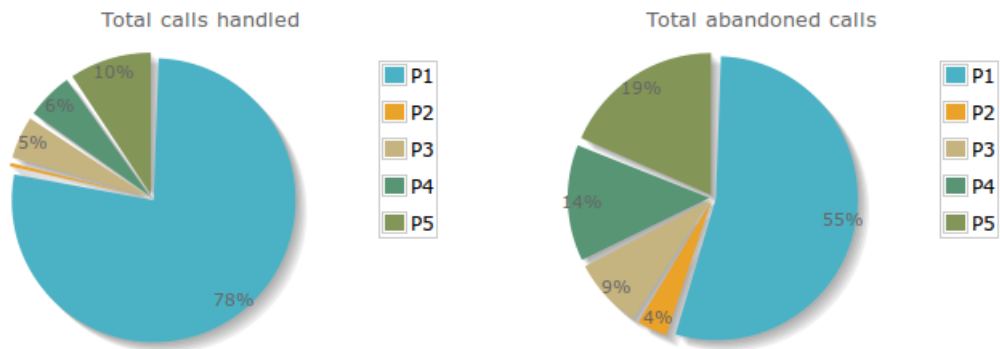
P1: 0-15s | P2: 16-20s | P3: 21-30s | P4: 31-60s | P5: 61s |

	Calls handled					Abandoned calls				
	P1	P2	P3	P4	P5	P1	P2	P3	P4	P5
27 week										
Monday	147	0	10	10	26	4	0	1	2	0
Tuesday	145	2	8	8	6	2	0	0	0	1
Wednesday	128	0	8	7	7	1	0	1	0	2
Thursday	122	2	5	11	23	2	1	0	0	1
Friday	0	0	0	0	0	0	0	0	0	0
28 week										
Monday	0	0	0	0	0	0	0	0	0	0
Tuesday	0	0	0	0	0	0	0	0	0	0
Wednesday	0	0	0	0	0	0	0	0	0	0
Thursday	0	0	0	0	0	0	0	0	0	0
Friday	0	0	0	0	0	0	0	0	0	0
29 week										
Monday	0	0	0	0	0	0	0	0	0	0
Tuesday	0	0	0	0	0	0	0	0	0	0
Wednesday	27	7	0	5	0	10	0	1	2	1
Thursday	0	0	0	0	0	0	0	0	0	0
Friday	0	0	0	0	0	0	0	0	0	0
30 week										
Monday	0	0	0	0	0	0	0	0	0	0
Tuesday	0	0	0	0	0	0	0	0	0	0
Wednesday	0	0	0	0	0	0	0	0	0	0
Thursday	0	0	0	0	0	0	0	0	0	0
Friday	0	0	0	0	0	0	0	0	0	0
31 week										
Monday	0	0	0	0	0	0	0	0	0	0
Tuesday	0	0	0	0	0	0	0	0	0	0
Wednesday	0	0	0	0	0	0	0	0	0	0
Total	569	11	31	41	62	19	1	3	4	5



P1: 0-15s | P2: 16-20s | P3: 21-30s | P4: 31-60s | P5: 61s |

	Calls handled					Abandoned calls				
	P1	P2	P3	P4	P5	P1	P2	P3	P4	P5
January	0	0	0	0	0	0	0	0	0	0
February	21	0	0	1	7	28	4	9	9	5
March	10	0	0	0	0	10	0	0	0	0
April	4	0	0	0	0	16	0	0	0	0
May	1	0	0	0	0	3	0	0	0	2
June	1570	14	119	121	214	23	2	4	13	21
July	569	11	31	41	62	19	1	3	4	5
August	0	0	0	0	0	0	0	0	0	0
September	0	0	0	0	0	0	0	0	0	0
October	0	0	0	0	0	0	0	0	0	0
November	0	0	0	0	0	0	0	0	0	0
December	0	0	0	0	0	0	0	0	0	0
Total	2028	25	140	153	257	95	7	15	24	33



Field	Values	Description
id	generated	
call-id	numeric value	This call id is also used in the CEL table and can be used to get call detail information
time	Call time	
ring-time		Ring duration time in seconds
talk-time		Talk time duration in seconds
wait-time		Wait time duration in seconds
status		See status description below
queue_id		Id of the queue, the name of the queue can be found in table <code>stat_queue</code> , using this name queue details can be found in table <code>queuefeatures</code>
agent_id		Id of the agent, the agent name can be found in table <code>stat_agent</code> , using this name agent details can be found in table <code>agentfeatures</code> using the number in the second part of the name Example : Agent/1002 is agent with number 1002 in table <code>agentfeatures</code>

Queue Call Status	Status	Description
	full	Call was not queued because queue was full, happens when the number of calls is greater than the maximum number of calls allowed to wait
	closed	Closed due to the schedule applied to the queue
	joinempty	No agents were available in the queue to take the call (follows the join empty parameter of the queue)
	leaveempty	No agents available while the call was waiting in the queue
	di-vert_ca_ratio	Call diverted because the ratio number of agent number of calls waiting configured was exceeded
	di-vert_waittime	Call diverted because the maximum expected waiting time configured was exceeded
	answered	Call was answered
	abandoned	Call hangup by the caller
	timeout	Call stayed longer than the maximum time allowed in queue parameter

stat_queue_periodic Table This table is an aggregation of the queue_log table.

This table contains counters on each queue for each given period. The granularity at the time of this writing is an hour and is not configurable. This table is then used to compute statistics for a given range of hours, days, week, month or year.

Field	Description
id	Generated id
time	time period, all counters are aggregated for an hour
answered	Number of answered calls during the period
abandoned	Number of abandoned calls during the period
total	Total calls received during the period
full	Number of calls received when queue was full
closed	Number of calls received on close
joinempty	Number of calls received no agents available
leaveempty	Number of calls diverted agents not available during the wait
di-vert_ca_ratio	Number of calls diverted due to the number of agent number versus calls waiting configured was exceeded
di-vert_waittime	Number of calls diverted because the maximum expected waiting time configured was exceeded
timeout	Number of calls diverted because the maximum time allowed in queue parameter was exceeded
queue_id	

stat_agent This table is used to match agents to an id that is different from the id in the agent configuration table. This is necessary to avoid losing statistics on a deleted agent. This also means that if an agent changes number ie. Agent/1001 to Agent/1202, the supervisor will have to take this information into account when viewing the statistics. Affecting an old number to a another agent also means that the supervisor will have to ignore entries for this given agent for the period before the number assignment to the new agent.

stat_queue This table is used to store queues in a table that is different from the queue configuration table. This is necessary to avoid losing statistics on a deleted queue. Renaming a queue is also not handled at this time.

High Availability (HA)

The HA (High Availability) solution in XiVO makes it possible to maintain basic telephony function whether your main XiVO server is running or not. When running a XiVO HA cluster, users are guaranteed to never experience a downtime of more than 5 minutes of their basic telephony service.

The HA solution in XiVO is based on a 2-nodes “master and slave” architecture. In the normal situation, both the master and slave nodes are running in parallel, the slave acting as a “hot standby”, and all the telephony services are provided by the master node. If the master fails or must be shutdown for maintenance, then the telephony devices automatically communicate with the slave node instead of the master one. Once the master is up again, the telephony devices failback to the master node. Both the failover and the failback operation are done automatically, i.e. without any user intervention, although an administrator might want to run some manual operations after failback as to, for example, make sure any voicemail messages that were left on the slave are copied back to the master.

Prerequisites

The HA in XiVO only works with telephony devices (i.e. phones) that support the notion of a primary and backup telephony server.

- Phones must be able to reach the master and the slave (take special care if master and slave are not in the same subnet)
- If firewalling, the master must be allowed to join the slave on ports 22 and 5432
- If firewalling, the slave must be allowed to join the master with an ICMP ping
- Trunk registration timeout (expiry) should be less than 300 seconds (5 minutes)
- The slave must have no provisioning plugins installed.

The HA solution is guaranteed to work correctly with *the following devices*.

Quick Summary

- You need two configured XiVO (wizard passed)
- Configure one XiVO as a master -> setup the slave address (VoIP interface)
- Restart services (xivo-service restart) on master
- Configure the other XiVO as a slave -> setup the master address (VoIP interface)
- Configure file synchronization by running the script `xivo-sync -i` on the master
- Start configuration synchronization by running the script `xivo-master-slave-db-replication <slave_ip>` on the master
- Resynchronize all your devices
- Configure the XiVO Clients

That's it, you now have a HA configuration, and every hour all the configuration done on the master will be reported to the slave.

Configuration Details

First thing to do is to *install 2 XiVO*.

Important: When you upgrade a node of your cluster, you must also upgrade the other so that they both are running the same version of XiVO. Otherwise, the replication might not work properly.

You must configure the HA in the Web interface (*Configuration → Management → High Availability* page).

You can configure the master and slave in whatever order you want.

You must also run `xivo-sync -i` on the master to setup file synchronization. Running `xivo-sync -i` will create a passwordless SSH key on the master, stored under the `/root/.ssh` directory, and will add it to the

/root/.ssh/authorized_keys file on the slave. The following directories will then be rsync'ed every hour:

- /etc/asterisk/extensions_extra.d
- /etc/xivo/asterisk
- /var/lib/asterisk/agi-bin
- /var/lib/asterisk/moh
- /var/lib/xivo/certificates
- /var/lib/xivo/sounds/acd
- /var/lib/xivo/sounds/playback

Warning: When the HA is configured, some changes will be automatically made to the configuration of XiVO.

SIP expiry value on master and slave will be automatically updated:

- min: 3 minutes
- max: 5 minutes
- default: 4 minutes

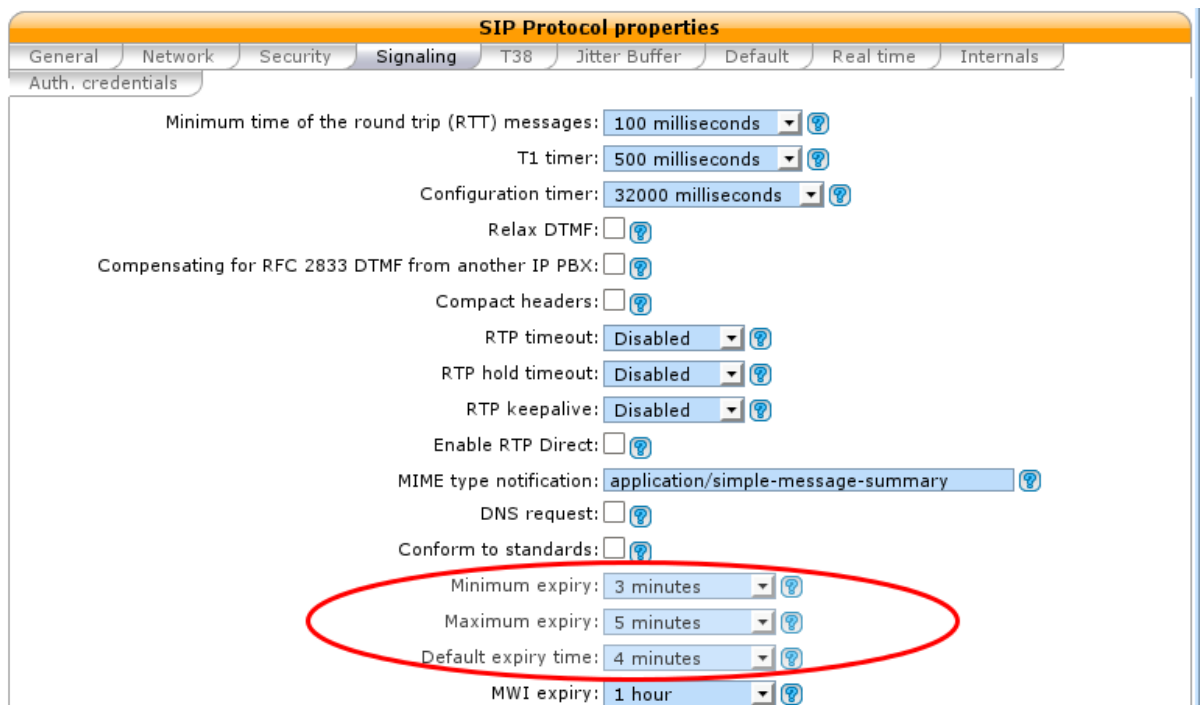


Fig. 1.99: Services → IPBX → General Settings → SIP Protocol

The provisioning server configuration will be automatically updated in order to allow phones to switch from XiVO power failure.

Warning: Do not change these values when the HA is configured, as this may cause problems. These values will be reset to blank when the HA is disabled.

Important: For the telephony devices to take the new proxy/registrar settings into account, you must *resynchronize the devices* or restart them manually.

Fig. 1.100: Configuration → Provisioning → Template Line → Edit default

Disable node Default status of HIGH AVAILABILITY (HA) is disabled:

Note: You can reset at any time by choosing a server mode (disabled)

Fig. 1.101: HA Dashboard Disabled (default state)

Important: You have to restart services (xivo-service restart) once the master node is disabled.

Master node In choosing the method `Master` you must enter the IP address **of the VoIP interface** of the slave node.

Fig. 1.102: HA Dashboard Master

Important: You have to restart all services (xivo-service restart) once the master node is configured.

Slave node In choosing the method `Slave` you must enter the IP address **of the VoIP interface** of the master node.

Replication Configuration Once master slave configuration is completed, XiVO configuration is replicated from the master node to the slave every hour (:00).

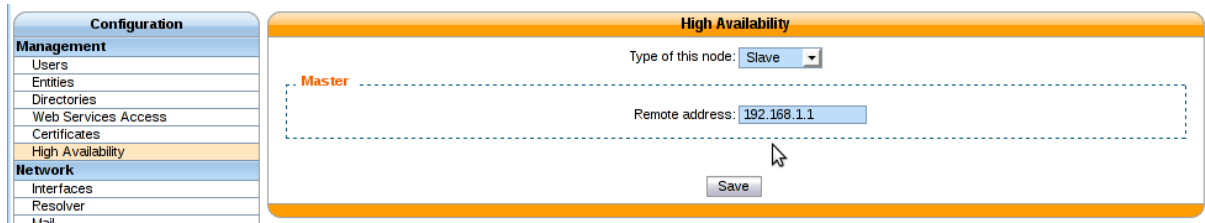


Fig. 1.103: HA Dashboard Slave

Replication can be started manually by running the replication scripts on the master:

```
xivo-master-slave-db-replication <slave_ip>
xivo-sync
```

The replication does not copy the full XiVO configuration of the master. Notably, these are excluded:

- All the network configuration (i.e. everything under the *Configuration* → *Network* section)
- All the support configuration (i.e. everything under the *Configuration* → *Support* section)
- Call logs
- Call center statistics
- Certificates
- HA settings
- Provisioning configuration
- Voicemail messages

Less importantly, these are also excluded:

- Queue logs
- CELs

XiVO Client You have to enter the master and slave address in the *Connection* tab of the XiVO Client configuration :

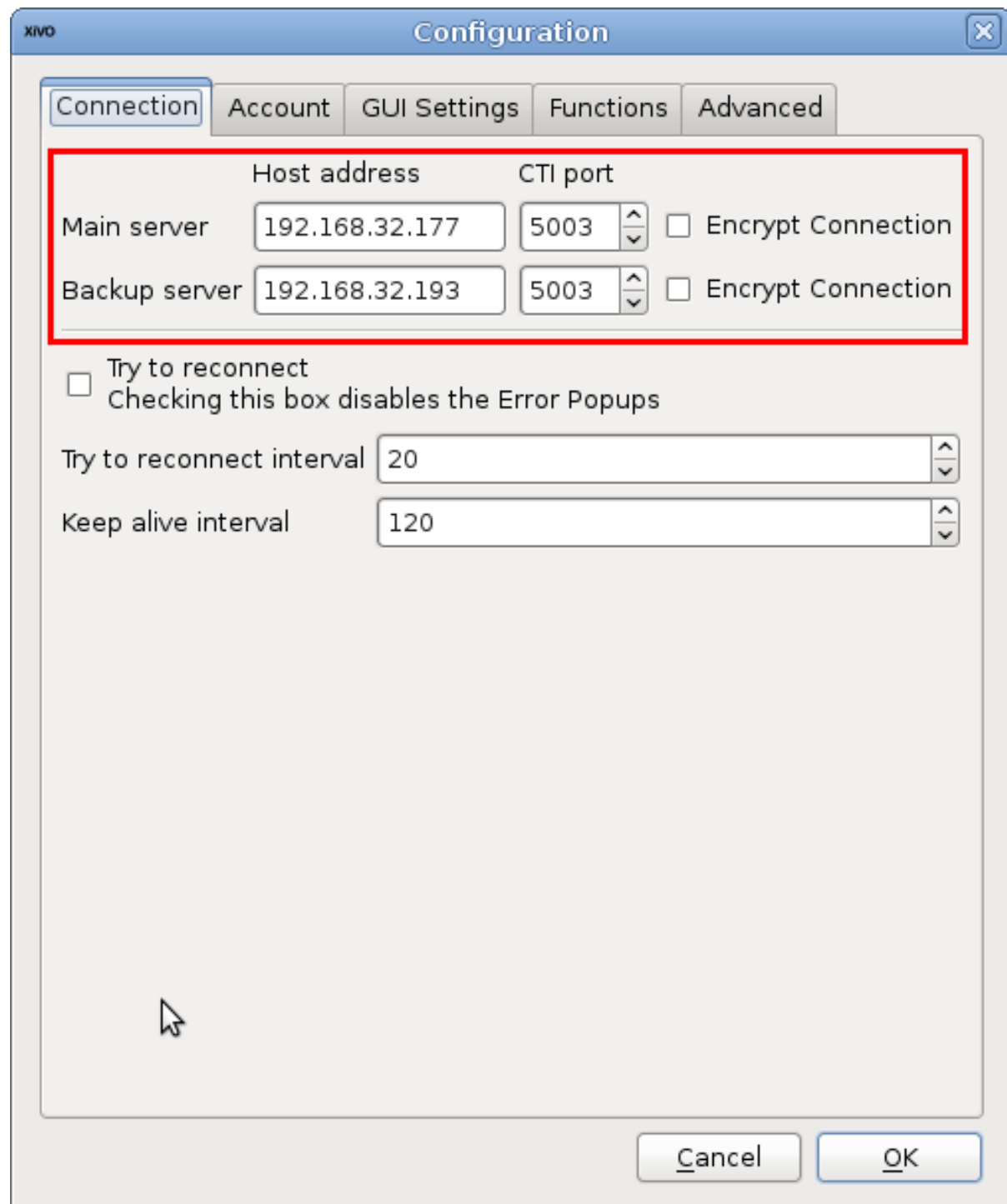
The main server is the master node and the backup server is the slave node.

When connecting the XiVO Client with the main server down, the login screen will hang for 3 seconds before connecting to the backup server.

Internals

4 scripts are used to manage services and data replication.

- `xivo-master-slave-db-replication <slave_ip>` is used on the master to replicate the master's data on the slave server. It runs on the master.
- `xivo-manage-slave-services {start,stop}` is used on the slave to start, stop monit and asterisk. The services won't be restarted after an upgrade or restart.
- `xivo-check-master-status <master_ip>` is used to check the status of the master and enable or disable services accordingly.
- `xivo-sync` is used to sync directories from master to slave.



Limitations

When the master node is down, some features are not available and some behave a bit differently. This includes:

- Call history / call records are not recorded.
- Voicemail messages saved on the master node are not available.
- Custom voicemail greetings recorded on the master node are not available.
- Phone provisioning is disabled, i.e. a phone will always keep the same configuration, even after restarting it.
- Phone remote directory is not accessible, because provisioned IP address points to the master.

Note that, on failover and on failback:

- DND, call forwards, call filtering, ..., statuses may be lost if changed recently.
- If you are connected as an agent, then you might need to reconnect as an agent when the master goes down. Since it's hard to know when the master goes down, if your CTI client disconnects and you can't reconnect it, then it's a sign the master might be down.

Additionally, only on failback:

- Voicemail messages are not copied from the slave to the master, i.e. if someone left a message on your voicemail when the master was down, you won't be able to consult it once the master is up again.
- More generally, custom sounds are not copied back. This includes recordings.

Here's the list of limitations that are more relevant on an administrator standpoint:

- The master status is up or down, there's no middle status. This means that if Asterisk is crashed the XiVO is still up and the failover will NOT happen.

Berofos Integration

Berofos Integration XiVO offers the possibility to integrate a [berofos failover switch](#) within a HA cluster.

This is useful if you have one or more ISDN lines (i.e. T1/E1 or T0 lines) that you want to use whatever the state of your XiVO HA cluster. To use a berofos within your XiVO HA installation, you need to properly configure both your berofos and your XiVOs, then the berofos will automatically switch your ISDN lines from your master node to your slave node if your master goes down, and vice-versa when it comes back up.

You can also use a Berofos failover switch to secure the ISDN provider lines when installing a XiVO in front of an existing PBX. The goal of this configuration is to mitigate the consequences of an outage of the XiVO : with this equipment the ISDN provider links could be switched to the PBX directly if the XiVO goes down.

XiVO does not offer natively the possibility to configure Berofos in this failover mode. The [Berofos Integration with PBX](#) section describes a workaround.

Installation and Configuration

Master Configuration There is nothing to be done on the master node.

Slave Configuration First, install the bntools package:

```
apt-get install bntools
```

This will make the `bnfos` command available.

You can then connect your berofos to your network and power it on. By default, the berofos will try to get an IP address via DHCP. If it is not able to get such address from a DHCP server, it will take the 192.168.0.2/24 IP address.

Note: The DHCP server on XiVO does not offer IP addresses to berofos devices by default.

Next step is to create the `/etc/bnfos.conf` file via the following command:

```
bnfos --scan -x
```

If no berofos device is detected using this last command, you'll have to explicitly specify the IP address of the berofos via the `-h` option:

```
bnfos --scan -x -h <berofos ip>
```

At this stage, your `/etc/bnfos.conf` file should contains something like this:

```
[fos1]
mac = 00:19:32:00:12:1D
host = 10.34.1.50
#login = <user>:<password>
```

It is advised to configure your berofos with a static IP address. You first need to put your berofos into *flash mode* :

- press and hold the black button next to the power button,
- power on your berofos,
- release the black button when the red LEDs of port D start blinking.

Then, you can issue the following command, by first replacing the network configuration with your one:

```
bnfos --netconf -f fos1 -i 10.34.1.20 -n 255.255.255.0 -g 10.34.1.1 -d 0
```

Note:

- `-i` is the IP address
 - `-n` is the netmask
 - `-g` is the gateway
 - `-d 0` is to disable DHCP
-

You can then update your berofos firmware to version 1.53:

```
wget http://www.beronet.com/downloads/berofos/bnfos_v153.bin
bnfos --flash bnfos_v153.bin -f fos1
```

Once this is done, you'll have to reboot your berofos in operationnal mode (that is in normal mode).

Then you must rewrite the `/etc/bnfos.conf` (mainly if you changed the IP address):

```
bnfos --scan -x -h <berofos ip>
```

Now that your berofos has proper network configuration and an up to date firmware, you might want to set a password on your berofos:

```
bnfos --set apwd=<password> -f fos1
bnfos --set pwd=1 -f fos1
```

You must then edit the `/etc/bnfos.conf` and replace the login line to something like:

```
login = admin:<password>
```

Next, configure your berofos for it to work correctly with the XiVO HA:

```
bnfos --set wdog=0 -f fos1
bnfos --set wdogdef=0 -f fos1
bnfos --set scenario=0 -f fos1
bnfos --set mode=1 -f fos1
bnfos --set modedef=1 -f fos1
```

This, among other things, disable the watchdog. The switching from one relay mode to the other will be done by the XiVO slave node once it detects the master node is down, and vice-versa.

Finally, you can make sure everything works fine by running the xivo-berofos command:

```
xivo-berofos master
```

The green LEDs on your berofos should be lighted on ports A and B.

Connection

Two XiVOs Here's how to connect the ISDN lines between your berofos with:

- two XiVOs in high availability

In this configuration you can protect **up to 4** ISDN lines. If more than 4 ISDN lines to protect, you must set up a *Multiple berofos* configuration.

Here's an example with 4 ISDN lines coming from your telephony provider:

```

ISDN lines (provider)
  | | | |
  | | | |
+-----+
|      A      B      C      D      |
| 1|2|3|4    1|2|3|4    1|2|3|4    1|2|3|4 |
+-----+
                | | | |
                | | | |
                +-----+
                | xivo-1 |
                +-----+
                +-----+
                | xivo-2 |
                +-----+

```

Two XiVOs and one PBX Here's how to connect your berofos with:

- two XiVOs in high availability,
- one PBX.

In this configuration you can protect **up to two 2** ISDN lines. If more than 2 ISDN lines to protect, you must set up a *Multiple berofos* configuration.

Logical view:

```

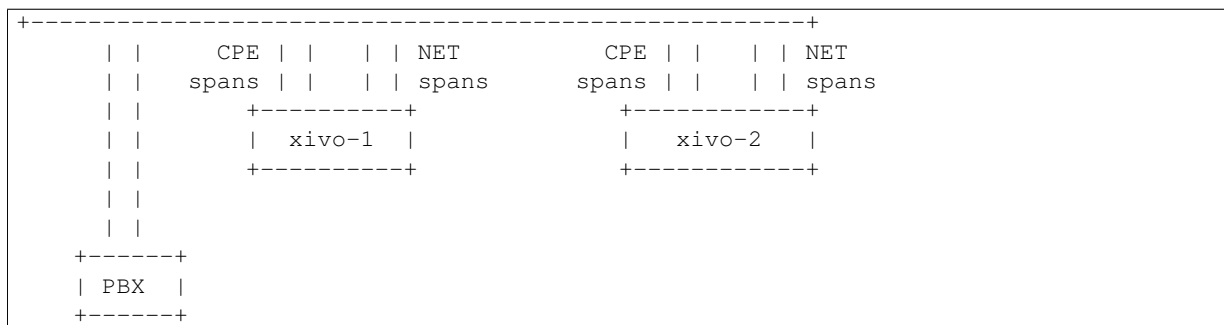
+-----+
-- Provider ----| xivo-1 | -- ISDN Interconnection --| PBX | -- Phones
+-----+
               | xivo-2 |
+-----+

```

This example shows the case where there are 2 ISDN lines coming from your telephony provider:

ISDN lines (provider)															
1		2		3		4		5							

A		B		C		D									
1	2	3	4	1	2	3	4	1	2	3	4	1	2	3	4



One XiVO and one PBX This case is not currently supported. You'll find a workaround in the *Berofos Integration with PBX* section.

Multiple berofos It's possible to use more than 1 berofos with XiVO.

For each supplementary berofos you want to use, you must first configure it properly like you did for the first one. The only difference is that you need to add a berofos declaration to the `/etc/bnfos.conf` file instead of creating/overwriting the file. Here's an example of a valid config file for 2 berofos:

```
[fos1]
mac = 00:19:32:00:12:1D
host = 10.100.0.201
login = admin:foobar

[fos2]
mac = 00:11:22:33:44:55
host = 10.100.0.202
login = admin:barfoo
```

Warning: berofos name must follow the pattern `fosX` where X is a number starting with 1, then 2, etc. The `bnfos` tool won't work properly if it's not the case.

Operation When your XiVO switch the relay mode of your berofos, it logs the event in the `/var/log/syslog` file.

Default mode Note that when the berofos is off, the A and D ports are connected together. This behavior is not customizable.

Uninstallation It is important to remove the `/etc/bnfos.conf` file on the slave node when you don't want to use anymore your berofos with your XiVOs.

Reset the Berofos You can reset the berofos configuration :

1. Power on the berofos,
2. When red and green LEDs are still lit, press & hold the black button,
3. Release it when the red LEDs of the D port start blinking fast
4. Reboot the beronet, it should have lost its configuration.

External links

- [berofos user manual](#)

Troubleshooting

When replicating the database between master and slave, if you encounter problems related to the system locale, see *PostgreSQL localization errors*.

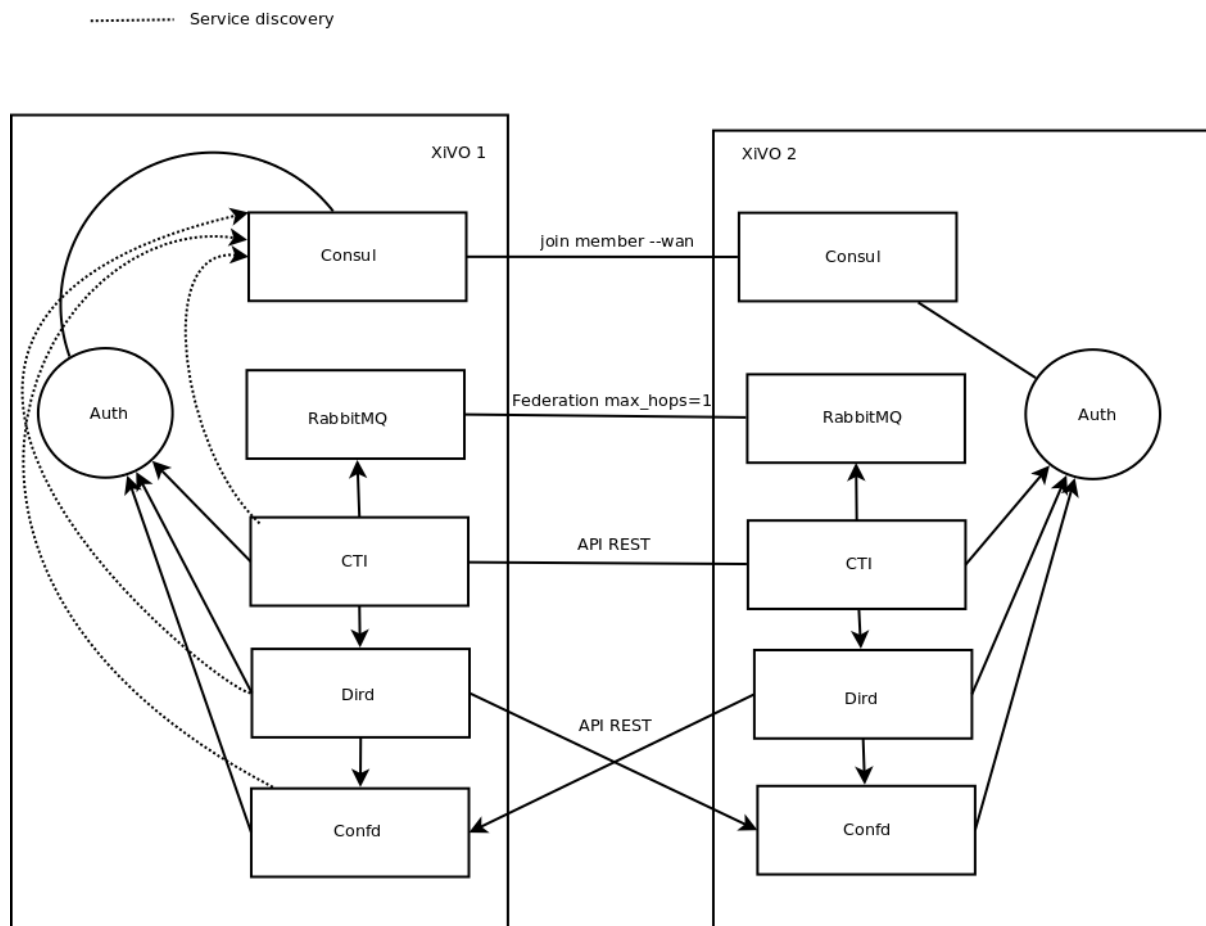
Scalability and Distributed Systems

This section gathers configuration that are possible using XiVO to feature rich scalable communication systems.

Contact and Presence Sharing

XiVO allow the administrator to share presence and statuses between multiple installations. For example, an enterprise could have a XiVO in each office and still be able to search, contact and view the statuses of colleagues in other offices.

This page will describe the steps that are required to configure such use case.



Prerequisite

1. All XiVO that you interconnect should be on the same version
2. This configuration is only possible with XiVO 15.19 and above
3. All ports necessary for communication should be open *Network*

Warning: If you are cloning a virtual machine or copy the database, the UUID of the two XiVO will be the same, you must regenerate them in the *infos* table of the *asterisk* database and restart all services. You must also remove all consul data that included the old UUID.

Warning: Telephony will be interrupted during the configuration period.

Warning: The configuration must be applied to each XiVO you want to interconnect. For example, if 6 different XiVO are to be connected, the configuration for all other XiVO should be added. This does not apply to the message bus which can use a ring policy, each XiVO talking to its two neighbours.

Warning: You should use your firewall to restrict access to the HTTP ports of consul and xivo-ctid, because they don't have any authentication mechanism enabled.

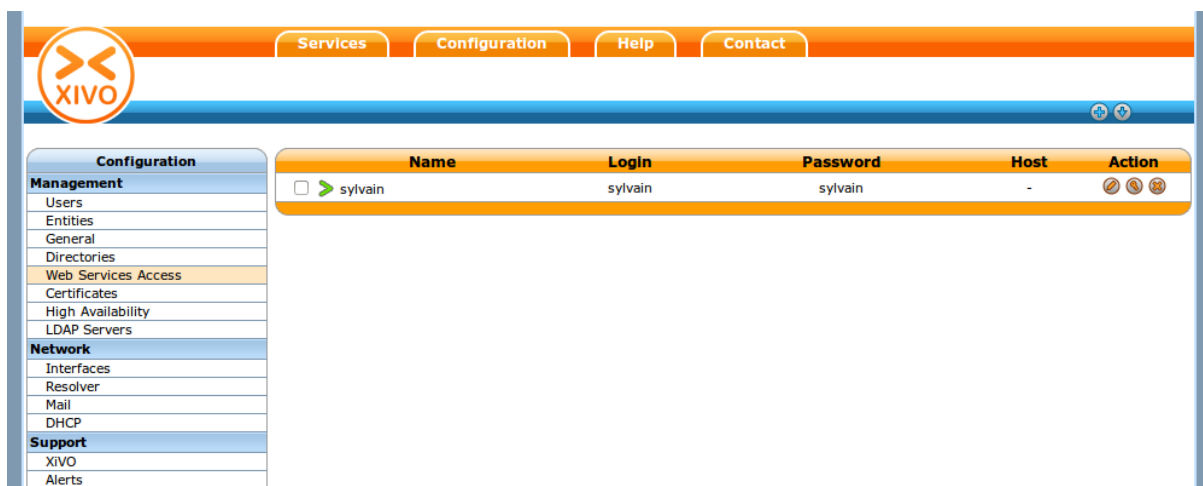
Note: In an architecture with a lot of XiVO, we recommend that you centralize some services, such as xivo-dird, to make your life easier. Don't forget redundancy. This applies also to RabbitMQ and Consul. In this case, the configuration will have to be done entirely manually in YAML config files.

For this procedure, the following name and IP addresses will be used:

- XiVO 1: 192.168.1.124
- XiVO 2: 192.168.1.125

Add a Web Service User The first thing is to make XiVO accept remote connections to your internal users directory. For this, you must create a *Web service access* by authorizing either an IP address or a login/password.

This can be done in *Configuration* → *Management* → *Web Services Access*



Configuring the directories

Add New Directory Sources for Remote XiVO For each remote XiVO a new directory has to be created in *Configuration* → *Management* → *Directories*

Note: We recommend doing a working configuration without certificate verification first. Once you get it working, enable certificate verification.

Access Web Services > Add

Name:

Login:

Password:

Host:

Description:

Services
Configuration
Help
Contact

Configuration

Management

Users

Entities

General

Directories

Web Services Access

Certificates

High Availability

LDAP Servers

Network

Interfaces

	Name	Uri	Action
<input type="checkbox"/>	phonebook	http://localhost/service/ipbx/json.php/private/pbx_services/phonebook	
<input type="checkbox"/>	XIVO	http://localhost:9487	
<input type="checkbox"/>	xivo-dev-2	https://192.168.1.125:9486	
<input type="checkbox"/>	xivo-dev-3	https://192.168.1.126:9486	

Directories Servers > Edit

Directory name:

Type:

URI:

XiVO directory

Username:

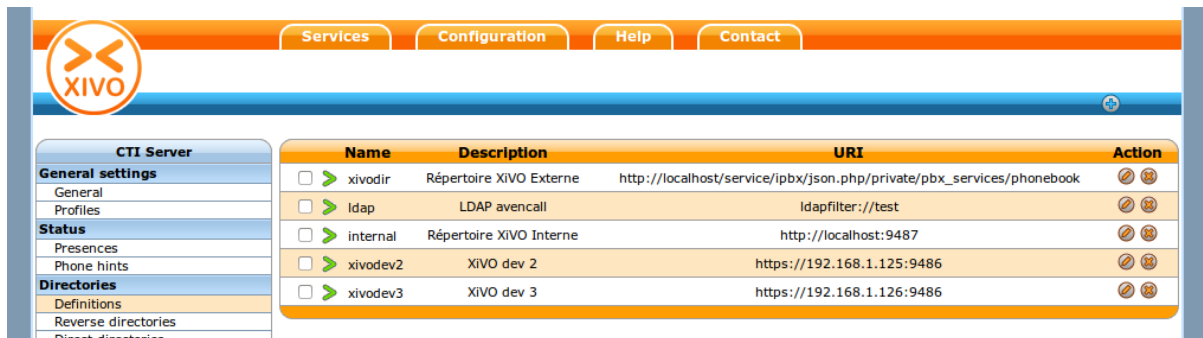
Password:

Verify certificate:

Custom CA certificate:

Description

Add a Directory Definition for Each New Directories To add a new directory definition, go to *Services* → *CTI Server* → *Directories* → *Definitions*



In each directory definition, add the fields to match the configured *Display filters*

Update directories

Name:
URI:
Delimiter:
Direct match:
Match reverse directories:

Mapped fields:

Fieldname	Value
<input type="text" value="firstname"/>	<input type="text" value="{firstname}"/>
<input type="text" value="lastname"/>	<input type="text" value="{lastname}"/>
<input type="text" value="phone"/>	<input type="text" value="{exten}"/>
<input type="text" value="name"/>	<input type="text" value="{firstname} {lastname}"/>
<input type="text" value="directory"/>	<input type="text" value="XiVO dev 2"/>

Description

XiVO dev 2

You need to restart the Dird server to apply changes.

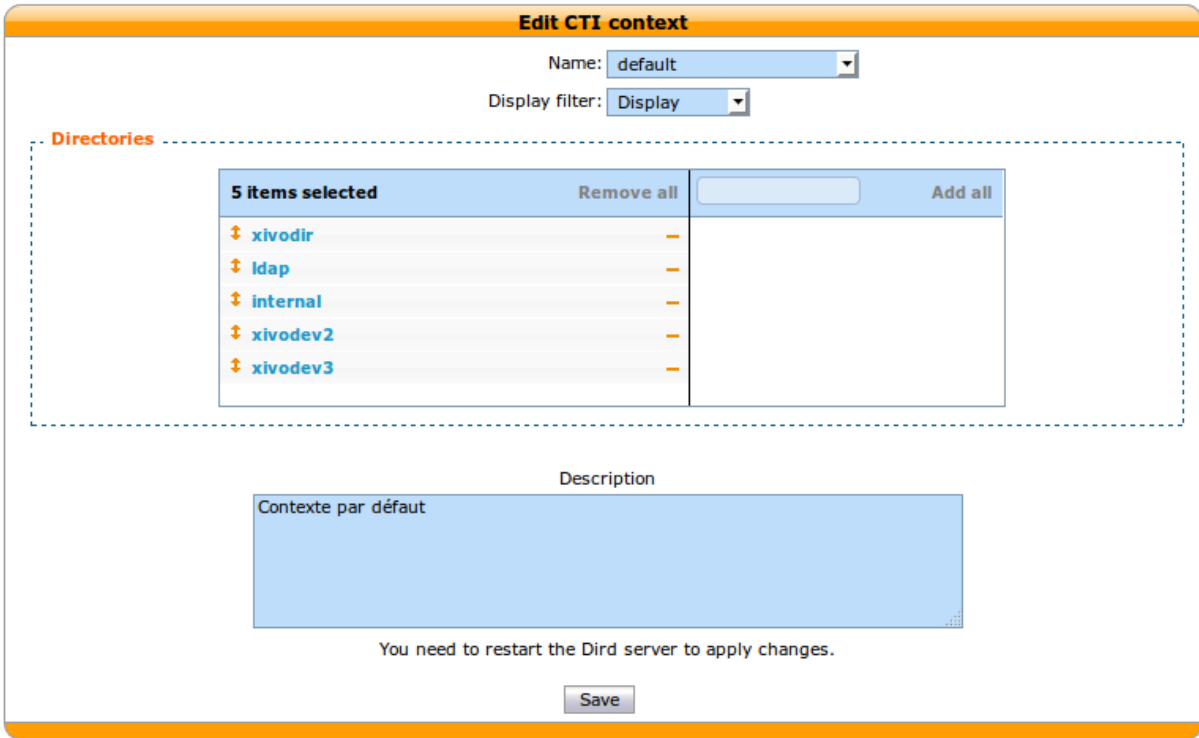
Save

Add the New Definitions to Your Dird Profiles At the moment of this writing xivo-dird profiles are mapped directly to the user's profile. For each internal context where you want to be able to see user's from other XiVO, add the new directory definitions in *Services* → *CTI Server* → *Directories* → *Direct directories*.

Restart xivo-dird To apply the new directory configuration, you can either restart from:

- *Services* → *IPBX*
- on the command line `service xivo-dird restart`

Check that the Configuration is Working At this point, you should be able to search for users on other XiVO from the *People Xlet*.



Configuring RabbitMQ

Create a RabbitMQ user

```
rabbitmqctl add_user xivo xivo
rabbitmqctl set_user_tags xivo administrator
rabbitmqctl set_permissions -p / xivo ".*" ".*" ".*"
rabbitmq-plugins enable rabbitmq_federation
```

Restart RabbitMQ

```
service rabbitmq-server restart
```

Setup Message Federation

```
rabbitmqctl set_parameter federation-upstream xivo-dev-2 '{"uri":"amqp://xivo:xivo@192.168.1.125"}'
rabbitmqctl set_policy federate-xivo 'xivo' '{"federation-upstream-set":"all"}' --priority 1 --ap
```

Configure xivo-ctid

Create a Custom Configuration File Create a configuration file for xivo-ctid, e.g `/etc/xivo-ctid/conf.d/interconnection.yml`

```
rest_api:
  http:
    listen: 0.0.0.0
service_discovery:
  advertise_address: auto
  advertise_address_interface: eth0 # Interface bearing the IP address of this XiVO, reachable f
```

Restart xivo-ctid

```
service xivo-ctid restart
```

Check That Service Discovery is Working

```
apt-get install consul-cli
```

```
consul-cli agent-services --ssl --ssl-verify=false
```

The output should include a service names *xivo-ctid* with an address that is reachable from other XiVO.

```
{"consul": {"ID": "consul",
  "Service": "consul",
  "Tags": [],
  "Port": 8300,
  "Address": ""},
  "e546a652-e290-47e2-8519-ec3642daa6e6": {"ID": "e546a652-e290-47e2-8519-ec3642daa6e6",
    "Service": "xivo-ctid",
    "Tags": ["xivo-ctid",
      "607796fc-24e2-4e26-8009-cbb48a205512"],
    "Port": 9495,
    "Address": "192.168.1.124"}}
```

Configure Consul

Stop XiVO

```
xivo-service stop
```

Remove All Consul Data

```
rm -rf /var/lib/consul/raft/
rm -rf /var/lib/consul/serf/
rm -rf /var/lib/consul/services/
rm -rf /var/lib/consul/tmp/
rm -rf /var/lib/consul/checks/
```

Configure Consul to be Reachable from Other XiVO Add a new configuration file `/etc/consul/xivo/interconnection.json` with the following content where `advertise_addr` is reachable from other XiVO.

```
{
  "client_addr": "0.0.0.0",
  "bind_addr": "0.0.0.0",
  "advertise_addr": "192.168.1.124" // The IP address of this XiVO, reachable from outside
}
```

Check that the Configuration is Valid

```
consul configtest --config-dir /etc/consul/xivo/
```

No output means that the configuration is valid.

Start Consul

```
service consul start
```

Start XiVO

```
xivo-service start
```

Join the Consul Cluster Join another member of the Consul cluster. Only one join is required as members will be propagated.

```
consul join -wan 192.168.1.125
```

Check that Consul Sees other Consul List other members of the cluster with the following command

```
consul members -wan
```

Check consul logs for problems

```
consul monitor
```

Check That Everything is Working There is no further configuration needed, you should now be able to connect your XiVO Client and search contacts from the People Xlet. When looking up contacts of another XiVO, you should see their phone status, their user availability, and agent status dynamically.

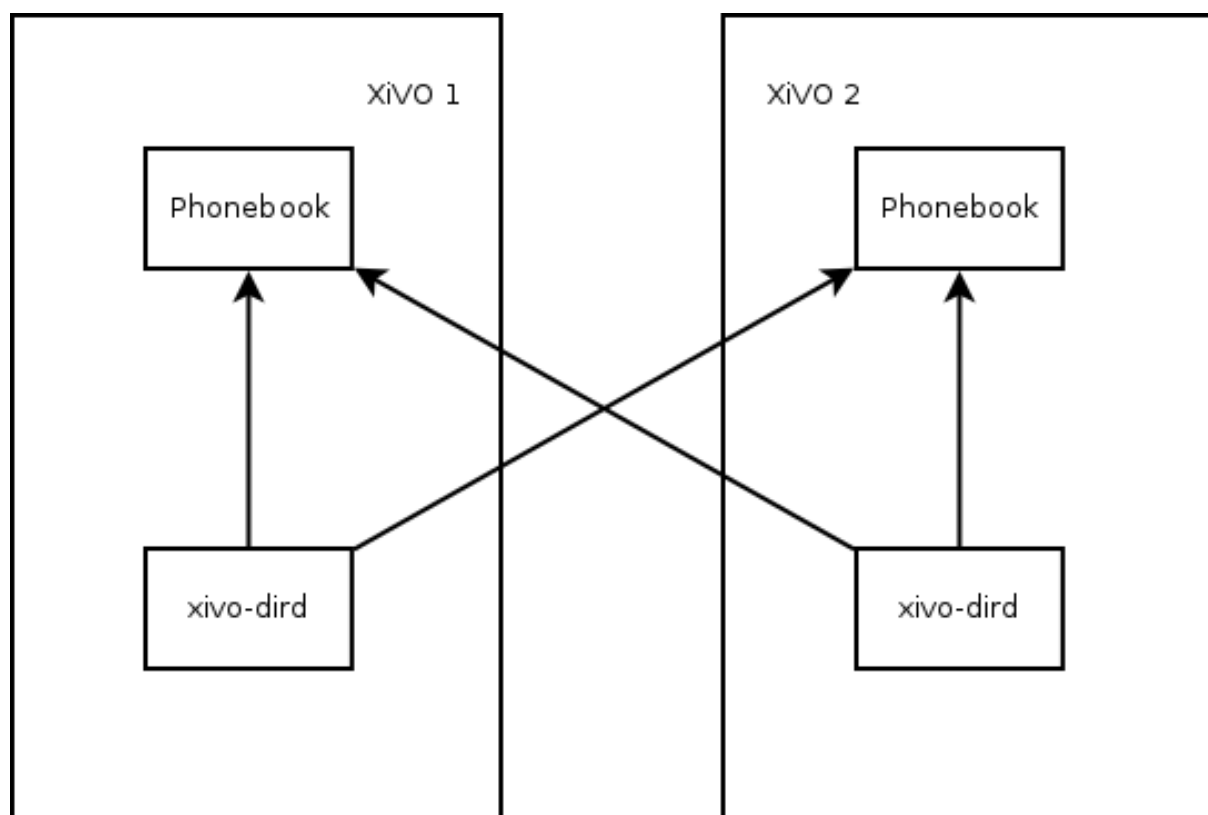
Troubleshooting Chances are that everything won't work the first time, here are some interesting commands to help you debug the problem.

```
tail -f /var/log/xivo-dird.log
tail -f /var/log/xivo-ctid.log
tail -f /var/log/xivo-confd.log
consul monitor
consul members -wan
consul-cli agent-services --ssl --ssl-verify=false
rabbitmqctl eval 'rabbit_federation_status:status().'
```

What's next? Once you get this part working, check out *Phonebook Sharing*.

Phonebook Sharing

Sharing phonebooks allows users of different XiVO servers to access the contacts in the *phonebooks* of the other XiVO servers.



This procedure follows the *Contact and Presence Sharing* (but it's not mandatory), so we will use the same conventions.

Open Phonebook Access On each XiVO, you must have a *Web Services User* that authorizes access from another host (not by login/password). The phonebook access does not support login/password authorization.

This Web Services user will allow other XiVO servers to access the phonebook of this XiVO.

Configuring the Directories For each remote XiVO a new phonebook has to be created in *Configuration → Management → Directories*

Note that the URL of the directory must contain `restricted`, not `private`, e.g:

Name	Uri	Action
<input type="checkbox"/> > phonebook	http://localhost/service/ipbx/json.php/private/pbx_services/phonebook	
<input checked="" type="checkbox"/> > phonebook-2	http://192.168.1.125/service/ipbx/json.php/protected/pbx_services/phonebook	
<input type="checkbox"/> > xivo	http://localhost:9487	
<input type="checkbox"/> > xivo-2	https://192.168.1.125:9486	

```
http://192.168.1.125/service/ipbx/json.php/restricted/pbx_services/phonebook
```

Add a Directory Definition for Each Phonebook To add a new directory definition, go to *Services* → *CTI Server* → *Directories* → *Definitions*

In each directory definition, add the fields to match the other phonebooks:

Add the New Definitions to Your Users We just defined the directories, now let's use them:

Check That Everything is Working There is no further configuration needed, you should now be able to connect your XiVO Client and search phonebook contacts from the People Xlet.

Troubleshooting Chances are that everything won't work the first time, here are some interesting commands to help you debug the problem.

```
tail -f /var/log/xivo-dird.log
tail -f /var/log/nginx/xivo.access.log
```

API and SDK

Message Bus

The message bus is used to receive events from XiVO. It is provided by an [AMQP 0-9-1](#) broker (namely, [RabbitMQ](#)) that is integrated in XiVO.

CTI Server

General settings

General

Profiles

Status

Presences

Phone hints

Directories

Definitions

Reverse directories

Name	Description	URI	Action
<input type="checkbox"/> > internal		http://localhost:9487	
<input type="checkbox"/> > xivodir		http://localhost/service/ipbx/json.php/private/pbx_services/phonebook	
<input type="checkbox"/> > phonebook2		http://192.168.1.125/service/ipbx/json.php/protected/pbx_services/phonebook	

CTI Server

General settings

General

Profiles

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Directories

Definitions

Reverse directories

Direct directories

Display filters

Sheets

Models

Events

Update directories

Name:

URI: ▼

Delimiter:

Direct match:

Match reverse directories:

Mapped fields:

Fieldname	Value
firstname	{phonebook.firstname}
lastname	{phonebook.lastname}
fullname	{phonebook.fullname}
name	{phonebook.fullname}
display_name	{phonebook.displayname}
phone	{phonebooknumber.office.number}
phone_mobile	{phonebooknumber.mobile.number}
phone_home	{phonebooknumber.home.number}
phone_other	{phonebooknumber.other.number}
company	{phonebook.society}
email	{phonebook.email}
reverse	{phonebook.fullname}

Description

Répertoire XiVO Externe

You need to restart the Dird server to apply changes.

CTI Server

General settings

General

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Directories

Definitions

Reverse directories

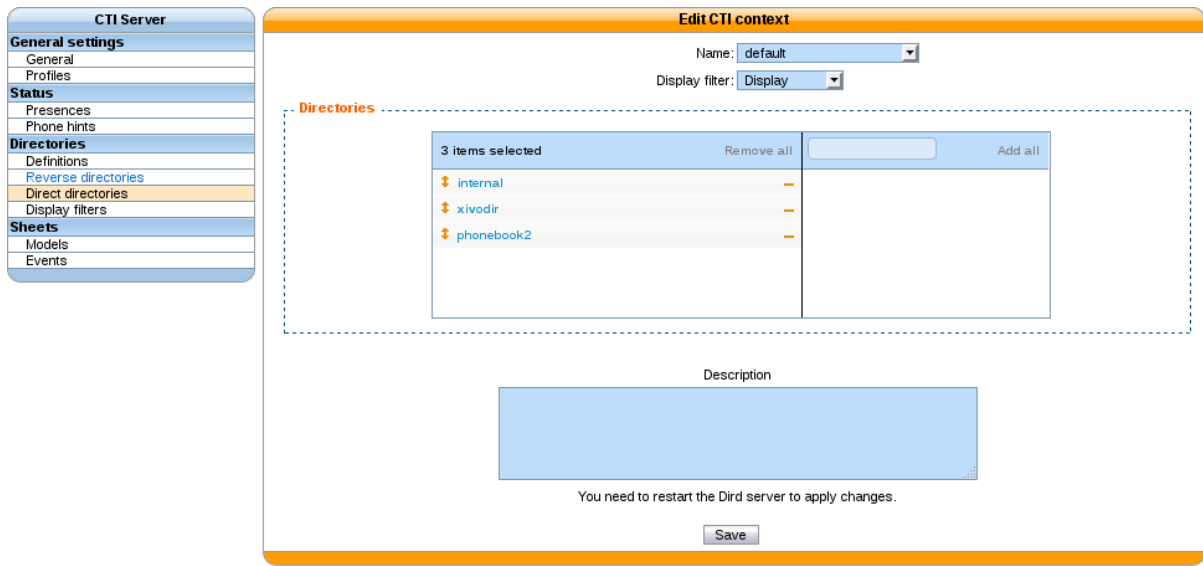
Direct directories

Display filters

Sheets

Context

Context	Description	Action
<input type="checkbox"/> > _switchboard_directory		
<input type="checkbox"/> > default	Contexte par défaut	



Warning: Interaction with the bus is presently experimental and some things might change in the next XiVO versions.

Usage At the moment, the AMQP broker only listen on the 127.0.0.1 address. This means that if you want to connect to the AMQP broker from a distant machine, you must modify the RabbitMQ server configuration, which is not yet an officially supported operation. All events are sent to the *xivo* exchange.

Otherwise, the default connection information is:

- Virtual host: /
- User name: guest
- User password: guest
- Port: 5672
- Exchange name: xivo
- Exchange type: topic

Example Here's an example of a simple client, in python, listening for the *call_form_result* CTI events:

```
import kombu

from kombu.mixins import ConsumerMixin

EXCHANGE = kombu.Exchange('xivo', type='topic')
ROUTING_KEY = 'call_form_result'

class C(ConsumerMixin):

    def __init__(self, connection):
        self.connection = connection

    def get_consumers(self, Consumer, channel):
        return [Consumer(kombu.Queue(exchange=EXCHANGE, routing_key=ROUTING_KEY),
                               callbacks=[self.on_message])]

    def on_message(self, body, message):
        print('Received:', body)
```

```

        message.ack()

def main():
    with kombu.Connection('amqp://guest:guest@localhost:5672//') as conn:
        try:
            C(conn).run()
        except KeyboardInterrupt:
            return

main()

```

If you are new to AMQP, you might want to look at the [RabbitMQ tutorial](#).

Notes Things to be aware when writing a client/consumer:

- The published messages are not persistent. When the AMQP broker stops, the messages that are still in queues will be lost.

Changelog

16.08

- The *call_held* bus message has been added.
- The *call_resumed* bus message has been added.
- The *user_status_update* bus message now uses the user's UUID instead of the user's ID.

16.07

- The *user_created* bus message has been added.
- The *user_edited* bus message has been added.
- The *user_deleted* bus message has been added.

15.20

- The *chat_message_event* bus message has been added.

15.17

- The *service_registered_event* and *service_deregistered_event* bus messages have been added.

Events Events that are sent to the bus use a JSON serialization format with the content-type *application/json*. For example, the CTI *call_form_result* event looks like this:

```

{"name": "call_form_result",
 "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
 "data": {...}}

```

All events have the same basic structure, namely, a JSON object with 4 keys:

name A string representing the name of the event. Each event type has a unique name.

required_acl (optional) Either a string or null. Currently used by xivo-websocketd to determine if a client can receive the event or not. See the *Events Access Control* section for more information.

origin_uuid The uuid to identify the message producer.

data The data specific part of the event. This is documented on a per event type; if not this is assumed to be null.

AMI events All AMI events are broadcasted on the bus.

- routing key: ami.<event name>
- event specific data: a dictionary with the content of the AMI event

Example event with binding key QueueMemberStatus:

```
{
  "name": "QueueMemberStatus",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "Status": "1",
    "Penalty": "0",
    "CallsTaken": "0",
    "Skills": "",
    "MemberName": "sip/m3ylhs",
    "Queue": "petak",
    "LastCall": "0",
    "Membership": "static",
    "Location": "sip/m3ylhs",
    "Privilege": "agent,all",
    "Paused": "0",
    "StateInterface": "sip/m4ylhs"
  }
}
```

call_form_result The call_form_result event is sent when a *custom call form* is submitted by a CTI client.

- routing key: call_form_result
- event specific data: a dictionary with 2 keys:
 - user_id: an integer corresponding to the user ID of the client who saved the call form
 - variables: a dictionary holding the content of the form

Example:

```
{
  "name": "call_form_result",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "user_id": 40,
    "variables": {
      "firstname": "John",
      "lastname": "Doe"
    }
  }
}
```

agent_status_update The agent_status_update is sent when an agent is logged in or logged out.

- routing key: status.agent
- required ACL: events.statuses.agents
- event specific data: a dictionary with 3 keys:
 - agent_id: an integer corresponding to the agent ID of the agent who's status changed
 - status: a string identifying the status
 - xivo_id: the uuid of the xivo

Example:

```
{
  "name": "agent_status_update",
  "required_acl": "events.statuses.agents",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "agent_id": 42,
    "xivo_id": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
    "status": "logged_in"
  }
}
```

call_created, call_updated, call_ended The events `call_created`, `call_updated`, `call_ended` are sent when a call handled by `xivo-ctid-ng` is received, connected or hung up.

- routing key: `calls.call.created`, `calls.call.updated`, `calls.call.ended`
- required ACL: `events.calls.<user_uuid>`
- event specific data: a dictionary with the same fields as the REST API model of Call (See <http://api.xivo.io>, section `xivo-ctid-ng`)

Example:

```
{
  "name": "call_created",
  "required_acl": "events.calls.2e752722-0864-4665-887d-a78a024cf7c7",
  "origin_uuid": "08c56466-8f29-45c7-9856-92bflba89b82",
  "data": {
    "bridges": [],
    "call_id": "1455123422.8",
    "caller_id_name": "Some One",
    "caller_id_number": "1001",
    "creation_time": "2016-02-10T11:57:02.592-0500",
    "status": "Ring",
    "talking_to": {},
    "user_uuid": "2e752722-0864-4665-887d-a78a024cf7c7"
  }
}
```

call_held This message is sent when a call is placed on hold

- routing key: `calls.hold.created`
- event specific data:
 - `call_id`: The asterisk channel unique ID

Example:

```
{ "name": "call_held",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": { "call_id": "1465572129.31" } }
```

call_resumed This message is sent when a call is resumed from hold

- routing key: `calls.hold.deleted`
- event specific data:
 - `call_id`: The asterisk channel unique ID

Example:

```
{
  "name": "call_resumed",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": { "call_id": "1465572129.31" }}
```

chat_message_event This message is used to send a chat message to a user

- routing key: chat.message.<xivo-uuid>.<user_id>
- event specific data:
 - alias: The nickname of the chatter
 - to: The destination's XiVO UUID and user UUID
 - from: The chatter's XiVO UUID and user UUID
 - msg: The message

Example:

```
{
  "name": "chat_message_event",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "alias": "Alice"
    "to": ["ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3", "fcb36731-c50a-453e-92c7-571297d41616"],
    "from": ["ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3", "4f2e2249-ae2b-4bc2-b5fc-ad42ee01ddaf"],
    "msg": "Hi!"
  }
}
```

endpoint_status_update The endpoint_status_update is sent when an end point status changes. This information is based on asterisk hints.

- routing key: status.endpoint
- required ACL: events.statuses.endpoints
- event specific data: a dictionary with 3 keys
 - xivo_id: the uuid of the xivo
 - endpoint_id: an integer corresponding to the endpoint ID
 - status: an integer corresponding to the asterisk device state

Example:

```
{
  "name": "endpoint_status_update",
  "required_acl": "events.statuses.endpoints",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "endpoint_id": 67,
    "xivo_id": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
    "status": 0
  }
}
```

user_created The *user_created* event is published when a new user is created.

- routing key: *config.user.created*
- event specific data: a dictionary with 2 keys
 - id: the ID of the created user

- uuid: the UUID of the created user

Example:

```
{
  "name": "user_created",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "id": 42,
    "uuid": "8e58d2a7-cfed-4c2e-ac72-14e0b5c26dc2"
  }
}
```

user_deleted The *user_deleted* event is published when a user is deleted.

- routing key: *config.user.deleted*
- event specific data: a dictionary with 2 keys
 - id: the ID of the deleted user
 - uuid: the UUID of the deleted user

Example:

```
{
  "name": "user_deleted",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "id": 42,
    "uuid": "8e58d2a7-cfed-4c2e-ac72-14e0b5c26dc2"
  }
}
```

user_edited The *user_edited* event is published when a user is modified.

- routing key: *config.user.edited*
- event specific data: a dictionary with 2 keys
 - id: the ID of the modified user
 - uuid: the UUID of the modified user

Example:

```
{
  "name": "user_edited",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "id": 42,
    "uuid": "8e58d2a7-cfed-4c2e-ac72-14e0b5c26dc2"
  }
}
```

user_status_update The *user_status_update* is sent when a user changes his CTI presence using the XiVO client.

- routing key: *status.user*
- required ACL: *events.statuses.users*
- event specific data: a dictionary with 3 keys
 - *xivo_id*: the uuid of the xivo

- user_uuid: the user's UUID
- status: a string identifying the status

Example:

```
{
  "name": "user_status_update",
  "required_acl": "events.statuses.users",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "user_uuid": "8e58d2a7-cfed-4c2e-ac72-14e0b5c26dc2",
    "xivo_id": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
    "status": "busy"
  }
}
```

users_forwards_<forward_name>_updated The users_forwards_<forward_name>_updated is sent when a user changes his forward using REST API.

- forward_name:
 - busy
 - noanswer
 - unconditional
- routing key: config.users.<user_uuid>.forwards.<forward_name>.updated
- required ACL: events.config.users.<user_uuid>.forwards.<forward_name>.updated
- event specific data: a dictionary with 3 keys
 - user_uuid: the user uuid
 - enabled: the state of the forward
 - destination: the destination of the forward

Example:

```
{
  "name": "users_forwards_busy_updated",
  "required_acl": "events.config.users.a1223fe6-bff8-4fb6-a982-f9157dea5094.forwards.busy.updated",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "user_uuid": "a1223fe6-bff8-4fb6-a982-f9157dea5094",
    "enabled": true
    "destination": "1234"
  }
}
```

users_services_<service_name>_updated The users_services_<service_name>_updated is sent when a user changes his service using REST API.

- service_name:
 - dnd
 - incallfilter
- routing key: config.users.<user_uuid>.services.<service_name>.updated
- required ACL: events.config.users.<user_uuid>.services.<service_name>.updated
- event specific data: a dictionary with 2 keys
 - user_uuid: the user uuid

- enabled: the state of the service

Example:

```
{
  "name": "users_services_dnd_updated",
  "required_acl": "events.config.users.a1223fe6-bff8-4fb6-a982-f9157dea5094.services.dnd.updated",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "user_uuid": "a1223fe6-bff8-4fb6-a982-f9157dea5094",
    "enabled": true
  }
}
```

service_registered_event The service_registered_event is sent when a service is started.

- routing key: service.registered.<service_name>
- event specific data: a dictionary with 5 keys
 - service_name: The name of the started service
 - service_id: The consul ID of the started service
 - address: The advertised address of the started service
 - port: The advertised port of the started service
 - tags: The advertised Consul tags of the started service

Example:

```
{
  "name": "service_registered_event",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "service_name": "xivo-ctid",
    "service_id": "8e58d2a7-cfed-4c2e-ac72-14e0b5c26dc2",
    "address": "192.168.1.42",
    "port": 9495,
    "tags": ["xivo-ctid", "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3", "Québec"]
  }
}
```

service_deregistered_event The service_deregistered_event is sent when a service is stopped.

- routing key: service.deregistered.<service_name>
- event specific data: a dictionary with 3 keys
 - service_name: The name of the stopped service
 - service_id: The consul ID of the stopped service
 - tags: The advertised Consul tags of the stopped service

Example:

```
{
  "name": "service_deregistered_event",
  "origin_uuid": "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3",
  "data": {
    "service_name": "xivo-ctid",
    "service_id": "8e58d2a7-cfed-4c2e-ac72-14e0b5c26dc2",
    "tags": ["xivo-ctid", "ca7f87e9-c2c8-5fad-ba1b-c3140ebb9be3", "Québec"]
  }
}
```

Queue logs

Queue logs are events logged by Asterisk in the queue_log table of the asterisk database. Queue logs are used to generate XiVO call center statistics.

Queue log sample

Agent callback login

time	callid	queuename	agent	event	
2012-07-03 15:27:23.896208	1341343640.4	NONE	Agent/3001	AGENTCALLBACKLOGIN	100

Agent callback logoff Agent/3001 is logged in queues q1 and q2.

time	callid	queuename	agent	event	
2012-07-03 15:28:07.348244	NONE	q2	Agent/3001	UNPAUSE	
2012-07-03 15:28:07.346320	NONE	q1	Agent/3001	UNPAUSE	
2012-07-03 15:28:07.327425	NONE	NONE	Agent/3001	UNPAUSEALL	
2012-07-03 15:28:06.249357	NONE	NONE	Agent/3001	AGENTCALLBACKLOGOFF	100

Call on a Queue with join empty conditions met

time	callid	queuename	agent	event	
2012-07-04 07:27:55.640421	1341401275.9	q1	NONE	JOINEMPTY	

Enter the queue and get answered by an agent

time	callid	queuename	agent	event	
2012-07-04 07:33:23.085718	1341401601.24	q1	Agent/3001	CONNECT	2
2012-07-04 07:33:21.165823	1341401601.24	q1	NONE	ENTERQUEUE	

Agent or caller ends the call after 12 seconds

time	callid	queuename	agent	event	
2012-07-04 07:37:46.601754	1341401851.34	q1	Agent/3001	COMPLETEAGENT	2

Call on a full queue

time	callid	queuename	agent	event	
2012-07-04 07:40:17.339945	1341402016.44	q1	NONE	FULL	

Call on a closed queue

time	callid	queuename	agent	event	
2012-07-04 07:48:03.455999	1341402482.49	q1	NONE	CLOSED	

Caller abandon before an answer

time	callid	queuename	agent	event	
2012-07-04 07:49:52.939802	1341402586.51	q1	NONE	ABANDON	1

REST API

The XiVO REST APIs are the privileged way to programmatically interact with XiVO.

Reference

xivo-agentd REST API You can view the API documentation at <http://api.xivo.io>.

Changelog

15.19

- Token authentication is now required for all routes, i.e. it is not possible to interact with xivo-agentd without a xivo-auth authentication token.

15.18

- xivo-agentd now uses HTTPS instead of HTTP.

15.15

- The resources returning agent statuses, i.e.:
 - GET /agents
 - GET /agents/by-id/{agent_id}
 - GET /agents/by-number/{agent_number}

are now returning an additional argument named “state_interface”, which is “the interface (e.g. SIP/alice) that is used to determine if an agent is in use or not”.

xivo-confd REST API

Note: REST API 1.1 for confd is currently evolving. New features and small fixes are regularly being added over time. We invite the reader to periodically check the [changelog](#) for an update on new features and changes.

xivo-confd REST API changelog

16.13

- New readonly parameters have been added to the trunks resource:
 - endpoint_sip
 - endpoint_sccp
- A new readonly parameter have been added to the endpoint_sip and endpoint_custom resource:
 - trunk
- A new API for associating an extension with an incall has been added:
 - DELETE /1.1/incalls/<incall_id>/extensions/<extension_id>
 - PUT /1.1/incalls/<incall_id>/extensions/<extension_id>
- Added incalls endpoints:
 - GET /1.1/incalls

- POST /1.1/incalls
- DELETE /1.1/incalls/<incall_id>
- GET /1.1/incalls/<incall_id>
- PUT /1.1/incalls/<incall_id>

16.12

- A new API for associating an endpoint with a trunk has been added:
 - DELETE /1.1/trunks/<trunk_id>/endpoints/sip/<endpoint_id>
 - PUT /1.1/trunks/<trunk_id>/endpoints/sip/<endpoint_id>
 - GET /1.1/trunks/<trunk_id>/endpoints/sip
 - GET /1.1/endpoints/sip/<endpoint_id>/trunks
 - DELETE /1.1/trunks/<trunk_id>/endpoints/custom/<endpoint_id>
 - PUT /1.1/trunks/<trunk_id>/endpoints/custom/<endpoint_id>
 - GET /1.1/trunks/<trunk_id>/endpoints/custom
 - GET /1.1/endpoints/custom/<endpoint_id>/trunks
- Added trunks endpoints:
 - GET /1.1/trunks
 - POST /1.1/trunks
 - DELETE /1.1/trunks/<trunk_id>
 - GET /1.1/trunks/<trunk_id>
 - PUT /1.1/trunks/<trunk_id>
- Added SIP general endpoints:
 - GET /1.1/asterisk/sip/general
 - PUT /1.1/asterisk/sip/general

16.11

- A new API for associating a user with an agent has been added:
 - DELETE /1.1/users/<user_id>/agents
 - GET /1.1/users/<user_id>/agents
 - PUT /1.1/users/<user_id>/agents/<agent_id>
- A new API to list lines associated to an extension
 - GET /1.1/extensions/<extension_id>/lines
- The following URLs have been deprecated. Please use the new API instead:
 - GET /1.1/extensions/<extension_id>/line

16.10

- Add possibility to associate many lines to the same user.
- Add possibility to associate many extensions to the same line (only if these lines are associated to the same user).
- A new API for associating a user with a voicemail has been added:

- DELETE /1.1/users/<user_id>/voicemails
 - GET /1.1/users/<user_id>/voicemails
 - PUT /1.1/users/<user_id>/voicemails
- A new API for associating a line with an extension has been added:
 - PUT /1.1/lines/<line_id>/extensions/<extension_id>
- A new API for associating a user with a line has been added:
 - PUT /1.1/users/<user_id>/lines/<line_id>
- The following URLs have been deprecated. Please use the new API instead:
 - DELETE /1.1/users/<user_id>/voicemail
 - GET /1.1/users/<user_id>/voicemail
 - POST /1.1/users/<user_id>/voicemail
 - POST /1.1/users/<user_id>/lines
 - POST /1.1/lines/<line_id>/extensions

16.09

- Added entities endpoints:
 - GET /1.1/entities
 - POST /1.1/entities
 - GET /1.1/entities/<entity_id>
 - DELETE /1.1/entities/<entity_id>
- A new API for updating all user's funckeys
 - PUT /1.1/users/<user_id>/funckeys
- A new parameter have been added to the users resource:
 - dtmf_hangup_enabled

16.06

- A new API for initializing a XiVO (passing the wizard):
 - GET /1.1/wizard
 - POST /1.1/wizard
 - GET /1.1/wizard/discover
- A new API for associating a user with an entity has been added:
 - GET /1.1/users/<user_id>/entities
 - PUT /1.1/users/<user_id>/entities/<entity_id>

16.05

- A new API for associating a user with a call permission has been added:
 - GET /1.1/users/<user_id>/callpermissions
 - PUT /1.1/users/<user_id>/callpermissions/<call_permission_id>
 - DELETE /1.1/users/<user_id>/callpermissions/<call_permission_id>
 - GET /1.1/callpermissions/<call_permission_id>/users

- Two new parameters have been added to the users resource:
 - call_permission_password
 - enabled
- A new API for user's forwards has been added:
 - PUT /1.1/users/<user_id>/forwards
- SIP endpoint: allow and disallow options are not split into multiple options anymore.
- SCCP endpoint: allow and disallow options are not split into multiple options anymore.

16.04

- The summary view has been added to /users (GET /users?view=summary)
- A new API for user's services has been added:
 - GET /1.1/users/<user_id>/services
 - GET /1.1/users/<user_id>/services/<service_name>
 - PUT /1.1/users/<user_id>/services/<service_name>
- A new API for user's forwards has been added:
 - GET /1.1/users/<user_id>/forwards
 - GET /1.1/users/<user_id>/forwards/<forward_name>
 - PUT /1.1/users/<user_id>/forwards/<forward_name>
- GET /1.1/users/export now requires the following header for CSV output:

```
Accept: text/csv; charset=utf-8
```

- Added call permissions endpoints:
 - GET /1.1/callpermissions
 - POST /1.1/callpermissions
 - GET /1.1/callpermissions/<callpermission_id>
 - PUT /1.1/callpermissions/<callpermission_id>
 - DELETE /1.1/callpermissions/<callpermission_id>

16.03

- Added switchboard endpoints:
 - GET /1.1/switchboards
 - GET /1.1/switchboards/<switchboard_id>/stats
- A new API for associating a line with a device has been added:
 - PUT /1.1/lines/<line_id>/devices/<device_id>
 - DELETE /1.1/lines/<line_id>/devices/<device_id>
- The following URLs have been deleted. Please use the new API instead:
 - GET /1.1/devices/<device_id>/associate_line/<line_id>
 - GET /1.1/devices/<device_id>/dissociate_line/<line_id>

16.02

- Added users endpoints in REST API:
 - GET /1.1/users/<user_uuid>/lines/main/associated/endpoints/sip

16.01

- The SIP API has been improved. `options` now accepts any extra parameter. However, due to certain database limitations, parameters that appear in *Supported parameters on SIP endpoints* may only appear once in the list. This limitation will be removed in future versions.
- A new API for custom endpoints has been added: `/1.1/endpoints/custom`
- A new API for associating custom endpoints has been added: `/1.1/lines/<line_id>/endpoints/custom/<endpoint_id>`

15.20

- A new API for mass updating users has been added: `PUT /1.1/users/import`
- A new API for exporting users has been added: `GET /1.1/users/export`

15.19

- A new API for mass importing users has been added: `POST /1.1/users/import`
- The following fields have been added to the `/users` API:
 - `supervision_enabled`
 - `call_transfer_enabled`
 - `ring_seconds`
 - `simultaneous_calls`

15.18

- Ports 50050 and 50051 have been removed. Please use 9486 and 9487 instead
- Added sccp endpoints in REST API:
 - GET /1.1/endpoints/sccp
 - POST /1.1/endpoints/sccp
 - DELETE /1.1/endpoints/sccp/<sccp_id>
 - GET /1.1/endpoints/sccp/<sccp_id>
 - PUT /1.1/endpoints/sccp/<sccp_id>
 - GET /1.1/endpoints/sccp/<sccp_id>/lines
 - GET /1.1/lines/<line_id>/endpoints/sccp
 - DELETE /1.1/lines/<line_id>/endpoints/sccp/<sccp_id>
 - PUT /1.1/lines/<line_id>/endpoints/sccp/<sccp_id>
- Added lines endpoints in REST API:
 - GET /1.1/lines/<line_id>/users

15.17

- A new API for SIP endpoints has been added. Consult the documentation on <http://api.xivo.io> for further details.
- The `/lines_sip` API has been deprecated. Please use `/lines` and `/endpoints/sip` instead.
- Due to certain limitations in the database, only a limited number of optional parameters can be configured. This limitation will be removed in future releases. Supported parameters are listed further down.
- Certain fields in the `/lines` API have been modified. List of fields are further down

Fields modified in the `/lines` API

Name	Replaced by	Editable ?	Required ?
id		no	
device_id		no	
name		no	
protocol		no	
device_slot	position	no	
provisioning_extension	provisioning_code	no	
context		yes	yes
provisioning_code		yes	
position		yes	
caller_id_name		yes	
caller_id_num		yes	

Supported parameters on SIP endpoints

- md5secret
- language
- accountcode
- amaflags
- allowtransfer
- fromuser
- fromdomain
- subscribemwi
- buggywmi
- call-limit
- callerid
- fullname
- cid-number
- maxcallbitrate
- insecure
- nat
- promiscdir
- usereqphone
- videosupport
- trustpid
- sendrpid
- allowsubscribe

- allowoverlap
- dtmfmode
- rfc2833compensate
- qualify
- g726nonstandard
- disallow
- allow
- autoframing
- mohinterpret
- useclientcode
- progressinband
- t38pt-udptl
- t38pt-usertpsource
- rtptimeout
- rtpholdtimeout
- rtpkeepalive
- deny
- permit
- defaultip
- setvar
- port
- regexten
- subscribecontext
- fullcontact
- vmexten
- callingpres
- ipaddr
- regseconds
- regserver
- lastms
- parkinglot
- protocol
- outboundproxy
- transport
- remotesecret
- directmedia
- callcounter
- busylevel
- ignoresdpversion

- session-timers
- session-expires
- session-minse
- session-refresher
- callbackextension
- timert1
- timerb
- qualifyfreq
- contactpermit
- contactdeny
- unsolicited_mailbox
- use-q850-reason
- encryption
- snom-aoc-enabled
- maxforwards
- disallowed-methods
- textsupport

15.16

- The parameter `skip` is now deprecated. Use `offset` instead for:
 - GET /1.1/devices
 - GET /1.1/extensions
 - GET /1.1/voicemails
 - GET /1.1/users
- The users resource can be referred to by `uuid`
 - GET /1.1/users/<uuid>
 - PUT /1.1/users/<uuid>
 - DELETE /1.1/users/<uuid>

15.15

- The field `enabled` has been added to the voicemail model
- A line is no longer required when associating a voicemail with a user
- Voicemails can now be edited even when they are associated to a user

15.14

- All optional fields on a user are now always null (sometimes they were empty strings)
- The caller id is no longer automatically updated when the firstname or lastname is modified. You must update the caller id yourself if you modify the user's name.
- Caller id will be generated if and only if it does not exist when creating a user.

14.16

- Association user-voicemail, when associating a voicemail whose id does not exist:
 - before: error 404
 - after: error 400

14.14

- Association line-extension, a same extension can not be associated to multiple lines

14.13

- Resource line, field `provisioning_extension`: type changed from `int` to `string`

REST API 1.1 examples

Create User for a line and a exten

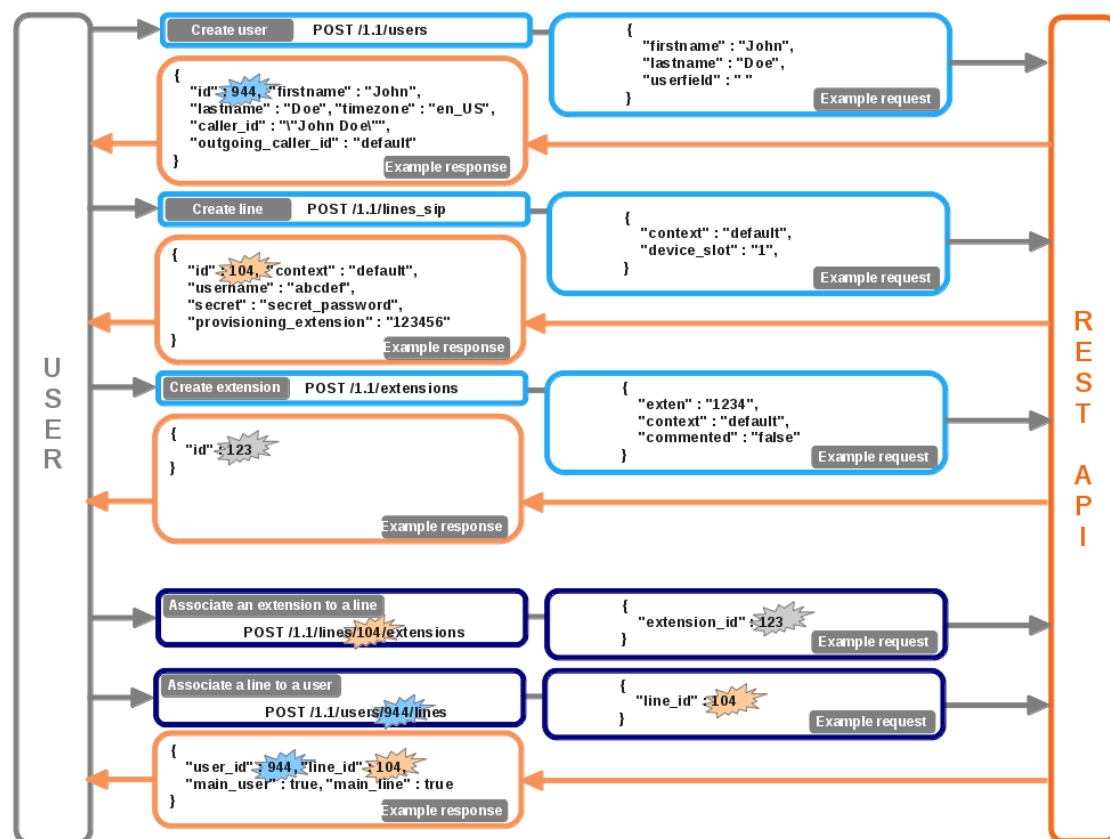


Fig. 1.104: Download source. (source)

Add user, line and exten with association

Add voicemail with association

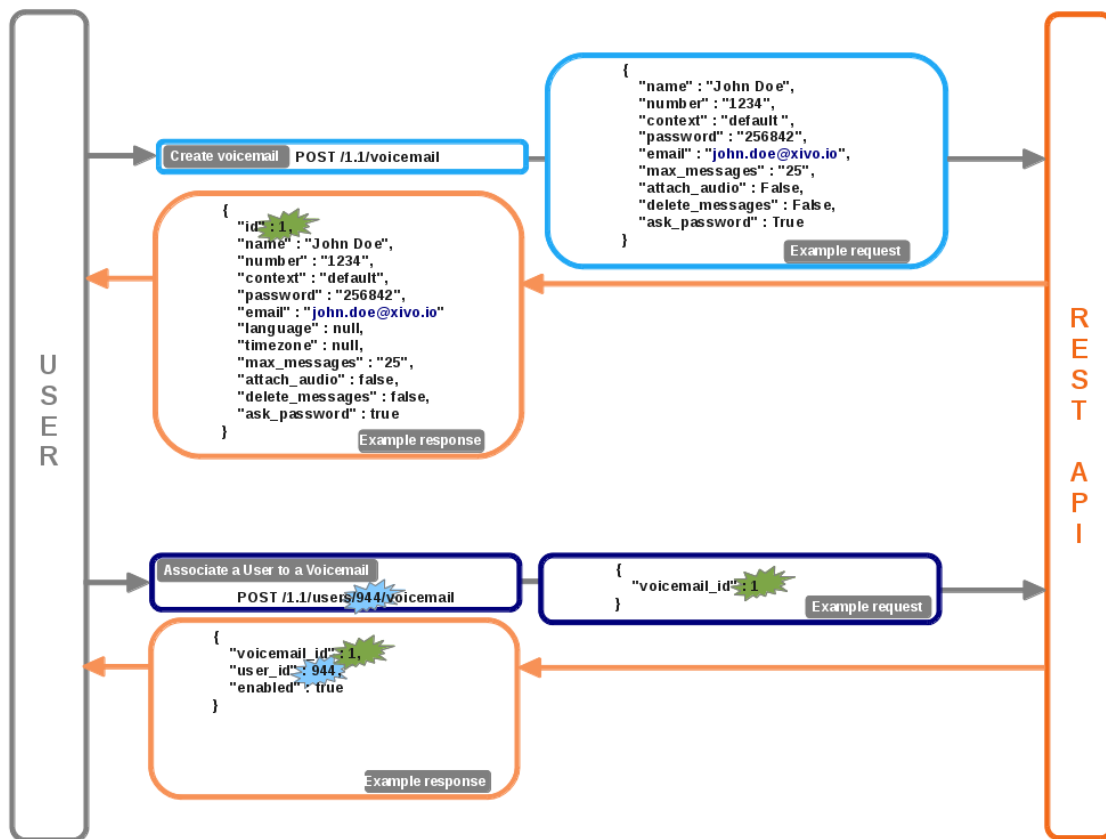


Fig. 1.105: Download source. (source)

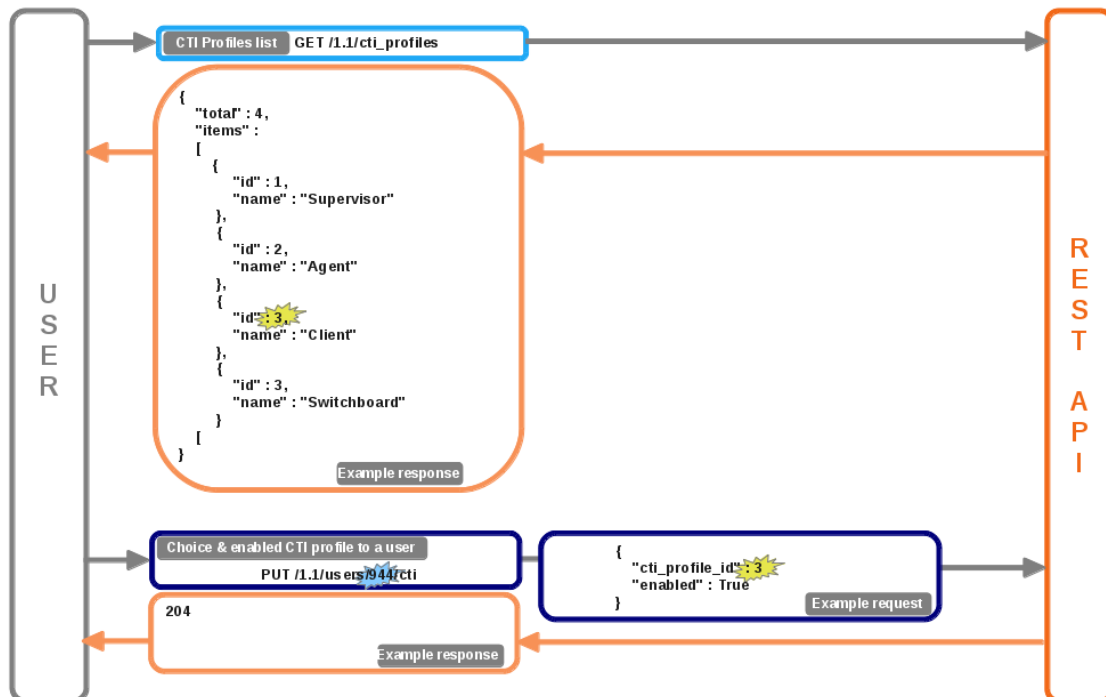


Fig. 1.106: Download source. (source)

Choice and add CTI profile with association

Multiple users for a line association

API reference API documentation is available on <http://api.xivo.io>. This section contains extended documentation for certain aspects of the API.

Function Keys Function keys can be used as shortcuts for dialing a number, or accomplishing other menial tasks, by pushing a button on the phone. A function key's action is determined by its destination.

Function keys can be added directly on a user, or in a template. Templates are useful for creating a set of common function keys that can be used by the same group of people.

This page only describes the data models used by the REST API. Consult the [API documentation](#) for further details on URLs.

Function Key Template

Parameters	Field	Type	Re- quired	Description
	name	string	No	A name for the template.
	keys	<i>Function Key</i>	No	A collection of function keys under the form {"position": "funckey"}. See the example for more details.

Example

```
{
  "name": "Example template",
  "keys": {
    "1": {
      "destination": {
        "type": "user",
        "user_id": 34
      }
    },
    "2": {
      "blf": true,
      "label": "Call mom",
      "destination": {
        "type": "custom",
        "exten": "5551234567"
      }
    }
  }
}
```

Function Key

Description	Field	Type	Required	Description
	blf	boolean	No	Turn on BLF when there is activity on the destination
	label	string	No	Label to display next to the function key
	destination	<i>Destination</i>	Yes	Destination to call

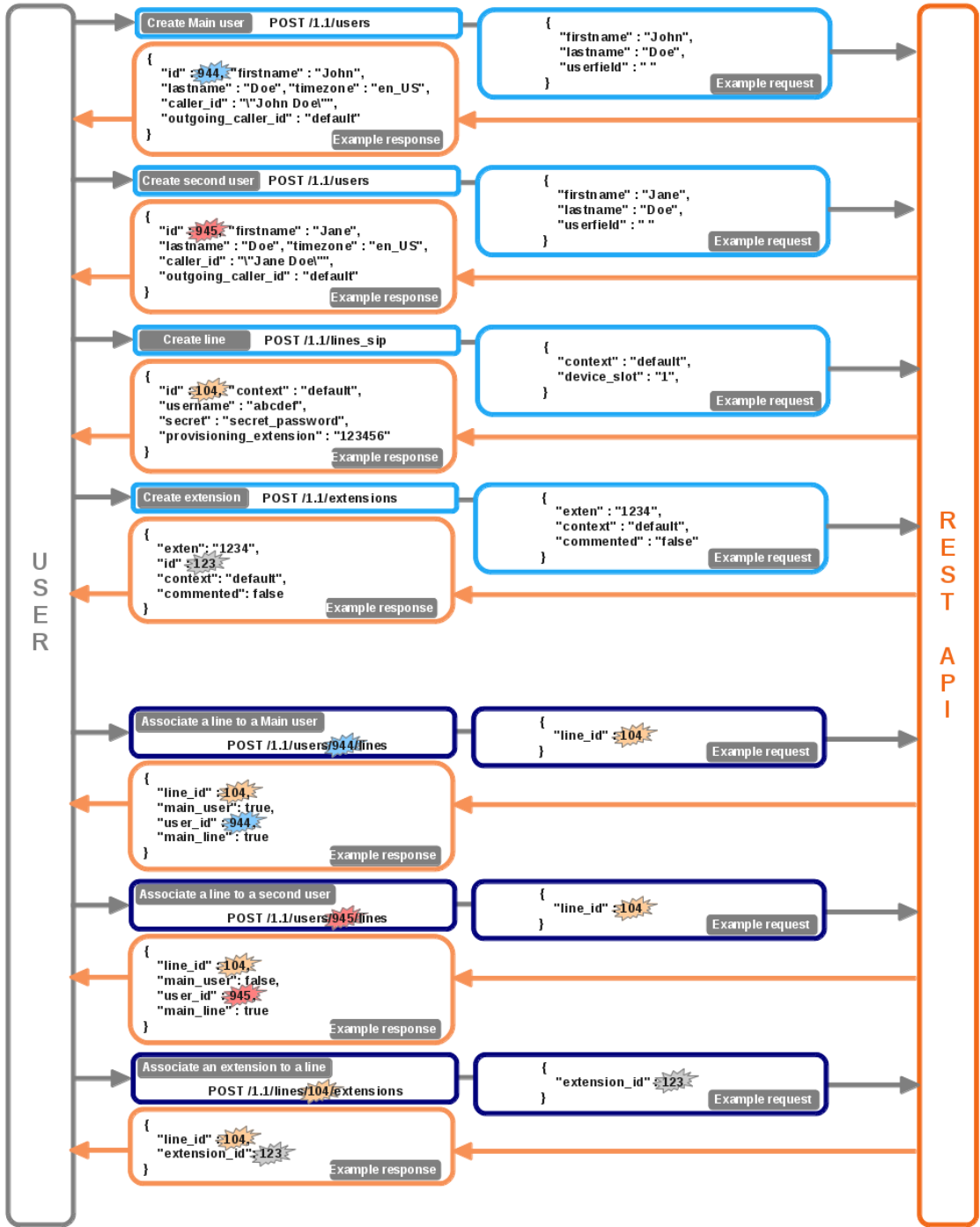


Fig. 1.107: Download source. (source)

Example

```
{
  "blf": True,
  "label": "Call john",
  "destination": {
    "type": "user",
    "user_id": 34
  }
}
```

Destination A destination determines the number to dial when using a function key. Destinations are composed of a parameter named `type` and any additional parameters required by its type.

Available destination types:

agent An agent

bsfilter Boss/Secretary filter

conference Conference room

custom A custom number to dial

forward Forward a call towards another number

group A group

onlinerec Record a conversation during a call

paging A paging

park Park a call

park_position Pick up a parked call

queue Call queue

service A call service

transfer Transfer a call

user A User

Here are the parameters required for each destination:

Agent	Field	Type	Value
	agent_id	numeric	Agents's id

BSFilter	Field	Type	Value
	filter_member_id	numeric	ID of the filter member

Conference	Field	Type	Value
	conference_id	numeric	Conference's id

Custom	Field	Type	Value
	exten	string	Number to dial

Forward	Field	Type	Value
	forward	string	Type of forward. Possible values: busy, noanswer, unconditional
	exten	string	Number to dial (optional)

Group	Field	Type	Value
	group_id	numeric	Group's id

Online call recording No parameters are required for this destination

Paging	Field	Type	Value
	paging_id	numeric	Pagings's id

Parking No parameters are required for this destination

Parking Position	Field	Type	Value
	position	numeric string	Position of the parking to pick up

Queue	Field	Type	Value
	queue_id	numeric	User's id

Service	Field	Type	Value
	service	string	Name of the service

Currently supported services:

phonestatus Phone Status

recsnd Sound Recording

callrecord Call recording

incallfilter Incoming call filtering

enablednd Enable "Do not disturb" mode

pickup Group Interception

calllistening Listen to online calls

directoryaccess Directory access

fwdundoall Disable all forwarding

enablevm Enable Voicemail

vmusermsg Consult the Voicemail

vmuserpurge Delete messages from voicemail

Transfer	Field	Type	Value
	transfer	string	Type of transfer. Possible values: blind, attended

User	Field	Type	Value
	user_id	numeric	User's id

CSV User Import Users and common related resources can be imported onto a XiVO server by sending a CSV file with a predefined *set of fields*.

This page only documents additional notes useful for API users. Consult the [API documentation](#) for more details.

Uploading files Files may be uploaded as usual through the web interface, or from a console by using HTTP utilities and the REST API. When uploading through the API, the header *Content-Type: text/csv charset=utf-8* must be set and the CSV data must be sent in the body of the request. A file may be uploaded using *curl* as follows:

```
curl -k -H "Content-Type: text/csv; charset=utf-8" -u username:password --data-binary "@file.csv"
```

The response can be reindented in a more readable format by piping the output through *python -m json.tool* in the following way:

```
curl (...) | python -m json.tool
```

Migration from 1.0 The API version 1.0 is no longer supported and has been removed. In most cases, code that used the old API can be migrated to version 1.1 without much hassle by updating the URL. For example, in 1.0, the URL to list users was:

```
/1.0/users/  
  
In 1.1, it is::  
  
/1.1/users
```

Please note that there are no trailing slashes in URLs for version 1.1.

For further details consult the documentation at <http://api.xivo.io>

xivo-ctid REST API

Note: The HTTP API 0.1 for xivo-ctid is currently evolving. New features and small fixes are regularly being added over time. We invite the reader to periodically check the [changelog](#) for an update on new features and changes.

xivo-ctid HTTP API changelog

16.08

- The GET `/0.1/users` now returns the `user_uuid` field

API reference API documentation is available on <http://api.xivo.io>.

xivo-ctid-ng REST API

Note: The HTTP API 1.0 for xivo-ctid-ng is currently evolving. New features and small fixes are regularly being added over time. We invite the reader to periodically check the [changelog](#) for an update on new features and changes.

xivo-ctid-ng HTTP API changelog

16.11

- A new API for getting the status of lines:
 - GET `/1.0/lines/{id}/presences`

16.10

- A new API for checking the status of the daemon:
 - GET /1.0/status

16.09

- A new API for updating user presences:
 - GET /1.0/users/{uuid}/presences
 - PUT /1.0/users/{uuid}/presences
 - GET /1.0/users/me/presences
 - PUT /1.0/users/me/presences
- New APIs for listing and hanging up calls of a user:
 - GET /1.0/users/me/calls
 - DELETE /1.0/users/me/calls/{id}
- New APIs for listing, cancelling and completing transfers of a user:
 - GET /1.0/users/me/transfers
 - DELETE /1.0/users/me/transfers/{transfer_id}
 - PUT /1.0/users/me/transfers/{transfer_id}/complete
- POST /1.0/users/me/transfers may now return 403 status code.
- Originates (POST /*/calls) now return 400 if an invalid extension is given.

16.08

- A new API for making calls from the authenticated user:
 - POST /1.0/users/me/calls
- A new API for sending chat messages:
 - POST /1.0/chats
 - POST /1.0/users/me/chats
- A new parameter for transfer creation (POST /1.0/transfers):
 - variables
- A new API for making transfers from the authenticated user:
 - POST /1.0/users/me/transfers

API reference API documentation is available on <http://api.xivo.io>.

xivo-provd REST API This section describes the REST API provided by the xivo-provd application.

If you want to interact with the REST API of the xivo-provd daemon that is executing as part of XiVO, you should be careful on which operation you are doing as to not cause stability problem to other parts of the XiVO ecosystem. Mostly, this means being careful when editing or deleting devices and configs.

By default, the REST API of xivo-provd is accessible only from localhost on port 8666. No authentication is required.

Warning: Major changes could happen to this API.

API The description of the API has been split into these sections:

Provd Management

Get the Provd Manager The provd manager resource represents the main entry point to the xivo-provd REST API.

It links to the following resources:

- The `dev` relation links to a *device manager*.
- The `cfg` relation links to a *config manager*.
- The `pg` relation links to a *plugin manager*.
- The `srv.configure` relation links to the provd manager *configuration service*.

Query

```
GET /provd
```

Example request

```
GET /provd HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "links": [
    {
      "href": "/provd/dev_mgr",
      "rel": "dev"
    },
    {
      "href": "/provd/cfg_mgr",
      "rel": "cfg"
    },
    {
      "href": "/provd/pg_mgr",
      "rel": "pg"
    },
    {
      "href": "/provd/configuration",
      "rel": "srv.configure"
    }
  ]
}
```

Devices Management

Get the Device Manager The device manager links to the following resources:

- The `dev.synchronize` relation links to the *device synchronization service*.
- The `dev.reconfigure` relation links to the *device reconfiguration service*.

- The `dev.dhcpinfo` relation links to the *device DHCP information service*.
- The `dev.devices` relation links to the *list of devices*.

Query

```
GET /provd/dev_mgr
```

Example request

```
GET /provd/dev_mgr HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "links": [
    {
      "href": "/provd/dev_mgr/synchronize",
      "rel": "dev.synchronize"
    },
    {
      "href": "/provd/dev_mgr/reconfigure",
      "rel": "dev.reconfigure"
    },
    {
      "href": "/provd/dev_mgr/dhcpinfo",
      "rel": "dev.dhcpinfo"
    },
    {
      "href": "/provd/dev_mgr/devices",
      "rel": "dev.devices"
    }
  ]
}
```

List Devices

Query

```
GET /provd/dev_mgr/devices
```

Query Parameters

Field	Description
q	A selector, encoded in JSON, describing which device should be returned. All devices are returned if not specified. Example: <code>q={"ip":"10.34.1.119"}</code>
fields	A list of fields, separated by comma. Example: <code>fields=mac,ip</code>
skip	An integer specifying the number of devices to skip. Example: <code>skip=10</code>
sort	The key on which to sort the results. Example: <code>sort=id</code>
sort_ord	The order of sort; either ASC or DESC.

Example request

```
GET /provd/dev_mgr/devices HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "devices": [
    {
      "added": "auto",
      "config": "38e5e08ffe804b468f5aa53b9536bb25",
      "configured": true,
      "description": "",
      "id": "38e5e08ffe804b468f5aa53b9536bb25",
      "ip": "10.34.1.122",
      "mac": "00:08:5d:33:e5:76",
      "model": "6731i",
      "plugin": "xivo-aastra-3.3.1-SP2",
      "remote_state_sip_username": "je5qtq",
      "vendor": "Aastra",
      "version": "3.3.1.2235"
    }
  ]
}
```

Create a Device**Query**

```
POST /provd/dev_mgr/devices
```

Example request

```
POST /provd/dev_mgr/devices HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{
  "device": {
    "ip": "192.168.1.1",
    "mac": "00:11:22:33:44:55",
    "plugin": "xivo-aastra-3.3.1-SP2"
  }
}
```

Example response

```
HTTP/1.1 201 Created
Content-Type: application/vnd.proformatique.provd+json
Location: /provd/dev_mgr/devices/68b10c99945b4fb889f22a7559fc3271

{"id": "68b10c99945b4fb889f22a7559fc3271"}
```

If the `id` field is not given, then an ID is automatically generated by the server.

Get a Device**Query**

```
GET /provd/dev_mgr/devices/<device_id>
```

Example request

```
GET /provd/dev_mgr/devices/68b10c99945b4fb889f22a7559fc3271 HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "device": {
    "added": "auto",
    "config": "38e5e08ffe804b468f5aa53b9536bb25",
    "configured": true,
    "description": "",
    "id": "38e5e08ffe804b468f5aa53b9536bb25",
    "ip": "10.34.1.122",
    "mac": "00:08:5d:33:e5:76",
    "model": "6731i",
    "plugin": "xivo-aastra-3.3.1-SP2",
    "remote_state_sip_username": "je5qtq",
    "vendor": "Aastra",
    "version": "3.3.1.2235"
  }
}
```

Update a Device**Query**

```
PUT /provd/dev_mgr/devices/<device_id>
```

Example request

```
PUT /provd/dev_mgr/devices/68b10c99945b4fb889f22a7559fc3271 HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{
  "device": {
    "added": "auto",
    "config": "38e5e08ffe804b468f5aa53b9536bb25",
    "configured": true,
    "description": "",
    "id": "38e5e08ffe804b468f5aa53b9536bb25",
    "ip": "10.34.1.122",
    "mac": "00:08:5d:33:e5:76",
    "model": "6731i",
    "plugin": "xivo-aastra-3.4",
    "remote_state_sip_username": "je5qtq",
    "vendor": "Aastra",
    "version": "3.3.1.2235"
  }
}
```

Example response

```
HTTP/1.1 204 No Content
```

Delete a Device

Query

```
DELETE /provd/dev_mgr/devices/<device_id>
```

Example request

```
DELETE /provd/dev_mgr/devices/68b10c99945b4fb889f22a7559fc3271 HTTP/1.1
Host: xivoserver
```

Example response

```
HTTP/1.1 204 No Content
```

Synchronize a Device

Query

```
POST /provd/dev_mgr/synchronize
```

Example request

```
POST /provd/dev_mgr/synchronize HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{
  "id": "d035bccaf0dd4a8396fc57a3329ca0a4"
}
```

Example response

```
HTTP/1.1 201 Created
Location: /provd/dev_mgr/synchronize/42
```

The URI returned in the `Location` header points to an *operation in progress* resource.

Reconfigure a Device

Query

```
POST /provd/dev_mgr/reconfigure
```

Errors	Error code	Error message	Description
	400	invalid device ID	

Example request

```
POST /provd/dev_mgr/reconfigure HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{
  "id": "d035bccaf0dd4a8396fc57a3329ca0a4"
}
```


Example response

```
HTTP/1.1 204 No Content
```

Push DHCP Request Information**Query**

```
POST /provd/dev_mgr/dhcpinfo
```

Example request

```
POST /provd/dev_mgr/dhcpinfo HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{
  "dhcp_info": {
    "ip": "192.168.1.100",
    "mac": "00:11:22:33:44:55",
    "op": "commit",
    "options": [
      "06066.6f.6f.62.61.72.a"
    ]
  }
}
```

Example response

```
HTTP/1.1 204 No Content
```

Configs Management

Get the Config Manager The config manager links to the following resources:

- The `cfg.configs` relation links to the *list of configs*.
- The `cfg.autocreate` relation links to the *config autocreate service*.

Query

```
GET /provd/cfg_mgr
```

Example request

```
GET /provd/cfg_mgr HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "links": [
    {
      "href": "/provd/cfg_mgr/configs",

```

```
        "rel": "cfg.configs"
      },
      {
        "href": "/provd/cfg_mgr/autocreate",
        "rel": "cfg.autocreate"
      }
    ]
  }
}
```

List Configs

Query

```
GET /provd/cfg_mgr/configs
```

Query Parameters These are the *same parameters as for the list devices* action.

Example request

```
GET /provd/cfg_mgr/configs HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "configs": [
    {
      "configdevice": "defaultconfigdevice",
      "deletable": true,
      "id": "38e5e08ffe804b468f5aa53b9536bb25",
      "parent_ids": [
        "base",
        "defaultconfigdevice"
      ],
      "raw_config": {
        "X_key": "",
        "exten_dnd": "*25",
        "exten_fwd_busy": "*23",
        "exten_fwd_disable_all": "*20",
        "exten_fwd_no_answer": "*22",
        "exten_fwd_unconditional": "*21",
        "exten_park": null,
        "exten_pickup_call": "*8",
        "exten_pickup_group": null,
        "exten_voicemail": "*98",
        "funckeys": {
          "1": {
            "label": "",
            "line": 1,
            "type": "speeddial",
            "value": "1005"
          }
        }
      },
      "protocol": "SIP",
      "sip_dtmf_mode": "SIP-INFO",
      "sip_lines": {
```

```

        "1": {
            "auth_username": "je5qtq",
            "display_name": "El\u00e8s 01",
            "number": "1001",
            "password": "T2S7C0",
            "proxy_ip": "10.34.1.11",
            "registrar_ip": "10.34.1.11",
            "username": "je5qtq"
        }
    }
}
]
}

```

Create a Config

Query

```
POST /provd/cfg_mgr/configs
```

Example request

```

POST /provd/cfg_mgr/configs HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{
  "config": {
    "parent_ids": [
      "base"
    ],
    "raw_config": {
      "sip": {
        "lines": {
          "1": {
            "auth_username": "100",
            "display_name": "Foo",
            "password": "100",
            "username": "100"
          }
        }
      }
    }
  }
}

```

Example response

```

HTTP/1.1 201 Created
Content-Type: application/vnd.proformatique.provd+json
Location: /provd/cfg_mgr/configs/77839d0f05c84662864b0ae5c27b33e4

{"id": "77839d0f05c84662864b0ae5c27b33e4"}

```

If the `id` field is not given, then an ID is automatically generated by the server.

Get a Config

Query

```
GET /provd/cfg_mgr/configs/<config_id>
```

Example request

```
GET /provd/cfg_mgr/configs/77839d0f05c84662864b0ae5c27b33e4 HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "config": {
    "id": "77839d0f05c84662864b0ae5c27b33e4",
    "parent_ids": [
      "base"
    ],
    "raw_config": {
      "sip": {
        "lines": {
          "1": {
            "auth_username": "100",
            "display_name": "Foo",
            "password": "100",
            "username": "100"
          }
        }
      }
    }
  }
}
```

Get a Raw Config

Query

```
GET /provd/cfg_mgr/configs/<config_id>/raw
```

Example request

```
GET /provd/cfg_mgr/configs/77839d0f05c84662864b0ae5c27b33e4/raw HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "raw_config": {
    "X_xivo_phonebook_ip": "10.34.1.11",
    "http_port": 8667,
    "ip": "10.34.1.11",
    "ntp_enabled": true,
    "ntp_ip": "10.34.1.11",
    "sip": {
```

```

        "lines": {
            "1": {
                "auth_username": "100",
                "display_name": "John",
                "password": "100",
                "username": "100"
            }
        },
        "tftp_port": 69
    }
}

```

Update a Config

Query

```
PUT /provd/cfg_mgr/configs/<config_id>
```

Example request

```

PUT /provd/cfg_mgr/configs/77839d0f05c84662864b0ae5c27b33e4 HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{
  "config": {
    "id": "77839d0f05c84662864b0ae5c27b33e4",
    "parent_ids": [
      "base"
    ],
    "raw_config": {
      "sip": {
        "lines": {
          "1": {
            "auth_username": "100",
            "display_name": "John",
            "password": "100",
            "username": "100"
          }
        }
      }
    }
  }
}

```

Example response

```
HTTP/1.1 204 No Content
```

Delete a Config

Query

```
DELETE /provd/cfg_mgr/configs/<config_id>
```

Example request

```
DELETE /provd/cfg_mgr/configs/77839d0f05c84662864b0ae5c27b33e4
Host: xivoserver
```

Example response

```
HTTP/1.1 204 No Content
```

Autocreate a Config This service is used to create a new config from the config that has the `autocreate` role.

Query

```
POST /provd/cfg_mgr/autocreate
```

Example request

```
POST /provd/cfg_mgr/autocreate HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{ }
```

Example response

```
HTTP/1.1 201 Created
Content-Type: application/vnd.proformatique.provd+json
Location: /provd/cfg_mgr/configs/autoprov1411400365

{ "id": "autoprov1411400365" }
```

Plugins Management

Get the Plugin Manager The plugin manager links to the following resources:

- The `srv.install` relation links to the plugin manager *installation service*. This installation service permits installing/uninstalling plugins.
- The `pg.plugins` relation links to the *list of plugins*.
- The `pg.reload` relation links to the *plugin reload service*.

Query

```
GET /provd/pg_mgr
```

Example request

```
GET /provd/pg_mgr HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "links": [
    {

```

```

        "href": "/provd/pg_mgr/install",
        "rel": "srv.install"
      },
      {
        "href": "/provd/pg_mgr/plugins",
        "rel": "pg.plugins"
      },
      {
        "href": "/provd/pg_mgr/reload",
        "rel": "pg.reload"
      }
    ]
  }
}

```

List Plugins List the installed plugins.

If you want to install/uninstall plugins, you need to go through the plugin installation service.

Query

```
GET /provd/pg_mgr/plugins
```

Example request

```

GET /provd/pg_mgr/plugins HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json

```

Example response

```

HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "plugins": {
    "xivo-aastra-3.3.1-SP2": {
      "links": [
        {
          "href": "/provd/pg_mgr/plugins/xivo-aastra-3.3.1-SP2",
          "rel": "pg.plugin"
        }
      ]
    },
    "xivo-cisco-sccp-9.0.3": {
      "links": [
        {
          "href": "/provd/pg_mgr/plugins/xivo-cisco-sccp-9.0.3",
          "rel": "pg.plugin"
        }
      ]
    }
  }
}

```

Get a Plugin The plugin links to the following resources:

- The `pg.info` relation links to the *plugin information*.
- The `srv.install` relation links to the *plugin installation service*. Plugins usually provided this service to install/uninstall firmware and language files.

Query

```
GET /provd/pg_mgr/plugins/<plugin_id>
```

Example request

```
GET /provd/pg_mgr/plugins/xivo-aastra-3.3.1-SP2 HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "links": [
    {
      "href": "/provd/pg_mgr/plugins/xivo-aastra-3.3.1-SP2/info",
      "rel": "pg.info"
    },
    {
      "href": "/provd/pg_mgr/plugins/xivo-aastra-3.3.1-SP2/install",
      "rel": "srv.install"
    }
  ]
}
```

Get Information of a Plugin

Query

```
GET /provd/pg_mgr/plugins/<plugin_id>/info
```

Example request

```
GET /provd/pg_mgr/plugins/xivo-aastra-3.3.1-SP2/info HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "plugin_info": {
    "capabilities": {
      "Aastra, 6730i, 3.3.1.5089": {
        "sip.lines": 6
      },
      "Aastra, 6731i, 3.3.1.2235": {
        "sip.lines": 6,
        "switchboard": true
      },
      "Aastra, 6735i, 3.3.1.5089": {
        "sip.lines": 9
      },
      "Aastra, 6737i, 3.3.1.5089": {
        "sip.lines": 9
      }
    }
  }
}
```



```

    "Aastra, 6739i, 3.3.1.2235": {
      "sip.lines": 9
    },
    "Aastra, 6753i, 3.3.1.2235": {
      "sip.lines": 9
    },
    "Aastra, 6755i, 3.3.1.2235": {
      "sip.lines": 9,
      "switchboard": true
    },
    "Aastra, 6757i, 3.3.1.2235": {
      "sip.lines": 9,
      "switchboard": true
    },
    "Aastra, 9143i, 3.3.1.2235": {
      "sip.lines": 9
    },
    "Aastra, 9480i, 3.3.1.2235": {
      "sip.lines": 9
    }
  },
  "description": "Plugin for Aastra 6730i, 6731i, 6735i, 6737i, 6739i, 6753i, 6755i, 6757i, 9143i, 9480i",
  "version": "1.1"
}

```

Reload a Plugin Reload the given plugin. This is mostly useful during plugin development, after changing the code of the plugin, instead of restarting the xivo-provd application.

Query

```
POST /provd/pg_mgr/reload
```

Example request

```

POST /provd/pg_mgr/reload HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{
  "id": "xivo-aastra-3.3.1-SP2"
}

```

Example response

```
HTTP/1.1 204 No Content
```

General Resources This section describes the resources that are available from more than one URI or are generic enough to not fit in a more specific section.

Operation In Progress This resource represents an operation in progress and is used to follow the progress of an underlying operation. Said differently, it is a monitor on an operation that can change over time.

Get Current Status

Query

```
GET <uri>
```

Example request

```
GET <uri> HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "status": "progress"
}
```

The status field describe the current status of the operation. The format is [label][state[/end]] (\(sub_oips\))* . Here's some examples:

- progress
- download|progress
- download|progress;10
- download|progress;10/100
- download|progress(file_1|progress;20/100)(file_2|waiting;0/50)
- download|progress;20/150(file_1|progress)(file_2|waiting)
- op|progress(op1|progress(op11|progress)(op12|waiting))(op2|progress)

The state of an operation is either waiting, progress, success or fail.

Delete Delete the “operation in progress” resource.

This does not cancel the underlying operation; it only deletes the monitor. Every monitor that is created should be deleted, else they won't be freed by the process and they will accumulate, taking memory.

Query

```
DELETE <uri>
```

Example request

```
DELETE <uri> HTTP/1.1
Host: xivoserver
```

Example response

```
HTTP/1.1 204 No Content
```

Configuration Service

Get the Configuration

Query

```
GET <uri>
```

Example request Example request for the configuration service of the *provd manager*.

```
GET /provd/configure HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "params": [
    {
      "description": "The plugins repository URL",
      "id": "plugin_server",
      "links": [
        {
          "href": "/provd/configure/plugin_server",
          "rel": "srv.configure.param"
        }
      ],
      "value": "http://provd.xivo.solutions/plugins/1/stable"
    },
    {
      "description": "The proxy for HTTP requests. Format is \"http://[user:password@]host:port\"",
      "id": "http_proxy",
      "links": [
        {
          "href": "/provd/configure/http_proxy",
          "rel": "srv.configure.param"
        }
      ],
      "value": null
    },
    {
      "description": "The proxy for FTP requests. Format is \"http://[user:password@]host:port\"",
      "id": "ftp_proxy",
      "links": [
        {
          "href": "/provd/configure/ftp_proxy",
          "rel": "srv.configure.param"
        }
      ],
      "value": null
    },
    {
      "description": "The proxy for HTTPS requests. Format is \"host:port\"",
      "id": "https_proxy",
      "links": [
        {
          "href": "/provd/configure/https_proxy",
          "rel": "srv.configure.param"
        }
      ],
      "value": null
    }
  ]
}
```

```
    "description": "The current locale. Example: fr_FR",
    "id": "locale",
    "links": [
      {
        "href": "/provd/configure/locale",
        "rel": "srv.configure.param"
      }
    ],
    "value": null
  },
  {
    "description": "Set to 1 if all the devices are behind a NAT.",
    "id": "NAT",
    "links": [
      {
        "href": "/provd/configure/NAT",
        "rel": "srv.configure.param"
      }
    ],
    "value": 0
  }
]
```

Get the Value of a Parameter

Query

```
GET <uri>
```

Example request Example request for the NAT option of the configuration service of the provd entry point.

```
GET /provd/configure/NAT HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "param": {
    "value": 0
  }
}
```

Set the Value of a Parameter

Query

```
PUT <uri>
```

Example request Example request for the NAT option of the configuration service of the *provd manager*.

```
PUT /provd/configure/NAT HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json
```

```
{
  "param": {
    "value": 1
  }
}
```

Example response

```
HTTP/1.1 204 No Content
Content-Type: application/vnd.proformatique.provd+json
```

Installation Service

Get the Installation Service

Query

```
GET <uri>
```

Example request Example request for the installation service of the *plugin manager*.

```
GET /provd/pg_mgr/install HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "links": [
    {
      "href": "/provd/pg_mgr/install/install",
      "rel": "srv.install.install"
    },
    {
      "href": "/provd/pg_mgr/install/uninstall",
      "rel": "srv.install.uninstall"
    },
    {
      "href": "/provd/pg_mgr/install/installed",
      "rel": "srv.install.installed"
    },
    {
      "href": "/provd/pg_mgr/install/installable",
      "rel": "srv.install.installable"
    },
    {
      "href": "/provd/pg_mgr/install/upgrade",
      "rel": "srv.install.upgrade"
    },
    {
      "href": "/provd/pg_mgr/install/update",
      "rel": "srv.install.update"
    }
  ]
}
```

The upgrade and update services are optional and not all installation service provide them.

Install a Package

Query

```
POST <uri>
```

Example request Example request for the installation service of the plugin manager.

```
POST /provd/pg_mgr/install/install HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{
  "id": "xivo-polycom-4.0.4"
}
```

Example response

```
HTTP/1.1 201 Created
Location: /provd/pg_mgr/install/install/1
Content-Type: application/vnd.proformatique.provd+json
```

The URI returned in the `Location` header points to an *operation in progress* resource.

Uninstall a Package

Query

```
POST <uri>
```

Example request Example request for the installation service of the plugin manager.

```
POST /provd/pg_mgr/install/uninstall HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{
  "id": "xivo-polycom-4.0.4"
}
```

Example response

```
HTTP/1.1 204 No Content
Content-Type: application/vnd.proformatique.provd+json
```

Upgrade a Package

Query

```
POST <uri>
```

Example request Example request for the installation service of the plugin manager.

```
POST /provd/pg_mgr/install/upgrade HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{
  "id": "xivo-polycom-4.0.4"
}
```

Example response

```
HTTP/1.1 201 Created
Location: /provd/pg_mgr/install/upgrade/1
Content-Type: application/vnd.proformatique.provd+json
```

The URI returned in the `Location` header points to an *operation in progress* resource.

Update the List of Installable Packages**Query**

```
POST <uri>
```

Example request Example request for the installation service of the plugin manager.

```
POST /provd/pg_mgr/install/update HTTP/1.1
Host: xivoserver
Content-Type: application/vnd.proformatique.provd+json

{}
```

Example response

```
HTTP/1.1 201 Created
Location: /provd/pg_mgr/install/update/1
Content-Type: application/vnd.proformatique.provd+json
```

The URI returned in the `Location` header points to an *operation in progress* resource.

List Installable Packages**Query**

```
GET <uri>
```

Example request Example request for the installation service of the plugin manager.

```
GET /provd/pg_mgr/install/installable HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "pkgs": {
    "null": {
```

```

    "capabilities": {
      "*, *, *": {
        "sip.lines": 0
      }
    },
    "description": "Plugin that offers no configuration service and rejects TFTP/HTTP requests",
    "dsize": 1073,
    "shasum": "90b2fb6c2b135a9d539488b6a85779dd95e0e876",
    "version": "1.0"
  },
  "xivo-aastra-3.3.1-SP2": {
    "capabilities": {
      "Aastra, 6730i, 3.3.1.5089": {
        "sip.lines": 6
      },
      "Aastra, 6731i, 3.3.1.2235": {
        "sip.lines": 6,
        "switchboard": true
      },
      "Aastra, 6735i, 3.3.1.5089": {
        "sip.lines": 9
      },
      "Aastra, 6737i, 3.3.1.5089": {
        "sip.lines": 9
      },
      "Aastra, 6739i, 3.3.1.2235": {
        "sip.lines": 9
      },
      "Aastra, 6753i, 3.3.1.2235": {
        "sip.lines": 9
      },
      "Aastra, 6755i, 3.3.1.2235": {
        "sip.lines": 9,
        "switchboard": true
      },
      "Aastra, 6757i, 3.3.1.2235": {
        "sip.lines": 9,
        "switchboard": true
      },
      "Aastra, 9143i, 3.3.1.2235": {
        "sip.lines": 9
      },
      "Aastra, 9480i, 3.3.1.2235": {
        "sip.lines": 9
      }
    },
    "description": "Plugin for Aastra 6730i, 6731i, 6735i, 6737i, 6739i, 6753i, 6755i, 6757i",
    "dsize": 9397,
    "shasum": "68dbed6afa87cf624a89166bdc6bdf7413cb84df",
    "version": "1.1"
  }
}

```

List Installed Packages

Query

```
GET <uri>
```

Example request Example request for the installation service of the plugin manager.


```
GET /provd/pg_mgr/install/installed HTTP/1.1
Host: xivoserver
Accept: application/vnd.proformatique.provd+json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/vnd.proformatique.provd+json

{
  "pkgs": {
    "xivo-aastra-3.3.1-SP2": {
      "capabilities": {
        "Aastra, 6730i, 3.3.1.5089": {
          "sip.lines": 6
        },
        "Aastra, 6731i, 3.3.1.2235": {
          "sip.lines": 6,
          "switchboard": true
        },
        "Aastra, 6735i, 3.3.1.5089": {
          "sip.lines": 9
        },
        "Aastra, 6737i, 3.3.1.5089": {
          "sip.lines": 9
        },
        "Aastra, 6739i, 3.3.1.2235": {
          "sip.lines": 9
        },
        "Aastra, 6753i, 3.3.1.2235": {
          "sip.lines": 9
        },
        "Aastra, 6755i, 3.3.1.2235": {
          "sip.lines": 9,
          "switchboard": true
        },
        "Aastra, 6757i, 3.3.1.2235": {
          "sip.lines": 9,
          "switchboard": true
        },
        "Aastra, 9143i, 3.3.1.2235": {
          "sip.lines": 9
        },
        "Aastra, 9480i, 3.3.1.2235": {
          "sip.lines": 9
        }
      },
      "description": "Plugin for Aastra 6730i, 6731i, 6735i, 6737i, 6739i, 6753i, 6755i, 6757i, 9143i, 9480i",
      "version": "1.1"
    }
  }
}
```

xivo-sysconfd REST API This service provides a public API that can be used to change the configuration that are on a XiVO.

Warning: The 0.1 API is currently in development. Major changes could still happen and new resources will be added over time.

API reference

Asterisk Voicemail

Delete voicemail

Query

```
GET /delete_voicemail
```

Parameters

Mandatory

name the voicemail name

Optional

context the voicemail context (default is 'default')

Errors	Error code	Error message	Description
	404	Not found	The voicemail does not exist

Example requests

```
GET /delete_voicemail HTTP/1.1
Host: xivoserver
Accept: application/json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/json

{
  nothing
}
```

Common configuration

Apply configuration

Query

```
GET /commonconf_apply
```

Generate configuration

Query

```
POST /commonconf_generate
```

Dhcpd configuration

Update configuration

Query

```
GET /dhcpd_update
```

Ethernet configuration

Discover interfaces

Query

```
GET /discover_netifaces
```

Example request

```
GET /discover_netifaces HTTP/1.1
Host: xivoserver
Accept: application/json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/json
{
  "lo":
  {
    "hwaddress": "00:00:00:00:00:00",
    "typeid": 24,
    "alias-raw-device": null,
    "network": "127.0.0.0",
    "family": "inet",
    "physicalif": false,
    "vlan-raw-device": null,
    "vlanif": false,
    "dummyif": false,
    "mtu": 65536,
    "broadcast": "127.255.255.255",
    "hwtypeid": 772,
    "netmask": "255.0.0.0",
    "carrier": true,
    "flags": 9,
    "address": "127.0.0.1",
    "vlan-id": null,
    "type": "loopback",
    "options": null,
    "aliasif": false,
    "name": "lo"
  },
  "eth0":
  {
    "alias-raw-device": null,
    "family": "inet",
    "hwaddress": "36:76:70:29:69:c2",
    "vlan-id": null,
    "network": "172.17.0.0",
    "physicalif": false,
    "vlan-raw-device": null,
    "vlanif": false,
    "type": "eth",
    "aliasif": false,
    "broadcast": "172.17.255.255",
    "netmask": "255.255.0.0",
```

```
    "address": "172.17.0.101",
    "typeid": 6,
    "name": "eth0",
    "hwtypeid": 1,
    "dummyif": false,
    "mtu": 1500,
    "carrier": true,
    "flags": 3,
    "options": null
  }
}
```

Get interface

Query

```
GET /netiface/<interface>
```

Example request

```
GET /netiface/eth0 HTTP/1.1
Host: xivoserver
Content-Type: application/json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/json
{
  "eth0":
  {
    "alias-raw-device": null,
    "family": "inet",
    "hwaddress": "36:76:70:29:69:c2",
    "vlan-id": null,
    "network": "172.17.0.0",
    "physicalif": false,
    "vlan-raw-device": null,
    "vlanif": false,
    "type": "eth",
    "aliasif": false,
    "broadcast": "172.17.255.255",
    "netmask": "255.255.0.0",
    "address": "172.17.0.101",
    "typeid": 6,
    "name": "eth0",
    "hwtypeid": 1,
    "dummyif": false,
    "mtu": 1500,
    "carrier": true,
    "flags": 3,
    "options": null
  }
}
```

Modify interface

Description

Field	Values	Description
iface	string	Interface name like eth0
method	list	static or dhcp
address	string	
netmask	string	
broadcast	string	
gateway	string	
mtu	int	
auto	boolean	
up	boolean	
options	list	dns-search and dns-nameservers

Query

```
PUT /modify_physical_eth_ipv4
```

Example request

```
PUT /modify_physical_eth_ipv4 HTTP/1.1
Host: xivoserver
Content-Type: application/json
{
  "ifname": "eth0",
  "method": "dhcp",
  "auto": "True"
}
```

Replace virtual interface**Query**

```
PUT /replace_virtual_eth_ipv4
```

Example request

```
PUT /replace_virtual_eth_ipv4 HTTP/1.1
Host: xivoserver
Content-Type: application/json
{
  "ifname": "eth0:0",
  "new_ifname": "eth0:1",
  "method": "dhcp",
  "auto": "True"
}
```

Modify interface**Query**

```
PUT /modify_eth_ipv4
```

Example request

```
PUT /modify_eth_ipv4 HTTP/1.1
Host: xivoserver
Content-Type: application/json
{
  'ifname' : 'eth0'
```

```
{
  'address': '192.168.0.1',
  'netmask': '255.255.255.0',
  'broadcast': '192.168.0.255',
  'gateway': '192.168.0.254',
  'mtu': 1500,
  'auto': True,
  'up': True,
  'options': [['dns-search', 'toto.tld tutu.tld'],
              ['dns-nameservers', '127.0.0.1 192.168.0.254']]
}
```

Change state

Query

```
PUT /change_state_eth_ipv4
```

Example request

```
PUT /change_state_eth_ipv4 HTTP/1.1
Host: xivoserver
Content-Type: application/json
{
  'ifname' : 'eth0',
  'state': True
}
```

Delete interface ipv4

Query

```
GET /delete_eth_ipv4/<interface>
```

Example request

```
GET /delete_eth_ipv4/eth0 HTTP/1.1
Host: xivoserver
Content-Type: application/json
```

HA configuration

Get HA configuration

Query

```
GET /get_ha_config
```

Update HA configuration

Query

```
POST /update_ha_config
```

network configuration

Get network configuration

Query

```
GET /network_config
```

Rename ethernet interface

Query

```
POST /rename_ethernet_interface
```

swap ethernet interface

Query

```
POST /swap_ethernet_interfaces
```

Routes

Query

```
POST /routes
```

OpenSSL configuration

List certificates

Query

```
GET /openssl_listcertificates
```

Get certificate infos

Query

```
GET /openssl_certificateinfos
```

Export public key

Query

```
GET /openssl_exportpubkey
```

Export SSL certificate

Query

```
GET /openssl_export
```

Create CA certificate

Query

```
POST /openssl_createcertificate
```

Create certificate

Query

```
POST / openssl_createcertificate
```

Delete certificate

Query

```
GET /openssl_deletertificate
```

Import SSL certificate

Query

```
POST /openssl_import
```

DNS configuration

Host configuration

Query

```
POST /hosts
```

Resolv.conf configuration

Query

```
POST /resolv_conf
```

Services daemon

Reload services

Query

```
POST /services
```

Xivo Services

Reload XiVO services

Query


```
POST /xivocctl
```

Handlers

Execute handlers

Query

```
POST /exec_request_handlers
```

Status check

Status

Query

```
GET /status_check
```

Example request

```
GET /status_check HTTP/1.1
Host: xivoserver
Content-Type: application/json
```

Example response

```
HTTP/1.1 200 OK
Content-Type: application/json
{
  "status": "up"
}
```

For other services, see <http://api.xivo.io>. This public instance does not allow you to directly test the requests (i.e. the “Try it out!” button will not work), but you may use the *embedded version of your XiVO*, where this button will work.

How to use the embedded REST API web interface (Swagger UI) Every XiVO server embeds its own copy of the Swagger UI exposed on <http://api.xivo.io>. The instance embedded in the XiVO allows you to directly try the requests with the in-page buttons.

For the rest of this article, we will consider that your XiVO is accessible under the hostname `MY_XIVO`.

The instance is available at: http://MY_XIVO/api

Before using the Swagger UI, there are a few prerequisites:

- Accept the HTTPS certificate for each service of the XiVO
- Add the permissions to use the REST API to a Web Services Access user
- Obtain an authentication token

HTTPS certificates For each service on the left menu that you want to try, you need to accept the HTTPS certificate for this service. To that end:

1. click on the service in the menu on the left
2. copy the URL you see in the text box at the top of the page, something like: `https://MY_XIVO:9497/0.1/api/api.json` and paste it in your browser
3. accept the HTTPS certificate validation exception
4. go back to http://MY_XIVO/api and select the service again (or click on the top-right “Explore” button)

You should now be able to see the different sections for the REST API of that service.

REST API permissions You must create a Web Services Access with the right permissions before using the REST API. See [Web Services Access](#).

Each endpoint has its own ACL, but you may add wildcard ACLs, like:

- `auth.#` to gain access to all `xivo-auth` REST API endpoints
- `confd.#` to gain access to all `xivo-confd` REST API endpoints
- `#` to gain access to every endpoint of every service.

Warning: Only use wildcards when doing tests, not with a production REST API access. You should always restrict the permissions to the bare minimum.

Obtain an authentication token The quick and easy way is to use <http://auth.xivo.io>. You may log-in with the following parameters:

- Host = https://MY_XIVO:9497 (you must have accepted the HTTPS certificate of `xivo-auth` first)
- Backend = XIVO Service
- Login = username of your Web Services Access
- Password = the associated password

Then click “Sign in!”, and you can get see the token. This token will expire after one hour, and you will need to re-authenticate to get a new token.

The other way you can get a token is via Swagger UI (what else?). Choose the `xivo-auth` service in the list of REST API. Under `tokens`, choose `POST /tokens`.

1. In the top-right text box of the page (left to the “Explore” button), fill “token” with the string `username:password` where those credentials come from the Web Services Access you created earlier.
2. Go back to the `POST /tokens` section and click on the yellow box to the right of the `body` parameter. This will pre-fill the `body` parameter.
3. In the `body` parameter, set:
 - `backend` to `xivo-service`
 - `expiration` to the number of seconds for the token to be valid (e.g. 60 for one hour). After the expiration time, you will need to re-authenticate to get a new token.
4. Click “Try it out” at the end of the section
5. In the response, you should see a `token` attribute.

For more informations about the backends of `xivo-auth`, see [xivo-auth plugins](#).

Use the authentication token To use the authentication token, choose the service for which you want to try the REST API, then paste the token in the top-right text box. You do not need to click “Explore” to apply the token change, the new token will be used automatically at the next request you send.

You can now choose a REST API endpoint and “Try it out”.

Access Each REST API is available via HTTPS on *different ports*.

Examples (xivo-confd)

```
# Get the list of users
curl --insecure \
-H 'Accept: application/json' \
-H 'X-Auth-Token: 17496bfa-4653-9d9d-92aa-17def0fa9826' \
https://xivo:9486/1.1/users

# Create a user
# When sending data, you need the Content-Type header.
curl --insecure \
-X POST \
-d '{"firstname": "hello-world"}' \
-H 'Accept: application/json' \
-H 'Content-Type: application/json' \
-H 'X-Auth-Token: 17496bfa-4653-9d9d-92aa-17def0fa9826' \
https://xivo:9486/1.1/users
```

Authentication For all REST APIs, the main way to authenticate is to use an access token obtained from *xivo-auth*. This token should be given in the X-Auth-Token header in your request. For example:

```
curl <options...> -H 'X-Auth-Token: 17496bfa-4653-9d9d-92aa-17def0fa9826' https://<xivo_address>:
```

Also, your token needs to have the right ACLs to give you access to the resource you want. See *REST API Permissions*.

REST API Permissions The tokens delivered by *xivo-auth* have a list of permissions associated (ACL), that determine which REST resources are authorized for this token. Each REST resource has an associated required ACL. When you try to access to a REST resource, this resource requests xivo-auth with your token and the required ACL to validate the access.

Syntax An ACL contains 3 parts separated by dot (.)

- *service*: name of service, without prefix xivo- (e.g. xivo-confd -> confd).
- *resource*: name of resource separated by dot (.) (e.g. /users/17/lines -> users.17.lines).
- *action*: action performed on resource. Generally, this is the following schema:
 - get -> read
 - put -> update
 - post -> create
 - delete -> delete

Substitutions There are 3 substitution values for an ACL.

- *: replace only one word between dot.
- #: replace one or multiple words.
- me: replace the user_uuid from sent token.

Example The ACL `confd.users.me.#.read` will have access to the following REST resources:

```
GET /users/{user_id}/cti
GET /users/{user_id}/funckeys
GET /users/{user_id}/funckeys/{position}
GET /users/{user_id}/funckeys/templates
GET /users/{user_id}/lines
GET /users/{user_id}/lines/{line_id}
GET /users/{user_id}/voicemail
```

- *service*: `confd`
- *resource*: `users.me.#`
- *action*: `read`

The ACL `confd.users.me.funckeys.*.*` will have access to the following REST resources:

```
DELETE /users/{user_id}funckeys/{position}
GET /users/{user_id}funckeys/{position}
PUT /users/{user_id}funckeys/{position}
GET /users/{user_id}funckeys/templates
```

- *service*: `confd`
- *resource*: `users.me.funckeys.*`
- *action*: `*`

Where `{user_id}` is the user uuid from the token.

Available ACLs The ACL corresponding to each resource is documented in <http://auth.xivo.io>. Some resources may not have any associated ACL yet, so you must use `{service}.#` instead.

See also *Service Authentication* for details about the token-based authentication process.

Other methods (xivo-confd)

Warning: DEPRECATED

For compatibility reason, `xivo-confd` may accept requests without an access token. For this, you must create a `webservices` user in the web interface (*Configuration* → *Management* → *Web Services Access*):

- if an IP address is specified for the user, no authentication is needed
- if you choose not to specify an IP address for the user, you can connect to the REST API with a HTTP Digest authentication, using the user name and password you provided. For instance, the following command line allows to retrieve XiVO users through the REST API, using the login **admin** and the password **passadmin**:

```
curl <options...> --digest --cookie '' -u admin:passadmin https://<xivo_address>:9486/1.1/users
```

HTTP status codes Standard HTTP status codes are used. For the full definition see [IANA definition](#).

- 200: Success
- 201: Created
- 400: Incorrect syntax
- 404: Resource not found
- 406: Not acceptable
- 412: Precondition failed
- 415: Unsupported media type
- 500: Internal server error

See also [Errors](#) for general explanations about error codes.

General URL parameters Example usage of general parameters:

```
GET http://<xivo_address>:9486/1.1/voicemails?limit=X&offset=Y
```

Parameters

order Sort the list using a column (e.g. “number”). See specific resource documentation for columns allowed.

direction ‘asc’ or ‘desc’. Sort list in ascending (asc) or descending (desc) order

limit total number of resources to show in the list. Must be a positive integer

offset number of resources to skip over before starting the list. Must be a positive integer

search Search resources. Only resources with a field containing the search term will be listed.

Data representation

Data retrieved from the REST server JSON is used to encode returned or sent data. Therefore, the following headers are needed:

- **when the request is supposed to return JSON:** Accept = application/json
- **when the request’s body contains JSON:** Content-Type = application/json

Note: Optional properties can be added without changing the protocol version in the main list or in the object list itself. Properties will not be removed, type and name will not be modified.

Getting object lists GET /1.1/objects

When returning lists the format is as follows:

- total - number of items in total in the system configuration (optional)
- items - returned data as an array of object properties list.

Other optional properties can be added later.

Response data format

```
{
  "total": 2,
  "items":
  [
    {
      "id": "1",
      "prop1": "test"
    },
    {
      "id": "2",
      "prop1": "ssd"
    }
  ]
}
```

Getting An Object Format returned is a list of properties. The object should always have the same attributes set, the default value being the equivalent to NULL in the content-type format.

GET /1.1/objects/<id>

Response data format

```
{
  "id": "1",
  "prop1": "test"
}
```

Data sent to the REST server The XiVO REST server implements POST and PUT methods for item creation and update respectively. Data is created using the POST method via a root URL and is updated using the PUT method via a root URL suffixed by /<id>. The server expects to receive JSON encoded data. Only one item can be processed per request. The data format and required data fields are illustrated in the following example:

Request data format

```
{
  "id": "1",
  "prop1": "test"
}
```

When updating, only the id and updated properties are needed, omitted properties are not updated. Some properties can also be optional when creating an object.

Errors A request to the web services may return an error. An error will always be associated to an HTTP error code, and eventually to one or more error messages. The following errors are common to all web services:

Error code	Error message	Description
406	empty	Accept header missing or contains an unsupported content type
415	empty	Content-Type header missing or contains an unsupported content type
500	list of errors	An error occurred on the server side; the content of the message depends of the type of errors which occurred

The 400, 404 and 412 errors depend on the web service you are requesting. They are separately described for each of them.

The error messages are contained in a JSON list, even if there is only one error message:

```
[ message_1, message_2, ... ]
```

Subroutine

What is it ? The preprocess subroutine allows you to enhance XiVO features through the Asterisk dialplan. Features that can be enhanced are :

- User
- Group
- Queue
- Meetme
- Incoming call
- Outgoing call

There are three possible categories :

- Subroutine for one feature

- Subroutine for global forwarding
- Subroutine for global incoming call to an object

Subroutines are called at the latest possible moment in the dialplan, so that the maximum of variables have already been set: this way, the variables can be read and modified at will before they are used.

Here is an example of the dialplan execution flow when an external incoming call to a user being forwarded to another external number (like a forward to a mobile phone):

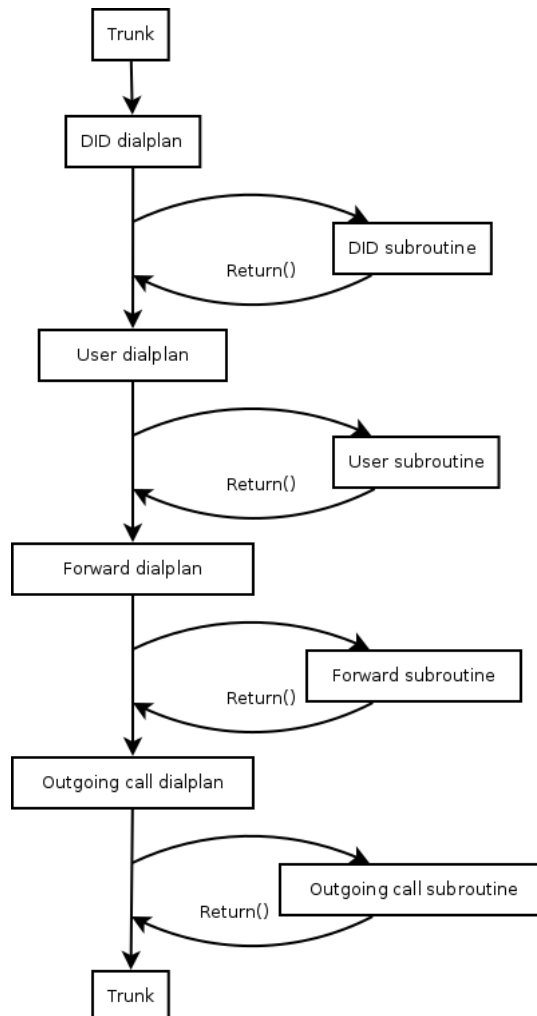


Fig. 1.108: Where subroutines are called in dialplan

Adding new subroutine If you want to add a new subroutine, we propose to edit a new configuration file in the directory `/etc/asterisk/extensions_extra.d`. You can also add this file by the web interface.

An example:

```

[myexample]
exten = s,1,NoOp(This is an example)
same = n,Return()
  
```

Subroutines should always end with a `Return()`. You may replace `Return()` by a `Goto()` if you want to completely bypass the XIVO dialplan, but this is not recommended.

To plug your subroutine into the XIVO dialplan, you must add `myexample` in the subroutine field in the web interface, e.g. *Services* → *IPBX* → *PBX Settings* → *Users* → *Edit* → *tab General* → *Preprocess subroutine*.

Global subroutine There is predefined subroutine for this feature, you can find the name and the activation in the `/etc/xivo/asterisk/xivo_globals.conf`. The variables are:

```
; Global Preprocess subroutine
XIVO_PRESUBR_GLOBAL_ENABLE = 1
XIVO_PRESUBR_GLOBAL_USER = xivo-subrgbl-user
XIVO_PRESUBR_GLOBAL_AGENT = xivo-subrgbl-agent
XIVO_PRESUBR_GLOBAL_GROUP = xivo-subrgbl-group
XIVO_PRESUBR_GLOBAL_QUEUE = xivo-subrgbl-queue
XIVO_PRESUBR_GLOBAL_MEETME = xivo-subrgbl-meetme
XIVO_PRESUBR_GLOBAL_DID = xivo-subrgbl-did
XIVO_PRESUBR_GLOBAL_OUTCALL = xivo-subrgbl-outcall
XIVO_PRESUBR_GLOBAL_PAGING = xivo-subrgbl-paging
```

So if you want to add a subroutine for all of your XiVO users you can do this:

```
[xivo-subrgbl-user]
exten = s,1,NoOp(This is an example for all my users)
same = n,Return()
```

Forward subroutine You can also use a global subroutine for call forward.

```
; Preprocess subroutine for forwards
XIVO_PRESUBR_FWD_ENABLE = 1
XIVO_PRESUBR_FWD_USER = xivo-subrfwd-user
XIVO_PRESUBR_FWD_GROUP = xivo-subrfwd-group
XIVO_PRESUBR_FWD_QUEUE = xivo-subrfwd-queue
XIVO_PRESUBR_FWD_MEETME = xivo-subrfwd-meetme
XIVO_PRESUBR_FWD_VOICEMAIL = xivo-subrfwd-voicemail
XIVO_PRESUBR_FWD_SCHEDULE = xivo-subrfwd-schedule
XIVO_PRESUBR_FWD_VOICEMENU = xivo-subrfwd-voicemenu
XIVO_PRESUBR_FWD_SOUND = xivo-subrfwd-sound
XIVO_PRESUBR_FWD_CUSTOM = xivo-subrfwd-custom
XIVO_PRESUBR_FWD_EXTENSION = xivo-subrfwd-extension
```

Dialplan variables Some of the XiVO variables can be used and modified in subroutines (non exhaustive list):

- **XIVO_CALLOPTIONS**: the value is a list of options to be passed to the Dial application, e.g. `hHtT`. This variable is available in agent, user and outgoing call subroutines. Please note that it may not be set earlier, because it will be overwritten.
- **XIVO_CALLORIGIN**: the value is:
 - `extern` for calls coming from a DID
 - `intern` for all other calls

This variable is used by xivo-agid when *selecting the ringtone* for ringing a user. This variable is available only in user subroutines.

- **XIVO_DSTNUM**: the value is the extension dialed, as received by XiVO (e.g. an internal extension, a DID, or an outgoing extension including the local prefix). This variable is available in all subroutines.
- **XIVO_GROUPNAME**: the value is the name of the group being called. This variable is only available in group subroutines.
- **XIVO_GROUPOPTIONS**: the value is a list of options to be passed to the Queue application, e.g. `hHtT`. This variable is only available in group subroutines.
- **XIVO_INTERFACE**: the value is the *Technology/Resource* pairs that are used as the first argument of the *Dial application*. This variable is only available in the user subroutines.
- **XIVO_MOBILEPHONENUMBER**: the value is the phone number of a user, as set in the web interface. This variable is only available in user subroutines.

- `XIVO_QUEUEENAME`: the value is the name of the queue being called. This variable is only available in queue subroutines.
- `XIVO_QUEUEOPTIONS`: the value is a list of options to be passed to the Queue application, e.g. `hHTT`. This variable is only available in queue subroutines.
- `XIVO_SRCNUM`: the value is the callerid number of the originator of the call: the internal extension of a user (outgoing callerid is ignored), or the public extension of an external incoming call. This variable is available in all subroutines.

WebSocket Event Service

XiVO offers a service to receive messages published on the *bus* (e.g. *RabbitMQ*) over an encrypted [WebSocket](#) connection. This ease in building dynamic web applications that are using events from your XiVO.

The service is provided by the `xivo-websocketd` component.

Getting Started First, you need a XiVO in version 16.03 or later.

Then, to use the service, you need to:

1. connect to it on port 9502 using an encrypted WebSocket connection.
2. authenticate to it by providing a `xivo-auth` token that has the `websocketd` ACL. If you don't know how to obtain a `xivo-auth` token from your XiVO, consult the [documentation on xivo-auth](#).

For example, if you want to use the service located at `example.org` with the token `some-token-id`, you would use the URL `wss://example.org:9502/?token=some-token-id`.

The [SSL/TLS certificate](#) that is used by the WebSocket server is the same as the one used by the XiVO web interface and the REST APIs. By default, this is a self-signed certificate, and web browsers will prevent connections from being successfully established for security reasons. On most web browsers, this can be circumvented by first visiting the `https://<xivo-ip>:9502/` URL and adding a security exception. Other solutions to this problem are described in the [connection section](#).

After a succesful connection and authentication to the service, the server will send the following message:

```
{ "op": "init", "code": 0, "msg": "" }
```

This indicate that the server is ready to accept more commands from the client. Had an error happened, the server would have closed the connection, possibly with one of the [service specific WebSocket close code](#).

The message you see is part of the small [JSON-based protocol](#) that is used for the client/server interaction.

To receive events on your WebSocket connection, you need to tell the server which type of events you are interested in, and then tell it to start sending you these events. For example, if you are interested in the *“endpoint_status_update”* events, you send the following command:

```
{ "op": "subscribe", "data": { "event_name": "endpoint_status_update" } }
```

If all goes well, the server will respond with:

```
{ "op": "subscribe", "code": 0, "msg": "" }
```

Once you have subscribed to all the events you are interested in, you ask the server to start sending you the matching events by sending the following command:

```
{ "op": "start" }
```

The server will respond with:

```
{ "op": "start", "code": 0, "msg": "" }
```

Once you have received this message, all the other messages you'll receive will be messages originating from the bus, in the same format as they were on the bus.

Example Here's a rudimentary example of a web page accessing the service:

```

1  <!DOCTYPE html>
2  <html>
3  <head>
4      <meta charset="utf-8">
5      <title>XiVO WebSocket Example</title>
6  <script>
7      var socket = null;
8      var started = false;
9
10     function connect() {
11         if (socket != null) {
12             console.log("socket already connected");
13             return;
14         }
15
16         var host = document.getElementById("host").value;
17         var token_id = document.getElementById("token").value;
18         socket = new WebSocket("wss://" + host + ":9502/?token=" + token_id);
19         socket.onclose = function(event) {
20             socket = null;
21             console.log("websocketd closed with code " + event.code + " and reason '" + event.reason + "'");
22         };
23         socket.onmessage = function(event) {
24             if (started) {
25                 console.log("message received: " + event.data);
26                 return;
27             }
28
29             var msg = JSON.parse(event.data);
30             switch (msg.op) {
31                 case "init":
32                     subscribe("*");
33                     start();
34                     break;
35                 case "start":
36                     started = true;
37                     console.log("waiting for messages");
38                     break;
39             }
40         };
41         started = false;
42     }
43
44     function subscribe(event_name) {
45         var msg = {
46             op: "subscribe",
47             data: {
48                 event_name: event_name
49             }
50         };
51         socket.send(JSON.stringify(msg));
52     };
53
54     function start() {
55         var msg = {
56             op: "start"
57         };
58         socket.send(JSON.stringify(msg));
59     }
60 </script>
61 </head>

```

```

62 <body>
63   <p>Open the web console to see what's happening.</p>
64   <form>
65     <div>
66       <label for="host">Host:</label>
67       <input type="text" id="host" autofocus>
68     </div>
69     <div>
70       <label for="token">Token ID:</label>
71       <input type="text" id="token" size="35">
72     </div>
73     <div>
74       <button type="button" onclick="connect();">Connect</button>
75     </div>
76   </form>
77 </body>
78 </html>

```

The page has a form for the user to enter a host and token ID, and has a connect button. When the button is clicked, the `connect` function is called, and the WebSocket connection is created at line 18 (using the [WebSocket API](#)):

```
socket = new WebSocket("wss://" + host + ":9502/?token=" + token_id);
```

Then, at line 23, a `onmessage` callback is set on the WebSocket object:

```

socket.onmessage = function(event) {
  if (started) {
    console.log("message received: " + event.data);
    return;
  }

  var msg = JSON.parse(event.data);
  switch (msg.op) {
    case "init":
      subscribe("endpoint_status_update");
      subscribe("user_status_update");
      start();
      break;
    case "start":
      started = true;
      console.log("waiting for messages");
      break;
  }
};

```

After a successful connection to the service, an “init” message will be received by the client. When the client receives this message, it sends two subscribe commands (e.g. `subscribe("endpoint_status_update")`) and a start command (e.g. `start()`). When the client receives the “start” message, it sets the `started` flag. After that, all the other messages it receives will be logged to the console.

Reference The WebSocket service is provided by `xivo-websocketd`, and its behaviour can be configured via its [configuration files](#) located under the `/etc/xivo-websocketd` directory. After modifying the configuration files, you need to restart `xivo-websocketd` with `systemctl restart xivo-websocketd`.

Connection The service is available on port 9502 on all network interfaces by default. This can be changed in the configuration file.

The canonical URL to reach the service is `wss://<host>:9502/`.

The connection is always encrypted. The certificate and private key used by the server can be changed in the configuration file. By default, since the certificate is self-signed, you’ll have to either:

- add a security exception on the client machines that access the service
- use a certificate signed by an untrusted CA and add the CA bundle on the system that access the service
- use a trusted certificate

See the [HTTPS certificate](#) section for more information on certificate configuration.

Authentication Authentication is done by passing a xivo-auth token ID in the `token` query parameter. Authentication is mandatory.

The token must have the `websocketd` ACL.

When the token expires, the server close the connection with the status code 4003. There is currently no way to change the token of an existing connection. A new connection must be made when the token expires.

Events Access Control Clients connected to `xivo-websocketd` only receive events that they are authorized to receive. For example, a client connected with a token obtained from the “xivo_user” `xivo-auth` backend will *not* receive call events of other users.

When a message is received from the bus by `xivo-websocketd`, it extracts the ACL from the `required_acl` key of the event. If the field is missing, no clients will receive the event. If the value is null, all subscribed clients will receive the event. If the value is a string, then all subscribed clients which have a matching ACL will receive the event.

No authorization check is done at subscription time. Checks are only done when an event is received by the server. This mean a client can subscribe to an event “foo”, but will never receive any of these events if it does not have the matching ACL.

See the [Events](#) section for more information on the required ACL of events which are available by default on XiVO.

Status Code The WebSocket connection might be closed by the server using one of following status code:

- 4001: No token ID was provided.
- 4002: Authentication failed. Either the token ID is invalid, expired, or does not have the necessary ACL.
- 4003: Authentication expired. The token has expired or was deleted.
- 4004: Protocol error. The server received a frame that it could not understand. For example, the content was not valid JSON, or was requesting an unknown operation, or a mandatory argument to an operation was missing.

The server also uses the [pre-defined WebSocket status codes](#).

Protocol A JSON-based protocol is used over the WebSocket connection to control which events are received by the client.

Client Messages The format of the messages sent by the client are all of the same format:

```
{ "op": "<operation-name>", "data": <operation-specific-value> }
```

The “op” key is mandatory, and the value is either “subscribe” or “start”. The “data” key is mandatory for the “subscribe” operation.

The “subscribe” message ask the server to subscribe the client to the given event. When a message with the same name is published on the “xivo” exchange of the bus, the server forwards the message to all the subscribed clients that are authorized to receive it. For this command, the “data” value is a dictionary with an “event_name” key (mandatory). Example:

```
{ "op": "subscribe", "data": { "event_name": "endpoint_status_update" } }
```

You can subscribe to any event. The special event name `*` can be used to match all events.

See the [Events](#) section for more information on the events which are available by default on XiVO.

The “start” message ask the server to start sending messages from the bus to the client. Example:

```
{ "op": "start" }
```

The server won’t forward messages from the bus to the client until it receives the “start” message from the client.

If the client send a message that the server doesn’t understand, the server closes the connection.

Server Messages The format of the messages sent by the server are all of the same format (until the server receives a “start” command):

```
{ "op": "<operation-name>", "code": <status-code>, "msg": "<error message>" }
```

The 3 keys are always present. The value of the “op” key can be one of “init”, “subscribe” or “start”. The value of the “code” key is an integer representing the status of the operation, 0 meaning there was no error, other values meaning there was an error. The “msg” is an empty string unless “code” is non-zero, in which case it’s a human-readable message of the error.

The “init” message is only sent after the connection is successfully established between the client and the server. It’s code is always zero; if the connection could not be established, the connection is simply closed. Example:

```
{ "op": "init", "code": 0, "msg": "" }
```

The “subscribe” message is sent as a response to a client “subscribe” message. The code is always zero. Example:

```
{ "op": "subscribe", "code": 0, "msg": "" }
```

The “start” message is sent as a response to a client “start” message. The code is always zero. Example:

```
{ "op": "start", "code": 0, "msg": "" }
```

After receiving the “start” message, the server switch into the “bus/started” mode, where all messages that the server will ever sent will be the body of the messages it received on the bus on behalf of the client.

Note that a client can subscribe to more events after sending its “start” message, but it won’t receive any response from the server, e.g. the server won’t send a corresponding “subscribe” message. Said differently, once the client has sent a “start” message, every message the client will ever receive are messages coming from the bus.

Contributors

General information:

Contributing to the Documentation

XiVO documentation is generated with Sphinx. The source code is available on GitHub at <https://github.com/xivo-pbx/xivo-doc>

Provided you already have Python installed on your system. You need first to install [Sphinx](#) : `easy_install -U Sphinx`¹.

Quick Reference

- <http://docutils.sourceforge.net/docs/user/rst/cheatsheet.txt>
- <http://docutils.sourceforge.net/docs/user/rst/quickref.html>
- http://openalea.gforge.inria.fr/doc/openalea/doc/_build/html/source/sphinx/rest_syntax.html

¹ `easy_install` can be found in the debian package `python-setuptools` : `sudo apt-get install python-setuptools`

Documentation guideline Here's the guideline/conventions to follow for the XiVO documentation.

Language The documentation must be written in english, and only in english.

Sections The top section of each file must be capitalized using the following rule: capitalization of all words, except for articles, prepositions, conjunctions, and forms of to be.

Correct:

```
The Vitamins are in My Fresh California Raisins
```

Incorrect:

```
The Vitamins Are In My Fresh California Raisins
```

Use the following punctuation characters:

- * with overline, for “file title”
- =, for sections
- -, for subsections
- ^, for subsubsections

Punctuation characters should be exactly as long as the section text.

Correct:

```
Section1  
=====
```

Incorrect:

```
Section2  
=====
```

There should be 2 empty lines between sections, except when an empty section is followed by another section.

Correct:

```
Section1  
=====
```

Foo.

```
Section2  
=====
```

Bar.

Correct:

```
Section1  
=====
```

Foo.

```
.. _target:
```

```
Section2  
=====
```

Bar.

Correct:

```
Section1
=====

Subsection1
-----

Foo.
```

Incorrect:

```
Section1
=====

Foo.

Section2
=====

Bar.
```

Lists Bullet lists:

```
* First item
* Second item
```

Autonumbered lists:

```
#. First item
#. Second item
```

Literal blocks Use `::` on the same line as the line containing text when possible.

The literal blocks must be indented with three spaces.

Correct:

```
Bla bla bla::
    apt-get update
```

Incorrect:

```
Bla bla bla:

::
    apt-get update
```

Inline markup Use the following roles when applicable:

- `:file:` for file, i.e.:

```
The :file:`/dev/null` file.
```

- `:menuselection:` for the web interface menu:

```
The :menuselection:`Configuration --> Management --> Certificates` page.
```

- `:guilabel:` for designating a specific GUI element:

The `:guilabel:`Action`` column.

Others

- There must be no warning nor error messages when building the documentation with `make html`.
- There should be one and only one newline character at the end of each file
- There should be no trailing whitespace at the end of lines
- Paragraphs must be wrapped and lines should be at most 100 characters long

Debugging Asterisk

Precondition To debug asterisk crashes or freezes, you need the following debug packages on your XiVO:

General rule	XiVO < 14.18	XiVO >= 14.18
Example version	14.12	14.18
Commands	<pre>apt-get install xivo-fai-14.12 apt-get update apt-get install gdb apt-get install -t xivo-14.12</pre>	<pre>xiVo-dist xivo-14.18 apt-get update apt-get install gdb apt-get install -t xivo-14.18</pre>

So There is a Problem with Asterisk. Now What ?

1. Find out the time of the incident from the people most likely to know
2. Determine if there was a segfault
 - (a) The command `grep segfault /var/log/syslog` should return a line such as the following:

```
Oct 16 16:12:43 xivo-1 kernel: [10295061.047120] asterisk[1255]: segfault at e ip b751aa
```

- (b) Note the exact time of the incident from the segfault line.
 - (c) Follow the [Debugging Asterisk Crash](#) procedure.
3. If you observe some of the following common symptoms, follow the [Debugging Asterisk Freeze](#) procedure.
 - The output of command `service asterisk status` says Asterisk PBX is running.
 - No more calls are distributed and phones go to No Service.
 - Command `core show channels` returns only headers (no data) right before returning
 4. Fetch Asterisk logs for the day of the crash (make sure file was not already logrotated):

```
cp -a /var/log/asterisk/full /var/local/`date +%Y%m%d`-`hostname`-asterisk-full.log
```

5. Fetch xivo-ctid logs for the day of the crash (make sure file was not already logrotated):

```
cp -a /var/log/xivo-ctid.log /var/local/`date +%Y%m%d`-`hostname`-xivo-ctid.log
```

6. Open a new issue on the [bugtracker](#) with following information
 - Tracker: Bug
 - Status: New
 - Category: Asterisk
 - In versions: The version of your XiVO installation where the crash/freeze happened
 - Subject: Asterisk Crash or Asterisk Freeze

- Description : Add as much context as possible, if possible, a scenario that lead to the issue, the date and time of issue, where we can fetch logs and backtrace
- Attach logs and backtrace (if available) to the ticket (issue must be saved, then edited and files attached to a comment).

Debugging Asterisk Crash When asterisk crashes, it usually leaves a core file in `/var/spool/asterisk/`.

You can create a backtrace from a core file named `core_file` with:

```
gdb -batch -ex "bt full" -ex "thread apply all bt" asterisk core_file > bt-threads.txt
```

Debugging Asterisk Freeze You can create a backtrace of a running asterisk process with:

```
gdb -batch -ex "thread apply all bt" asterisk $(pidof asterisk) > bt-threads.txt
```

If your version of asterisk has been compiled with the `DEBUG_THREADS` flag, you can get more information about locks with:

```
asterisk -rx "core show locks" > core-show-locks.txt
```

Note: Debugging freeze without this information is usually a lot more difficult.

Optionally, other information that can be interesting:

- the output of `asterisk -rx 'core show channels'`
- the verbose log of asterisk just before the freeze

Recompiling Asterisk It's relatively straightforward to recompile the asterisk version of your XIVO with the `DEBUG_THREADS` and `DONT_OPTIMIZE` flag, which make debugging an asterisk problem easier.

The steps are:

1. Uncomment the `deb-src` line for the XIVO sources:

```
sed -i 's/^# *deb-src/deb-src/' /etc/apt/sources.list.d/xivo*
```

2. Fetch the asterisk source package:

```
mkdir -p ~/ast-rebuild
cd ~/ast-rebuild
apt-get update
apt-get install -y build-essential
apt-get source asterisk
```

3. Install the build dependencies:

```
apt-get build-dep -y asterisk
```

4. Enable the `DEBUG_THREADS` and `DONT_OPTIMIZE` flag:

```
cd <asterisk-source-folder>
vim debian/rules
```

5. Update the changelog by appending `+debug1` in the package version:

```
vim debian/changelog
```

6. Rebuild the asterisk binary packages:

```
dpkg-buildpackage -us -uc
```

This will create a couple of `.deb` files in the parent directory, which you can install via `dpkg`.

Recompiling a vanilla version of Asterisk It is sometimes useful to produce a “vanilla” version of Asterisk, i.e. a version of Asterisk that has none of the XiVO patches applied, to make sure that the problem is present in the original upstream code. This is also sometimes necessary before opening a ticket on the [Asterisk issue tracker](#).

The procedure is similar to the one described above. Before calling `dpkg-buildpackage`, you just need to:

1. Make sure `quilt` is installed:

```
apt-get install -y quilt
```

2. Unapply all the currently applied patches:

```
quilt pop -a
```

3. Remove all the lines in the `debian/patches/series` file:

```
truncate -s0 debian/patches/series
```

When installing a vanilla version of Asterisk on a XiVO 16.08 or earlier, you’ll need to stop `monit`, otherwise it will restart asterisk every few minutes.

Running Asterisk under Valgrind

1. Install `valgrind`:

```
apt-get install valgrind
```

2. Recompile asterisk with the `DONT_OPTIMIZE` flag.
3. Edit `/etc/asterisk/modules.conf` so that asterisk doesn’t load unnecessary modules. This step is optional. It makes asterisk start (noticeably) faster and often makes the output of `valgrind` easier to analyze, since there’s less noise.
4. Edit `/etc/asterisk/asterisk.conf` and comment the `highpriority` option. This step is optional.
5. Stop `monit` and `asterisk`:

```
monit quit
service asterisk stop
```

6. Stop all unneeded XiVO services. For example, it can be useful to stop `xivo-ctid`, so that it won’t interact with asterisk via the AML.
7. Copy the `valgrind.supp` file into `/tmp`. The `valgrind.supp` file is located in the `contrib` directory of the asterisk source code.
8. Execute `valgrind` in the `/tmp` directory:

```
cd /tmp
valgrind --leak-check=full --log-file=valgrind.txt --suppressions=valgrind.supp --vgdb=no ast
```

Note that when you terminate asterisk with `Control-C`, asterisk does not unload the modules before exiting. What this means is that you might have lots of “possibly lost” memory errors due to that. If you already know which modules is responsible for the memory leak/bug, you should explicitly unload it before terminating asterisk.

Running asterisk under `valgrind` takes a lots of extra memory, so make sure you have enough RAM.

External links

- <https://wiki.asterisk.org/wiki/display/AST/Debugging>
- <http://blog.xivo.io/visualizing-asterisk-deadlocks.html>
- <https://wiki.asterisk.org/wiki/display/AST/Valgrind>

Debugging Daemons

To activate debug mode, add `debug: true` in the daemon *configuration file*). The output will be available in the daemon's *log file*.

It is also possible to run the XiVO daemon, in command line. This will allow to run in foreground and debug mode. To see how to use it, type:

```
xivo-{name} -h
```

Note that it's usually a good idea to stop `monit` before running a daemon in foreground:

```
systemctl stop monit.service
```

xivo-confgend

```
twistd -no -u xivo-confgend -g xivo-confgend --python=/usr/bin/xivo-confgend --logger xivo_confgend
```

No debug mode in `confgend`.

xivo-provd

```
twistd -no -u xivo-provd -g xivo-provd -r epoll --logger provd.main.twistd_logs xivo-provd -s -v
```

- `-s` for logging to stderr
- `-v` for verbose

consul

```
sudo -u consul /usr/bin/consul agent -config-dir /etc/consul/xivo -pid-file /var/run/consul/consul
```

There is no log file, but you can consult the output of `consul` with:

```
consul monitor
```

```
2015/08/03 09:48:25 [INFO] consul: cluster leadership acquired
2015/08/03 09:48:25 [INFO] consul: New leader elected: this-xivo
2015/08/03 09:48:26 [INFO] raft: Disabling EnableSingleNode (bootstrap)
2015/08/03 11:04:08 [INFO] agent.rpc: Accepted client: 127.0.0.1:41545
```

Generate your own prompts

If you want your XiVO to speak in your language that is not supported by XiVO, and you don't want to record the whole package of sounds in a studio, you may generate them yourself with some text-to-speech services.

The following procedure will generate prompts for `pt_BR` (portuguese from Brazil) based on the Google TTS service.

Note: There are two sets of prompts: the *Asterisk prompts* and the XiVO prompts. This procedure only covers the XiVO prompts, but it may be adapted for Asterisk prompts.

1. Create an account on Transifex and join the team of translation of XiVO.
2. Translate the prompts in the `xivo-prompts` resource.
3. Go to https://www.transifex.com/proformatique/xivo/xivo-prompt/pt_BR/download/for_use/ and download the file on your XiVO. You should have a file named like `for_use_xivo_xivo-prompt_pt_BR.ini`.
4. On your XiVO, download the tool to automate the use of Google TTS:

```
wget https://github.com/zaf/asterisk-googlelets/raw/master/cli/googlelets-cli.pl
chmod +x googlelets-cli.pl
```

5. Then run the tool, and generate the sound files (set LANGUAGE and COUNTRY to your own language):

```
LANGUAGE=pt
COUNTRY=BR
mkdir -p wav/{digits,letters}
cat for_use_xivo_xivo-prompt_${LANGUAGE}_${COUNTRY}.ini | while IFS=' ' read file text ; do
    echo $file
    ./googlelets-cli.pl -t "$text" -l ${LANGUAGE}-${COUNTRY} -s 1.4 -r 8000 -o wav/$file.wav
done
```

6. Install the prompts on your system:

```
mv wav /usr/share/asterisk/sounds/${LANGUAGE}_${COUNTRY}
```

7. Make your language available in the web interface:

```
sed -i "s/'nl_NL'/'\0, '${LANGUAGE}_${COUNTRY}'/" /usr/share/xivo-web-interface/lib/i18n.inc
systemctl restart spawn-fcgi
```

Note that this last modification may be erased after running `xivo-upgrade`.

And that's it, you can configure a user to use your new language and he will hear the prompts in your language. You may also want to use the [xivo-confd HTTP API](#) to mass-update your users.

XiVO Guidelines

Inter-process communication Our current goal is to use only two means of communication between XiVO processes:

- a REST API over HTTP for synchronous commands
- a software bus (RabbitMQ) for asynchronous events

Each component should have its own REST API and its own events and can communicate with every other component from across a network only via those means.

Service API The current [xivo-dao](#) Git repository contains the basis of the future services Python API. The API is split between different resources available in XiVO, such as users, groups, schedules... For each resource, there are different modules :

- service: the public module, providing possible actions. It contains only business logic and no technical logic. There must be no file name, no SQL queries and no URLs in this module.
- dao: the private Data Access Object. It knows where to get data and how to update it, such as SQL queries, file names, URLs, but has no business logic.
- model: the public class used to represent the resource. It must be self-contained and have almost no methods, except for computed fields based on other fields in the same object.
- notifier: private, it knows to whom and in which format events must be sent.
- validator: private, it checks input parameters from the service module.

Definition of XiVO Daemon The goal is to make XiVO as elastic as possible, i.e. the XiVO services need to be able to run on separate machines and still talk to each other.

To be in accordance with our goal, a XiVO daemon must (if applicable):

- Offer a REST API (with encryption, authentication and accepting cross-site requests)
- Be able to read and send events on a software bus

- Be able to run inside a container, such as Docker, and be separated from the XiVO server
- Offer a configuration file in YAML format.
- Access the XiVO database through the `xivo-dao` library
- Have a configurable level of logging
- Have its own log file
- Be extendable through the use of plugins
- Not run with system privileges
- Be installable from source
- Service discovery with consul

Currently, none of the XiVO daemons meet these expectations; it is a work in progress.

Network

Configuration for daemon Network Flow table (IN) :

Daemon Name	Service	Protocol	Port	Listen	Authentication	Enabled
-	ICMP	ICMP	-	0.0.0.0	no	yes
postfix	SMTP	TCP	25	0.0.0.0	yes	yes
isc-dhcpd	DHCP	UDP	67	0.0.0.0	no	no
isc-dhcpd	DHCP	UDP	68	0.0.0.0	no	no
xivo-provd	TFTP	UDP	69	0.0.0.0	no	yes
ntpd	NTP	UDP	123	0.0.0.0	yes	yes
monit	HTTP	TCP	2812	127.0.0.1	no	yes
asterisk	SIP	UDP	5060	0.0.0.0	yes	yes
asterisk	IAX	UDP	4569	0.0.0.0	yes	yes
asterisk	SCCP	TCP	2000	0.0.0.0	yes	yes
asterisk	AMI	TCP	5038	127.0.0.1	yes	yes
asterisk	HTTP	TCP	5039	127.0.0.1	yes	yes
asterisk	HTTPS	TCP	5040	127.0.0.1	yes	yes
sshd	SSH	TCP	22	0.0.0.0	yes	yes
nginx	HTTP	TCP	80	0.0.0.0	yes	yes
nginx	HTTPS	TCP	443	0.0.0.0	yes	yes
munin	HTTP	TCP	4949	127.0.0.1	no	yes
xivo-ctid	XiVO-CTI/S	TCP	5003	0.0.0.0	yes	yes
postgresql	SQL	TCP	5432	127.0.0.1	yes	yes
rabbitMQ	AMQP	TCP	5672	0.0.0.0	yes	yes
consul	Consul RPC	TCP	8300	127.0.0.1	yes	yes
consul	Consul Serf LAN	TCP/UDP	8301	127.0.0.1	yes	yes
consul	Consul Serf WAN	TCP/UDP	8302	127.0.0.1	yes	yes
consul	Consul HTTPS	TCP	8500	127.0.0.1	both	yes
xivo-provd	HTTP	TCP	8666	127.0.0.1	no	yes
xivo-provd	HTTP	TCP	8667	0.0.0.0	no	yes
xivo-confgend	HTTP	TCP	8669	127.0.0.1	no	yes
xivo-sysconfd	HTTP	TCP	8668	127.0.0.1	no	yes
xivo-confd	HTTPS	TCP	9486	0.0.0.0	yes	yes
xivo-confd	HTTP	TCP	9487	127.0.0.1	no	yes
xivo-dird	HTTPS	TCP	9489	0.0.0.0	yes	yes
xivo-amid	HTTPS	TCP	9491	0.0.0.0	yes	yes
xivo-agentd	HTTPS	TCP	9493	0.0.0.0	yes	yes
xivo-ctid	HTTP	TCP	9495	127.0.0.1	no	yes

Continued on next page

Table 1.9 – continued from previous page

Daemon Name	Service	Protocol	Port	Listen	Authentication	Enabled
xivo-auth	HTTPS	TCP	9497	0.0.0.0	both	yes
xivo-dird-phoned	HTTP	TCP	9498	0.0.0.0	IP filtering	yes
xivo-dird-phoned	HTTPS	TCP	9499	0.0.0.0	IP filtering	yes
xivo-ctid-ng	HTTPS	TCP	9500	0.0.0.0	yes	yes
xivo-websocketd	WSS	TCP	9502	0.0.0.0	yes	yes

Debian packaging for XiVO

Adding a package from backports

1. Download the package:

```
apt-get download name-of-package/jessie-backports
```

2. Copy the .deb on to the mirror:

```
scp name-of-package.deb mirror.xivo.solutions:/tmp
```

3. Add package to distribution on mirror:

```
ssh mirror.xivo.solutions
cd /data/reprepo/xivo
reprepro includedeb xivo-dev /tmp/name-of-package.deb
```

Profiling Python Programs

Profiling CPU/Time Usage Here’s an example on how to profile xivo-ctid for CPU/time usage:

1. Stop the monit daemon:

```
service monit stop
```

2. Stop the process you want to profile, i.e. xivo-ctid:

```
service xivo-ctid stop
```

3. Start the service in foreground mode running with the profiler:

```
python -m cProfile -o test.profile /usr/bin/xivo-ctid -f
```

This will create a file named `test.profile` when the process terminates.

To profile xivo-confend, you must use this command instead of the one above:

```
twistd -p test.profile --profiler=cprofile --savestats -no --python=/usr/bin/xivo-confend
```

Note that profiling multi-threaded program (xivo-agid, xivo-confd) doesn’t work reliably.

The [Debugging Daemons](#) section documents how to launch the various XiVO services in foreground/debug mode.

4. Examine the result of the profiling:

```
$ python -m pstats test.profile
Welcome to the profile statistics browser.
% sort time
% stats 15
...
% sort cumulative
% stats 15
```

Measuring Code Coverage Here's an example on how to measure the code coverage of xivo-ctid.

This can be useful when you suspect a piece of code to be unused and you want to have additional information about it.

1. Install the following packages:

```
apt-get install python-pip build-essential python-dev
```

2. Install coverage via pip:

```
pip install coverage
```

3. Run the program in foreground mode with `coverage run`:

```
service monit stop
service xivo-ctid stop
coverage erase
coverage run /usr/bin/xivo-ctid -f
```

The *Debugging Daemons* section documents how to launch the various XiVO service in foreground/debug mode.

4. After the process terminates, use `coverage html` to generate an HTML coverage report:

```
coverage html --include='*xivo_cti*'
```

This will generate an `htmlcov` directory in the current directory.

5. Browse the coverage report.

Either copy the directory onto your computer and open it with a web browser, or start a web server on the XiVO:

```
cd htmlcov
python -m SimpleHTTPServer
```

Then open the page from your computer (i.e. not on the xivo):

```
firefox http://<xivo-hostname>:8000
```

External Links

- [Official python documentation](#)
- [PyMOTW](#)
- [coverage.py](#)

Style Guide

Syntax

License Python files start with a UTF8 encoding comment and the GPLv3 license. A blank line should separate the license from the imports

Example:

```
# -*- coding: utf-8 -*-
# Copyright 2016 AvenCALL
# SPDX-License-Identifier: GPL-3.0+

import argparse
```

Spacing

- Lines should not go further than 80 to 100 characters.
- In python, indentation blocks use 4 spaces
- In PHP, indentation blocks use tabs
- Imports should be ordered alphabetically
- Separate module imports and `from` imports with a blank line

Example:

```
import argparse
import datetime
import os
import re
import shutil
import tempfile

from StringIO import StringIO
from urllib import urlencode
```

PEP8 When possible, use pep8 to validate your code. Generally, the following errors are ignored :

- E501 (max 80 chars per line)

Example:

```
pep8 --ignore=E501 xivo_cti
```

When possible, avoid using backslashes to separate lines.

Bad Example:

```
user = session.query(User).filter(User.firstname == firstname)\
    .filter(User.lastname == lastname)\
    .filter(User.number == number)\
    .all()
```

Good Example:

```
user = (session.query(User).filter(User.firstname == firstname)
    .filter(User.lastname == lastname)
    .filter(User.number == number)
    .all())
```

Strings Avoid using the `+` operator for concatenating strings. Use string interpolation instead.

Bad Example:

```
phone_interface = "SIP" + "/" + username + "-" + password
```

Good Example:

```
phone_interface = "SIP/%s-%s" % (username, password)
```

Comments Redundant comments should be avoided. Instead, effort should be put on making the code clearer.

Bad Example:

```
#Add the meeting to the calendar only if it was created on a week day
#(monday to friday)
if meeting.day > 0 and meeting.day < 7:
    calendar.add(meeting)
```


Good Example:

```
def created_on_week_day(meeting):
    return meeting.day > 0 and meeting.day < 7

if created_on_week_day(meeting):
    calendar.add(meeting)
```

Conditions Avoid using parenthesis around if statements, unless the statement expands on multiple lines or you need to nest your conditions.

Bad Examples:

```
if(x == 3):
    print "condition is true"

if(x == 3 and y == 4):
    print "condition is true"
```

Good Examples:

```
if x == 3:
    print "condition is true"

if x == 3 and y == 4:
    print "condition is true"

if (extremely_long_variable == 3
    and another_long_variable == 4
    and yet_another_variable == 5):

    print "condition is true"

if (2 + 3 + 4) - (1 + 1 + 1) == 6:
    print "condition is true"
```

Consider refactoring your statement into a function if it becomes too long, or the meaning isn't clear.

Bad Example:

```
if price * tax - bonus / reduction + fee < money:
    product.pay(money)
```

Good Example:

```
def calculate_price(price, tax, bonus, reduction, fee):
    return price * tax - bonus / reduction + fee

final_price = calculate_price(price, tax, bonus, reduction, fee)

if final_price < money:
    product.pay(money)
```

Naming

- Class names are in CamelCase
- File names are in lower_underscore_case

Conventions for functions prefixed by *find*:

- Return None when nothing is found
- Return an object when a single entity is found

- Return the first element when multiple entities are found

Example:

```
def find_by_username(username):
    users = [user1, user2, user3]
    user_search = [user for user in users if user.username == username]

    if len(user_search) == 0:
        return None

    return user_search[0]
```

Conventions for functions prefixed by *get*:

- Raise an Exception when nothing is found
- Return an object when a single entity is found
- Return the first element when multiple entities are found

Example:

```
def get_user(userid):
    users = [user1, user2, user3]
    user_search = [user for user in users if user.userid == userid]

    if len(user_search) == 0:
        raise UserNotFoundError(userid)

    return user_search[0]
```

Conventions for functions prefixed by *find_all*:

- Return an empty list when nothing is found
- Return a list of objects when multiple entites are found

Example:

```
def find_all_users_by_username(username):
    users = [user1, user2, user3]
    user_search = [user for user in users if user.username == username]

    return user_search
```

Magic numbers Magic numbers should be avoided. Arbitrary values should be assigned to variables with a clear name

Bad example:

```
class TestRanking(unittest.TestCase):

    def test_ranking(self):
        rank = Rank(1, 2, 3)

        self.assertEqual(rank.position, 1)
        self.assertEqual(rank.grade, 2)
        self.assertEqual(rank.session, 3)
```

Good example:

```
class TestRanking(unittest.TestCase):

    def test_ranking(self):
        position = 1
```

```

grade = 2
session = 3

rank = Rank(position, grade, session)

self.assertEqual(rank.position, position)
self.assertEqual(rank.grade, grade)
self.assertEqual(rank.session, session)

```

Tests Tests for a package are placed in their own folder named “tests” inside the package.

Example:

```

package1/
__init__.py
mod1.py
tests/
    __init__.py
    test_mod1.py
package2/
__init__.py
mod9.py
tests/
    __init__.py
    test_mod9.py

```

Unit tests should be short, clear and concise in order to make the test easy to understand. A unit test is separated into 3 sections :

- Preconditions / Preparations
- Thing to test
- Assertions

Sections are separated by a blank line. Sections that become too big should be split into smaller functions.

Example:

```

class UserTestCase(unittest.TestCase):

    def test_fullname(self):
        user = User(firstname='Bob', lastname='Marley')
        expected = 'Bob Marley'

        fullname = user.fullname()

        self.assertEqual(expected, fullname)

    def _prepare_expected_user(self, firstname, lastname, number):
        user = User()
        user.firstname = firstname
        user.lastname = lastname
        user.number = number

        return user

    def _assert_users_are_equal(self, expected_user, actual_user):
        self.assertEqual(expected_user.firstname, actual_user.firstname)
        self.assertEqual(expected_user.lastname, actual_user.lastname)
        self.assertEqual(expected_user.number, actual_user.number)

    def test_create_user(self):
        expected = self._prepare_expected_user('Bob', 'Marley', '4185551234')

```

```
user = create_user('Bob', 'Marley', '4185551234')

self._assert_users_are_equal(expected, user)
```

Exceptions Exceptions should not be used for flow control. Raise exceptions only for edge cases, or when something that isn't usually expected happens.

Bad Example:

```
def is_user_available(user):
    if user.available():
        return True
    else:
        raise Exception("User isn't available")

try:
    is_user_available(user)
except Exception:
    disable_user(user)
```

Good Example:

```
def is_user_available(user):
    if user.available():
        return True
    else:
        return False

if not is_user_available(user):
    disable_user(user)
```

Avoid throwing Exception. Use one of Python's built-in Exceptions, or create your own custom Exception. A list of exceptions is available on [the Python documentation website](#).

Bad Example:

```
def get_user(userid):
    user = session.query(User).get(userid)

    if not user:
        raise Exception("User not found")
```

Good Example:

```
class UserNotFoundError(LookupError):

    def __init__(self, userid):
        message = "user with id %s not found" % userid
        LookupError.__init__(self, message)

def get_user(userid):
    user = session.query(User).get(userid)

    if not user:
        raise UserNotFoundError(userid)
```

Never use `except:` without specifying any exception type. The reason is that it will also catch important exceptions, such as `KeyboardInterrupt` and `OutOfMemory` exceptions, making your program unstopable or continuously failing, instead of stopping when wanted.

Bad Example:

```
try:
    get_user(user_id)
except:
    logger.exception("There was an error")
```

Good Example:

```
try:
    get_user(user_id)
except UserNotFoundError as e:
    logger.error(e.message)
    raise
```

Translating XiVO

French and English are maintained by Avencall. Other languages are provided by the community.

Asterisk and XiVO Prompts Avencall is in contact with several studios for different languages and prompts. The information for those languages are :

- French : Super Sonic productions (supersonicprod@wanadoo.fr)
- English : Asterisk voice (allison@theasteriskvoice.com)
- German : ATS studio
- Italian : ATS studio

Prompts transcripts are listed in [Transifex](#) (*-prompts). You may translate them there.

The prompts used in XiVO are stored in [xivo-sounds](#) git repository. You may also want to *generate your own sound files*.

XiVO Client All translations are in [Transifex](#) (xivo-client). The source language is English. Translations are synchronised with the code before every release.

Web Interface Translations are currently available in French and English. There are no plans to translate the Web interface in other languages.

XiVO Package File Structure

Package naming Let's assume we want to organise the files for xivo-confd.

- Git repo name: xivo-confd
- Binary file name: xivo-confd
- Python package name: xivo_confd

```
xivo-confd
|-- bin
|   |-- xivo-confd
|-- contribs
|   |-- docker
|       |-- ...
|           |-- prod
|               |-- ...
|-- debian
|   |-- ...
|-- Dockerfile
|-- docs
```

```
|  `-- ...
|-- etc
|  `-- ...
|-- integration-tests
|  `-- ...
|-- LICENSE
|-- README.md
|-- requirements.txt
|-- setup.cfg
|-- setup.py
|-- test-requirements.txt
|-- .travis.yml
`-- xivo_confd
    `-- ...
```

Sources

etc/ Contains default configuration files.

docs/ Contains technical documentation for this package: API doc, architecture doc, diagrams, ... Should be in RST format using Sphinx.

bin/ Contains the binaries. Not applicable for pure libraries.

integration-tests/ Contains the tests bigger than unit-tests. Tests should be runnable simply, e.g. `nosetests integration-tests`.

README.md Read me in markdown (Github flavor).

LICENSE License (GPLv3)

.travis.yml Travis CI configuration file

Python Standard files:

- setup.py
- setup.cfg
- requirements.txt
- test-requirements.txt
- xivo_confd/ (the main sources)

Debian

debian/ Contains the Debian packaging files (control, rules, ...)

Docker

Dockerfile Used to build a docker image for a working production version

contribs/docker/prod/ Contains the files necessary for running xivo-confd inside a production Docker image

contribs/docker/other/ Contains the Dockerfile and other files to run xivo-confd inside Docker with specific configuration

File naming

- PID file: `/var/run/xivo-confd/xivo-confd.pid`
- WSGI socket file: `/var/run/xivo-confd/xivo-confd.sock`
- Config file: `/etc/xivo-confd/config.yml`
- Log file: `/var/log/xivo-confd.log`
- Static data files: `/usr/share/xivo-confd`
- Storage data files: `/var/lib/xivo-confd`

Component specific information:

CTI Server

This section describes the informations and tools for CTI Server.

CTI Proxy Here's how to run the various CTI client-server development/debugging tools. These tools can be found on GitHub, in the [XiVO project](#).

You can get the scripts by using Git:

```
$ git clone https://github.com/xivo-pbx/xivo-tools.git
```

General Information Both the `ctispy`, `ctisave` and `ctistat` tools work in a similar way. They both are proxies that need to be inserted between the CTI client and the CTI server message flow.

To do this, you first start the given tool on your development machine, giving it the CTI server hostname as the first argument. You then configure your CTI client to connect to the tool on port 50030 (notice the trailing 0). The tool should then accept the connection from the client, and once this is done, will make a connection to the server, thereby being able to process all the information sent between the client and the server.

In the following examples, we suppose that the CTI server is located on the host named `xivo-new`.

Tools

ctispy `ctispy` can be used to see the message flow between the client and the server in “real-time”.

The simplest invocation is:

```
$ cti-proxy/ctispy xivo-new
```

You can pretty-print the messages if you want by using the `--pretty-print` option:

```
$ cti-proxy/ctispy xivo-new --pretty-print
```

By default, each message is displayed separately even though more than one message can be in a single TCP packet. You can also use the `--raw` option if you want to see the raw traffic between the client and the server:

```
$ cti-proxy/ctispy xivo-new --raw
```

Note that when using the `--raw` option, some other option doesn't work because the messages are not decoded/analyzed.

If you want to remove some fields from the messages, you can use the `--strip` option:

```
$ cti-proxy/ctispy xivo-new --strip timenow --strip commandid --strip replyid
```

If you want to see only messages matching a certain key and value, use the `--include` option:

```
$ cti-proxy/ctispy xivo-new --include class=getlist
```

Finally, you can ignore all the messages from the client or the server by using the `--no-client` or `--no-server` option respectively.

By default, ctispy will exit after the connection with the client is closed. You can bypass this behavior with the `--loop` option, that will make the CTI proxy continue, whether the client is connected or not.

ctisave ctisave save the messages from the client and the server in two separate files. This is useful to do more careful post-analysis.

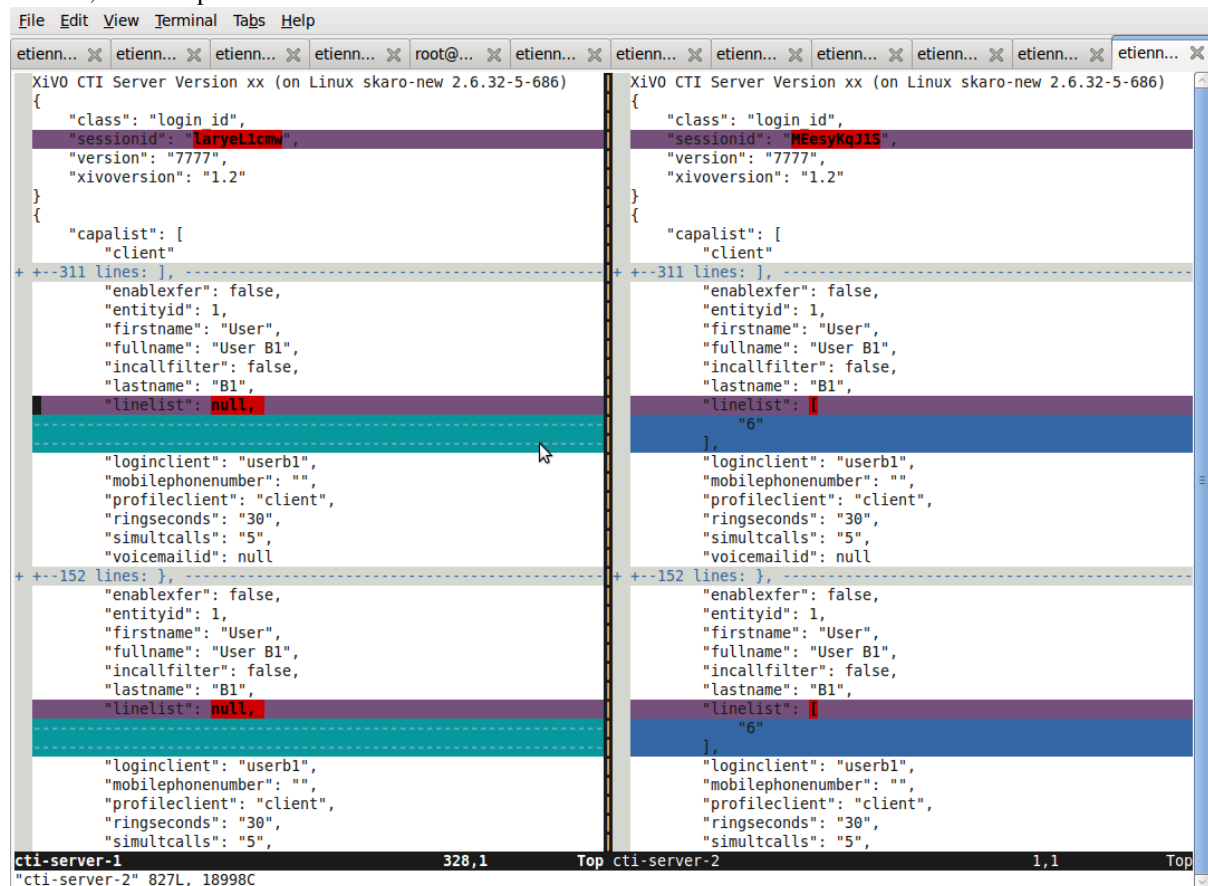
The simplest invocation is:

```
$ cti-proxy/ctisave xivo-new /tmp/cti-client /tmp/cti-server
```

To do comparison, it's often useful to strip some fields:

```
$ cti-proxy/ctisave xivo-new /tmp/cti-client /tmp/cti-server --strip timenow
--strip commandid --strip replyid
```

One useful thing to do with files generated from different ctisave invocation is to compare them with a tool like vimdiff, for example:



ctistat ctistat display various statistic about a CTI “session” when it ends.

The simplest invocation is:

```
$ cti-proxy/ctistat xivo-new
```

CTI Protocol

Protocol Changelog The versions below indicate the xivo version followed by the protocol version.

Warning: The CTI server protocol is subject to change without any prior warning. If you are using this protocol in your own tools please be sure to check that the protocol did not change before upgrading XiVO

16.11 - 2.2

- the *user_id* field has been added back to the *User status update*

16.09 - 2.2

- the *Register user status update* now uses the *user_uuid* instead of the *user_id*
- the *User status update* now uses the *user_uuid* instead of the *user_id*

16.04 - 2.1

- the *Chitchat* command *to* and *from* fields are now a list of two strings, *xivo_uuid* and *user_uuid*.

16.01 - 2.0

- the *lastconnswins* field has been removed from the *Login capas* command
- the *loginkind* field has been removed from the *Login capas* command
- the *ipbxcommands* and *regcommands* capakinds have been removed from *Login capas* command
- the *Login password* command has been modified. The *hashedpassword* has been replaced by the *password* field which is now sent verbatim.

15.20 - 1.2

- the *STARTTLS* command has been added

15.19 - 1.2

- the *Chitchat* command *to* field is now a list of two elements, *xivo_uuid* and *user_id*.
- the *getList* command has been removed for the *channels* listname.
- many fields have been removed from the *getList* command.
 - users list
 - * *enableclient*
 - * *profileclient*
 - phones
 - * *context*
 - * *protocol*
 - * *simultcalls*
 - * *channels*
 - voicemails
 - * *email*
 - * *fullname*
 - * *old*

- * waiting
 - agents
 - * phonenumber
- some ipbxcommands have been removed:
 - mailboxcount
 - atxfer
 - transfer
 - hangup
 - originate

15.18 - 1.2

- add the *Attended transfer to voicemail* command
- add the *Blind transfer to voicemail* command
- the *Send fax* command now include the size and data field.
- the *filetransfer* command has been removed.

15.16 - 1.2

- the *Get relations* command was added.
- the *Relations* message was added.

15.14 - 1.2

- the `people_purge_personal_contacts` message was added.
- the `people_personal_contacts_purged` message was added.
- the `people_personal_contact_raw` message was added.
- the `people_personal_contact_raw_result` message was added.
- the `people_edit_personal_contact` message was added.
- the `people_personal_contact_raw_update` message was added.
- the `people_import_personal_contacts_csv` message was added.
- the `people_import_personal_contacts_csv_result` message was added.
- the `people_export_personal_contacts_csv` message was added.
- the `people_export_personal_contacts_csv_result` message was added.
- for messages `people_personal_contact_deleted` and `people_favorite_update` there are no longer data sub-key.

15.13 - 1.2

- for channel status update message:
 - the value of `commstatus` have been changed from `linked-caller` and `linked-called` to `linked`.
 - the key `direction` have been removed.
 - the key `talkingto_kind` have been removed.

- the `people_personal_contacts` message was added.
- the `people_personal_contacts_result` message was added.
- the `people_create_personal_contact` message was added.
- the `people_personal_contact_created` message was added.
- the `people_delete_personal_contact` message was added.
- the `people_personal_contact_deleted` message was added.

15.12 - 1.2

- `people_search_result` has a new key in `relations`: `source_entry_id`
- the `people_favorites` message was added.
- the `people_favorites_result` message was added.
- the `people_set_favorite` message was added.
- the `people_favorite_update` message was added.

15.11 - 1.2

- the `fax_progress` message was added.

15.09 - 1.2

- for messages of class `history` the client cannot request by mode anymore. The server returns all calls and the mode is now metadata for each call.

14.24 - 1.2

- for messages of class `ipbxcommand`, the `command record` and `sipnotify` have been removed.
- the `logfromclient` message has been removed

14.22 - 1.2

- for messages of class `faxsend`, the steps `file_decoded` and `file_converted` have been removed.

14.06 - 1.2

- the `dial_success` message was added

14.05 - 1.2

- the `unhold_switchboard` command was renamed `resume_switchboard`.

13.22 - 1.2

- the `actionfiche` message was renamed `call_form_result`.

13.17 - 1.2

- for messages of class `login_capas` from server to client: the key `presence` has been removed.

13.14 - 1.2

- for messages of class `getlist`, list agents and function `updatestatus`: the key availability in the status object/dictionary has changed values:
 - deleted values: `on_call_non_acd_incoming` and `on_call_non_acd_outgoing`
 - added values: `* on_call_non_acd_incoming_internal *`
`on_call_non_acd_incoming_external` `* on_call_non_acd_outgoing_internal`
`* on_call_non_acd_outgoing_external`

13.12 - 1.2

- for messages of class `getlist`, list agents and function `updatestatus`: the key availability in the status object/dictionary has changed values:
 - deleted value: `on_call_non_acd`
 - added values: `on_call_non_acd_incoming` and `on_call_non_acd_outgoing`

13.10 - 1.2

- for messages of class `getlist` and function `updateconfig`, the config object/dictionary does not have a `rules_order` key anymore.

Commands Objects have the format: “<type>:<xivoid>/<typeid>”

- <type> can take any of the following values: user, agent, queue, phone, group, meetme, ...
- <xivoid> indicates on which server the object is defined
- <typeid> is the object id, type dependant

e.g. `user:xivo-test/5` I'm looking for the user that has the ID 5 on the xivo-test server.

Here is a non exhaustive list of types:

- `exten`
- `user`
- `vm_consult`
- `voicemail`

Agent

Login agent Client -> Server

```
{ "agentphonenumber": "1000", "class": "ipbxcommand", "command": "agentlogin", "commandid": 733366 }
```

`agentphonenumber` is the physical phone set where the agent is going to log on.

Server > Client

- Login successfull :

```
{ "function": "updateconfig",
  "listname": "queuemembers",
  "tipbxid": "xivo",
  "timenow": 1362664323.94,
  "tid": "Agent/2002,blue",
  "config": { "paused": "0",
              "penalty": "0",
              "membership": "static",
```

```

        "status": "1",
        "lastcall": "",
        "interface": "Agent/2002",
        "queue_name": "blue",
        "callstaken": "0"},
    "class": "getlist"}

{"function": "updatestatus",
 "listname": "agents",
 "tipbxid": "xivo",
 "timenow": 1362664323.94,
 "status": {"availability_since": 1362664323.94,
  "queues": [],
  "on_call": false,
  "availability": "available",
  "channel": null},
  "tid": 7,
  "class": "getlist"}

```

- The phone number is already used by an other agent :

```

{"class": "ipbxcommand", "error_string": "agent_login_exten_in_use", "timenow": 1362664158.14}

```

Logout agent Client -> Server

```

{"class": "ipbxcommand", "command": "agentlogout", "commandid": 552759274}

```

Pause On all queues

Client -> Server

```

{"class": "ipbxcommand", "command": "queuepause", "commandid": 859140432, "member": "agent:xivo/1"}

```

Un pause agent On all queues

Client -> Server

```

{"class": "ipbxcommand", "command": "queueunpause", "commandid": 822604987, "member": "agent:xivo/1"}

```

Add an agent in a queue Client -> Server

```

{"class": "ipbxcommand", "command": "queueadd", "commandid": 542766213, "member": "agent:xivo/3"}

```

Remove an agent from a queue Client -> Server

```

{"class": "ipbxcommand", "command": "queueremove", "commandid": 742480296, "member": "agent:xivo/3"}

```

Listen to an agent Client -> Server

```

{"class": "ipbxcommand", "command": "listen", "commandid": 1423579492, "destination": "xivo/1", "member": "agent:xivo/1"}

```

Configuration The following messages are used to retrieve XiVO configuration.

Common fields

- class : getlist
- function : listid
- commandid
- tipbxid
- listname : Name of the list to be retrieved : users, phones, agents, queues, voicemails, queuemembers

```
{
  "class": "getlist",
  "commandid": 489035169,
  "function": "listid",
  "tipbxid": "xivo",
  "listname": "....."
}
```

Users configuration Return a list of configured user id's

Client -> Server

```
{"class": "getlist", "commandid": 489035169, "function": "listid", "listname": "users", "tipbxid": "xivo", "timenow": 1362735061.17}
```

Server -> Client

```
{
  "class": "getlist",
  "function": "listid", "listname": "users",
  "list": ["11", "12", "14", "17", "1", "3", "2", "4", "9"],
  "tipbxid": "xivo", "timenow": 1362735061.17
}
```

User configuration Return a user configuration

- tid is the userid returned by *Users configuration* message

Client -> Server

```
{
  "class": "getlist",
  "function": "updateconfig",
  "listname": "users",
  "tid": "17",
  "tpbxid": "xivo", "commandid": 5}

```

Server -> Client

```
{
  "class": "getlist",
  "function": "updateconfig",
  "listname": "users",
  "tid": "17",
  "tipbxid": "xivo",
  "timenow": 1362741166.4,
  "config": {
    "enablednd": 0, "destrna": "", "enablerna": 0, "enableunc": 0, "destunc": "", "destbusy": 0,
    "firstname": "Alice", "lastname": "Bouzat", "fullname": "Alice Bouzat",
    "voicemailid": null, "incallfilter": 0, "enablevoicemail": 0, "agentid": 2, "linelist": 1
  }
}
```

Phones configuration Client -> Server

```
{"class": "getlist", "commandid": 495252308, "function": "listid", "listname": "phones", "tipbxid": "xivo"}
```

Server > Client

```
{"class": "getlist", "function": "listid", "list": ["1", "3", "2", "5", "14", "7", "6", "9", "8"],  
  "listname": "phones", "timenow": 1364994093.38, "tipbxid": "xivo"}
```

Individual phone configuration request:

```
{"class": "getlist", "commandid": 704096693, "function": "updateconfig", "listname": "phones", "tid": "3", "tipbxid": "xivo"}
```

Server > Client

```
{"class": "getlist",  
  "config": {"allowtransfer": null, "identity": "SIP/ihvbur", "iduserfeatures": 1,  
    "initialized": null, "number": "1000"},  
  "function": "updateconfig", "listname": "phones", "tid": "3", "timenow": 1364994093.43, "tipbxid": "xivo"}
```

Agents configuration Client -> Server

```
{"class": "getlist", "commandid": 1431355191, "function": "listid", "listname": "agents", "tipbxid": "xivo"}
```

Queues configuration Client -> Server

```
{"class": "getlist", "commandid": 719950939, "function": "listid", "listname": "queues", "tipbxid": "xivo"}
```

Server -> Client

```
{"function": "listid", "listname": "queues", "tipbxid": "xivo",  
  "list": ["1", "10", "3", "2", "5", "4", "7", "6", "9", "8"], "timenow": 1382704649.64, "class": "getlist"}
```

Queue configuration tid is the id returned in the list field of the getlist response message

Client -> Server

```
{"commandid": 7, "class": "getlist", "tid": "3", "tipbxid": "xivo", "function": "updateconfig", "listname": "queues", "tipbxid": "xivo"}
```

Server -> Client

```
{  
  "function": "updateconfig", "listname": "queues", "tipbxid": "xivo", "timenow": 1382704649.69, "config":  
    "config":  
      {"displayname": "red", "name": "red", "context": "default", "number": "3002"},  
  "class": "getlist"}
```

Voicemails configuration Client -> Server

```
{"class": "getlist", "commandid": 1034160761, "function": "listid", "listname": "voicemails", "tipbxid": "xivo"}
```

Queue members configuration Client -> Server

```
{"class": "getlist", "commandid": 964899043, "function": "listid", "listname": "queuemembers", "tipbxid": "xivo"}
```

Server -> Client

```
{ "function": "listid", "listname": "queuemembers", "tipbxid": "xivo",  
  "list": [ "Agent/2501,blue", "Agent/2500,yellow", "Agent/2002,yellow", "Agent/2003,___switchboard",  
            "Agent/2003,blue", "Agent/108,blue", "Agent/2002,blue" ],  
  "timenow": 1382717016.23,  
  "class": "getlist" }
```

Fax

Send fax Client -> Server

```
{ "class": "faxsend",  
  "filename": "contract.pdf",  
  "destination", 41400,  
  "size": 100000,  
  "data": "<base64 of the fax content>" }
```

Fax status Server -> Client

- pages: number of pages sent (NULL if FAILED)
- status
 - FAILED: Failed to send fax.
 - PRESENDFAX: Fax number exist and converting pdf->tiff has been done.
 - SUCCESS: Fax sent with success.

```
{ "class": "fax_progress", "status": "SUCCESS", "pages": 2 }
```

Call control commands

Dial

- destination can be any number
- destination can be a pseudo URL of the form “type:ibpx/id”

Client -> Server

```
{  
  "class": "ipbxcommand",  
  "command": "dial",  
  "commandid": <commandid>,  
  "destination": "exten:xivo/<extension>"  
}
```

For example :

```
{  
  "class": "ipbxcommand",  
  "command": "dial",  
  "commandid": 1683305913,  
  "destination": "exten:xivo/1202"  
}
```

The server will answer with either an error or a success:

```
{  
  "class": "ipbxcommand",  
  "error_string": "unreachable_extension:1202",  
}
```



```
{
  "class": "dial_success",
  "exten": "1202"
}
```

Attended transfer to voicemail Transfer the current call to a given voicemail and listen to the message before completing the transfer.

Client -> Server

```
{
  "class": "attended_transfer_voicemail",
  "voicemail": "<voicemail number>"
}
```

Blind transfer to voicemail Transfer the current call to a given voicemail.

Client -> Server

```
{
  "class": "blind_transfer_voicemail",
  "voicemail": "<voicemail number>"
}
```

Login Once the network is connected at the socket level, the login process requires three steps. If one of these steps is omitted, the connection is reset by the cti server.

- login_id, the username is sent as a login to the cti server, cti server answers by giving a sessionid
- login_pass, the password is sent to the cti server, cti server answers by giving a capaid
- login_capas, the capaid is returned to the server with the user's availability, cti server answers with a list of info relevant to the user

```
{
  "commandid": <commandid>,
  "class": "login_id",
}
```

- class: defined what class of command use.
- commandid : a unique integer number.

Login ID Client -> Server

```
{
  "class": "login_id",
  "commandid": 1092130023,
  "company": "default",
  "ident": "X11-LE-24079",
  "lastlogout-datetime": "2013-02-19T11:13:36",
  "lastlogout-stopper": "disconnect",
  "userlogin": <userlogin>,
  "xivoversion": "<cti protocol version>"
}
```

Server -> Client

```
{
  "class": "login_id",
  "sessionid": "21UaGDfst7",
}
```

```
{
  "timenow": 1361268824.64,
  "xivoversion": "<cti protocol version>"
}
```

Note: sessionid is used to calculate the hashed password in next step

Login password Client -> Server

```
{
  "class": "login_pass",
  "password": "secret",
  "commandid": <commandid>
}
```

Server -> Client

```
{
  "capalist": [
    2
  ],
  "class": "login_pass",
  "replyid": 1646064863,
  "timenow": 1361268824.68
}
```

If no CTI profile is defined on XiVO for this user, the following message will be sent:

```
{
  "error_string": "capaid_undefined",
  "class": "login_pass",
  "replyid": 1646064863,
  "timenow": 1361268824.68
}
```

Note: the first element of the capalist is used in the next step login_capas

Login capas Client -> Server

```
{
  "capaid": 3,
  "commandid": <commandid>,
  "state": "available",
  "class": "login_capas"
}
```

Server -> Client

First message, describes all the capabilities of the client, configured at the server level

- presence : actual presence of the user
- userid : the user id, can be used as a reference
- capas
 - **userstatus** [a list of available statuses]
 - * status name
 - * color

- ```
{
 "class": "login_capas"
 "presence": "available",
 "userid": "3",
 "ipbxid": "xivo",
 "timenow": 1361440830.99,
 "replyid": 3,
 "capas": {
 "preferences": false,
 "userstatus": {
 "available": { "color": "#08FD20",
 "allowed": ["available", "away", "outtolunch",
 "actions": {"enablednd": "false"}, "longname"
 },
 "berightback": { "color": "#FFB545",
 "allowed": ["available", "away", "outtolun",
 "actions": {"enablednd": "false"}, "longna"
 },
 "disconnected": { "color": "#202020",
 "actions": {"agentlogoff": ""}, "longname"
 },
 /* a list of other status depends on the cti server configura
 },
 "services": ["fwdrna", "fwdbusy", "fwdunc", "enablednd"],
 "phonestatus": {
 "16": {"color": "#F7FF05", "longname": "En Attente"},
 "1": {"color": "#FF032D", "longname": "En ligne OU appel"},
 "0": {"color": "#0DFF25", "longname": "Disponible"},
 "2": {"color": "#FF0008", "longname": "Occup\u00e9"},
 "-1": {"color": "#000000", "longname": "D\u00e9sactiv\u00e9"},
 "4": {"color": "#FFFFFF", "longname": "Indisponible"},
 "-2": {"color": "#030303", "longname": "Inexistant"},
 "9": {"color": "#FF0526", "longname": "(En Ligne OU Appe"},
 "8": {"color": "#1B0AFF", "longname": "Sonne"}
 }
 },
 "capaxlets": [{"identity", "grid"}, {"search", "tab"}, {"customerinfo", "tab", "1"},
 "appliname": "Client",
}
```

```
{
 "function": "updateconfig",
 "listname": "users",
 "tipbxid": "xivo",
 "timenow": 1361440830.99,
 "tid": "3",
 "config": {"enablednd": false},
 "class": "getlist"
}
```

Third message describes the current user status

```
{
 "function": "updatestatus",
 "listname": "users",
 "status": {"availstate": "available"},
 "tipbxid": "xivo",
 "tid": "3",
 "class": "getlist",
 "timenow": 1361440830.99
}
```

## Others

**call\_form\_result** This message is received when a *call form* is submitted from a client to the XiVO.

Client -> Server

```
{
 "class": "call_form_result",
 "commandid": <commandid>,
 "infos": {"buttonname": "saveandclose",
 "variables": {"XIVOFORM_varname1": "value1",
 "XIVOFORM_varname2": "value2"}}
}
```

## History

- size : Size of the list to be sent by the server

Client -> Server

```
{
 "class": "history",
 "commandid": <commandid>
 "size": "8",
 "xuserid": "<xivoid>/<userfeaturesid>",
}
```

Server > Client

Send back a table of calls :

- duration in seconds
- extension: caller/destination extension
- fullname: caller ID name
- mode
  - 0 : sent calls
  - 1 : received calls
  - 2 : missed calls

```
{
 "class": "history",
 "history": [
 {"calldate": "2013-03-29T08:44:35.273998",
 "duration": 30.148765,
 "extension": "*844201",
 "fullname": "Alice Wonderland",
 "mode": 0},
]
}
```

```

 {"calldate": "2013-03-28T16:56:48.071213",
 "duration": 58.134744,
 "extension": "41400",
 "fullname": "41400"}
 "mode": 1},
],
 "replyid": 529422441,
 "timenow": 1364571477.33
}
```

### Chitchat Client > Server

```

{
 "class": "chitchat",
 "alias": "Alice",
 "text": "Lorem ipsum dolor sit amet, consectetur adipiscing elit. Suspendisse venenatis velit",
 "to": [<xivo_uuid>, <user_uuid>],
 "commandid": <commandid>
}
```

Server > Client

The following message is received by the remote XiVO client

```

{
 "class": "chitchat",
 "from": [<xivo_uuid>, <user_uuid>],
 "to": [<xivo_uuid>, <user_uuid>]
 "alias": "Alice",
 "text": "Lorem ipsum dolor sit amet, consectetur adipiscing elit. Suspendisse venenatis velit"
}
```

**Directory** Request directory information, names matching pattern ignore case.

Client -> Server

```

{
 "class": "directory",
 "commandid": 1079140548,
 "pattern": "pau"
}
```

Server > Client

```

{
 "class": "directory",
 "headers": ["Nom", "Num\u00e9ro", "Mobile", "Autre num\u00e9ro", "E-mail", "Fonction", "Site",
 "replyid": 1079140548,
 "resultlist": ["Claire Mapaurtal;;+33644558899;31256;cmapaurtal@societe.com;;;",
 "Paul Salvadier;+33445236988;+33678521430;31406;psalvadier@societe.com;;;"],
 "status": "ok",
 "timenow": 1378798928.26
}
```

parking

keepalive

availstate

getipbxlist

```
{
 "class": "getipbxlist",
 "commandid": <commandid>
}
```

## People

**Get relations** This command will trigger a *Relations* message.

Client -> Server

```
{
 "class": "get_relations"
}
```

**People headers** Client -> Server

```
{
 "class": "people_headers",
}
```

Server -> Client

```
{
 "class": "people_headers_result",
 "column_headers": ["Status", "Name", "Number"],
 "column_types": [null, null, "number"],
}
```

**People Search** Client -> Server

```
{
 "class": "people_search",
 "pattern": <pattern>,
}
```

Server -> Client

```
{
 "class": "people_search_result",
 "term": "Bob",
 "column_headers": ["Firstname", "Lastname", "Phone number", "Mobile", "Fax", "Email", "Agent"],
 "column_types": [null, "name", "number_office", "number_mobile", "fax", "email", "relation_agent"],
 "results": [
 {
 "column_values": ["Bob", "Marley", "5555555", "5556666", "5553333", "mail@example.com", null],
 "relations": {
 "agent_id": null,
 "user_id": null,
 "endpoint_id": null,
 "source_entry_id": null
 },
 "source": "my_ldap_directory"
 }, {
 "column_values": ["Charlie", "Boblin", "5555556", "5554444", "5552222", "mail2@example.com", null],
 "relations": {
 "agent_id": 12,
 "user_id": 34,
 "endpoint_id": 56,
 "source_entry_id": "34"
 }
 }
]
}
```

```

 },
 "source": "internal"
 }
]
}

```

**Relations** This message can currently only be received as a response to the *Get relations* command.

- The *xivo\_uuid* is the id of the server
- The *user\_id* is the id of the current user.
- The *endpoint\_id* is the id of the line of the current user or null.
- The *agent\_id* is the id of the agent of the current user or null.

Server -> Client

```

{
 "class": "relations",
 "data": { "xivo_uuid": <the xivo uuid>,
 "user_id": <the user id>,
 "endpoint_id": <the endpoint id>,
 "agent_id": <the agent id> }
}

```

**Favorites list** Client -> Server

```

{
 "class": "people_favorites",
}

```

Server -> Client

```

{
 "class": "people_favorites_result",
 "column_headers": ["Firstname", "Lastname", "Phone number", "Mobile", "Fax", "Email", "Agent", "Relation agent"],
 "column_types": [null, "name", "number_office", "number_mobile", "fax", "email", "relation_agent"],
 "results": [
 {
 "column_values": ["Bob", "Marley", "5555555", "5556666", "5553333", "mail@example.com", null],
 "relations": {
 "agent_id": null,
 "user_id": null,
 "endpoint_id": null,
 "source_entry_id": "55"
 },
 "source": "my_ldap_directory"
 }, {
 "column_values": ["Charlie", "Boblin", "5555556", "5554444", "5552222", "mail2@example.com", null],
 "relations": {
 "agent_id": 12,
 "user_id": 34,
 "endpoint_id": 56,
 "source_entry_id": "34"
 },
 "source": "internal"
 }
]
}

```

**Set favorite** Client -> Server

```
{
 "class": "people_set_favorite",
 "source": "my_ldap_directory"
 "source_entry_id": "55"
 "favorite": true
}
```

Server -> Client

```
{
 "class": "people_favorite_update",
 "source": "my_ldap_directory"
 "source_entry_id": "55"
 "favorite": true
}
```

**STARTTLS** The STARTTLS command is used to upgrade a connection to use SSL. Once connected, the server send a starttls offer to the client which can reply with a starttls message including the status field. The server will then send a starttls message back to the client with the same status and start the handshake if the status is true.

Server -> Client

```
{
 "class": "starttls"
}
```

Client -> Server -> Client

```
{
 "class": "starttls",
 "status": true
}
```

---

**Note:** a client which does not reply to the starttls offer will keep it's unencrypted connection.

---

**Personal contacts list** Client -> Server

```
{
 "class": "people_personal_contacts"
}
```

Server -> Client

```
{
 "class": "people_personal_contacts_result",
 "column_headers": ["Firstname", "Lastname", "Phone number", "Mobile", "Fax", "Email", "Agent", "Relation agent"],
 "column_types": [null, "name", "number_office", "number_mobile", "fax", "email", "relation_agent"],
 "results": [
 {
 "column_values": ["Bob", "Marley", "5555555", "5556666", "5553333", "mail@example.com", null],
 "relations": {
 "agent_id": null,
 "user_id": null,
 "endpoint_id": null,
 "source_entry_id": "abcd-12"
 },
 "source": "personal"
 }, {
 "column_values": ["Charlie", "Boblin", "5555556", "5554444", "5552222", "mail2@example.com", null]
 }
]
}
```



```

 "relations": {
 "agent_id": null,
 "user_id": null,
 "endpoint_id": null,
 "source_entry_id": "efgh-34"
 },
 "source": "personal"
 }
]
}

```

#### Personal contact purge Client -> Server

```

{
 "class": "people_purge_personal_contacts",
}

```

#### Server -> Client

```

{
 "class": "people_personal_contacts_purged",
}

```

#### Personal contact raw Client -> Server

```

{
 "class": "people_personal_contact_raw",
 "source": "personal",
 "source_entry_id": "abcd-1234"
}

```

#### Server -> Client

```

{
 "class": "people_personal_contact_raw_result",
 "source": "personal",
 "source_entry_id": "abcd-1234",
 "contact_infos": {
 "firstname": "Bob",
 "lastname": "Wonderland"
 ...
 }
}

```

#### Create personal contact Client -> Server

```

{
 "class": "people_create_personal_contact",
 "contact_infos": {
 "firstname": "Bob",
 "lastname": "Wonderland",
 ...
 }
}

```

#### Server -> Client

```

{
 "class": "people_personal_contact_created"
}

```

**Delete personal contact** Client -> Server

```
{
 "class": "people_delete_personal_contact",
 "source": "personal",
 "source_entry_id": "abcd-1234"
}
```

Server -> Client

```
{
 "class": "people_personal_contact_deleted",
 "source": "personal",
 "source_entry_id": "abcd-1234"
}
```

**Edit personal contact** Client -> Server

```
{
 "class": "people_edit_personal_contact",
 "source": "personal",
 "source_entry_id": "abcd-1234",
 "contact_infos": {
 "firstname": "Bob",
 "lastname": "Wonderland",
 ...
 }
}
```

Server -> Client

```
{
 "class": "people_personal_contact_raw_update",
 "source": "personal",
 "source_entry_id": "abcd-1234"
}
```

**Import personal contacts** Client -> Server

```
{
 "class": "people_import_personal_contacts_csv",
 "csv_contacts": "firstname,lastname\r\nBob,the Builder\r\n,Alice,Wonderland\r\n,BobMissingFields"
}
```

Server -> Client

```
{
 "class": "people_import_personal_contacts_csv_result",
 "created_count": 2,
 "failed": [
 {
 "line": 3,
 "errors": [
 "missing fields"
]
 }
]
}
```

**Export personal contacts** Client -> Server

```
{
 "class": "people_export_personal_contacts_csv",
}
```

Server -> Client

```
{
 "class": "people_export_personal_contacts_csv_result",
 "csv_contacts": "firstname,lastname\r\nBob,the Builder\r\n,Alice,Wonderland\r\n"
}
```

## Service

- class : featuresput

## Call Filtering

- function : incallfilter
- value : true, false activate deactivate filtering

Client -> Server

```
{"class": "featuresput", "commandid": 1326845972, "function": "incallfilter", "value": true}
```

Server > Client

```
{
 "class": "getlist",
 "config": {"incallfilter": true},
 "function": "updateconfig",
 "listname": "users",
 "tid": "2",
 "timenow": 1361456398.52, "tipbxid": "xivo" }
}
```

## DND

- function : enablednd
- value : true, false activate deactivate DND

Client -> Server

```
{"class": "featuresput", "commandid": 1088978942, "function": "enablednd", "value": true}
```

Server > Client

```
{
 "class": "getlist",
 "config": {"enablednd": true},
 "function": "updateconfig",
 "listname": "users",
 "tid": "2",
 "timenow": 1361456614.55, "tipbxid": "xivo"}
}
```

## Recording

- function : enablerecording
- value : true, false

Activate / deactivate recording for a user, extension call recording has to be activated : *Services->IPBX->IPBX services->Extension*

Client -> Server

```
{ "class": "featuresput", "commandid": 1088978942, "function": "enablerecording", "value": true, "tid": "7", "timenow": 1361456614.55, "tipbxid": "xivo" }
```

Server > Client

```
{
 "class": "getlist",
 "config": { "enablerecording": true },
 "function": "updateconfig",
 "listname": "users",
 "tid": "7",
 "timenow": 1361456614.55, "tipbxid": "xivo" }
```

**Unconditional Forward** Forward the call at any time, call does not reach the user

- function : fwd

Client -> Server

```
{
 "class": "featuresput", "commandid": 2082138822, "function": "fwd",
 "value": { "destunc": "1002", "enableunc": true }
}
```

Server > Client

```
{
 "class": "getlist",
 "config": { "destunc": "1002", "enableunc": true },
 "function": "updateconfig",
 "listname": "users",
 "tid": "2",
 "timenow": 1361456777.98, "tipbxid": "xivo" }
```

**Forward On No Answer** Forward the call to another destination if the user does not answer

- function : fwd

Client -> Server

```
{
 "class": "featuresput", "commandid": 1705419982, "function": "fwd",
 "value": { "destrna": "1003", "enablerna": true }
}
```

Server > Client

```
{
 "class": "getlist",
 "config": { "destrna": "1003", "enablerna": true },
 "function": "updateconfig",
 "listname": "users",
 "tid": "2",
 "timenow": 1361456966.89, "tipbxid": "xivo" }
```

**Forward On Busy** Forward the call to another destination when the user is busy

- function : fwd

Client -> Server

```
{
 "class": "featuresput", "commandid": 568274890, "function": "fwd",
 "value": {"destbusy": "1009", "enablebusy": true}
}
```

Server > Client

```
{
 "class": "getlist",
 "config": {"destbusy": "1009", "enablebusy": true},
 "function": "updateconfig",
 "listname": "users",
 "tid": "2",
 "timenow": 1361457163.77, "tipbxid": "xivo"
}
```

## Statistics

**Subscribe to queues stats** This message can be sent from the client to enable statistics update on queues

Client -> Server

```
{"commandid":36,"class":"subscribetoqueuesstats"}
```

``Server > Client``

**Get queues stats** When statistic update is enable by sending message *Subscribe to queues stats*.

The first element of the message is the queue id

```
{"stats": {"10": {"Xivo-LoggedAgents": 0}},
 "class": "getqueuesstats", "timenow": 1384509582.88}
{"stats": {"1": {"Xivo-WaitingCalls": 0}},
 "class": "getqueuesstats", "timenow": 1384509582.89}
{"stats": {"1": {"Xivo-TalkingAgents": "0", "Xivo-AvailableAgents": "1", "Xivo-EWT": "6"}},
 "class": "getqueuesstats", "timenow": 1384512350.25}
```

**Status** These messages can also be received without any request as unsolicited messages.

**User status** User status is to manage user presence

- Request user status update

Client -> Server

```
{"class": "getlist", "commandid": 107712156,
 "function": "updatestatus",
 "listname": "users",
 "tid": "14", "tipbxid": "xivo"}
```

Server > Client

```
{"class": "getlist",
 "function": "updatestatus",
 "listname": "users",
 "status": {"availstate": "outtolunch", "connection": "yes"},
 "tid": "1", "timenow": 1364994093.48, "tipbxid": "xivo"}
```

- Change User status

Client -> Server

```
{ "availstate": "away",
 "class": "availstate",
 "commandid": 1946092392,
 "ipbxid": "xivo",
 "userid": "1" }
```

Server > Client

```
{ "class": "getlist",
 "function": "updatestatus",
 "listname": "users",
 "status": { "availstate": "away" },
 "tid": "1", "timenow": 1370523352.6, "tipbxid": "xivo" }
```

### Phone status

- tid is the line id, found in linelist from message *User configuration*

Client -> Server

```
{ "class": "getlist", "commandid": 107712156,
 "function": "updatestatus",
 "listname": "phones", "tid": "8", "tipbxid": "xivo" }
```

Server > Client

```
{ "class": "getlist",
 "function": "updatestatus",
 "listname": "phones",
 "status": { "hintstatus": "0" },
 "tid": "1",
 "timenow": 1364994093.48,
 "tipbxid": "xivo" }
```

### Queue status Client -> Server

```
{ "commandid": 17, "class": "getlist", "tid": "8", "tipbxid": "xivo", "function": "updatestatus", "listname": "queues" }
```

Server > Client

```
{ "function": "updatestatus", "listname": "queues", "tipbxid": "xivo", "timenow": 1382710430.54,
 "status": { "agentmembers": ["1", "5"], "phonemembers": ["8"] },
 "tid": "8", "class": "getlist" }
```

### Agent status

- tid is the agent id.

Client -> Server

```
{ "class": "getlist",
 "commandid": <random_integer>,
 "function": "updatestatus",
 "listname": "agents",
 "tid": "635",
 "tipbxid": "xivo" }
```

Server > Client

```
{ "class": "getlist",
 "listname": "agents",
 "function": "updatestatus",
```

```

"tipbxid": "xivo",
"tid": 635,
"status": {
 "availability": "logged_out",
 "availability_since": 1370868774.74,
 "channel": null,
 "groups": [],
 "on_call_acd": false,
 "on_call_nonacd": false,
 "on_wrapup": false,
 "phonenumber": null,
 "queues": [
 "113"
]
}
}
}

```

- availability can take the values:
  - logged\_out
  - available
  - unavailable
  - on\_call\_nonacd\_incoming\_internal
  - on\_call\_nonacd\_incoming\_external
  - on\_call\_nonacd\_outgoing\_internal
  - on\_call\_nonacd\_outgoing\_external
- availability\_since is the timestamp of the last availability change
- queues is the list of queue ids from which the agent receives calls

## Switchboard

**Answer** This allows the switchboard operator to answer an incoming call or unhold a call on-hold.

```
{ "class": "answer", "uniqueid": "12345667.89" }
```

**Unsolicited Messages** These messages are received whenever one of the following corresponding event occurs: sheet message on incoming calls, or updatestatus when a phone status changes.

**Sheet** This message is received to display customer information if configured at the server side

```

{
 "timenow": 1361444639.61,
 "class": "sheet",
 "compressed": true,
 "serial": "xml",
 "payload": "AAADnnicndPBToNAEAbgVln3XgFN1AP.....",
 "channel": "SIP/e6fhff-00000007"
}

```

How to decode payload :

```

>>> b64content = base64.b64decode(<payload content>)
>>> # 4 first cars are the encoded lenght of the xml string (in Big Endian format)
>>> xmlflen = struct.unpack('>I',b64content[0:4])
>>> # the rest is a compressed xml string
>>> xmlcontent = zlib.decompress(toto[4:])

```

```
>>> print xmlcontent

<?xml version="1.0" encoding="utf-8"?>
 <profile>
 <user>
 <internal name="ipbxid"><![CDATA[xivo]]></internal>
 <internal name="where"><![CDATA[dial]]></internal>
 <internal name="channel"><![CDATA[SIP/barometrix_jyldev-00000009]]></internal>
 <internal name="focus"><![CDATA[no]]></internal>
 <internal name="zip"><![CDATA[1]]></internal>
 <sheet_qtui order="0010" name="qtui" type="None"><![CDATA[]]></sheet_qtui>
 <sheet_info order="0010" name="Nom" type="title"><![CDATA[0230210083]]></sheet_info>
 <sheet_info order="0030" name="Origine" type="text"><![CDATA[extern]]></sheet_info>
 <sheet_info order="0020" name="Num\\©ro" type="text"><![CDATA[0230210083]]></sheet_info>
 <systray_info order="0010" name="Nom" type="title"><![CDATA[Maric\\© Sapr\\¯tcl]]></systray_info>
 <systray_info order="0030" name="Origine" type="body"><![CDATA[extern]]></systray_info>
 <systray_info order="0020" name="Num\\©ro" type="body"><![CDATA[0230210083]]></systray_info>
 </user>
 </profile>
```

The xml file content is defined by the following xsd file: `xivo-javactilib/src/main/xsd/sheet.xsd` ([online version](#))

**Phone status update** Received when a phone status change

- class : getlist
- function : updatestatus
- listname : phones

```
{
 "class": "getlist",
 "function": "updatestatus",
 "listname": "phones",
 "tipbxid": "xivo",
 "timenow": 1361447017.29,

}
```

tid is the the object identification

Example of phone messages received when a phone is ringing :

```
{... "status": {"hintstatus": "0"}, "tid": "3"}
{... "status": {"hintstatus": "8"}, "tid": "3"}
```

## Update notification

**Register agent status update** The *register\_agent\_status\_update* command is used to register to the status updates of a list of agent. Once registered to a agent's status, the client will receive all *Agent status update* events for the registered agents.

This command should be sent when an agent is displayed in the people xlet to be able to update the agent status icon.

The *Unregister agent status update* command should be used to stop receiving updates.

Client -> Server

```
{
 "class": "register_agent_status_update",
 "agent_ids": [{"<xivo-uuid>", "<agent-id1>"}],
```



```

 ["<xivo-uuid>", "<agent-id2>"],
 ...,
 ["<xivo-uuid>", "<agent-idn>"]],
 "commandid": <commandid>
}

```

**Unregister agent status update** The *unregister\_agent\_status\_update* command is used to unregister from the status updates of a list of agent.

Once unregistered, the client will stop receiving the *Agent status update* events for the specified agents.

Client -> Server

```

{
 "class": "unregister_agent_status_update",
 "agent_ids": [
 ["<xivo-uuid>", "<agent-id1>"],
 ["<xivo-uuid>", "<agent-id2>"],
 ...,
 ["<xivo-uuid>", "<agent-idn>"]
],
 "commandid": <commandid>
}

```

**Agent status update** The *agent\_status\_update* event is received when the presence of an agent changes.

To receive this event, the user must first register to the event for a specified agent using the *Register agent status update* command.

To stop receiving this event, the user must send the *Unregister agent status update* command.

- data, a dictionary containing 3 fields:
  - agent\_id, is an integer containing the ID of the user affected by this status change
  - xivo\_uuid: a string containing the UUID of the XiVO that sent the status update
  - status: a string containing the new status, “logged\_in” or “logged\_out”

Server -> Client

```

{
 "class": "agent_status_update",
 "data": {
 "agent_id": 42,
 "xivo_uuid": "<the-xivo-uuid>",
 "status": "<status-name>"
 }
}

```

The *agent\_status\_update* event contains the same data as the *agent\_status\_update*. The latter should be preferred to the former for uses that do not require a persistent connection to xivo-ctid.

**Register endpoint status update** The *register\_endpoint\_status\_update* command is used to register to the status updates of a list of lines. Once registered to an endpoint’s status, the client will receive all *Endpoint status update* events for the registered agents.

This command should be sent when an endpoint is displayed in the people xlet to be able to update the agent status icon.

The *Unregister endpoint status update* command should be used to stop receiving updates.

Client -> Server

```
{
 "class": "register_endpoint_status_update",
 "endpoint_ids": [["<xivo-uuid>", "<endpoint-id1>"],
 ["<xivo-uuid>", "<endpoint-id2>"],
 ...,
 ["<xivo-uuid>", "<endpoint-idn>"]],
 "commandid": <commandid>
}
```

**Unregister endpoint status update** The *unregister\_endpoint\_status\_update* command is used to unregister from the status updates of a list of agent.

Once unregistered, the client will stop receiving the *Endpoint status update* events for the specified agents.

Client -> Server

```
{
 "class": "unregister_endpoint_status_update",
 "endpoint_ids": [["<xivo-uuid>", "<endpoint-id1>"],
 ["<xivo-uuid>", "<endpoint-id2>"],
 ...,
 ["<xivo-uuid>", "<endpoint-idn>"]],
 "commandid": <commandid>
}
```

**Endpoint status update** The *endpoint\_status\_update* event is received when the status of a line changes.

To receive this event, the user must first register to the event for a specified endpoint using the *Register endpoint status update* command.

To stop receiving this event, the user must send the *Unregister endpoint status update* command.

- data, a dictionary containing 3 fields:
  - endpoint\_id, is an integer containing the ID of the line affected by this status change
  - xivo\_uuid: a string containing the UUID of the XiVO that sent the status update
  - status: an integer matching an entry in the cti hint configuration

Server -> Client

```
{
 "class": "endpoint_status_update",
 "data": {
 "endpoint_id": 42,
 "xivo_uuid": "<the-xivo-uuid>",
 "status": <hint-status>
 }
}
```

The *endpoint\_status\_update* event contains the same data as the *endpoint\_status\_update*. The latter should be preferred to the former for uses that do not require a persistent connection to xivo-ctid.

**Register user status update** The *register\_user\_status\_update* command is used to register to the status updates of a list of user. Once registered to a user's status, the client will receive all *User status update* events for the registered users.

This command should be sent when a user is displayed in the people xlet to be able to update the presence status icon.

The *Unregister user status update* command should be used to stop receiving updates.

Client -> Server

```
{
 "class": "register_user_status_update",
 "user_ids": [["<xivo-uuid>", "<user-uuid1>"],
 ["<xivo-uuid>", "<user-uuid2>"],
 ...,
 ["<xivo-uuid>", "<user-uuidn>"]],
 "commandid": <commandid>
}
```

**Unregister user status update** The *unregister\_user\_status\_update* command is used to unregister from the status updates of a list of user.

Once unregistered, the client will stop receiving the *User status update* events for the specified users.

Client -> Server

```
{
 "class": "unregister_user_status_update",
 "user_ids": [["<xivo-uuid>", "<user-uuid1>"],
 ["<xivo-uuid>", "<user-uuid2>"],
 ...,
 ["<xivo-uuid>", "<user-uuidn>"]],
 "commandid": <commandid>
}
```

**User status update** The *user\_status\_update* event is received when the presence of a user changes.

To receive this event, the user must first register to the event for a specified user using the *Register user status update* command.

To stop receiving this event, the user must send the *Unregister user status update* command.

- data, a dictionary containing the following fields:
  - user\_uuid, a string containing the UUID of the user.
  - user\_id, an integer containing the ID of the user.
  - xivo\_uuid: a string containing the UUID of the XiVO that sent the status update
  - status: a string containing the new status of the user based on the cti profile configuration

---

**Note:** When multiple XiVO share user statuses, the cti profile configuration for presences and phone statuses should match on all XiVO to be displayed properly

---

Server -> Client

```
{
 "class": "user_status_update",
 "data": {
 "user_uuid": "<the-user-uuid>",
 "user_id": <the-user-id>,
 "xivo_uuid": "<the-xivo-uuid>",
 "status": "<status-name>"
 }
}
```

**Warning:** The *user\_id* field is **DEPRECATED** and **should not be used**. Use the *user\_uuid* field instead.

**CTI server implementation** In the git repository `git://github.com/xivo-pbx/xivo-ctid.git`

- *cti\_config* handles the configuration coming from the WEBI
- *interfaces/interface\_ami*, together with *asterisk\_ami\_definitions*, *amiinterpret* and *xivo\_ami* handle the AMI connections (asterisk)
- *interfaces/interface\_info* handles the CLI-like connections
- *interfaces/interface\_webi* handles the requests and signals coming from the WEBI
- *interfaces/interface\_cti* handles the clients' connections, with the help of *client\_connection*, and it often involves *cti\_command* too
- *innerdata* is meant to be the place where all statuses are computed and stored

The main loop uses *select()* syscall to dispatch the tasks according to miscellaneous incoming requests.

Requirements for *innerdata*:

- the properties fetched from the WEBI configuration shall be stored in the relevant *xod\_config* structure
- the properties fetched from elsewhere shall be stored in the relevant *xod\_status* structure
- at least two kinds of objects are not “predefined” (as are the phones or the queues, for instance)
  - the channels (in the asterisk SIP/345-0x12345678 meaning)
  - the group and queue members shall be handled in a special way each

The purpose of the ‘relations’ field, in the various structures is to keep track of relations and cross-relations between different objects (a phone logged in as an agent, itself in a queue, itself called by some channels belonging to phones ...).

**CTI server Message flow** Messages sent from the CTI clients to the server are received by the CTIServer class. The CTIServer then calls *interface\_cti.CTI* class *manage\_connection* method. The *interface\_cti* uses his *\_cti\_command\_handler* member to parse and run the command. The *CTICommandHandler* get a list of classes that handle this message from the *CTICommandFactory*. Then the *interface\_cti.CTI* calls *run\_commands* on the handler, which returns a list of all commands replies.

To implement a new message in the protocol you have to create a new class that inherits the *CTICommand* class. Your new class should have a static member *caller\_required\_fields* which is a list of required fields for this class. Your class should also have a *conditions* static member which is a list of tuples of conditions to detect that an incoming message matches this class. The *\_\_init\_\_* of your class is responsible for the initialization of it's fields and should call *super(<ClassName>, self).\_\_init\_\_(msg)*. Your class should register itself to the *CTICommandFactory*.

```
from xivo_cti.cti.cti_command import CTICommand
from xivo_cti.cti.cti_command_factory import CTICommandFactory

class InviteConfroom(CTICommand):
 required_fields = ['class', 'invitee']
 conditions = [('class', 'invite_confroom')]
 def __init__(self):
 super(InviteConfroom, self).__init__(msg)
 self._invitee = msg['invitee']

CTICommandFactory.register_class(InviteConfroom)
```

Each CTI commands has a callback list that you can register to from anywhere. Each callback function will be called when this message is received with the command as parameter.

Refer to *MeetmeList.\_\_init\_\_* for a callback registration example and to *MeetmeList.invite* for the implementation of a callback.

```
from xivo_cti.cti.commands.invite_confroom import InviteConfroom

class MySuperClass(object):
 def __init__(self):
 InviteConfroom.register_callback(self.invite_confroom_handler)

 def invite_confroom_handler(self, invite_confroom_command):
 # Do your stuff here.
 if ok:
 return invite_confroom_command.get_message('Everything is fine')
 else:
 return invite_confroom_command.get_warning('I don't know you, go away', True)
```

**Note:** The client's connection is injected in the command instance before calling callbacks functions. The client's connection is an `interface_cti.CTI` instance.

## Database

**Adding a Migration Script** Strating with XiVO 14.08, the database migration is handled by [alembic](#).

The XiVO migration scripts can be found in the [xivo-manage-db](#) repository.

On a XiVO, they are located in the `/usr/share/xivo-manage-db` directory.

To add a new migration script from your developer machine, go into the root directory of the `xivo-manage-db` repository. There should be an `alembic.ini` file in this directory. You can then use the following command to create a new migration script:

```
alembic revision -m "<description>"
```

This will create a file in the `alembic/versions` directory, which you'll have to edit.

When the migration scripts are executed, they use a connection to the database with the role/user `asterisk`. This means that new objects that are created in the migration scripts will be owned by the `asterisk` role and it is thus not necessary (nor recommended) to explicitly grant access to objects to the `asterisk` role (i.e. no "GRANT ALL" command after a "CREATE TABLE" command).

## Diagrams

**Agent states** Graphs representing states and transitions between agent states. Used in Agent status dashboard and agent list.

Download (DIA)

## Architecture

## Provisioning

This section describes the informations and tools for `xivo-provd`.

**Managing DHCP server configuration** This page considers the configuration files of the DHCP server in `/etc/dhcp/dhcpd_update/`.

**Who modifies the files** The files are updated with the command `dhcpd-update`, which is also run when updating the provisioning plugins. This commands fetches configurations files from the `provdxivo.solutions` server.

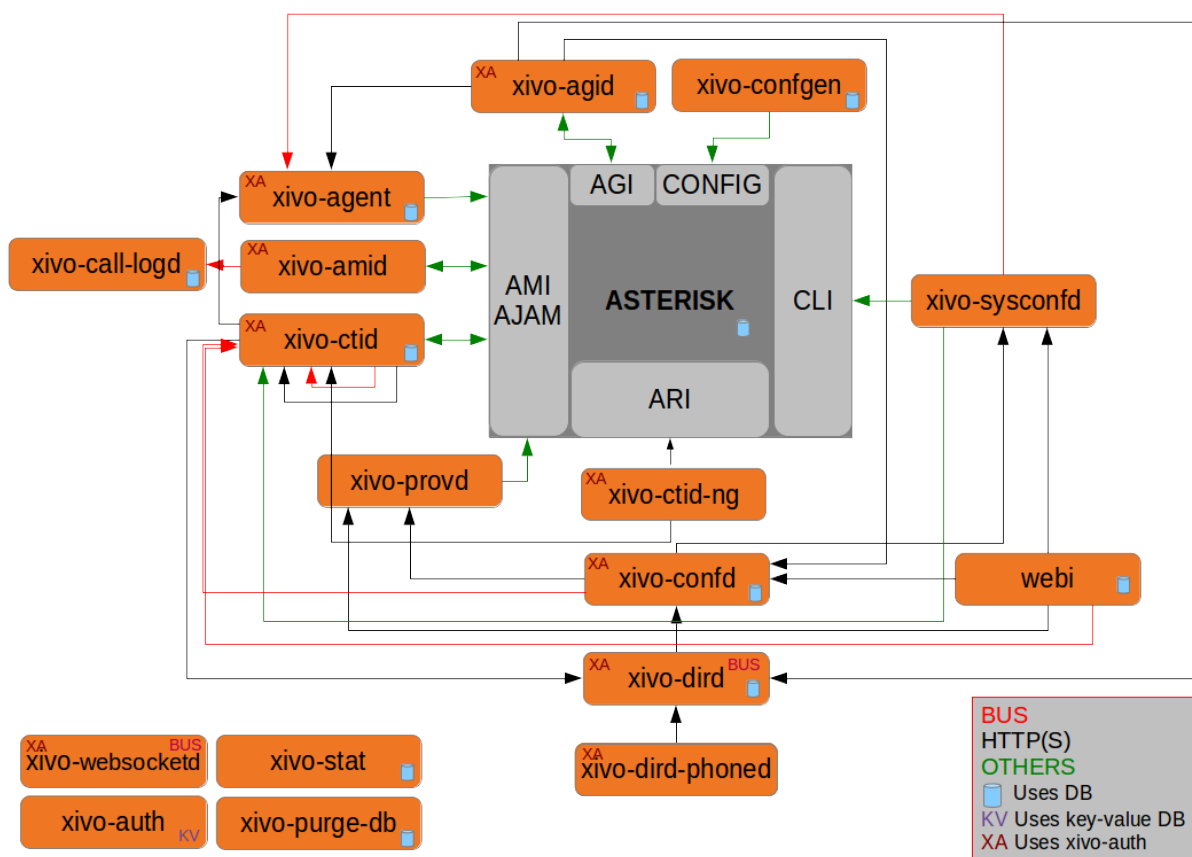


Fig. 1.109: Relationships between the components of XiVO. (source)

## How to update the source files

### Ensure your modifications are working

- On a XiVO, edit manually the file `/etc/dhcp/dhcpd_update/*.conf`
- `service isc-dhcp-server restart`
- If errors are shown in `/var/log/daemon.log`, check your modifications

### Edit the files

- Edit the files in the Git repo `xivo-provd-plugins`, directory `dhcp/`
- Push your modifications
- Go in `dhcp/`
- Run `make upload` to push your modifications to `provd.xivo.solutions`. There is no testing version of these files. Once the files are uploaded, they are available for all XiVO installations.

## Managing Plugins

**Git Repository** Most plugin-related files are available in the [xivo-provd-plugins repository](#). Following examples are relative to the repository directory tree. Any modifications should be preceeded by a *git pull*.

**Updating a Plugin** We will be using the *xivo-cisco-spa* plugins family as an example on this page

There is one directory per family. Here is the directory structure for `xivo-cisco-spa`:

```
plugins/xivo-cisco-spa/
+-- model_name_xxx
+-- model_name_xxx
+-- common
+-- build.py
```

Every plugin has a folder called `common` which regroups common ressources for each model. Every model has its own folder with its version number.

After modifying a plugin, you must increment the version number. You can modify the file `plugin-info` to change the version number:

```
plugins/xivo-cisco-spa/
+-- model_name_xxx
 +-- plugin-info
```

---

**Important:** If ever you modify the folder `common`, you must increment the version number of all the models.

---

**Use Case: Update Firmwares for a given plugin** Let us suppose we want to update firmwares for `xivo-snom` from 8.7.3.25 to 8.7.3.25.5. Here are the steps to follow :

1. Copy folder `plugins/xivo-snom/8.7.3.25` to `plugins/xivo-snom/8.7.3.25.5`
2. Update VERSION number in `plugins/xivo-snom/8.7.3.25.5/entry.py`
3. Update VERSION number in `plugins/xivo-snom/8.7.3.25.5/plugin-info`
4. Download new firmwares (.bin files from [snom website](#))
5. Update VERSION number and URIs in `plugins/xivo-snom/8.7.3.25.5/pkgs/pkgs.db` (with uris of downloaded files from `snom website`)

6. Update sizes and shasums in `plugins/xivo-snom/8.7.3.25.5/pkgs/pkgs.db` (using helper script `xivo-tools/dev-tools/check_fw`)
7. Update `plugins/xivo-snom/build.py` (duplicate and update section 8.7.3.25 > 8.7.3.25.5)

**Test your changes** You have three different methods to test your changes on your development machine.

**Always increase plugin version (easiest)** If the production version is 0.4, change the plugin version to 0.4.01, make your changes and upload to testing (see below).

Next modification will change the plugin version to 0.4.02, etc. When you are finished making changes, change the version to 0.5 and upload one last time.

**Edit directly on XiVO** Edit the files in `/var/lib/xivo-provd/plugins`.

To apply your changes, go in `xivo-provd-cli` and run:

```
plugins.reload('xivo-cisco-spa-7.5.4')
```

**Disable plugin caching** Edit `/etc/xivo/provd/provd.conf` and add the line:

```
cache_plugin: True
```

Empty `/var/cache/xivo-provd` and restart `provd`.

Make your changes in `provd-plugins`, update the plugin version to the new one and upload to testing (see below). Now, every time you uninstall/install the plugin, the new plugin will be fetched from testing, instead of being cached, even without changing the version.

**Uploading to testing** Before updating a plugin, it must be passed through the testing phase. Once it has been approved it can be uploaded to the production server

---

**Important:** Before uploading a plugin in the testing `provd` repository, make sure to git pull the `xivo-provd-plugins` git repository.

---

To upload the modified plugin in the testing repo on *provd.xivo.solutions*, you can execute the following command:

```
$ make upload
```

Afterwards, in the web-interface, you must modify the URL in section *Configuration* → *Provisioning* → *General* to:

```
`http://provd.xivo.solutions/plugins/1/testing/`
```

You can then update the list of plugins and check the version number for the plugin that you modified. Don't forget to install the plugin to test it.

**Mass-install all firmwares related to a given plugin** Using `xivo-provd-cli` on a xivo server, one can mass-install firmwares. Following example installs all firmwares for xivo-snom 8.7.3.25.5 plugin (note the auto-completion):

```
xivo-provd-cli> plugins.installed().keys()
[u'xivo-snom-8.7.3.15',
 u'xivo-cisco-sccp-legacy',
 u'xivo-snom-8.4.35',
 u'xivo-snom-8.7.3.25',
 u'xivo-aastra-switchboard',
 u'xivo-aastra-3.2.2-SP3',
 u'xivo-aastra-3.2.2.1136',
```



```
u'xivo-cisco-sccp-9.0.3',
u'null',
u'xivo-snom-8.7.3.25.5']
xivo-provd-cli> p = plugins['xivo-snom-8.7.3.25.5']
xivo-provd-cli> p.install_all()
```

**Uploading to stable** Once checked, you must synchronize the plugin from *testing* to *stable*. If applicable, you should also update the archive repo.

To download the stable and archive plugins:

```
$ make download-stable
$ make download-archive
```

Go to the *plugins/\_build* directory and delete the plugins that are going to be updated. Note that if you are not updating a plugin but you are instead removing it “once and for all”, you should instead move it to the archive directory:

```
$ rm -fi stable/xivo-cisco-spa*
```

Copy the files from the directory *testing* to *stable*:

```
$ cp testing/xivo-cisco-spa* stable
```

Go back to the *plugins* directory and upload the files to the stable and archive repo:

```
$ make upload-stable
$ make upload-archive
```

The file are now up to date and you can test by putting back the *stable* url in the web-interface’s configuration:

```
`http://provd.xivo.solutions/plugins/1/stable/`
```

**Testing a new SIP phone** Let’s suppose you have received a brand new SIP phone that is not supported by the provisioning system of XiVO. You would like to know if it’s possible to add auto-provisioning support for it. That said, you have never tested the phone before.

This guide will help you get through the different steps that are needed to add auto-provisioning support for a phone to XiVO.

**Prerequisites** Before continuing, you’ll need the following:

- a private LAN where only your phones and your test machines are connected to it, i.e. a LAN that you fully control.

**Configuring a test environment** Although it’s possible to do all the testing directly on a XiVO, it’s more comfortable and usually easier to do on a separate, dedicated machine.

That said, you’ll still need a XiVO near, since we’ll be doing the call testing part on it and not on a separate asterisk.

So, for the rest of this guide, we’ll suppose you are doing your tests on a *Debian jessie* with the following configuration:

- Installed packages:

```
isc-dhcp-server tftpd-hpa apache2
```

- Example content of the `/etc/dhcp/dhcpd.conf` file (restart `isc-dhcp-server` after modification):

```
ddns-update-style none;

default-lease-time 7200;
max-lease-time 86400;

log-facility local7;

subnet 10.34.1.0 netmask 255.255.255.0 {
 authoritative;

 range 10.34.1.200 10.34.1.250;

 option subnet-mask 255.255.255.0;
 option broadcast-address 10.34.1.255;
 option routers 10.34.1.6;

 option ntp-servers 10.34.1.6;
 option domain-name "my-domain.example.org";
 option domain-name-servers 10.34.1.6;

 log(concat("[VCI: ", option vendor-class-identifier, "]"));
}
```

- Example content of the `/etc/default/tftpd-hpa` file (restart `tftpd-hpa` after modification):

```
TFTP_USERNAME="tftp"
TFTP_DIRECTORY="/srv/tftp"
TFTP_ADDRESS="0.0.0.0:69"
TFTP_OPTIONS="--secure --verbose"
```

With this configuration, files served via TFTP will be in the `/srv/tftp` directory and those served via HTTP in the `/var/www` directory.

**Testing** Adding auto-provisioning support for a phone is mostly a question of finding answers to the following questions.

1. *Is it worth the time adding auto-provisioning support for the phone ?*

Indeed. Adding quality auto-provisioning support for a phone to XiVO requires a non negligible amount of work, if you don't meet any real problem and are comfortable with provisioning in XiVO. Not all phones are born equal. Some are cheap. Some are old and slow. Some are made to work on proprietary system and will only work in degraded mode on anything else.

That said, if you are uncertain, testing will help you clarifying your idea.

2. *What is the vendor, model, MAC address and firmware version (if available) of your phone ?*

Having the vendor and model name is essential when looking for documentation or other information. The MAC address will be needed later on for some tests, and it's always good to know the firmware version of the phone if you are trying to upgrade to a newer firmware version and you're having some troubles, and when reading the documentation.

3. *Is the official administrator guide/documentation available publicly on the vendor web site ? Is it available only after registering and login to the vendor web site ?*

Having access to the administrator guide/documentation of the phone is also essential. Once you've found it, download it and keep the link to the URL. If you can't find it, it's probably not worth going further.

4. *Is the latest firmware of the phone available publicly on the vendor web site ? Is it available only after registering and login to the vendor web site ?*

Good auto-provisioning support means you need to have an easy way to download the latest firmware of the phone. Ideally, this mean the firmware is downloadable from an URL, with no authentication whatsoever. In the worst case, you'll need to login on some web portal before being able to download the firmware,

which will be cumbersome to automatize and probably fragile. If this is the case, it's probably not worth going further.

5. *Does the phone need other files, like language files ? If so, are these files available publicly on the vendor web site ? After registering ?*

Although you might not be able to answer to this question yet because you might not know if the phone needs such files to be either in English or in French (the two officially supported language in XiVO), you'll need to have an easy access to these files if its the case.

6. *Does the phone supports auto-provisioning via DHCP + HTTP (or TFTP) ?*

The provisioning system in XiVO is based on the popular method of using a DHCP server to tell the phone where to download its configuration files, and a HTTP (or TFTP) server to serve these configuration files. Some phones support other methods of provisioning (like TR-069), but that's of no use here. Also, if your phone is only configurable via its web interface, although it's technically possible to configure it automatically by navigating its web interface, it's an **extremely bad** idea since it's impossible to guarantee that you'll still be able to provision the phone on the next firmware release.

If the phone supports both HTTP and TFTP, pick HTTP, it usually works better with the provisioning server of XiVO.

7. *What are the default usernames/passwords on the phone to access administrator menus (phone UI and web UI) ? How do you do a factory reset of the phone ?*

Although this step is optional, it might be handy later to have these kind of information. Try to find them now, and note them somewhere.

8. *What are the DHCP options and their values to send to the phones to tell it where its configuration files are located ?*

Once you know that the phone supports DHCP + HTTP provisioning, the next question is what do you need to put in the DHCP response to tell the phone where its configuration files are located. Unless the admin documentation of the phone is really poor, this should not be too hard to find.

Once you have found this information, the easiest way to send it to the phone is to create a custom host declaration for the phone in the `/etc/dhcp/dhcpd.conf` file, like in this example:

```
host my-phone {
 hardware ethernet 00:11:22:33:44:55;
 option tftp-server-name "http://169.254.0.1/foobar.cfg";
}
```

9. *What are the configuration files the phone needs (filename and content) and what do we need to put in it for the phone to minimally be able to make and receive calls on XiVO ?*

Now that you are able to tell your phone where to look for its configuration files, you need to write these files with the right content in it. Again, at this step, you'll need to look through the documentation or examples to answer this question.

Note that you only want to have the most basic configuration here, i.e. only configure 1 line, with the right SIP registrar and proxy, and the associated username and password.

10. *Do basic telephony services, like transfer, works correctly when using the phone buttons ?*

On most phones, it's possible to do transfer (both attended and direct), three-way conferences or put someone on hold directly from the phone. Do some tests to see if it works correctly.

Also at this step, it's a good idea to check how the phone handle non-ascii characters, either in the caller ID or in its configuration files.

11. *Does other "standard" features work correctly on the phone ?*

For quality auto-provisioning support, you must find how to configure and make the following features work:

- NTP server

- MWI
- function keys (speed dial, BLF, directed pickup / call interception)
- timezone and DST support
- multi language
- DTMF
- hard keys, like the voicemail hard key on some phone
- non-ASCII labels (line name, function key label)
- non-ASCII caller ID
- backup proxy/registrar
- paging

Once you have answered all these questions, you'll have a good idea on how the phone works and how to configure it. Next step would be to start the development of a new provd plugin for your phone for a specific firmware version.

#### IOT Phones FK = Funckey

HK = HardKey

Y = Supported

MN = Menu

N = Not supported

NT = Not tested

NYT = Not yet tested

SK = SoftKey

	model
Provisioning	Y
H-A	Y
Directory XIVO	Y
Funckey	8
<b>Supported programmable keys</b>	
User with supervision function	Y
Group	Y
Queue	Y
Conference Room with supervision function	Y
<b>General Functions</b>	
Online call recording	N
Phone status	Y
Sound recording	Y
Call recording	Y
Incoming call filtering	Y
Do not disturb	Y
Group interception	Y
Listen to online calls	Y
Directory access	Y
Filtering Boss - Secretary	Y
<b>Transfers Functions</b>	
Blind transfer	HK
Indirect transfer	HK
Continued on next page	

Table 1.10 – continued from previous page

	model
<b>Forwards Functions</b>	
Disable all forwarding	Y
Enable/Disable forwarding on no answer	Y
Enable/Disable forwarding on busy	Y
Enable/Disable forwarding unconditional	Y
<b>Voicemail Functions</b>	
Enable voicemail with supervision function	Y
Reach the voicemail	Y
Delete messages from voicemail	Y
<b>Agent Functions</b>	
Connect/Disconnect a static agent	Y
Connect a static agent	Y
Disconnect a static agent	Y
<b>Parking Functions</b>	
Parking	Y
Parking position	Y
<b>Paging Functions</b>	
Paging	Y

**Configuring a NAT Environment** This is a configuration example to simulate the case of a hosted XiVO, i.e. an environment where:

- the XiVO has a public IP address
- the phones are behind a NAT

In this example, we'll reproduce the following environment:

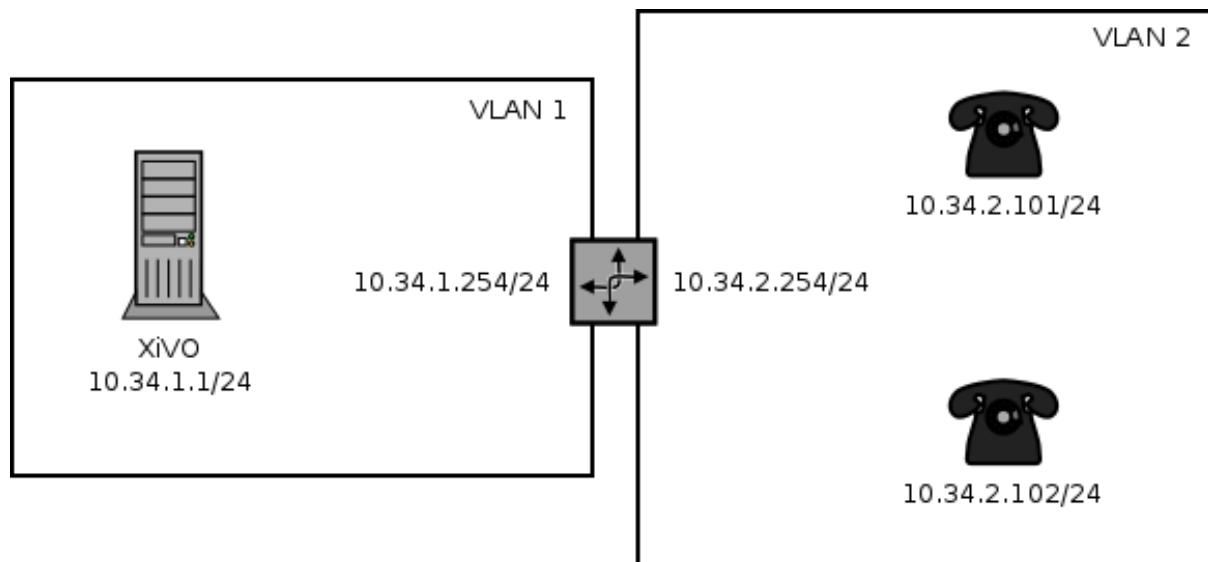


Fig. 1.110: Phones behind a NAT

Where:

- the XiVO is installed inside a virtual machine
- the host machine is used as a router, a NAT and a DHCP server for the phones
- the phones are in a separate VLAN than the XiVO, and when they want to interact with it, they must pass through the NAT

With this setup, we could also put some phones in the same VLAN as the XiVO. We would then have a mixed environment, where some phones are behind the NAT and some phones aren't.

Also, it's easy to go from a non-NAT environment to a NAT environment with this setup. What you usually have to do is only to switch your phone from the "XiVO" VLAN to the "phones" VLAN, and reconfiguring the lines on your XiVO.

The instruction in this page are written for Debian jessie and VirtualBox.

**Prerequisite** On the host machine:

- 1 VLAN network interface for the XiVO. In our example, this will be `eth0.341`, with IP 10.34.1.254/24.
- 1 VLAN network interface for the phones. In our example, this will be `eth0.342`, with IP 10.34.2.254/24.

On the guest machine, i.e. on the XiVO:

- 1 network adapter attached to the "XiVO" VLAN network interface. In our example, this interface inside the virtual machine will have the IP 10.34.1.1/24.

### Configuration

1. On the host, install the ISC DHCP server:

```
apt-get install isc-dhcp-server
```

2. If you do not want it to always be started:

```
systemctl disable isc-dhcp-server.service
```

3. Edit the DHCP server configuration file `/etc/dhcp/dhcpd.conf`. We need to configure the DHCP server to serve network configuration for the phones (Aastra and Snom in this case):

```
ddns-update-style none;

default-lease-time 3600;
max-lease-time 86400;

log-facility daemon;

option space Aastra6700;
option Aastra6700.cfg-server-name code 2 = text;
option Aastra6700.contact-rcs code 3 = boolean;

class "Aastra" {
 match if substring(option vendor-class-identifier, 0, 6) = "Aastra";

 vendor-option-space Aastra6700;
 option Aastra6700.cfg-server-name = "http://10.34.1.1:8667/Aastra";
 option Aastra6700.contact-rcs false;
}

class "Snom" {
 match if substring(option vendor-class-identifier, 0, 4) = "snom";

 option tftp-server-name = "http://10.34.1.1:8667";
 # the domain-name-servers option must be provided for the Snom 715 to work properly
 option domain-name-servers 10.34.1.1;
}

subnet 192.168.32.0 netmask 255.255.255.0 {
}

subnet 10.34.1.0 netmask 255.255.255.0 {
```

```

}

subnet 10.34.2.0 netmask 255.255.255.0 {
 authoritative;

 range 10.34.2.100 10.34.2.199;

 option subnet-mask 255.255.255.0;
 option broadcast-address 10.34.2.255;
 option routers 10.34.2.254;

 option ntp-servers 10.34.1.1;
}

```

4. If you have many network interfaces on your host machine, you might also want to edit `/etc/default/isc-dhcp-server` to only include the “phones” VLAN network interface in the “INTERFACES” variable.
5. Start the `isc-dhcp-server`:

```
systemctl start isc-dhcp-server.service
```

6. Add an `iptables` rules to do NAT:

```
iptables -t nat -A POSTROUTING -o eth0.341 -j MASQUERADE
```

7. Make sure that IP forwarding is enabled:

```
sysctl -w net.ipv4.ip_forward=1
```

8. Put all the phones in the “phones” VLAN on your switch
9. Activate the NAT and Monitoring options on the *Services* → *IPBX* → *General settings* → *SIP Protocol* page of your XiVO.

Note that the `iptables` rules and the IP forwarding setting are not persistent. If you don’t make them persistent (not documented here), don’t forget to reactivate them each time you want to recreate a NAT environment.

## SCCP

`xivo-libsccp` is an alternative SCCP channel driver for Asterisk. It was originally based on `chan_skinny`.

This page is intended for developers and people interested in using `xivo-libsccp` on something other than XiVO.

### Installation from the git repository

**Warning:** If you just want to use your SCCP phones with XiVO, refer to [SCCP Configuration](#)

The following packages are required to compile `xivo-libsccp` on Debian.

- `build-essential`
- `asterisk-dev`

```
apt-get update && apt-get install build-essential asterisk-dev
```

```

git clone https://github.com/xivo-pbx/xivo-libsccp.git
cd xivo-libsccp
make
make install

```

## Configuration

**Warning:** If you just want to use your SCCP phones with XiVO, refer to *SCCP Configuration* instead.

See [sccp.conf.sample](#) for a configuration file example.

## FAQ

Q. When is this \*feature X\* will be available?

A. The order in which we implement features is based on our client needs. Write us an email that clearly explain your setup and what you would like to do and we will see what we can do. We don't provide any timeline.

Q. I want to use the Page() application to call many phones at the same time.

A. Here a Page() example for a one way call (half-duplex):

```
exten => 1000,1,Verbose(2, Paging to external cisco phone)
same => n,Page(sccp/100/autoanswer&sccp/101/autoanswer,i,120)
```

...for a two-way call (full-duplex):

```
exten => 1000,1,Verbose(2, Paging to external cisco phone)
same => n,Page(sccp/100/autoanswer&sccp/101/autoanswer,di,120)
```

**Network Configuration for 7920/7921** Here's how to to configure a hostapd based AP on a Debian host so that both a 7920 and 7921 Wi-Fi phone can connect to it.

The 7920 is older than the 7921 and is pretty limited in its Wi-Fi fonctionnality:

- 802.11b
- WPA (no WPA2)
- TKIP (no CCMP/AES)

Which means that the most secure WLAN you can set up if you want both phones to connect to it is not that secure.

1. Make sure you have a wireless NIC capable of master mode.
2. If needed, install the firmware-<vendor> package. For example, if you have a ralink card like I do:

```
apt-get install firmware-ralink
```

3. Install the other dependencies:

```
apt-get install wireless-tools hostapd bridge-utils
```

4. Create an hostapd configuration file in /etc/hostapd/hostapd.sccp.conf with content: hostapd.sccp.conf

5. Update the following parameters (if applicable) in the configuration file:

- interface
- ssid
- channel
- wpa\_passphrase

6. Create a new stanza in /etc/network/interfaces:

```
iface wlan-sccp inet manual
 hostapd /etc/hostapd/hostapd.sccp.conf
```

7. Up the interface:



```
ifup wlan0=wlan-sccp
```

8. Configure your 7920/7921 to connect to the network.

To unlock the phone's configuration menu on the 7921:

- Press the Navigation Button downwards to enter SETTINGS mode
- Navigate to and select Network Profiles
- Unlock the IP phone's configuration menu by pressing **\*\*#**. The padlock icon on the top-right of the screen will change from closed to open.

When asked for the authentication mode, select something like "Auto" or "AKM".

You don't have to enter anything for the username/password.

9. You'll probably want to bridge your wlan0 interface with another interface, for example a VLAN interface:

```
brctl addbr br0
brctl addif br0 wlan0
brctl addif br0 eth0.341
ip link set br0 up
```

10. If you are using virtualbox and your guest interface is bridged to eth0.341, you'll need to change its configuration and bridge it with br0 instead, else it won't work properly.

**Adding Support for a New Phone** This section describes the requirements to consider that a SCCP phone is working with XiVO libsccp.

#### Basic functionality

- Register on Asterisk
- SCCP reset [restart]
- Call history
- Date time display
- HA

**Telephony** These test should be done with and without direct media enabled

- Emit a call
- Receive a call
- Receive and transfer a call
- Emit a call and transfer the call
- Hold and resume a call
- Features (\*0 and others)
- Receive 2 calls simultaneously
- Emit 2 calls simultaneously
- DTMF on an external IVR

## Function keys

- Redial
- DND
- Hold
- Resume
- New call
- End call
- Call forward (Enable)
- Call forward (Disable)
- Try each button in each mode (on hook, in progress, etc)

## Optional options to test and document

- Phone book
- Caller ID and other display i18n
- MWI
- Speeddial/BLF

## Web Interface

**Configuration for development** Default error level for XiVO web interface is E\_ALL & ~E\_DEPRECATED & ~E\_USER\_DEPRECATED & ~E\_RECOVERABLE\_ERROR & ~E\_STRICT

If you want to display warning or other error in your browser, edit the /etc/xivo/web-interface/xivo.ini and replace report\_type level to 3:

```
[error]
level = E_ALL
report_type = 3
report_mode = 1
report_func = 1
email = john.doe@example.com
file = /var/log/xivo-web-interface/error.log
```

You may also edit /etc/xivo/web-interface/php.ini and change the error level, but you will need to restart the cgi:

```
service spawn-fcgi restart
```

## Interactive debugging in Eclipse Instructions for Eclipse 4.5.

On your XiVO:

1. Install php5-xdebug:

```
apt-get install php5-xdebug
```

2. Edit the /etc/php5/cgi/conf.d/20-xdebug.ini (or /etc/php5/conf.d/20-xdebug.ini on wheezy) and add these lines at the end:

```
xdebug.remote_enable=1
xdebug.remote_host="<dev_host_ip>"
```

where <dev\_host\_ip> is the IP address of your machine where Eclipse is installed.

## 3. Restart spawn-fcgi:

```
service spawn-fcgi restart
```

On your machine where Eclipse is installed:

1. Make sure you have Eclipse PDT installed
2. Create a PHP project named `xivo-web-interface`:
  - Choose “Create project at existing location”, using the `xivo-web-interface` directory
3. In the Window / Preferences / PHP menu:
  - Add a new PHP server with the following information:
    - Name: anything you want
    - Base URL: `https://<xivo_ip>`
    - Path Mapping:
      - \* Path on Server: `/usr/share/xivo-web-interface`
      - \* Path in Workspace: `/xivo-web-interface/src`
4. Create a new PHP Web Application debug configuration:
  - Choose the PHP server you created in last step
  - Pick some file, which can be anything if you don’t “break at first line”
  - Uncheck “Auto Generate”, and set the path you want your browser to open when you’ll launch this debug configuration.

Then, to start a debugging session, set some breakpoints in the code and launch your debug configuration. This will open the page in your browser, and when the code will hit your breakpoints, you’ll be able to go through the code step by step, etc.

## XiVO Client

### Building the XiVO Client

**Building the XiVO Client on Windows platforms** This page explains how to build an executable of the XiVO Client from its sources for Windows.

### Windows Prerequisites

**Cygwin** [Cygwin Web site](#)

Click the “setup” link and execute.

During the installer, check the package:

- Devel > git

**Qt SDK** You need the development files of the Qt 5 library, available on the [Qt website](#). The currently supported Qt version is 5.5.0.

**NSIS (installer only)** You will only need NSIS installed if you want to create an installer for the XiVO Client.

[NSIS download page](#)

During the installer, choose the full installation.

The XiVO Client NSIS script file uses two plug-ins:

- the NSIS Application Association Registration Plug-in ([download page](#))
- the NsProcess Plug-in ([download page](#))

For each plug-in, download and extract the plug-in and place:

- the DLL from /Plugins in the NSIS/Plugins directory
- the .nsh from /Include in the NSIS/Include directory

**Get sources** In a Cygwin shell:

```
git clone git://github.com/xivo-pbx/xivo-client-qt.git
cd xivo-client-qt
touch xivoclient/qt-solutions/qtsingleapplication/src/{QtSingleApplication,QtLockedFile}
```

## Building

**Path configuration** You must change the values in C:\Cygwin\home\user\xivo-client-qt\build-deps to match the paths of your installed programs. You must use an editor capable of understanding Unix end of lines, such as [Notepad++](#).

Replace C:\ with /cygdrive/c and backslashes (\) with slashes (/). You must respect the case of the directory names. Paths containing spaces must be enclosed in double quotes (").

For example, if you installed NSIS in C:\Program Files (x86)\nsis, you should write:

```
WIN_NSIS_PATH="/cygdrive/c/Program files (x86)/nsis"
```

**Build** In a Cygwin shell:

```
source build-deps
export PATH=$WIN_QT_PATH/bin:$WIN_MINGW_PATH/bin:$PATH

qmake
mingw32-make SHELL=
```

Binaries are available in the bin directory.

The version of the executable is taken from the `git describe` command.

**Launch** You can launch the built executable with:

```
source build_deps
PATH=$WIN_QT_PATH/bin:$PATH bin/xivoclient
```

**Package** To create the installer:

```
mingw32-make pack
```

This will result in a .exe file in the current directory.

**Build options** To add a console:

```
qmake CONFIG+=console
```

To generate debug symbols:

```
mingw32-make SHELL= DEBUG=yes
```

**Clean**

```
mingw32-make distclean
```

**Building the XiVO Client on GNU/Linux platforms** This page explains how to build an executable of the XiVO Client from its sources for GNU/Linux.

### Prerequisites

- Qt5 library development files: [Qt website](#) (Ubuntu packages `qt5-default` `qt5-qmake` `qttools5-dev-tools` `qttools5-dev` `libqt5svg5-dev`). The currently supported Qt version is 5.5.0.
- OpenGL development library - libGL (Debian package `libgl1-mesa-dev`)
- Git (Debian package `git`)
- Generic software building tools : `make`, `g++` ... (Debian package `build-essential`)

**Get sources** In a bash shell:

```
$ git clone git://github.com/xivo-pbx/xivo-client-qt.git
```

**Building** You need to have the Qt5 binaries (`qmake`, `lrelease`, ...) in your `$PATH`.

Launch `qmake` to generate the Makefile:

```
$ cd xivo-client-qt
$ /path/to/qt5/bin/qmake
```

This will also generate a file `versions.mak` that contains version informations about the code being compiled. It is necessary for compilation and packaging.

You can then launch `make`:

```
$ make
```

Binaries are available in the `bin` directory.

The version of the executable is taken from the `git describe` command.

**Build options** To generate debug symbols:

```
$ make DEBUG=yes
```

To compile the unit tests of the XiVO Client:

```
$ qmake CONFIG+=tests
```

or, if you have a recent version of Google Mock:

```
$ qmake CONFIG+=tests CONFIG+=gmock
```

To compile the XiVO Client ready for functional tests:

```
$ make FUNCTESTS=yes
```

### Cleaning

```
$ make distclean
```

**Launch** You can launch the built executable with:

```
$ LD_LIBRARY_PATH=bin bin/xivoclient
```

**Package** To create the Debian package, usable on Debian and Ubuntu, you first need to modify `build-deps` to locate the Qt 5 installation directory:

```
$ /path/to/qt5/bin/qmake -spec linux-g++
$ make
$ make pack
```

This will result in a `.deb` file in the current directory.

The version of the package is taken from the `git describe` command.

**Building the XiVO Client on Mac OS** This page explains how to build an executable of the XiVO Client from its sources for Mac OS.

### Mac OS Prerequisites

**Developer tools** You will need an Apple developer account to get development tools, such as GCC. To log in or sign in, go to the [Developer portal of Apple](#). In the Downloads section, get the Command line Tools for XCode and install them. You might want to get XCode too, but it is rather big.

**Qt SDK** You need the development files of the Qt 5 library, available on the [Qt website](#). The currently supported Qt version is 5.5.0.

**Get sources** In a bash shell, enter:

```
$ git clone git://github.com/xivo-pbx/xivo-client-qt.git
```

**Building** Launch `qmake` to generate the Makefile:

```
$ cd xivo-client-qt
$ /path/to/qt5/bin/qmake -spec macx-g++
```

This will also generate a file `versions.mak` that contains version informations about the code being compiled. It is necessary for compilation and packaging.

You can then launch `make`:

```
$ make
```

Binaries are available in the `bin` directory.

The version of the executable is taken from the `git describe` command.

**Debug build** Add `DEBUG=yes` on the command line:

```
$ make DEBUG=yes
```

### Cleaning

```
$ make distclean
```

**Launch** You can launch the built executable with:

```
$ DYLD_LIBRARY_PATH=bin bin/xivoclient.app/Contents/MacOS/xivoclient
```

**Package** You need to have the bin directory of Qt in your `$PATH`.

To create the app bundle:

```
$ make pack
```

This will result in a `.dmg` file in the current directory.

The version of the package is taken from the `git describe` command.

## Building old versions

### Building old versions

#### 1.1.23 - Gallifrey

##### Build

- Download `this patch`
- `git checkout xivo-client-1.1.23`
- `git apply xivoclient-1.1.23.patch`
- Edit Makefile and set the variable `QMAKE` to the path of your `qmake`
- `make all`

##### Package (macos)

- Edit `cross/macos-pack.sh` and set `QT_PATH`
- `./cross/macos-pack.sh`

#### 1.0.15 - Dalek

##### Build (windows)

- Download `this patch`
- Edit the patch and set the paths to Qt, NSIS, etc.
- (cygwin) `git checkout xivo-client-1.0.15`
- (cygwin) `make all-win32`
- (qt cmd) `mingw32-make win32-baselib`

- (qt cmd) mingw32-make win32-xivoclient
- (qt cmd) mingw32-make win32-plugins

### Package (windows)

- (cygwin) make win32packdyn-xivoclient

### Coding the XiVO Client

#### Project folder map

**baselib** The folder *baselib* contains all files necessary to build the baselib. It contains the necessary code and data structures to communicate with the XiVO CTI server.

This library is designed to be reusable by other XiVO CTI clients. If you want to build it without the rest of the XiVO Client, go in its folder and type:

```
$ qmake && make
```

The library will be available in the new bin folder.

**xivoclient** The folder *xivoclient* contains all other source files included in the XiVO Client.

*src* contains the source code files, *images* contains the images, *i18n* contains the translation files and *qtaddons* contains some Qt addons used by the XiVO Client.

**src** The source files are separated in three categories :

- the XiVO Client itself, the source files are directly in *src*.
- the XLet library (*xletlib*) contains the code common to multiple XLets (plugins), like the XLet base class and mainly GUI stuff.
- the XLets themselves (*xlets*), each one is in a *xlets/something* subfolder.

Each XLet is compiled into a dynamic library, but some XLets are still compiled within the xivoclient executable instead of in a separated library. They are marked with a *\*-builtin* subfolder name.

**delivery** This folder contains all license informations necessary for the XiVO Client to be redistributed, i.e. the GNU GPLv3 and the additional requirements.

**Configuration access** The settings of the application are stored in BaseEngine for runtime and in files when the client is closed :

- *~/.config/XiVO* on GNU/Linux systems
- (what about other platforms?)

There are now 3 sets of functions from BaseEngine that you can use to read/store settings :

**getConfig() / setConfig()** They are proxy methods to use the BaseConfig object inside BaseEngine. They use QVariantMap to store the settings values. They are currently used to store/retrieve options used in the ConfigWidget.

**You can find the available keys to access data in the detailed Doxygen documentation of BaseEngine, or in *baseengine.h*.**

Note that the settings stored in BaseConfig won't be written in the configuration file if BaseEngine is not aware of their existence (loaded in *loadSettings* and saved in *saveSettings*).



**getSettings()** Through this function, you can access the lowest level of configuration storage, QSettings. It also contains the options stored in BaseConfig, but is less easy to use.

This direct access is used for purely graphical settings, only used to remember the appearance of the GUI until the next launch. These settings don't have to be shared with other widgets, and storing them directly in QSettings avoids writing code to import/export to/from BaseConfig.

**getProfileSetting() / setProfileSetting()** This pair of methods allow you to read/write settings directly in QSettings, but specifically for the current configuration profile.

**Configuration profiles** When starting XiVO Client with an argument, this argument is interpreted as a profile name. This profile name allows you to separate different profiles, with different configuration options.

For example, configuration profile “profileA” will auto-connect with user A and password B and “profileB” will not auto-connect, but is set to connect with user C, no password remembered. To invoke these profiles, use :

```
$ xivoclient profileA
$ xivoclient profileB
```

The default configuration profile is default-user.

**Recognizing / extracting phone numbers** Of course, working on XiVO Client implies working with phone numbers. But how to interpret them easily, when we are not sure of the format they're in?

You can use the PhoneNumber namespace (*baselib/src/phonenumbers.h*) to do that, it contains routines for recognition/extraction of phone numbers, that way you don't have to parse manually.

These subroutines are pretty basic for the moment, if you need/want to improve them, feel free to do it.

**Retrieving CTI server infos** Informations are synchronized from the server to the BaseEngine when the client connects.

It is stored in BaseEngine in “lists”. It is stored in a format close to the one used to transmit it, so you can see the CTI protocol definition for further documentation.

Each list contains objects of different type. These types are :

- channel
- user
- phone
- trunk
- agent
- queue
- group
- meetme
- voicemail
- queuemember
- parking

Each type corresponds to a class derived from XInfo, e.g. channel infos are stored in ChannelInfo objects.

The basic attributes of all objects are 3 strings: the IPBX ID, the XiVO object ID and the extended ID of the object, which is the two previous attributes linked with a “/”.

**Listen to IPBX events** If you want your XLet to receive IPBX/CTI events, you can do so by inheriting the `IPBXListener` interface.

You must specify which type of events you want to listen. This depends of the implemented functions in the CTI server. You can register to listen these events by calling the `IPBXListener` method :

```
registerListener(xxx);
```

For now, `xxx`, the event type, can take the values : `* chitchat` `* history` `* records_campaign` `* queuestats`

On reception of the specified type of event, `BaseEngine` will call the `IPBXListener` method `parseCommand(QVariantMap)`.

You should then reimplement this method to make it process the event data, stored in the `QVariantMap` parameter.

**The parking XLet** There are two concepts here : `* Parked calls`: These calls have been parked by a switchboard or an operator. They are waiting to be answered by a specific person, unlike a queue, where calls will be answered by one of the agents of the group associated to the queue. Each parked call is given a phone number so that the call can be answered by everyone.

- `Parking lots`: They are containers for parked calls. Each parking lot has a phone number, used to identify where to send the call we want to park.

`ParkingWidget` represents a parking lot and contains a table that stores all parked calls.

**Adding new XLets** When you want to add a new XLet, you can use the basic `XLetNull`, that only prints “Hello World”. Here is a little script to accelerate the copy from `XLetNull`.

```
#!/usr/bin/env sh

newname="newname" # Replaces xletnull
NewName="NewName" # Replaces XLetNull & XletNull
NEWNAME="NEWNAME" # Replaces XLETNULL

if [! -d xletnull] ; then

 echo "Please execute this script in XIVO_CLIENT/plugins"
 echo $newname
 exit 1
fi

cp -r xletnull $newname
cd $newname
rm -f moc* *.o Makefile

for f in $(find . -type f -print) ; do
 mv $f `echo $f | sed s/xletnull/$newname/`
done

find . -type f -exec sed -i "s/xletnull/$newname/g;s/X[Ll]etNull/$NewName/g;s/XLETNULL/$NEWNAME/g"
```

Before executing the script, just replace the first three variables with the name of the new XLet.

Then, you must add a line in `xivoclient/xlets.pro` to add your new directory to the `SUBDIRS` variable.

Then you can start implementing your new class. The `<xletname>Plugin` class is only an interface between the main app and your XLet.

**Translations** If you want to localize your XLet, there are four steps.

**Modify the sources** In the `<xletname>Plugin` constructor, add the line :

```
b_engine->registerTranslation("/: <xletname>_%1");
```

before the return instruction.

**Modify the project file** Add these lines in the `.pro` file in your XLet directory :

```
TRANSLATIONS = <xletname>_fr.ts TRANSLATIONS += <xletname>_nl.ts
```

```
RESOURCES = res.qrc
```

Replace `fr` and `nl` with the languages you want.

**Create the resource file** In a file `res.qrc` in your XLet directory, put these lines :

```
<!DOCTYPE RCC><RCC version="1.0">
 <qresource>
 <file><xletname>_fr.qm</file>
 <file><xletname>_nl.qm</file>
 </qresource>
</RCC>
```

These files will be embedded in the Xlet library binary.

**Create the translation files** In your XLet directory, run :

```
lupdate <xletname>.pro
```

This creates as much `.ts` translation files as specified in the `.pro` file. You can now translate strings in these file.

The XLet will now be compiled and translated.

**Add a new XLet** For now, it is not possible to add easily an XLet without changing the CTI server configuration files.

If you just want to test your new XLet, you can add the following line in `baseengine.cpp` :

```
m_capaxlets.push_back(QVariantList() << QVariant("<xletname>") << QVariant("tab"));
```

right after the line

```
m_capaxlets = datamap.value("capaxlets").toList();
```

You can replace `"tab"` with `"grid"` or `"dock"`.

**Add a translation** This is definitely not something funny and not easy to automatize.

You have to add, in every `.pro` file of the project (except `xlets.pro` and all those that don't need translations), a line

```
TRANSLATIONS += <project>_<lang>.ts
```

Replace `<project>` with the project name (`xivoclient`, `baselib`, `xlet`) and `<lang>` by the identifier of your language (`en`, `fr`, `nl`, ...) Then you have to add, in every `.qrc` file, the `.qm` files corresponding to the ones you added in the `.pro` files, such as :

```
<file><project>_<lang>.qm</file>
```

in the `<qresource>` section of these XML `.qrc` files.

After that, you have to run, in the XiVO Client root directory, something like :

```
find . -name *.pro -exec lupdate { } ;
```

This will create or update all `.ts` translation files registered in the `.pro` files.

You can then start translating the strings in these files, in the `xivoclient/i18n` folder.

**Code modification** If you want to be able to select your new language from within the XiVO Client, you have to add it in the interface.

For that, you can add your new language in the `m_locale_cbox` QCombobox in ConfigWidget.

**CTI debugging tool** If you have a problem and you want to see what is going on between the CTI server and client, you can use a specific script, designed specifically for XiVO, instead of using something like Wireshark to listen network communications.

**Profiling** To get profiling informations on the XiVO Client:

- Compile the XiVO Client with debugging symbols
- Run the command:

```
LD_LIBRARY_PATH=bin valgrind --tool=callgrind bin/xivoclient
```

- Quit the client
- Open the generated file `callgrind.out.<pid>` with KCacheGrind

**Automatic checking tools** We use two tools to check the source code of the XiVO Client: CppCheck et Valgrind.

**CppCheck** Usage:

```
cppcheck -I baselib/src -I xivoclient/src .
```

**Valgrind (Memcheck)** Usage:

```
LD_LIBRARY_PATH=bin valgrind --leak-check=full --suppressions=valgrind.supp --num-callers=30 --ge
```

You need to fill a file `valgrind.supp` with Valgrind suppressions, to avoid displaying errors in code you have no control over.

Here is a template `valgrind.supp` you can use. All memory in the XiVO Client is allocated using the new operator, so all calls to `malloc` and `co.` must come from libraries:

```
{
 malloc
 Memcheck:Leak
 fun:malloc
 ...
}

{
 calloc
 Memcheck:Leak
 fun:calloc
 ...
}

{
 realloc
 Memcheck:Leak
 fun:realloc
 ...
}

{
 memalign
 Memcheck:Leak
```



**Updating translations on transifex** Run the following commands from the root of the xivo-client-qt project:

```
make pushttr
```

After this command, you can visit [Transifex](#), and check that the xivo-client is 100% translated for your language. Once all the translations have been checked, run the 3 following commands:

```
make pulltr
git commit
git push
```

**Warning:** Under Arch Linux, you must have qt5 installed and prepend `QT_PATH=/usr/bin` before `make {pull,push}tr`.

**Add a new XiVO Client locale** Localizing the XiVO Client goes through four steps :

- Creating the new translation in Transifex
- Generating the translation files
- Embedding the translation in the binaries
- Displaying the new locale to be chosen

**Creating the new translation in Transifex** Log into Transifex and click the `Create language` option.

**Generate translation files** The translation files will be automatically generated from the source code.

For the command to create files for your locale, you need to ensure it is listed in the project file.

There are a few project files you should edit, each one will translate a module of the XiVO Client :

- `baselib/baselib.pro`
- `xivoclient/xivoclient.pro`
- `xivoclient/xletlib.pro`
- `xivoclient/src/xlets/*/*.pro`

In these files, you should add a line like this one:

```
TRANSLATIONS += $$ROOT_DIR/i18n/xivoclient_fr.ts
```

This line adds a translation file for french. Please replace `fr` by the code of your locale. The `$$ROOT_DIR` variable references either `xivoclient` or `baselib`.

You can use a command like the following to automate this (`$LANG` is the new language):

```
find . -name '*.pro' -exec sed -i -e 's|^TRANSLATIONS += $$\{?\?ROOT_DIR\}\?/i18n/\(..*\)_en.ts|\0\nT
```

To actually create the files, you will have to use the translation managing script. But first, you must tell the script about your new locale. Edit the `utils/translations.sh` file and add your locale to the `LOCALES` variable. Then, you can run the script:

```
$ make pulltr
```

**Embed the translation files** For each project previously edited, you should have a corresponding `.qrc` file. These resource files list all files that will be embedded in the XiVO Client binaries. You should then add the corresponding translation files like below:

```
<file>obj/xivoclient_fr.qm</file>
```

This embeds the French translation of the `xivoclient` module, corresponding to the translation file above. The path is changed to `obj/` because the `.qm` file will be generated from the `.ts` file.

You can use a command like the following to automate this (\$LANG is the new language):

```
find . -name '*.qrc' -exec sed -i -e 's|^(\ *)<file>\(.*\)obj/\(.*\)_fr.qm</file>|\0\n\1<file>\2
```

**Display the new locale** You have to edit the source file `xivoclient/src/configwidget.cpp` and add the entry corresponding to your locale in the locale-choosing combobox.

## Quality assurance

### Testing architecture

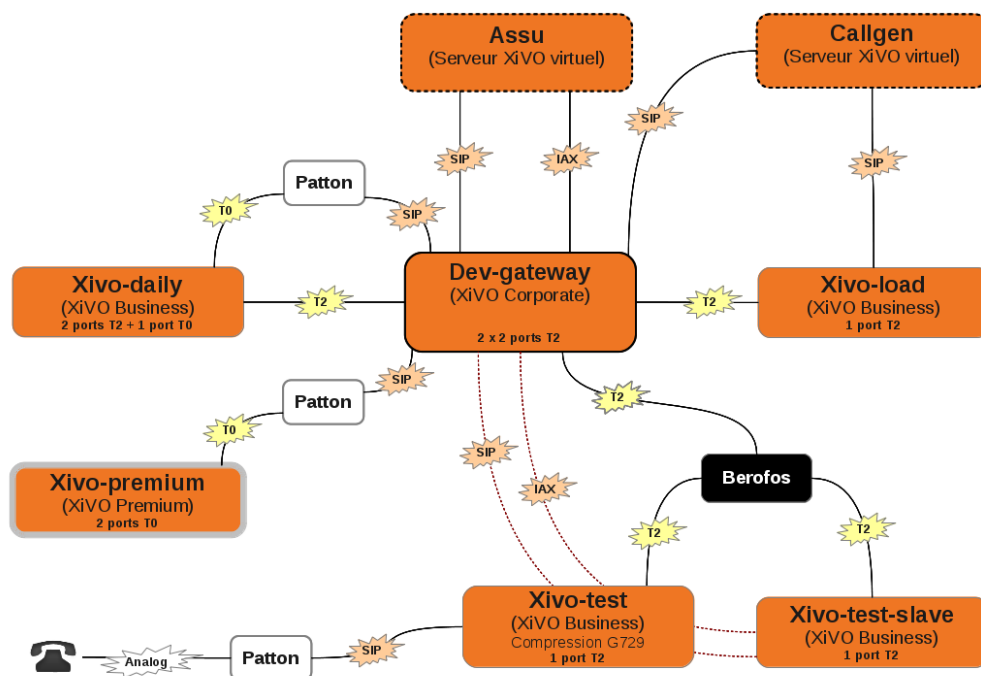


Fig. 1.113: Testing architecture

Legend:

- `assu` is our production XIVO, used to make calls in the company. We also use it as a source of “external” calls to the test servers.
- `dev-gateway` is a simple gateway, to link all other servers.
- `xivo-daily` is reinstalled every day and runs all the automatic tests in `xivo-acceptance`.
- `xivo-load` handles a lot of calls all day long, and we monitor the system metrics while it does.
- `callgen` makes the calls towards `xivo-load`
- `xivo-test` and `xivo-test-slave` are used for manual tests we run before each release
- `xivo-premium` (not yet installed) will allow us to test the new `xivo-premium` hardware

## Troubleshooting

The list of current bugs can be found on [the official XiVO issue tracker](#).

### Transfers using DTMF

When transferring a call using DTMF (\*1) you get an *invalid extension* error when dialing the extension.

The workaround to this problem is to create a preprocess subroutine and assign it to the destinations where you have the problem.

Under *Services* → *IPBX* → *IPBX configuration* → *Configuration files* add a new file containing the following dialplan:

```
[allow-transfer]
exten = s,1,NoOp(## Setting transfer context ##)
same = n,Set(__TRANSFER_CONTEXT=<internal-context>)
same = n,Return()
```

Do not forget to substitute <internal-context> with your internal context.

Some places where you might want to add this preprocess subroutine is on queues and outgoing calls to be able to transfer the called person to another extension.

### Fax detection

XiVO **does not currently support Fax detection**. The following describe a workaround to use this feature. The behavior is to answer all incoming (external) call, wait for a number of seconds (4 in this example) : if a fax is detected, receive it otherwise route the call normally.

**Note:** This workaround works only :

- on incoming calls towards an User (and an User only),
- if the incoming trunk is a DAHDI or a SIP trunk,
- if the user has a voicemail which is activated and with the email field filled
- XiVO >= 13.08 (needs asterisk 11)

Be aware that this workaround will probably not survive any upgrade.

1. In the Web Interface and under *Services* → *IPBX* → *IPBX configuration* → *Configuration files* add a new file named *fax-detection.conf* containing the following dialplan:

```
;; Fax Detection
[pre-user-global-faxdetection]
exten = s,1,NoOp(Answer call to be able to detect fax if call is external AND user has an email)
same = n,GotoIf("${XIVO_CALLORIGIN}" = "extern"?:return)
same = n,GotoIf("${XIVO_USEREMAIL}?:return)
same = n,Set(FAXOPT(faxdetect)=yes) ; Activate dynamically fax detection
same = n,Answer()
same = n,Wait(4) ; You can change the number of seconds it will wait for fax (4 to 6 is good)
same = n,Set(FAXOPT(faxdetect)=no) ; If no fax was detected deactivate dynamically fax detection
same = n(return),Return()

exten = fax,1,NoOp(Fax detected from ${CALLERID(num)} towards ${XIVO_DSTNUM} - will be sent u
same = n,GotoIf("${CHANNEL(channeltype)}" = "DAHDI"?changeechocan:continue)
same = n(changeechocan),Set(CHANNEL(echocan_mode)=fax) ; if chan type is dahdi set echo
same = n(continue),Gosub(faxtomail,s,1("${XIVO_USEREMAIL}))
```



2. In the file `/etc/xivo/asterisk/xivo_globals.conf` set the global user subroutine to `pre-user-global-faxdetection`: this subroutine will be executed each time a user is called:

```
XIVO_PRESUBR_GLOBAL_USER = pre-user-global-faxdetection
```

3. Reload asterisk configuration (both for dialplan and dahdi):

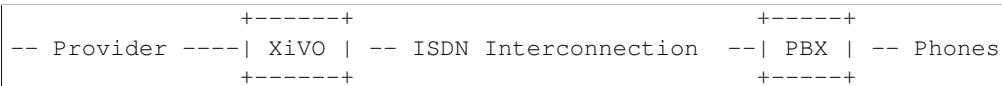
```
asterisk -rx 'core reload'
```

## Berofos Integration with PBX

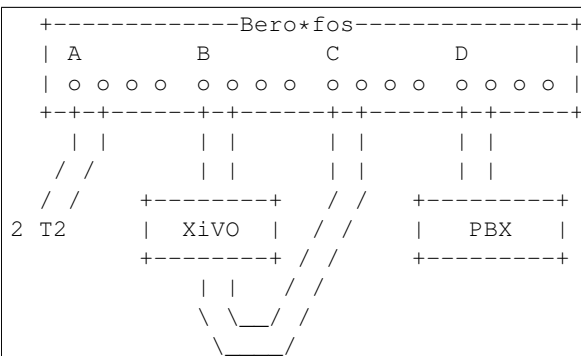
You can use a Berofos failover switch to secure the ISDN provider lines when installing a XiVO in front of an existing PBX. The goal of this configuration is to mitigate the consequences of an outage of the XiVO : with this equipment the ISDN provider links could be switched to the PBX directly if the XiVO goes down.

**XiVO does not offer natively** the possibility to configure Berofos in this failover mode. This section describes a workaround.

Logical view:



Connection:



The following describes how to configure your XiVO and your Berofos.

1. Follow the Berofos general configuration (firmware, IP, login/password) described in the the [Berofos Installation and Configuration](#) page.
2. When done, apply these specific parameters to the berofos:

```
bnfos --set scenario=1 -h 10.105.2.26 -u admin:berofos
bnfos --set mode=1 -h 10.105.2.26 -u admin:berofos
bnfos --set modedef=1 -h 10.105.2.26 -u admin:berofos
bnfos --set wdog=1 -h 10.105.2.26 -u admin:berofos
bnfos --set wdogdef=1 -h 10.105.2.26 -u admin:berofos
bnfos --set wdogitime=60 -h 10.105.2.26 -u admin:berofos
```

3. Add the following script `/usr/local/sbin/berofos-workaround`:

```
#!/bin/bash
Script workaround for berofos integration with a XiVO in front of PABX

res=$(/usr/sbin/service asterisk status)
does_ast_run=$?
if [$does_ast_run -eq 0]; then
 /usr/bin/logger "$0 - Asterisk is running"
 # If asterisk is running, we (re)enable wdog and (re)set the mode
 /usr/bin/bnfos --set mode=1 -f fos1 -s
 /usr/bin/bnfos --set modedef=1 -f fos1 -s

```

```

/usr/bin/bnfos --set wdog=1 -f fos1 -s

Now 'kick' berofos ten times each 5 seconds
for ((i == 1; i <= 10; i += 1)); do
 /usr/bin/bnfos --kick -f fos1 -s
 /bin/sleep 5
done
else
 /usr/bin/logger "$0 - Asterisk is not running"
fi

```

4. Add execution rights to script:

```
chmod +x /usr/local/sbin/berofos-workaround
```

5. Create a cron to launch the script every minutes /etc/cron.d/berofos-cron-workaround:

```

Workaround to berofos integration
MAILTO=""

*/1 * * * * root /usr/local/sbin/berofos-workaround

```

## Upgrading from XiVO 1.2.3

1. There is an issue with xivo-libsccp and pf-xivo-base-config during an upgrade from 1.2.3:

```

dpkg: error processing /var/cache/apt/archives/pf-xivo-base-config_13%3a1.2.4-1_all.deb (--un
trying to overwrite '/etc/asterisk/sccp.conf', which is also in package xivo-libsccp 1.2.3.1-
...
Errors were encountered while processing:
/var/cache/apt/archives/pf-xivo-base-config_13%3a1.2.4-1_all.deb
E: Sub-process /usr/bin/dpkg returned an error code (1)

```

2. You have to remove /var/lib/dpkg/info/xivo-libsccp.conf files:

```
rm /var/lib/dpkg/info/xivo-libsccp.conf files
```

3. You have to edit /var/lib/dpkg/info/xivo-libsccp.list and remove the following line:

```
/etc/asterisk/sccp.conf
```

4. and remove /etc/asterisk/sccp.conf:

```
rm /etc/asterisk/sccp.conf
```

5. Now, you can launch xivo-upgrade to finish the upgrade process

## CTI server is unexpectedly terminating

If you observe that your CTI server is sometimes unexpectedly terminating with the following message in /var/log/xivo-ctid.log:

```
(WARNING) (main): AMI: CLOSING
```

Then you might be in the case where asterisk generates lots of data in a short period of time on the AMI while the CTI server is busy processing other thing and is not actively reading from its AMI connection. If the CTI server takes too much time before consuming some data from the AMI connection, asterisk will close the AMI connection. The CTI server will terminate itself once it detects the connection to the AMI has been lost.

There's a workaround to this problem called the ami-proxy, which is a process which buffers the AMI connection between the CTI server and asterisk. This should only be used as a last resort solution, since this increases the latency between the processes and does not fix the root issue.

To enable the ami-proxy, you must:

1. Add a file `/etc/systemd/system/xivo-ctid.service.d/ami-proxy.conf`:

```
mkdir -p /etc/systemd/system/xivo-ctid.service.d
cat >/etc/systemd/system/xivo-ctid.service.d/ami-proxy.conf <<EOF
[Service]
Environment=XIVO_CTID_AMI_PROXY=1
EOF
systemctl daemon-reload
```

2. Restart the CTI server:

```
systemctl restart xivo-ctid.service
```

If you are on a XiVO cluster, you must do the same procedure on the slave if you want the ami-proxy to also be enabled on the slave.

To disable the ami-proxy:

```
rm /etc/systemd/system/xivo-ctid.service.d/ami-proxy.conf
systemctl daemon-reload
systemctl restart xivo-ctid.service
```

## Agents receiving two ACD calls

An agent can sometimes receive more than 1 ACD call at the same time, even if the queues he's in have the "ringinuse" parameter set to no (default).

This behaviour is caused by a bug in asterisk: <https://issues.asterisk.org/jira/browse/ASTERISK-16115>

It's possible to workaround this bug in XiVO by adding an agent *subroutine*. The subroutine can be either set globally or per agent:

```
[pre-limit-agentcallback]
exten = s,1,NoOp()
same = n,Set(LOCKED=${LOCK(agentcallback-${XIVO_AGENT_ID})})
same = n,GotoIf(${LOCKED}?not-locked,1)
same = n,Set(GROUP(agentcallback)=${XIVO_AGENT_ID})
same = n,Set(COUNT=${GROUP_COUNT(${XIVO_AGENT_ID}@agentcallback)})
same = n,NoOp(${UNLOCK(agentcallback-${XIVO_AGENT_ID})})
same = n,GotoIf(${COUNT} <= 1]?too-many-calls,1)
same = n,Return()

exten = not-locked,1,NoOp()
same = n,Log(ERROR,Could not obtain lock)
same = n,Wait(0.5)
same = n,Hangup()

exten = too-many-calls,1,NoOp()
same = n,Log(WARNING,Not calling agent ID/${XIVO_AGENT_ID} because already in use)
same = n,Wait(0.5)
same = n,Hangup()
```

This workaround only applies to queues with agent members; it won't work for queues with user members.

Also, the subroutine prevent asterisk from calling an agent twice by hangingup the second call. In the agent statistics, this will be shown as a non-answered call by the agent.

## PostgreSQL localization errors

The database and the underlying *database cluster* used by XiVO is sensitive to the system locale configuration. The locale used by the database and the database cluster is set when XiVO is installed. If you change your system

locale without particular attention to PostgreSQL, you might make the database and database cluster temporarily unusable.

When working with locale and PostgreSQL, there's a few useful commands and things to know:

- `locale -a` to see the list of currently available locales on your system
- `locale` to display information about the current locale of your shell
- `grep ^lc_ /etc/postgresql/9.4/main/postgresql.conf` to see the locale configuration of your database cluster
- `sudo -u postgres psql -l` to see the locale of your databases
- the `/etc/locale.gen` file and the associated `locale-gen` command to configure the available system locales
- `systemctl restart postgresql.service` to restart your database cluster
- the PostgreSQL log file located at `/var/log/postgresql/postgresql-9.4-main.log`

---

**Note:** You can use any locale with XiVO as long as it uses an UTF-8 encoding.

---

**Database cluster is not starting** If the database cluster doesn't start and you have the following errors in your log file:

```
LOG: invalid value for parameter "lc_messages": "en_US.UTF-8"
LOG: invalid value for parameter "lc_monetary": "en_US.UTF-8"
LOG: invalid value for parameter "lc_numeric": "en_US.UTF-8"
LOG: invalid value for parameter "lc_time": "en_US.UTF-8"
FATAL: configuration file "/etc/postgresql/9.4/main/postgresql.conf" contains errors
```

Then this usually means that the locale that is configured in `postgresql.conf` (here `en_US.UTF-8`) is not currently available on your system, i.e. does not show up the output of `locale -a`. You have two choices to fix this issue:

- either make the locale available by uncommenting it in the `/etc/locale.gen` file and running `locale-gen`
- or modify the `/etc/postgresql/9.4/main/postgresql.conf` file to set the various `lc_*` options to a locale that is available on your system

Once this is done, restart your database cluster.

**Can't connect to the database** If the database cluster is up but you get the following error when trying to connect to the asterisk database:

```
FATAL: database locale is incompatible with operating system
DETAIL: The database was initialized with LC_COLLATE "en_US.UTF-8", which is not recognized by
HINT: Recreate the database with another locale or install the missing locale.
```

Then this usually means that the database locale is not currently available on your system. You have two choices to fix this issue:

- either make the locale available by uncommenting it in the `/etc/locale.gen` file, running `locale-gen` and restarting your database cluster
- or *recreate the database using a different locale*

**Error during the upgrade** Then you are mostly in one of the cases described above. Check your log file.

**Error while restoring a database backup** If during a database restore, you get the following error:

```
pg_restore: [archiver (db)] Error while PROCESSING TOC:
pg_restore: [archiver (db)] Error from TOC entry 4203; 1262 24745 DATABASE asterisk asterisk
pg_restore: [archiver (db)] could not execute query: ERROR: invalid locale name: "en_US.UTF-8"
 Command was: CREATE DATABASE asterisk WITH TEMPLATE = template0 ENCODING = 'UTF8' LC_COLLATE = 'en_US.UTF-8' LC_CTYPE = 'en_US.UTF-8';
```

Then this usually means that your database backup has a locale that is not currently available on your system. You have two choices to fix this issue:

- either make the locale available by uncommenting it in the `/etc/locale.gen` file, running `locale-gen` and restarting your database cluster
- or if you want to restore your backup using a different locale (for example `fr_FR.UTF-8`), then restore your backup using the following commands instead:

```
sudo -u postgres dropdb asterisk
sudo -u postgres createdb -l fr_FR.UTF-8 -O asterisk -T template0 asterisk
sudo -u postgres pg_restore -d asterisk asterisk-*.dump
```

**Error during master-slave replication** Then the slave database is most likely not using an UTF-8 encoding. You'll need to *recreate the database using a different locale*

**Changing the locale (LC\_COLLATE and LC\_CTYPE) of the database** If you have decided to change the locale of your database, you must:

- make sure that you have enough space on your hard drive, more precisely in the file system holding the `/var/lib/postgresql` directory. You'll have, for a moment, two copies of the asterisk database.
- prepare for a service interruption. The procedure requires the services to be restarted twice, and the system performance will be degraded while the database with the new locale is being created, which can take a few hours if you have a really large database.
- make sure the new locale is available on your system, i.e. shows up in the output of `locale -a`

Then use the following commands (replacing `fr_FR.UTF-8` by your locale):

```
xivo-service restart all
sudo -u postgres createdb -l fr_FR.UTF-8 -O asterisk -T template0 asterisk_newlocale
sudo -u postgres pg_dump asterisk | sudo -u postgres psql -d asterisk_newlocale
xivo-service stop
sudo -u postgres psql <<'EOF'
DROP DATABASE asterisk;
ALTER DATABASE asterisk_newlocale RENAME TO asterisk;
EOF
xivo-service start
```

You should also modify the `/etc/postgresql/9.4/main/postgresql.conf` file to set the various `lc_*` options to the new locale value.

For more information, consult the [official documentation on PostgreSQL localization support](#).

### Originate a call from the Asterisk console

It is sometimes useful to ring a phone from the asterisk console. For example, if you want to call the 1234 extension in context default:

```
channel originate Local/1234@default extension 42@xivo-callme
```

## WebRTC

- *http.conf* - asterisk's webserver must accept connection from outside, the listen address must be updated, for the sake of simplicity let's use 0.0.0.0, you can also pick an address of one of the network interfaces:

```
[general]
enabled=yes
bindaddr=0.0.0.0
bindport=5039
prefix=
tlsenable=yes
tlsbindaddr=127.0.0.1:5040
tlscertfile=/usr/share/xivo-certs/server.crt
tlsprivatekey=/usr/share/xivo-certs/server.key
servername=XiVO PBX
```

Do not forget to reload the configuration by the *module reload http* command on the Asterisk CLI.

- *rtp.conf* - the ICE support must be activated:

```
;
; RTP Configuration
;
[general]
;
; RTP start and RTP end configure start and end addresses
;
; Defaults are rtpstart=5000 and rtpend=31000
;
rtpstart=10000
rtpend=20000
;
; Whether to enable or disable UDP checksums on RTP traffic
;
; rtpchecksums=no
;
; The amount of time a DTMF digit with no 'end' marker should be
; allowed to continue (in 'samples', 1/8000 of a second)
;
; dtmftimeout=3000
icesupport=yes
stunaddr=stun.l.google.com:19302
```

The configuration is reloaded by *module reload res\_rtp\_asterisk.so*.

- WebRTC requires DTLS keys to be generated in */etc/asterisk/keys*. If you need to manually generate the DTLS certificates following instructions on the Asterisk Wiki: <https://wiki.asterisk.org/wiki/display/AST/Secure+Calling+Tutorial>. You just need to generate the TLS certificates (first call of *ast\_tls\_cert*), other steps are not necessary. Make sure asterisk can read files by executing: `chown -R asterisk.asterisk /etc/asterisk/keys`

## Community Documentation

This page provides links to resources on various topics around XiVO. They have been generously created by people from the community.

## Tutorials

Please note that these resources are provided on an “as is basis”. They have not been reviewed by the XiVO team, therefore the information presented may be inaccurate. We also accept resources provided in other languages besides English.

Unless specified, the license is [CC BY-SA](#).

Tutorial	Language	Level	Author	XiVO Version
Définition de XiVO pour la communauté et tutoriel (video)	English	Beginner	XiVO	2015
Xivo pour les nuls	French	Beginner	Nicolas	2012
Installing XiVO (YouTube series)	English	Beginner	VoIP-Nuiz	14.20
Start: how to create a user with a SIP line (YouTube series)	French	Beginner	VoIP-Nuiz	2014
Start: how to popup an URL (Document)	French	Beginner		
Start: how to create a context, users, voicemails, ring group, music on hold, conf.call	French	Beginner	Networklab	2014
Tips: post-installation of XiVO on Kimsufi	French	Intermediate	NyXD Systems	2015
Tips: username and password on XiVO	French	Intermediate	NyXD Systems	2015
Tips: self-hosting and telephony with XiVO	French	Intermediate	NyXD Systems	2015
XiVO provisioning + pfSense + siproxd + OVH	French	Intermediate	NyXD Systems	2015
SCCP provisioning, unsupported phones and no DHCP	French	Intermediate	NyXD Systems	2015
Date format on SCCP 7941	French	Intermediate	NyXD Systems	2015
Installing XiVO on Raspberry Pi (Raspivo)	French	Intermediate	Iris Network	2015
How to popup an url with CTIClient	French	Intermediate	Assonance	14.17
How to backup XiVO to external FTP with backup-ftp.sh	French	Intermediate	Yohan Vitu	2015
How to create a XiVO Client	French	Intermediate	Yohan Vitu	2015
How to configure a C610P IP on XiVO	French	Intermediate	Yohan Vitu	2015
How to export the phonebook of XiVO with phonebook_csv_export.py	French	Intermediate	Yohan Vitu	2015
How to use openVPN on XiVO	French	Expert	Yohan Vitu	2015
How configure SNOM M700 DECT	French	Intermediate	Jonathan Thomas	2015
Scripted provisioning for SNOM M700 DECT with specific scripts	French	Intermediate	Jonathan Thomas	2015
How to configure XiVO with Untangle firewall	English	Intermediate	Scott McCarthy (SMS IT Group)	16.04
<b>464</b>		ate	<b>Chapter 1. Table of Contents</b>	
How to use Keepalived with XiVO (high availability)	English	Expert	Eric Viel (Iper Telecom)	16.11
Getting Started with XiVO	English	Beginner	Nerd Vittles	16.07



## Contribute

We gladly accept new contributions. There are two ways to contribute:

- The preferred way: open a pull request on [Github](#) and add a line to this page (see: [Contributing to the Documentation](#)).
- You can also open a contribution ticket on the [bug tracker](#).

Note that we only accept documents in open formats, such as PDF or ODF.

### 1.1.2 Indices and tables

- [genindex](#)
- [search](#)

## 1.2 XiVO-CC Documentation



XiVO-CC is an application suite developed by [Avenacall](#) Group, and provides enhancements of the [XiVO](#) PBX contact center functionalities.

### 1.2.1 Table of Contents

#### Introduction

XiVO-CC provides enhancements of the XiVO PBX contact center functionalities. It gives especially access to outsourced statistics, real-time supervision screens, third-party CTI integration and recording facilities.

#### Installation and system configuration

The XiVO-CC software suite is made of several independent components. Depending on your system size, they can be installed on separate virtual or physical machines. In this section, we will explain how to install these components on a single machine.

#### Installation

**Installation** This page describes how to install the *XiVO CC*.

It describes the installation with the debian package of the whole *XiVO CC*.

---

**Note:** As a reference, the manual installation page is here [Manual configuration and installation](#).

---

#### Warning:

- the wizard **MUST** be passed on the *XiVO PBX*
- *XiVO PBX* will be reconfigured during the installation and must be restarted. You may accept the automatic restart during the installation or you need to restart it manually later before starting the docker containers.
- If you configure HA on XiVO, you have to reconfigure postgres for CC ...

**Overview** The following components will be installed :

- XuC : outsourced CTI server providing telephony events, statistics and commands through a WebSocket
- XuC Management : supervision web pages based on the XuC
- Pack Reporting : statistic summaries stored in a PostgreSQL database
- Totem Support : near-real time statistics based on [ElasticSearch](#)
- SpagoBI : BI suite with default statistic reports based on the Pack Reporting
- Recording Server : web server allowing to search recorded conversations
- Xuc Rights Management : permission provider used by XuC and Recording Server to manage the user rights

**Prerequisites** We will assume your **XiVO CC** server meets the following requirements:

- OS : Debian 8 (jessie), 64 bits.
- you have a *XiVO PBX* installed in a compatible version (basically the two components *XiVO* and *XiVO CC* have to be in the *same* version).
- the *XiVO PBX* is reachable on the network.
- the *XiVO PBX* **is setup** (wizard must be passed) with users, queues and agents, you must be able to place and answer calls.

For the rest of this page, we will make the following assumptions :

- the *XiVO PBX* has the IP 192.168.0.1
- some data (incoming calls, internal calls etc.) might be available on *XiVO* (otherwise, you will not see *anything* in the [check-list](#) below).
- the *XiVO CC* server has the IP 192.168.0.2
- the package *xivo-recording* is available on a custom Debian mirror. If this is not the case, you will need to skip the *apt-get install* commands and build the packages yourself.

**XiVO PBX Restrictions and Limitations** *XiVO PBX* enables a wide range of configuration, *XiVO-CC* is tested and validated with a number of restriction concerning configurations of *XiVO PBX*:

### General Configuration

- Do not activate Contexts Separation in *xivo-ctid* Configuration
- Users deactivation is not supported

### Queue Configuration

- Queue ringing strategy should not be *Ring All*
- Do not use pause on one queue status, in queue advanced configuration, autopause should be No or All
- Do not activate Call a member already on (*Asterisk ringinuse*) on *xivo* queue advanced configuration

### User And Agent Configuration

- All users and queues have to be in the same context
- Agent and Supervisors profiles should use the same Presence Group
- Agents and Phones should be in the same context for mobile agents
- Agents must not have a password in *XiVO* agent configuration page

**Install from repository** The installation and configuration of *XiVO CC* (with its *XiVO PBX* part) is handled by the *xivocc-installer* package which is available in the repository.

**Install process overview** The install process consists of three parts:

1. The first part is to manually run the `install-docker.sh` script to install docker and docker compose.
2. The second part is the installation of *XiVO CC* itself.
3. The third part is to install the extra package for the recording.

The installation is automatic and you will be asked few questions during the process:

- When asked to generate a pair of authentication keys, leave the password field empty.
- Before copying the authentication keys, you will be prompted for the *XiVO PBX* root password.
- Enter IP addresses of *XiVO PBX* and *XiVO CC*.
- *XiVO PBX* must restart, the question will prompt you to restart during the process or to restart later.

### Install Docker and Docker Compose

**Note:** To be run on the *XiVO CC* server

On a fresh debian install you will probably need to install the `ca-certificates` package:

```
apt-get install ca-certificates
```

Now you can download the script which will install docker and docker compose.

1. First, download the `install-docker.sh` script (in the following URL, replace **CURRENT\_VERSION** with the current version, e.g. *2016.04*):

```
wget https://gitlab.com/xivoxc/packaging/raw/CURRENT_VERSION/install/install-docker.sh -O ins
```

2. And execute the script:

```
chmod +x install-docker.sh
./install-docker.sh
```

### Install ntp server

**Note:** To be run on the *XiVO CC* server

The *XiVO CC* server and the *XiVO PBX* server must be synchronized to the same NTP source.

```
apt-get install ntp
```

Recommended configuration : you should configure the NTP server of the *XiVO CC* server towards the *XiVO PBX*. In our example it means to add the following line in the file `/etc/ntp.conf`:

```
server 192.168.0.1 iburst
```

### Install XiVO CC

**Note:** To be run on the *XiVO CC* server

This step will install the *XiVO CC* components via the `xivocc-installer` package. It is required to restart *XiVO PBX* during or after the setup process. The installer will ask whether you wish to restart *XiVO PBX* later.

**Warning:**

- This package must be installed on the XivoCC server.
- Wizard **MUST** be passed on the *XiVO PBX*.
- *XiVO PBX* services will need to be restarted. The installer will ask whether you wish to restart *XiVO PBX* during or after the setup process.

Also, check that you have following information:

- *XiVO PBX* root password;
- OpenSSH `PermitRootLogin` set to `yes` (you could revert to `no` after installation of XivoCC);
- *XiVO PBX*'s IP address;
- *XiVO CC* IP address (the one visible *by XiVO PBX*);
- Number of weeks to keep statistics;
- Number of weeks to keep recordings (beware of space disk);

Install the *xivocc-installer* package via *apt*:

1. Add the sources list with the key (in the following URL, replace **CURRENT\_VERSION** with the current version, e.g. *2016.04*):

```
echo "deb http://mirror.xivo.solutions/archive/ xivo-solutions-CURRENT_VERSION main" > /etc/a
wget http://mirror.xivo.solutions/xivo_current.key -O - | apt-key add -
```

2. Update your source list and install the package:

```
apt-get update
apt-get install xivocc-installer
```

## Install the recording

**Note:** To be run on the **XiVO PBX** server

---

To be able to install the package you must have the XiVO Solutions repository on your **XiVO PBX**. Then install on the **XiVO PBX** the debian package available in the repository.

```
apt-get install xivo-recording
```

During the installation, you will be asked for :

- the recording server IP (192.168.0.2)
- the XiVO name (it must not contain any space or “-” character).

If you have more than one XiVO, you must give a different name to each of them.

This package installs two dialplan subroutines :

- `xivo-incall-recording` : used to record incoming calls
- `xivo-outcall-recording` : used to record outgoing calls

To use the subroutines, you must edit the configuration file `/etc/xivo/asterisk/xivo_globals.conf` and assign them to chosen preprocess subroutines. E.g. :

```
XIVO_PRESUBR_FWD_QUEUE = xivo-incall-recording
...
XIVO_PRESUBR_GLOBAL_QUEUE = xivo-incall-recording
```

If you want to record on a gateway used with Xivo, you must not use the `xivo-recording` package but `gateway-recording`.

If you want to use call recording filtering, please install also:

```
apt-get install call-recording-filtering
```

During the installation, you will be asked for :

- the recording server address with protocol and port (<http://192.168.0.2:9400>)

Using the XivoCC configuration manager : <http://192.168.0.2:9100/> add user xuc as administrator to be able to get call history in web assistant.

**After-install steps** After the successful installation, start docker containers by an alias which was added to `~/.bashrc`

```
source ~/.bashrc
dcomp up -d
```

If you selected to restart *XiVO PBX* later, please do so when possible to apply the modifications made by the installer. The *XiVO CC* server will not be able to connect correctly to the database on *XiVO PBX*.

To restart XiVO services, on *XiVO PBX* server run

```
xivo-service restart all
```

**Reinstallation** To reinstall the package, it is required to `apt-get purge xivocc-installer` followed by `apt-get install xivocc-installer`. This will re-run the configuration of the package, download the docker compose template and setup *XiVO PBX*.

Purging the package will also **remove** the *xuc* and *stats* users from the *XiVO PBX* database.

## Known Issues

**To avoid problems when uninstalling, you should:**

- to uninstall, please use `apt-get purge xivocc-installer`
- if the process is aborted, it will break the installation, please `apt-get purge` and `apt-get install` again

**Checking Installed Version** Component version can be find in the log files, on the web pages for web components. You may also get the version from the docker container itself by typing :

```
docker exec -ti xivocc_xucmgt_1 cat /opt/docker/conf/appli.version
```

Change `xivocc_xucmgt_1` by the component version you want to check

**Using XivoCC** The various applications are available on the following addresses:

- Xuc-related applications: <http://192.168.0.2:8070/>
- SpagoBI: <http://192.168.0.2:9500/>
- Config Management: <http://192.168.0.2:9100/>
- Recording server: <http://192.168.0.2:9400/>
- Kibana: <http://192.168.0.2/>

## Post Installation



## User Configuration

- Using the configuration manager : <http://192.168.0.2:9100/> (default user avencall/superpass) add a user to be able to use the recording interface with proper rights.

---

**Note:** Xuc server default user is xuc, add xuc as administrator to be able to get call history in web assistant.

---

**Warning:** If you change the cti login username in xivo configuration, user has to be recreated with appropriate rights in configuration manager.

## SpagoBI

- Go to <http://192.168.0.2:9500/SpagoBI> (by default login: biadmin, password: biadmin)
- Update default language : go to “ Resources” > “Configuration management” > in the “Select Category” field, chose “LANGUAGE\_SUPPORTED” and change value of the label “SPAGOB.LANGUAGE\_SUPPORTED.LANGUAGE.default” in your language : fr,FR , en,US , ...
- Download the standard reports from [https://gitlab.com/xivocc/sample\\_reports/raw/master/spagobi/standardreports.zip](https://gitlab.com/xivocc/sample_reports/raw/master/spagobi/standardreports.zip)
- Import zip file in SpagoBI: ” Repository Management” > Click on “Browse” and choose the previous downloaded zip file > Click on “Import” (All default options, with Jasper Report Engine as Engine associations).

**XivoCC Default Report Sample** Use the database status report to check if replication and reporting generation is working :

**ACD outgoing calls** XivoCC agent can make outgoing calls through an outgoing queue. This brings the statistics and supervision visualization for outgoing ACD calls. However, some special configuration steps are required:

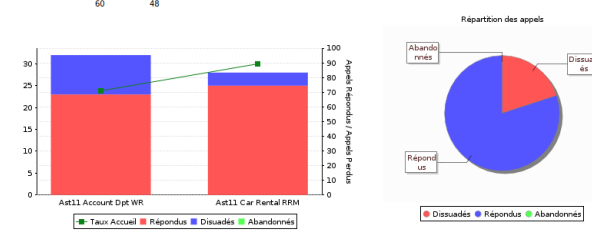
- You need to create an outgoing queue with a name starting with ‘out’, e.g. outgoing\_queue.

## Files d'attentes mensuelle 12

sauf samedi et dimanche

T1	15
T2	20

	Présentes	Répondus	TMC	Dissuadés	Raccrochés en file d'attente	Abandonnés en moins de T1	TMA	Transférés	Taux d'Accueil
Ast11 Account Dpt WR	32	23	00:02:49	9	0	0	00:00:05	0	71%
Ast11 Car Rental RRM	28	25	00:01:44	3	0	0	00:00:07	0	89%



## Etat de la base des statistiques

Dernier CEL

Dernier Queue log

339050470 12/16/15 3:40 PM

id	time
47057935	2015-12-16 15:40:39.027702

Queue specific

Queue periodic

time	queue ref	nb offered
12/16/15 3:15 PM	sales	10

time	queue	total
12/16/15 3:15 PM	sales	10

Agent periodic

Agent specific

Agent queue specific

time	agent	login time
12/16/15 3:15 PM	1645	0 years 0 mons 0 days 0 hours 14

time	agent num	nb offered
12/16/15 3:15 PM	1564	2

time	agent num	queue ref
12/16/15 3:15 PM	1603	travels

Call data

Call on queue

start time	src num	dst num	status	uniqueid
12/16/15 3:40 PM	loadtester	84552	null	1450276822.4231
12/16/15 3:40 PM	loadtester	84557	answer	1450276801.4227
12/16/15 3:39 PM	loadtester	84554	answer	1450276795.4223

queue time	queue re	agent nu	status
12/16/15 3:40 PM	travels	null	null
12/16/15 3:40 PM	carrental	1563	answered

Objets dans la configuration

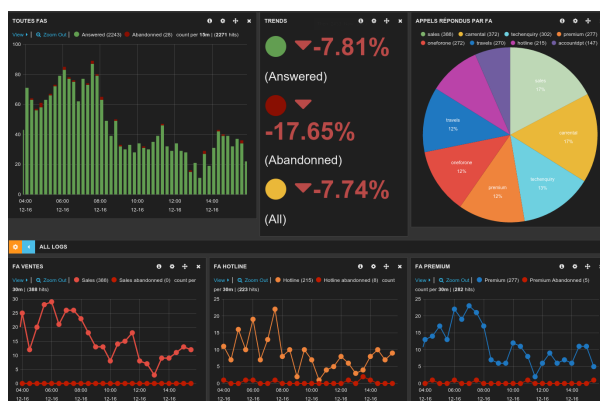
Agents	Queues	Agent groups	Extensions
173	19	8	420

- This queue must be configured with preprocess subroutine `xuc_outcall_acd`, without On-Hold Music (tab General), Ringing Time must be 0 and Ring instead of On-Hold Music must be activated (both tab Application).
- The subroutine must be deployed on the Xivo server (to `/etc/asterisk/extension_extra.d/` or through the web interface), the file is available from [https://gitlab.com/xivoxc/xucserver/raw/master/xivo/outbound/xuc\\_outcall\\_acd.conf](https://gitlab.com/xivoxc/xucserver/raw/master/xivo/outbound/xuc_outcall_acd.conf), with owner `asterisk:www-data` and rights `660`.
- You must also deploy the file [https://gitlab.com/xivoxc/xucserver/raw/master/xivo/outbound/generate\\_outcall\\_skills.py](https://gitlab.com/xivoxc/xucserver/raw/master/xivo/outbound/generate_outcall_skills.py) to `/usr/local/sbin/`, with owner `root:root` and rights `755`.
- Furthermore, you must replace the file `/etc/asterisk/queueskills.conf` by the following one <https://gitlab.com/xivoxc/xucserver/raw/master/xivo/outbound/queueskills.conf> (be sure to backup the original one), without changing the owner or rights
- And finally you need to add a new skill rule on the Xivo server: Services -> Call center -> Skill rules -> Add, with name `'select_agent'` and rules `'$agent > 0'`.

Once done, calls requested by an agent through the Cti.js with more than 6 digits are routed via the outgoing queue. You can change the number of digits using the parameter `xuc.outboundLength` in the `xuc`'s configuration.

**Totem Panels** Data replication can take some time if there are a lot of data in `xivo cel` and `queue log` tables. You may check `xivo-db-replication` log files (`/var/log/xivocc/xivo-db-replication.log`).

Preconfigured panels are available on <http://@IP/kibana/#/dashboard/file/queues.json> et <http://@IP/kibana/#/dashboard/file/agents.json> to be able to save this panels in elasticsearch database you have to sign on on request user `admin/Kibana`



## Post Installation Check List

- All components are running : `dcomp ps`
- Xuc internal database is synchronized with xivo check status page with <http://xivoccserver:8090/>
- CCManager is running, log a user and check if you can see and manage queues : <http://xivoccserver:8070/ccmanager>
- Web agent is running, log an agent and check if you can change the status : <http://xivoccserver:8070/agent>
- Web assistant is running, and you get call history : <http://xivoccserver:8070/>
- Check database replication status using spagobi system report
- Check elasticsearch database status (totem panels) [http://xivoccserver:9200/queue/logs/\\_status](http://xivoccserver:9200/queue/logs/_status)
- Check that you can listen to recordings <http://xivoccserver:9400/>
- Check totem panels <http://192.168.85.102/kibana>

##### reminder: Make sure to have few calls made in your XiVO, despite you will not see **anything** in totem or spagobi.



## Ldap Authentication

**Xuc** Configure LDAP authent for CCmanager, Web Assistant and Web Agent

You need to include in the compose.yml file a link to a specific configuration file by adding in xuc section a specific volume and an environment variable to specify the alternate config file location

```
xuc:

....

environment:
....
- CONFIG_FILE=/conf/xuc.conf

volumes:
- /etc/docker/xuc:/conf
```

Edit in /etc/docker/xuc/ a configuration file named xuc.conf to add ldap configuration (empty by default)

```
include "application.conf"

authentication {
 ldap {
 managerDN = "uid=company,ou=people,dc=company,dc=com" # user with read rights on the who
 managerPassword = "xxxxxxxxxx" # password for this user
 url = "ldap://ldap.company.com:389" # ldap URI
 searchBase = "ou=people,dc=company,dc=com" # ldap entry to use as search base
 userSearchFilter = "uid=%s" # filter to use to search users by
 }
}
```

Recreate the container : *dcomp up -d xuc*

**Manual configuration and installation** This section describes the manual installation of the XiVO CC components. In most cases you **SHOULD NOT** follow this page, and install the XiVO CC components via the *xivocc-installer* package (see [Installation](#)).

**Note:** We leave this page here :

- as a reference,
- and when one must install only a subset of the XiVO CC components (since it is not currently possible via the *xivocc-installer* package).

**Prerequisites** We will assume your XiVO CC server meets the following requirements:

- OS : Debian 8 (jessie), 64 bit
- the latest stable version of [Docker](#) is installed
- the latest stable version of [Docker-compose](#) is installed
- the XiVO PBX is reachable on the network
- the XiVO PBX is setup with users, queues and agents, you must be able to place and answer calls.

Note : Install only stable version of docker and docker compose.

We will make the following assumptions :

- the XiVO PBX has the IP 192.168.0.1

- some data (incoming calls, internal calls etc.) might be available on XiVO (otherwise, you will not see *anything* in the *Post Installation Check List*).
- the XiVO CC server has the IP 192.168.0.2
- the package xivo-recording is available on a custom Debian mirror. If this is not the case, you will need to skip the *apt-get install* commands and build the packages yourself.

### XiVO PBX configuration

**PostgreSQL configuration** Firstly, allow access to PostgreSQL from the outside. Edit `/etc/postgresql/9.4/main/postgresql.conf`:

```
listen_addresses = '*'
```

Add this line to `/etc/postgresql/9.4/main/pg_hba.conf`:

```
host asterisk all 192.168.0.2/32 md5
```

Create a user *stats* with read permissions :

```
sudo -u postgres psql asterisk << EOF
CREATE USER stats WITH PASSWORD 'stats';
GRANT SELECT ON ALL TABLES IN SCHEMA PUBLIC TO stats;
EOF
```

And run *xivo-service restart all* to apply these modifications.

### AMI configuration

- (XiVO PBX >= 15.18) Add a file `xuc.conf` in `/etc/asterisk/manager.d` directory with :

```
[xuc]
secret = xucpass
deny=0.0.0.0/0.0.0.0
permit=X.X.X.0/255.255.255.0
read = system,call,log,verbose,command,agent,user,dtmf,originate,dialplan
write = system,call,log,verbose,command,agent,user,dtmf,originate,dialplan
```

Replace `X.X.X.0` by your xivocc network

And reload the AMI :

```
asterisk -rx "manager reload"
asterisk -rx "manager show user xuc" and check your if previous configuration is displayed.
```

**CEL Configuration** Add some events in the CEL. Edit `/etc/asterisk/cel.conf`:

- For Asterisk 13:

```
[general]
enable = yes
apps = dial,park,queue
events = APP_START,CHAN_START,CHAN_END,ANSWER,HANGUP,BRIDGE_ENTER,BRIDGE_EXIT,USER_DEFINED,LINKED

[manager]
enabled = yes
```

and reload the cel module in Asterisk :

```
asterisk -rx "module reload cel"
```

**Customizations in the web interface** Create a user Xuc in *Services -> IPBX -> Users* with the following parameters:

- CTI login : xuc
- CTI password : 0000
- profil supervisor

Create a Web Services user in *Configuration -> Web Services Access* with the following parameters :

- Login : xivows
- Password : xivows
- Host : 192.168.0.2

Make sure **Multiqueues call stats sharing** is enabled in *Services -> IPBX -> Advanced configuration* tab.

**Phone integration** Do not forget to follow configuration steps detailed in [Required configuration for phone integration](#).

**Packages for the recording** Still on the xivo, install the package which will handle the recording :

```
apt-get update
apt-get install xivo-recording
```

During the installation, you will be asked for :

- the recording server IP (i.e. 192.168.0.2)
- and the XiVO name (it **must** not contain any space or “-” character).

If you have several XiVO, you **must** give a different name to each of them.

This package has installed two dialplan sub-routines :

- xivo-incall-recording : used to record incoming calls
- xivo-outcall-recording : used to record outgoing calls

You have to manually place them where you want.

If you want to record on a gateway used with Xivo, you must not use the xivo-recording package but gateway-recording.

If you want to use call recording filtering, please install also:

```
apt-get install call-recording-filtering
```

During the installation, you will be asked for :

- the recording server address with protocol and port (i.e. <http://192.168.0.2:9400>)

**XiVO CC configuration** Now we switch to the installation of the XiVO CC server.

### Install ntp server

```
apt-get install ntp
```

*XiVO CC* server and *XiVO PBX* server must be synchronized to the same source.

**Enable Docker LogRotate** Docker container log output to /dev/stdout and /dev/stderr. The Docker container log file is saved in /var/lib/docker/containers/[CONTAINER ID]/[CONTAINER\_ID]-json.log.

Create a new Logrotate config file for your Docker containers in the Logrotate folder /etc/logrotate.d/docker-container.

```
/var/lib/docker/containers/*/*.log {
 rotate 7
 daily
 compress
 missingok
 delaycompress
 copytruncate
}
```

You can test it with `logrotate -fv /etc/logrotate.d/docker-container`. You should get some output and a new log file with suffix [CONTAINER ID]-json.log.1 should be created. This file is compressed in next rotation cycle.

Retrieve the configuration script and launch it:

### Containers installation

```
wget https://gitlab.com/xivoxc/packaging/raw/master/install/install-docker-xivocc.sh
bash install-docker-xivocc.sh
```

During the installation, you will be asked for :

- the XiVO IP address (e.g. 192.168.0.1)
- the number of weeks to keep for the statistics
- the number of weeks to keep for the recording files
- the external IP of the machine (i.e. the address used afterwards for http URLs)

Create the following alias in your `.bashrc` file:

```
vi ~/.bashrc
alias dcomp='docker-compose -p xivocc -f /etc/docker/compose/docker-xivocc.yml'
```

**Containers modification** The yml file /etc/docker/compose/docker-xivocc.yml should have the correct tag version for each image.

Check also that the **XIVO\_CTI\_VERSION** is correct for the xuc container.

```
xivo_replic :
 image: xivoxc/xivo-db-replication:2016.03.latest

xivo_stats :
 image: xivoxc/xivo-full-stats:2016.03.latest

pack_reporting:
 image: xivoxc/pack-reporting:2016.03.latest

config_mgt:
 image: xivoxc/config-mgt:2016.03.latest

recording_server:
 image: xivoxc/recording-server:2016.03.latest

xuc:
 image: xivoxc/xuc:2016.03.latest

environment:
 - XIVO_CTI_VERSION=2.1
```

```
xucmgt:
image: xivoxc/xucmgt:2016.03.latest
```

**Starting XivoCC** Then you can launch the XiVO CC with the following command :

```
dcomp up -d
```

List XivoCC services :

# dcomp ps	Name	Command	State	Ports
xivocc_config_mgt_1		bin/config-mgt-docker	Up	0.0.0.0:9100->9000/tcp
xivocc_elasticsearch_1		/docker-entrypoint.sh elas ...	Up	0.0.0.0:9200->9200/tcp, 0.0.0.0:9300->9300/tcp
xivocc_fingerboard_1		/bin/sh -c /usr/bin/tail - ...	Up	
xivocc_kibana_volumes_1		/bin/sh -c /usr/bin/tail - ...	Up	
xivocc_nginx_1		nginx -g daemon off;	Up	443/tcp, 0.0.0.0:80->80/tcp
xivocc_pack_reporting_1		/bin/sh -c echo ...	Up	
xivocc_pgxivocc_1		/docker-entrypoint.sh postgres	Up	0.0.0.0:5443->5432/tcp
xivocc_postgresvols_1		/bin/bash	Exit 0	
xivocc_recording_server_1		bin/recording-server-docker	Up	0.0.0.0:9400->9000/tcp
xivocc_reporting_rsync_1		/usr/local/sbin/run-rsync.sh	Up	0.0.0.0:873->873/tcp
xivocc_spagobi_1		/bin/sh -c /root/start.sh	Up	0.0.0.0:9500->8080/tcp
xivocc_timezone_1		/bin/bash	Exit 0	
xivocc_xivo_replic_1		/usr/local/bin/start.sh /o ...	Up	
xivocc_xivo_stats_1		/usr/local/bin/start.sh /o ...	Up	
xivocc_xivocclogs_1		/bin/bash	Exit 0	
xivocc_xuc_1		bin/xuc_docker	Up	0.0.0.0:8090->9000/tcp
xivocc_xucmgt_1		bin/xucmgt_docker	Up	0.0.0.0:8070->9000/tcp

**Desktop Assistant** The desktop assistant is available through the xucmgt application so you need to deploy this container first.

**Windows (64bits)** To download the latest version available on your environment, just open the following url from your computer:

```
http://<xucmgt_host>:<xucmgt_port>/install/win64
```

and then start the downloaded program.

**Linux (Debian 64bits)** To install the latest version, you need to add a repository linked to the xucmgt host. Edit your /etc/apt/sources.list and add the following line:

```
deb http://<xucmgt_host>:<xucmgt_port>/updates/debian jessie contrib
```

Then run

```
sudo apt-get update
sudo apt-get install xivo-desktop-assistant
```

**Note:** This repository is currently not signed at all.

## Upgrade

## Upgrade

**Overview** The following components will be upgraded :

- Docker images
- xivocc-installer package

**Warning:** This upgrade procedure applies only to XiVO CC installed via the `xivocc-installer` package.

**Prerequisites** Before upgrading you have to check or change your sources list. It should be located in the file `/etc/apt/sources.list.d/xivo-solutions.list`.

There are two cases :

1. Upgrade to *latest* version,
2. Upgrade to a **specific** version (or an archive version)

**Upgrade to *latest* version** To upgrade to the latest version the sources list must point towards *debian* URI and *xivo-solutions* suite:

```
deb http://mirror.xivo.solutions/debian/ xivo-solutions main
```

**Upgrade to specific version** To upgrade to a **specific** version the sources list must point towards *archive* URI and *xivo-solutions-VERSION* suite.

For example if you want to upgrade to **2016.03** version you should have:

```
deb http://mirror.xivo.solutions/archive/ xivo-solutions-2016.03 main
```

Note the `/archive/` and `-2016.03` above.

**Upgrade** When you have checked the `sources.list` you can upgrade with the following commands:

```
apt-get update
apt-get install xivocc-installer
```

The current `docker-compose.yml` file will be renamed to `docker-compose.yml.dpkg-old` and new template downloaded. A new `docker-compose.yml` file will be rendered from the template using the current xivocc version.

Then you run the new version by `dcomp up -d` eventually preceded by the `dcomp pull` to download the new images.

### Upgrade notes

**2016.04** Consult the [2016.04 Roadmap](#)

#### System

- Parameters for `/etc/docker/compose/docker-xivocc.yml` are now stored in `/etc/docker/compose/.env` file. Important parameter is `XIVO_AMI_SECRET`, which holds Ami password.
- To be able to use the `/etc/docker/compose/.env` file, a new `dcomp` alias is generated in `.bashrc`. You must run:

```
source .bashrc
```

before running `dcomp` again.

**Note:** If you are using `docker-compose` instead of recommended alias `dcomp`, make sure your current directory is `/etc/docker/compose`, otherwise `/etc/docker/compose/.env` won't be used. i.e.:

```
cd /etc/docker/compose
docker-compose ...
```

### Web/Desktop Assistant

- For displaying search result, compatibility with *xivo-dird* of *XiVO PBX* has been enhanced. After upgrade you must verify the configuration of your CTI directory Display in *XiVO PBX* as described in [Directories](#) and [Views](#).

**Note:** Integration note: the *Web* and *Desktop Assistant* support only the display of

- 1 field for name (the one of type *name* in the directory display)
- 3 numbers (the one of type *number* and the first two of type *callable*)
- and 1 email

### Callbacks (CCManager)

- Default csv separator has been changed from pipe '`|`' to comma '`,`' for the callback export.

**2016.03** No behavior changes.

### Reporting and statistics

**Introduction** Pack reporting is a part of the XivoCC, but can also be installed separately. It aims at computing historical statistics, which are stored in the **xivo\_stats** database. Sample reports based on them are accessible in **SpagoBI**.

#### Standalone installation

**Warning:** Full installation of the pack reporting requires restarting XiVO services, so telephone communications will be cut.

1. Install docker by following installation instructions: <http://docs.docker.com/installation/>
2. Execute the following commands:

```
wget https://gitlab.com/xivoxc/packaging/raw/master/install/install-docker-reporting.sh
bash install-docker-reporting.sh
docker-compose -f /etc/docker/compose/docker-reporting.yml up -d
```

During installation you will be asked for: \* the XiVO IP address \* the number of weeks to keep in history

At the end of the installation some configuration must be done on the XiVO:

1. edit `/var/lib/postgresql/9.1/main/postgresql.conf` and set `listen_addresses` to `*`
2. edit `/var/lib/postgresql/9.1/main/pg_hba.conf` and add the following line: **host asterisk stats PACK\_REPORTING\_IP/32 md5**
3. add the following events to `/etc/asterisk/cel.conf`: **HOLD,UNHOLD,BLINDTRANSFER,ATTENDEDTRANSFER**
4. execute the following command:

```
sudo -u postgres psql asterisk << EOF
CREATE USER stats WITH PASSWORD 'stats';
GRANT SELECT ON ALL TABLES IN SCHEMA PUBLIC TO stats;
EOF
```

5. Finish the installation by a full restart of XiVO:

```
xivo-service restart all
```

## Checking the installation

### Historical statistics

- Docker containers *compose\_xivo\_replic\_1*, *compose\_xivo\_stats\_1* and *compose\_pack\_reporting\_1* should be started
- There should be no errors in */var/log/xivocc/xivo-db-replication/xivo-db-replication.log* and */var/log/xivocc/xivo-full-stats/xivo-full-stats.log*

### Kibana / TOTEM

- Data replication can take a long time, so you may need to be patient before finding data in the reports
- Some panels are preconfigured :
  - <http://@IP/kibana/#/dashboard/file/queues.json>
  - <http://@IP/kibana/#/dashboard/file/agents.json>
- To save these panels in Elasticsearch and make them accessible through Kibana menu, you will have to authenticate with **admin/Kibana**

### Known limitations

- Queue members should only be agents. If users are members of a queue, their statistics will be incomplete.
- Configuration modifications on the XiVO (such as an agent deletion) are replicated on the statistics server, and their previous value is not kept. However, statistics history is preserved.
- POPC statistics are wrong.
- If two agents are associated to the same call, they will have the same hold time for this call.
- Transfer statistics limitation : given two queues Q1 and Q2, two agents A1 and A2, and an external caller C.
  - C calls Q1 and A1 answers
  - A1 transfers to Q2 and A2 answers
  - A2 transfers to the outside

Then the second transfer is seen as a transfer to the outside.

**Attached Data** The pack reporting allows to attach as much data as wished to a given call, in order to find them in the reporting database for future use. This data must be in the form of a set of key-value pairs.

To attach data to a call, you must use the dialplan's **CELGenUserEvent** application:

```
exten = s,n,CELGenUserEvent (ATTACHED_DATA,my_key=my_value)
```

This will insert the following tuple in the **attached\_data** table:

key	value
my_key	my_value



**Upgrade notes** These notes include upgrade procedures for old versions of the **Pack reporting**, before **XivoCC** starts and before it was packaged with Docker. In those cases, run the following command to find the installed version of the pack reporting:

```
dpkg -l|grep pack-reporting
```

#### From version < 1.6

- data retention time will be lost during upgrade : save it and write it back in `/etc/xivo-reporting-db.conf`
- the upgrade is likely to be long if there is a lot of data in `queue_log`. Purge old data out of this table if possible in order to accelerate the upgrade
- at the end of the upgrade, run `apt-get autoremove` (deletion of xivo-stat, xivo-libdao and xivo-lib-python)

#### From version < 1.8

- XiVO in version < 14.08 is not supported anymore
- if it is required, the upgrade of the XiVO must be done before the upgrade of the pack reporting, and no call must be performed between the two upgrades

#### From a version using Debian packaging to a version using Docker

- **Beware:** this will require a migration of the original PostgreSQL database to the Dockerised one. For this you need to have free disk space : the amount of free disk space must equal the size of `/var/lib/postgresql`. This check must be performed after docker images have been pulled.
- Run the following commands:

```
apt-get update
apt-get install pack-reporting xivo-full-stats xivo-reporting-db xivo-db-replication db-utils
service xivo-db-replication stop
service xivo-full-stats stop
wget https://gitlab.com/xivoxc/packaging/raw/master/install/install-docker-reporting.sh
bash install-docker-reporting.sh
docker-compose -f /etc/docker/compose/docker-reporting.yml up -d pgxivocc
Database migration. CHECK THE FREE DISK SPACE
sudo -u postgres pg_dump --format c xivo_stats | docker exec -i xivocc_pgxivocc_1 pg_restore -U p
docker-compose -f /etc/docker/compose/docker-reporting.yml up -d
```

#### From a dockerized version before callbacks

- Run the following commands:

```
docker exec -ti compose_pgxivocc_1 psql -U postgres -c 'CREATE EXTENSION IF NOT EXISTS "uuid-osp
docker exec -ti compose_pgxivocc_1 psql -U postgres -c 'CREATE EXTENSION IF NOT EXISTS "uuid-osp
```

#### Database schema

**call\_data** Calls list

Column	Type	Description
id	INTEGER	
uniqueid	VARCHAR	Call unique reference, generated by Asterisk
dst_num	VARCHAR	Called number
start_time	TIMESTAMP	Call start time
answer_time	TIMESTAMP	Call answer time
end_time	TIMESTAMP	Call end time
status	status_type	Call status. Beware: only <i>answered</i> is properly filled.
ring_duration	INTEGER	Ring time of the endpoint answering the call, in seconds
transferred	BOOLEAN	True if the call has been transferred
call_direction	call_direction_type	Call direction (“incoming” : call from the outside, received by XiVO ; “outgoing” : call to the outside, originated by an endpoint associated to XiVO ; “internal” : call taking place entirely inside the XiVO)
src_num	VARCHAR	Calling number
transfer_direction	call_direction_type	Indicates the transfer direction, if relevant
src_agent	VARCHAR	Agent originating the call
dst_agent	VARCHAR	Agent receiving the call, if it is a direct call on an agent. Not filled when the call is destined to a queue
src_interface	VARCHAR	Interface originating the call (in the Asterisk sense, ex : SCCP/01234)

**attached\_data** Data attached to the call (cf. [Attached Data](#))

Column	Type	Description
id	INTEGER	
id_call_data	INTEGER	Id of the associated tuple in <i>call_data</i>
key	VARCHAR	Name of the attached data
value	VARCHAR	Value of the attached data

**call\_element** Part of a call matching the reaching of an endpoint

Column	Type	Description
id	INTEGER	
call_data_id	INTEGER	Id of the associated tuple in <i>call_data</i>
start_time	TIMESTAMP	Time at which the endpoint was called
answer_time	TIMESTAMP	Answer time for the endpoint
end_time	TIMESTAMP	End time of this call part
interface	VARCHAR	Endpoint interface

**call\_on\_queue** Calls on a queue

Column	Type	Description
id	INTEGER	
callid	VARCHAR	Call unique reference, generated by Asterisk
queue_time	TIMESTAMP	Time of entrance in the queue
total_ring_seconds	INTEGER	Total ring time, in seconds (includes ringing of non-answered calls)
answer_time	TIMESTAMP	Answer time
hangup_time	TIMESTAMP	Hangup time
status	call_exit_type	Call status ( <i>full</i> : full queue; <i>closed</i> : closed queue; <i>joinempty</i> : call arrived on empty queue; <i>leaveempty</i> : exit when queue becomes empty; <i>divert_ca_ratio</i> : call redirected because the ratio waiting calls/agents was exceeded ; <i>divert_waittime</i> : call redirected because estimated waiting time was exceeded; <i>answered</i> : call answered ; <i>abandoned</i> : call abandoned; <i>timeout</i> : maximum waiting time exceeded)
queue_ref	VARCHAR	Technical queue name
agent_num	VARCHAR	Number of the agent taking the call, if relevant

#### hold\_periods Hold periods

Column	Type	Description
id	INTEGER	
linkedid	VARCHAR	Call unique reference, generated by Asterisk
start	TIMESTAMP	Hold start time
end	TIMESTAMP	Hold end time

#### stat\_queue\_periodic Statistics aggregated by queue and time interval (15 minutes)

Column	Type	Description
id	INTEGER	
time	TIMESTAMP	Start time of the considered interval
queue	VARCHAR	Queue technical name
answered	INTEGER	Number of answered calls
abandoned	INTEGER	Number of abandoned calls
total	INTEGER	Total number of calls received on the queue (which excludes the calls dissuaded before entering the queue)
full	INTEGER	Number of calls arrived on a full queue (diversion before entering the queue)
closed	INTEGER	Number of calls arrived on a closed queue, outside of the configured schedules (diversion before entering the queue)
joinempty	INTEGER	Number of calls arrived on an empty queue (diversion before entering the queue)
leaveempty	INTEGER	Number of calls redirected because of a queue becoming empty
di-vert_ca_ratio	INTEGER	Number of calls arrived when the calls / available agents ratio is exceeded (diversion before entering the queue)
di-vert_waittime	INTEGER	Number of calls arrived when the estimated waiting time is exceeded (diversion before entering the queue)
timeout	INTEGER	Nombre of calls redirecting because maximum waiting time is exceeded

**stat\_agent\_periodic** Statistics aggregated by agent and time interval (15 minutes)

Column	Type	Description
id	INTEGER	
time	TIMESTAMP	Start time of the considered interval
agent	VARCHAR	Agent number
login_time	INTERVAL	Login time
pause_time	INTERVAL	Pause time
wrapup_time	INTERVAL	Wrapup time

**stat\_queue\_specific** Statistics aggregated by queue, called number and time interval (15 minutes)

Column	Type	Description
time	TIMESTAMP	Start time of the considered interval
queue_ref	VARCHAR	Technical name of the queue
dst_num	VARCHAR	Called number
nb_offered	INTEGER	Number of presented calls
nb_abandoned	INTEGER	Number of abandoned calls
sum_resp_delay	INTEGER	Wait time, in seconds
answer_less_t1	INTEGER	Number of calls answered in less than t1 seconds
abandoned_btw_t1_t2	INTEGER	Number of calls abandoned between t1 and t2 seconds
answer_btw_t1_t2	INTEGER	Number of calls answered between t1 and t2 seconds
abandoned_more_t2	INTEGER	Number of calls answered in more than t2 seconds
communication_time	INTEGER	Total communication time in seconds
hold_time	INTEGER	Total hold time in seconds
wrapup_time	INTEGER	Total wrapup time in seconds

The thresholds t1 and t2 are configurable:

- in the table `queue_specific_time_period` for the default values in seconds. Installation values are `t1=15` seconds and `t2=20` seconds. Data is saved in the form of (*name, seconds*) pairs, for example : ('t1', 15).
- in the table `queue_threshold_time` for values specific to a queue. Data is saved in the form of a tuple (queue name, t1, t2).

**stat\_agent\_specific** Statistics aggregated by agent and time interval (15 minutes)

Column	Type	Description
time	TIMESTAMP	Start time of the considered interval
agent_num	VAR-CHAR	Agent number
nb_offered	INTEGER	Number of calls presented from a queue
nb_answered	INTEGER	Number of calls answered from a queue
conversation_time	INTEGER	Conversation time on incoming calls from a queue, in seconds
ringing_time	INTEGER	Ring time on incoming calls from a queue, in seconds
nb_outgoing_calls	INTEGER	Number of calls emitted to the outside
conversation_time_outgoing_calls	INTEGER	Conversation time in calls emitted to the outside, in seconds
hold_time	INTEGER	Hold time for calls from a queue, in seconds
nb_received_internal_calls	INTEGER	Number of received internal calls
conversation_time_received_internal_calls	INTEGER	Conversation time on received internal calls, in seconds
nb_transferred_intern	INTEGER	Number of calls coming from a queue and transferred to an internal destination
nb_transferred_extern	INTEGER	Number of calls coming from a queue and transferred to an external destination
nb_emitted_internal_calls	INTEGER	Number of emitted internal calls
conversation_time_emitted_internal_calls	INTEGER	Conversation time on emitted internal calls, in seconds
nb_incoming_calls	INTEGER	Number of received incoming calls
conversation_time_incoming_calls	INTEGER	Conversation time on received incoming calls, in seconds

**stat\_agent\_queue\_specific** Statistics aggregated by queue, called number, agent and time interval (15 minutes)

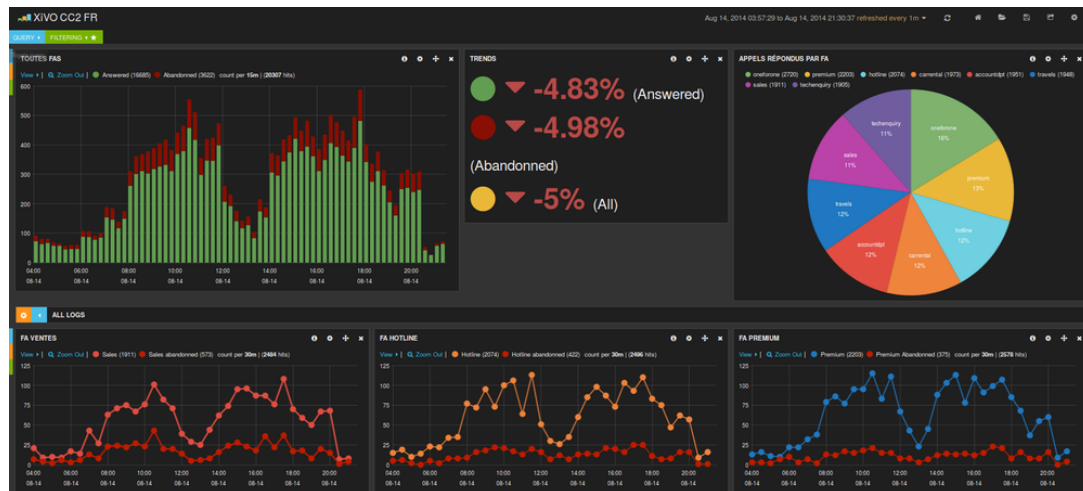
Column	Type	Description
time	TIMESTAMP	Start time of the considered interval
agent_num	VARCHAR	Agent number
queue_ref	VARCHAR	Technical name of the queue
dst_num	VARCHAR	Called number
nb_answered_calls	INTEGER	Number of answered calls
communication_time	INTEGER	Communication time, in seconds
hold_time	INTEGER	Hold time, in seconds
wrapup_time	INTEGER	Wrapup time, in seconds

Tables **call\_data**, **call\_on\_queue** et **hold\_periods** can be linked together by doing a join on a column holding the call reference. The columns are the following:

Table	Reference column
call_data	uniqueid
call_on_queue	callid
hold_periods	linkedid

D'autre part, les tables **attached\_data** et **call\_element** contiennent une clef étrangère référençant la colonne **id** de **call\_data**.

**Using Kibana** Kibana is a web tool used to compute statistics based on Elasticsearch content. The reports packaged with the Pack reporting give you an outline of your recent call center activity. Here is a Kibana sample panel:



Graphs are based on the `queue_log` table, enriched with agent names and agent groups, and inserted into an Elasticsearch index. It contains events about calls placed on queues, and events about agent presences.

For each entry in the `queue_log` index, the following attributes are available:

- `queuedisplayname` : Queue display name
- `data1` : basic `queue_log` data, with a different meaning according to the event
- `callid` : Call unique identifier, generated by Asterisk
- `event` : Call or agent status event - please see below
- `agentnumber` : Agent number
- `queueaname` : Technical queue name
- `groupname` : Agent group name
- `queuetime` : Time of the event
- `agentname` : Name of the agent, if available

The event can be one of the following (for a detailed explanation, please refer to <https://wiki.asterisk.org/wiki/display/AST/Queue+Logs>):

- Call events:
  - FULL
  - CONNECT
  - EXITEMPTY
  - CLOSED
  - EXITWITHTIMEOUT
  - JOINEMPTY
  - ABANDON
  - ENTERQUEUE
  - TRANSFER
  - COMPLETEAGENT
  - COMPLETECALLER

- RINGNOANSWER
- Agent or queue event:
  - ADDMEMBER
  - PAUSEALL
  - PAUSE
  - WRAPUPSTART
  - UNPAUSE
  - UNPAUSEALL
  - PENALTY
  - CONFIGRELOAD
  - AGENTCALLBACKLOGIN
  - AGENTCALLBACKLOGOFF
  - REMOVEMEMBER
  - PRESENCE
  - QUEUESTART

### Phone integration

XUC based web applications like agent interface or xivo client web integrates buttons for phone control. This section details necessary configuration, supported phones and limitations.

Note: The voip vlan network have to be accessible by the xivocc xuc server

Supported phones	Manufacturer	Function					
		Answer	Hangup	Hold	Conference	Attended Transfer	Direct Transfer
	Snom 7XX	OK	OK	OK	OK	OK	OK
	Polycom VVX	OK	OK	OK	NO	OK	OK
	Yealink T4X	OK	OK	OK	NO	OK	OK

- NO - Not available

### Required configuration

**Customize templates for Polycom phones** To enable phone control buttons on web interfaces you must update the basic template of Polycom phones:

- go to the plugin directory: `/var/lib/xivo-provd/plugins/xivo-polycom-VERSION`
- copy the default template from `templates/base.tpl` to `var/templates/`
- then you must update `app.push` parameters in the else section (**do not replace switchboard settings**) as follows:

```
apps.push.messageType="5"
apps.push.username="guest"
apps.push.password="guest"
```

**Customize templates for Yealink phones** To enable phone control buttons on web interfaces you must update the basic template of Yealink phones:

- go to the plugin directory: `/var/lib/xivo-provd/plugins/xivo-yealink-VERSION`
- copy the default template from `templates/base.tpl` to `var/templates/`
- enable sip notify even for non switchboard profiles (**do not replace switchboard settings**)

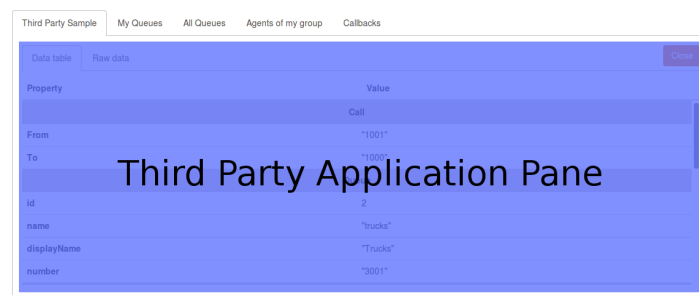
```
{% if XX_options['switchboard'] -%}
push_xml.sip_notify = 1
call_waiting.enable = 0
{% else -%}
push_xml.sip_notify = 1
call_waiting.enable = 1
{% endif %}
```

### Update Device Configuration

- to update device configuration you must run `xivo-provd-cli -c 'devices.using_plugin("xivo-polycom-VERSION").reconfigure()'`
- and finally you must resynchronize the device: `xivo-provd-cli -c 'devices.using_plugin("xivo-polycom-VERSION").synchronize()'`
- refer to [provisioning](#) documentation for more details
- if the phone synchronization fails check if the phone uses the version of the plugin you have updated, you can use `xivo-provd-cli -c 'devices.find()'`

### Third Party Integration

Third party web application integration is possible inside the XucMgt Agent application since XucMgt version 1.49.0. Upon each call, you can display a custom tab inside the agent interface:



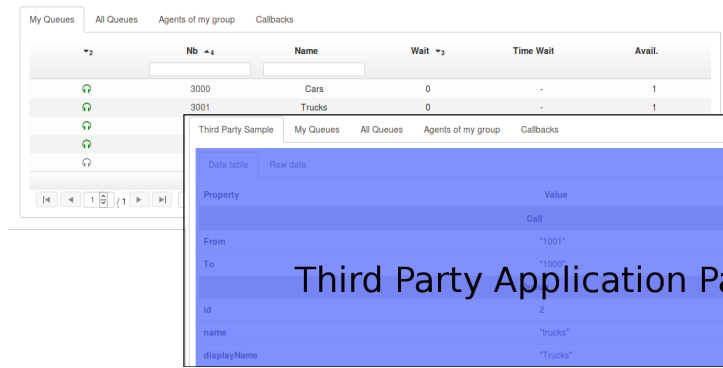
**Workflow** When a call is ringing on the agent phone, the Application will call the external web service (see [Configuration](#) below). The web service response will dictate the behaviour of the integration. For example, if the specified action is to open the application when the call is hung up, a new tab will be created and opened inside the agent interface, showing the content specified by the web service response. (see [Web Service API](#) for available options).

When the work is complete in the integrated application, the application must post a message to terminate the third party application pane inside the agent application (see [Completion](#)).

**Configuration** You need to specify the third party application web service url to integrate this application inside the XucMgt Agent interface. This can be done by specifying a `THIRD_PARTY_URL` environment variables.

For example, inside the dockerfile, in the XucMgt section:





```
environment:
...
- THIRD_PARTY_URL=http://some.url.com/ws/endpoint
```

The specified URL must be accessible from the client browser (i.e. the end user of the Agent application). The call will be made from his browser.

**Web Service API** The Web Service url specified in the :*Configuration* must conform to the following behaviour. The service will receive a POST request with a payload as `application/json`, for example:

```
{
 "user": {
 "userId": 4,
 "agentId": 1,
 "firstName": "James",
 "lastName": "Bond",
 "fullName": "James Bond"
 },
 "callee": "1000",
 "caller": "1001",
 "queue": {
 "id": 2,
 "name": "trucks",
 "displayName": "Trucks",
 "number": "3001"
 },
 "userData": {
 "XIVO_CONTEXT": "default",
 "XIVO_USERID": "2",
 "XIVO_SRCNUM": "1001",
 "XIVO_DSTNUM": "3001"
 }
}
```

- `user` contains the connected user information
- `callee` contains the number called
- `queue` queue properties
- `userData` call data presented by Xivo

The Web service must answer with an `application/json` content. For example:

```
{
 "action": "open",
 "event": "EventReleased",
 "url": "/thirdparty/open/6bd37819-b4a6-43d3-8fa3-6eb6489bb705",
 "autoplay": true,
```

```
"title": "Third Party Sample"
}
```

or:

```
{
 "action": "none"
}
```

- `action` is one of "open" or "none"
- `event` is one of "EventRingin", "EventEstablished", "EventReleased". The third party application will be opened when one the specified event occurs
- `url` should be the url to open inside the application. This url should point to a valid web application that can be specific for each call.
- `autoplay` if set to true, the agent will be put on pause when the application pane is opened and back to ready when the application is completed.
- `title` will set the title of the tabs that will be opened.

**Warning**, when the XucMgt application and the integrated application are on different server, domain, url,... (which should be common case), You may get [CORS](#) errors. To workaround this issue, you should implement the OPTIONS request on your web service. This method will be called by the browser before issuing the POST request to ensure the target web server allows calls from the original application. You application must set at least the following headers in order to overcome the [CORS](#) errors:

- `Access-Control-Allow-Origin`: \* or the domain hosting the XucMgt application
- `Access-Control-Allow-Methods`: POST, OPTIONS (at least)
- `Access-Control-Allow-Headers`: Origin, X-Requested-With, Content-Type, Accept (at least)

**Completion** Once the work is complete inside the third party application, it should post a completion message (`closeThirdParty`) to the application using the [Web Messaging API](#).

For example, here is how to define a close method in javascript to send the message to the hosting application and bind it to a simple button:

```
(function () {
 function close() {
 parent.window.postMessage("closeThirdParty", "*");
 }

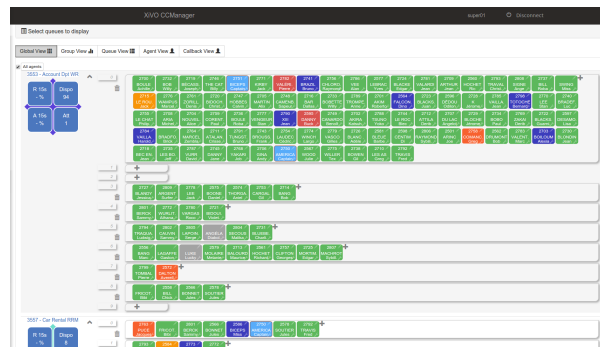
 document.getElementById("close").addEventListener("click", close, false);
})();
```

## Features

### Contact center management

**Introduction** CCmanager is a web application to manage and supervise a contact center

- Display queues
- Display agents / agents status
- Move or add agents in queue / penalty
- Move of add group of agents in queue / penalty
- **Action on agents**
  - Login / Logout



- Pause / Unpause
- Listen <sup>1</sup>

Start the application : <http://<xuc:port>/ccmanager>

Number	Name	Penalty	default
3554	Sales RRM	0	0
3552	Travels Lin	0	7
4553	Ast11 Account Dpt WR	0	2
4500	blue	1	1
4557	Ast11 Car Rental RRM	1	1

**Single Agent Edition** This interface allows a user to change queue assignment and the associated penalty. The queue table display the following columns

- “Number”: The queue number
- “Name”: The queue name
- “Penalty”: The active penalty for the corresponding queue
- “default”: The default penalty for the corresponding queue

The queue/active penalty couples can be saved as default configuration by clicking the “Set default” button, then “Save”. The queue/default penalty couples can be saved as active configuration by clicking the “Set current” button, then “Save”.

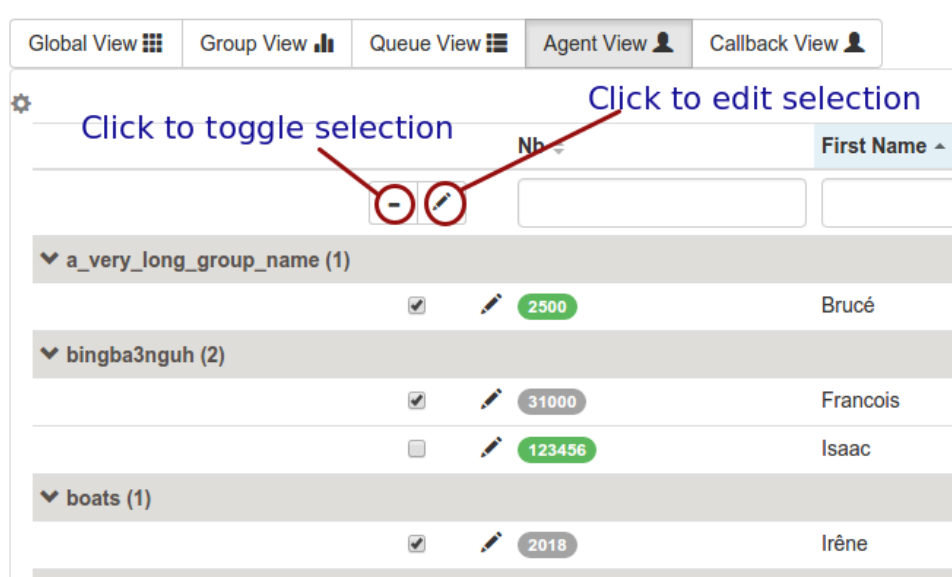
## Notes

- Emptying the penalty textbox and saving will remove the queue from the active configuration for the agent.

<sup>1</sup> Only supervisors which have their own lines can listen to agents, no supported on mobile supervisors, a line has to be affected to supervisors in xivo

- Emptying the default textbox and saving will remove the queue from the default configuration for the agent.

**Multiple Agent Selection** From agent view you are able to add or remove more than one agent at the same time.



Once the agent selection is done, click on the edit button to display the configuration window

Click on the plus button to add a queue for selection, click on the minus button to remove a queue to the selection. Once queue to add or removed are chosen, click on save button to apply your configuration change.

Click on “Apply default configuration” to apply existing default configuration to all selected users and make it the active configuration. This action only affects users with an existing default configuration, agents without default configuration remain unchanged.

**Create base configuration for a set of agents** From the agent view, after selecting one or more agents, you can create a base configuration by clicking on one of the menu item in the following drop down:

- ‘Create base configuration’ will allow you to create a base configuration from scratch for all the selected agents.
- ‘Create base configuration from active configuration’ will allow you to create a base configuration using the selected agents active configuration. The queue membership and penalty populated will be built based on the merged membership of all the selected agents. In case of conflict, the lowest penalty will be used.

In both cases, you will be able to review your changes before applying them. The ‘Create base configuration’ popup is similar to the single agent edition popup:

The queue table display the following columns:

- “Number”: The queue number
- “Name”: The queue name
- “Penalty”: The active penalty for the corresponding queue

Click on the plus button to add a queue for selection. Once your configuration is complete, click on save button to apply your configuration change.

**Thresholds** Color thresholds can be defines for the waitinig calls counter and the maximum waiting time counter

Applies to the queue view and the global view

Update agents

Blanc Sec  
(1626)Didascaux  
(1599)Vee  
(1686)

Add in queues

Number	Name	Penalty
--------	------	---------

Remove from queues

Number	Name
--------	------

Queue

Penalty 0

+

-

Apply default configuration

Save

Cancel

	Nb	Group	First Name	Last Name	Login	Log
<div><div></div><div></div></div>		Select				
<div><div></div><div></div></div>				K	-	-
<div><div></div><div></div></div>				Willis	-	-
<div><div></div><div></div></div>	2007	default	Carrie	Mathison	-	-

Create base configuration

K (2006)

Mathison (2007)

Willis (2001)

	Number	Name	Penalty
	3000	Cars	1
	3001	Trucks	3
	3003	Bikes	1
	3006	Boats	4

Queue

Penalty

0

+

Save

cancel

Select queues to display

Global View

Group View

Queue View

Agent View

Callback View

Nb

Name

Wait

Time Wait

EWT

%Ans. 15s

Avail.

Talk.

Total

Answ.

Waiting Calls

Max Waiting Time

1

3

30

55

Nb	Name	Wait	Time Wait	EWT	%Ans. 15s	Avail.
3012	BI Record	0		1	100.0	1
3014	BI Record Pause	0		0		0
3000	Blue Ocean	0		0	50.0	1

**Callbacks** This view allows to manage callback request : importing a new list of callbacks, monitoring them and downloading the associated tickets.

Sample list - Wisconsin

Phone numbers	Full name	Company	Description	Taken by
1003 -				
0587963214 - 0789654123	Alice O'Neill	YourSociety		
0230210092 - 0689746321	John Doe	MyCompany	Call back quickly	
1003 -				

**Callbacks import** Callbacks can be imported from a CSV file into a [callback list](#).

Line delimiter must be a new line character and column separator must be one of: 'l' or ';' or ','. Columns can be optionally enclosed by double-quote "".

The file must look like the following:

```
phoneNumber|mobilePhoneNumber|firstName|lastName|company|description|dueDate|period
0230210092|0689746321|John|Doe|MyCompany|Call back quickly||
0587963214|0789654123|Alice|O'Neill|YourSociety||2016-08-01|Afternoon
```

The header line must contain the exact field named described below:

- **phoneNumber**: The number to call (at least either **phoneNumber** or **mobilePhoneNumber** is required)
- **mobilePhoneNumber**: Alternate number to call
- **firstName**: The contact first name (optional)
- **lastName**: The contact last name (optional)
- **company**: The contact company name (optional)
- **description**: A text that will appear on the agent [callback pane](#)
- **dueDate**: The date when to callback, using ISO format: YYYY-MM-DD, ie. 2016-08-01 for August, 1st, 2016. If not present the next day will be used as **dueDate** (optional)
- **period**: The name of the period as defined in [callback list](#). If not present, the default period will be used (optional)

When an agent takes a callback, the column **Taken by** is updated with the number of the agent. The callback disappears when it is processed.

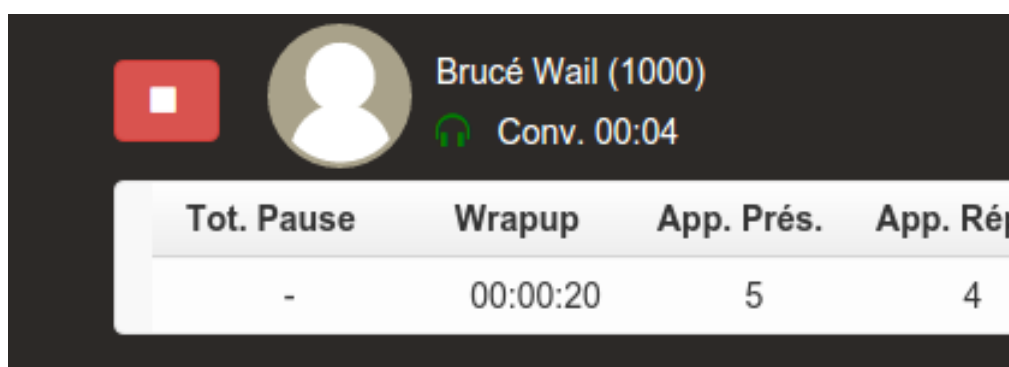
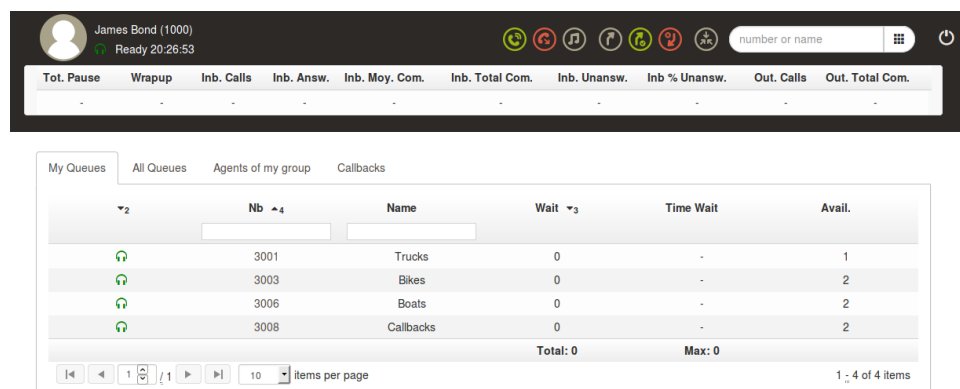
**Callbacks export** The tickets of the processed callbacks can be downloaded by clicking on the **Download tickets** button.

The downloaded file is a csv file with the comma ',' as delimiter.

## Agent environment

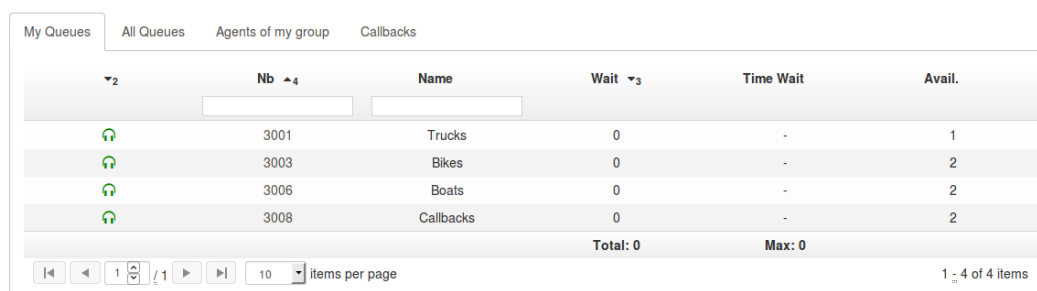
Web application for contact center agents

**Configuration** Recording can be paused or started by an agent, this feature can be disabled by changing **showRecordingControls** option in **application.conf**, you can also set the environment variable **SHOW\_RECORDING\_CONTROLS** to false for your xucmgt container in docker compose yml file. When disabled the recording status is not displayed any more



Callbacks panel can be removed using by changing showCallbacks option in application.conf, you can also use SHOW\_CALLBACKS environment variable in docker compose yml file.

By using the showQueueControls option in application.conf, you may allow an agent to enter or leave a queue. You can also use SHOW\_QUEUE\_CONTROLS environment variable in docker compose yml file.



**Login** Enter your XiVO client username, password and phone set you want to logged to on the login page. If you are already logged on an other application (CCmanager, Web Assistant) you only need to enter the phone number.


**Taking Callbacks** The agent can see the *callbacks* related to the queues he is logged on. They are available in the Callbacks tab, beside the Agents of my group tab.


On this page, the agent only has access to basic information about the callback: the phone number to call, the person's name and its company name. On the left of each callback line, a colored clock indicates the temporal status of this callback:


- yellow if the callback is to be processed later
- green if we are currently inside the callback period





XiVO Agent Login












 Connect

### XiVO Agent Login









.....

You are already connected, you have only  
to enter your phone number



Phone Number

 Connect

- red if the callback period is over

My Queues All Queues Agents of my group **Callbacks**

Period ▲3	Phone	Name	Queue	
🟢 12-08 Après-midi	1002	Jack Bauer	Callbacks	➔
🟡 13-08 Après-midi	1002	Jack Bauer	Callbacks	➔
🟡 04-09 Matin	1001	James Bond	Callbacks	➔
🟡 15-09 Après-midi	1002	Jack Bauer	Callbacks	➔
🟡 12-10 Toute la journée	1002	Jason Bourne	Callbacks	➔

10 items per page 1 - 5 of 5 items

To process one of these callbacks, the agent must click on one of the callbacks line. This will remove the callback from the other agents' list, and trigger the following screen:

My Queues All Queues Agents of my group **Callbacks**

**Callbacks - Jack Bauer - The Company**  
 🕒 Requested on 12-08 Après-midi (14:00:00 17:00:00)

**Callback**  
 📞 1002 📠 1003

**Description**  
 Some description

**Status**  
 ▼

**Comment**

You must callback by clicking on the phone number before saving

Cancel Save

To launch the call, the agent must click on one of the available phone numbers. Once the callback is launched, the status can be changed and a comment can be added.

If you set 'Callback' as status, the callback can be rescheduled at a later time and another period:

Clicking on the calendar icon next to the "New due date" field, will popup a calendar to select another callback date.

**Screen Popup** It is possible to display customer information in an external web application using Xivo [sheet](#) mechanism.

You must define a sheet with two fields

- **folderNumber** have to be defined. Can be calculated or use a default value not equal to "--"
- **popupUrl** The url to open when call arrives : i.e. <http://mycrm.com/customerInfo?folder=> the folder number will be automatically appended to the end of the URL

Example : Using the caller number to open a customer info web page

- Define folderNumber with any default value i.e. 123456
- Define popupUrl with a display value of <http://mycrm.com/customerInfo?nb={xivo-calleridnum}&fn=> when call arrives web page <http://mycrm.com/customerInfo?nb=1050&fn=123456> will be displayed

## Configuration Management

**Callbacks** The callback system in XivoCC aims at performing outgoing calls to specific numbers, to which some information can be associated such as a description or a personal name.

The core object of the callback system is the **callback request**. A callback request is made of the following fields:

- First name of person to call
- Last name
- Phone number
- Mobile phone number
- Company name
- Description
- Due date

Each callback request is associated to a predefined **callback period**, which represents the preferred interval of the day in which the call should be performed.

A callback request cannot exist on its own: it must be stored in a **callback list**, which is itself associated to a queue.

Once a callback request has been performed, it generates a **callback ticket**. This ticket sums up the original information of the callback request, but adding some new fields:

- Start date: date at which the callback request was actually performed
- Last update: date of the last modification of the ticket
- Comment
- Status : the result of the callback
- Agent: the Call Center agent who performed the callback

**Callback Lists** A callback list is an object which will contain callback request. It is associated to a queue, and several callback lists can be associated to the same queue.

Once created, a list can be populated whether through the *Callbacks tab* of the CCManager, or programmatically through the web services of the configuration server.

Listes de rappel

Périodes de rappel

Nom	File d'attente	Nombre de rappels
Liste de test	Wisconsin (wisconsin)	3

**Callback Periods** A callback period represents an interval of the day, bounded by a start date and an end date. It can be set as the default interval, so that a newly created callback request will be associated to this period if none is specified.

Listes de rappel

Périodes de rappel

Nom	Heure de début	Heure de fin	
Après-midi	14:00:00	17:00:00	<div></div> <div></div>
Matin	08:00:00	12:10:00	<div></div> <div></div>
Toute la journée	07:00:00	17:00:00	<div>✓</div> <div></div>

## Desktop Assistant

On first launch the application will display the settings page and ask for the xucmgt application address.



**Navigation** The top menu allows you to navigate either to the application or to the settings page. If you did not enter any setting, the application will redirect you to the settings page.

**Settings** This page allows you to specify the protocol and address of the xucmgt application.



- Check “Secure” if you use “https” protocol to connect to the xucmgt application or check “Unsecure” otherwise.
- Enter the xucmgt application host address and port.

**Update** On Windows, the application will check at startup for a new version of the application and offer to upgrade if one is available.

Desktop Assistant

x

Navigation



## PARAMÈTRES

**Protocole**

☒ Sécurisé  
☐ Non Sécurisé

**Veillez saisir le domaine de votre web-assistant**

webassistant-limonest.avencall.com

Enregistrer



On Debian, the update relies on the package manager behaviour. However you can check for any update by issuing the following commands:

```
sudo apt-get update
apt-cache policy xivo-desktop-assistant
```

**Options** The Desktop Assistant can be started with following options:

- `-d` to enable debug menu items
- `-ignore-certificate-errors` to disable certificate verification, this option is meant **only** for test purposes. You can use it with self-signed certificates.

**WebRTC integration** The *Desktop Assistant* can be used by users with WebRTC configuration, without physical phone.

For configuration and requirements, see [WebRTC Requirements](#).

### Web Assistant

The Web Assistant enables a user to:

- search contacts,
- call them,
- manage its favorites,
- manage its forward.

**Login** To login, one must have a user configured on the *XiVO PBX* with:

- XiVO Client enabled,
- Login, password and profile configured

**Search** You can use the search section to lookup for people in the company:

For this to work, one must configure the directories in the *XiVO PBX* as described in [Directories](#) and [Views](#).

---

**Note:** Integration note: the *Web Assistant* support only the display of

- 1 field for name (the one of type *name* in the directory display)
  - 3 numbers (the one of type *number* and the first two of type *callable*)
  - and 1 email
- 

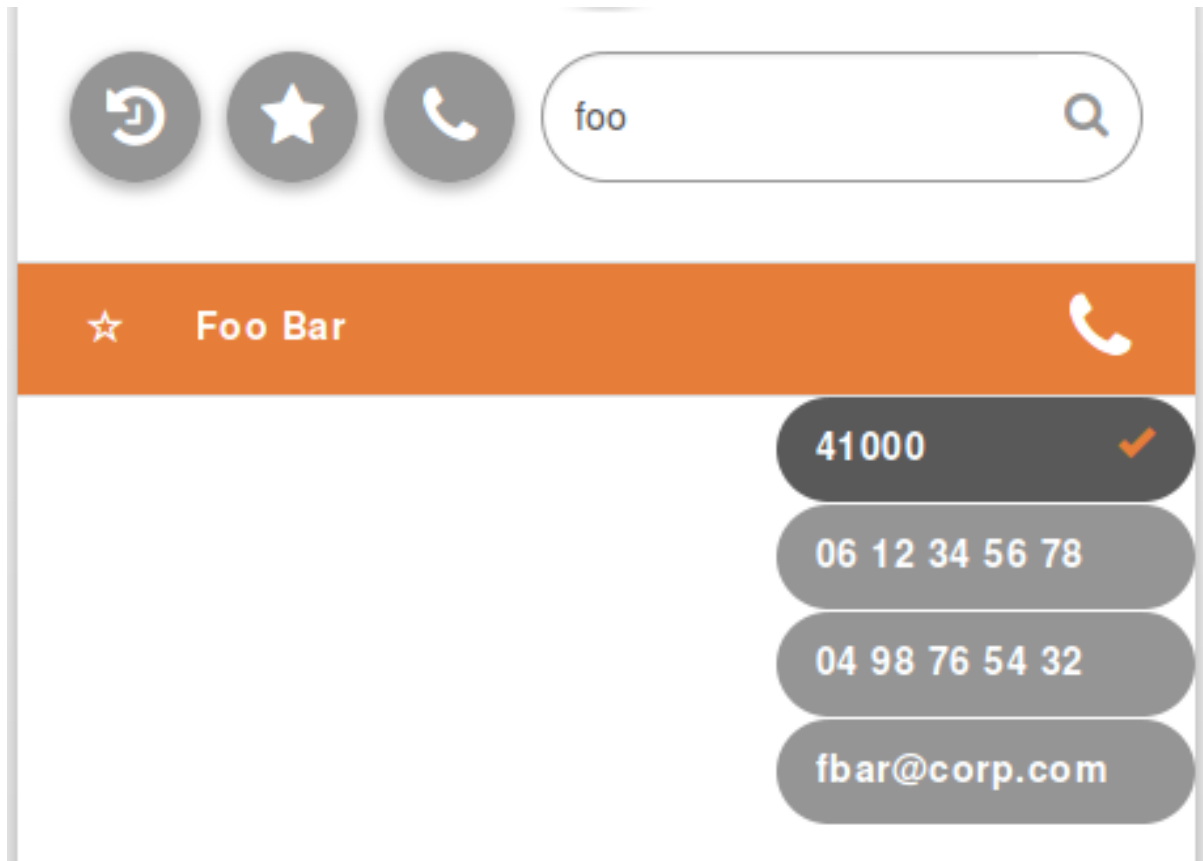
**Favorites** One can clic on the star to put a contact in its list of favorites.

For this to work, favorites must be configured in the *XiVO PBX* as described in [Favorites](#).

**Phone integration** The *Web Assistant* can integrate with the phone to :

- call,
- put on hold,
- transfer
- etc.





For this feature to work one must use a [Supported phones](#) and follow the [Required configuration](#) page.

**WebRTC integration** The *Web Assistant* can be used by users with WebRTC configuration, without physical phone.

For configuration and requirements, see [WebRTC Requirements](#).

## WebRTC

---

**Note:** added in version 2016.04

---

From version 2016.04 one can use WebRTC with *XiVO PBX* and *XiVO CC* in the following environment:

- LAN network (currently no support for WAN environment),
- with the:
  - *Web Assistant* with Chrome browser version 55 (tested on 55.0.2883.87 m 64-bit),
  - or *Desktop Assistant*

**WebRTC Requirements** The **requirements** are:

- to have a microphone and headphones for your PC,
- to configure, in the *XiVO PBX*, a user with a WebRTC line (see: [Configuration of user with WebRTC line](#)),
- have a SSL/TLS certificate signed by a certification authority installed on the nginx of *XiVO CC*,
- and use *https*:

- *Web Assistant*: you must connect to the *Web Assistant* via *https* protocol,
- *Desktop Application*: you must check *Protocol* -> *Secure* in the application parameters.

---

**Note:** Currently you can not have a user configured for both WebRTC and a phone set at the same time.

---

## Administration

### Log

The log of each components can be found in the `/var/log/xivocc` directory. Currently (it may change) the structure looks like this :

```
/var/log/xivocc :
-- purge-reporting-database.log
-- specific-stats.log
-- xivo-db-replication.log
-- xivo-full-stats.log
 -- recording-server
| -- downloads.log
| -- downloads.log
| -- recording-server.log
-- xuc
| -- xuc.log
-- xucmgt
 -- xucmgt.log
```

### Backup

You may backup your statistic database by using a similar command as below

```
docker run --rm --link demo_pgxivocc_1:db -v $(pwd):/backup -e PGPASSWORD=xivocc postgres pg_dump
```

### Restore

You may restore a backup using a similar command (to be adapted)

```
docker run --rm -it --link pgxivoccdemo_pgxivocc_1:db -v $(pwd):/backup postgres pg_restore -h db
```

## Xuc Xivo Unified Communication Framework

Xuc is an application suite developed by [Avencall](#) Group, based on several free existing components including [XiVO](#), and our own developments to provide communication services api and application to businesses. Xuc is build on [Play](#) using intensively [Akka](#) and written in [Scala](#)

XiVO is [free software](#). Most of its distinctive components, and Xuc as a whole, are distributed under the *LGPLv3 license*.

Xuc is providing

- Javascript API
- Web services
- Sample application
- Simple agent application

- Simple unified communication application pratix
- Contact center supervision
- Contact center statistics

The proposed applications are available in English and French. The list of preferred langs sent by the browser is analyzed and the first known lang is used, so if the browser requests it, en and fr the page will be server in en. The fallback language is French. Contributions are welcome, start with opening an issue on gitlab project page.

Xuc is composed of 3 modules

- The server module
- The core module
- The statistic module.

## Developer

### Building and packaging

#### Dependencies

- **Xivo Java Cti lib** ; <https://gitorious.org/xivo/xivo-javactilib>
  - mvn install
- **theatrus/akka-quartz** [<https://github.com/theatrus/akka-quartz>]
  - sbt publish-local

(sudo apt-get install devscripts)

#### Update change log

- dch -i in project root directory, parent of debian/changelog
- edit changelog to add version

#### Update documentation site

- update src/sphinx/conf.py with new version
- activator make-site
- copy target/sphinx/docs content to public/doc

#### Update xuc\_logger.xml with new version

- Create debian package : activator debian:genChanges

#### Docker Building docker image:

```
activator docker:publish

or

activator docker:publishLocal

docker tag xivo/xuc:2.4.32 xivo/xuc:latest

activator clean test docker:publishLocal; docker tag -f xivo/xuc:1.9.0 xivo/xuc:latest; docker push
```

**Documentation Guidelines** The Xuc documentation uses `reStructuredText` as its markup language and is built using `Sphinx`.

**Sphinx** For more details see [The Sphinx Documentation](#)

**reStructuredText** For more details see [The reST Quickref](#)

Quick Reference

- <http://docutils.sourceforge.net/docs/user/rst/cheatsheet.txt>
- <http://docutils.sourceforge.net/docs/user/rst/quickref.html>
- [http://openalea.gforge.inria.fr/doc/openalea/doc/\\_build/html/source/sphinx/rest\\_syntax.html](http://openalea.gforge.inria.fr/doc/openalea/doc/_build/html/source/sphinx/rest_syntax.html)

**Sections** Section headings are very flexible in reST. We use the following convention in the Xuc documentation:

- # (over and under) for module headings
- = for sections
- – for subsections
- ^ for subsubsections
- ~ for subsubsubsections

**Cross-referencing** Sections that may be cross-referenced across the documentation should be marked with a reference. To mark a section use `.. _ref-name:` before the section heading. The section can then be linked with `:ref: 'ref-name'`. These are unique references across the entire documentation.

For example:

```
.. _xuc-module:

#####
Xuc Module
#####

This is the module documentation.

.. _xuc-section:

Xuc Section
=====

Xuc Subsection

Here is a reference to "xuc section": :ref:`xuc-section` which will have the
name "Xuc Section".
```

**Build the documentation** First install `Sphinx`. See below.

**Building** For the html and pdf version of the docs:

```
activator make-site

open <project-dir>/target/sphinx/docs/index.html
open <project-dir>/target/sphinx/docs/Xuc-doc.pdf
```

**Installing Sphinx and other tools** To be able to generate pdf and documentation you need install Sphinx and other tools:

```
sudo easy_install -U Sphinx
sudo apt-get install texlive-latex-base texlive-latex-recommended texlive-latex-extra texlive-fonts-recommended texlive-fonts-extra
```

## Javascript API

**Introduction** The Xuc javascript API enables you to integrate enterprise communication functions to your business application. It exposes Cti functions using javascript methods calls.

You may add your own handlers for your application to react to telephony / contact center events.

This API is using [websockets](#), and therefore needs a modern browser supporting them ([firefox](#), [chrome](#) ...)

## Integration Principles

- Include the Cti and Callback javascript API from the Xuc Server

```
<script src="http://<xucserver>:<xucport>/assets/javascripts/shotgun.js" type="text/javascript"></script>
<script src="http://<xucserver>:<xucport>/assets/javascripts/cti.js" type="text/javascript"></script>
<script src="http://<xucserver>:<xucport>/assets/javascripts/callback.js" type="text/javascript"></script>
<script src="http://<xucserver>:<xucport>/assets/javascripts/membership.js" type="text/javascript"></script>
```

- Include also the xc\_webrtc and SIPml5 javascript APIs for the webRTC support:

```
<script src="http://<xucserver>:<xucport>/assets/javascripts/xc_webrtc.js" type="text/javascript"></script>
<script src="http://<xucserver>:<xucport>/assets/javascripts/SIPml-api.js" type="text/javascript"></script>
```

- Connect to the Xuc server using XiVO client username and password

```
var wsurl = "ws://" + server + "/ctichannel?username=" + username + "&agentNumber=" + phoneNumber + "&";
Cti.WebSocket.init(wsurl, username, phoneNumber);
```

- Setup event handlers to be notified on

- Phone state changes
- Agent state changes
- Statistics
- ...

- Eventually also webRTC handlers

- general
- register
- incoming
- outgoing

- Once web socket communication is established you are able to call XuC Cti javascript methods.

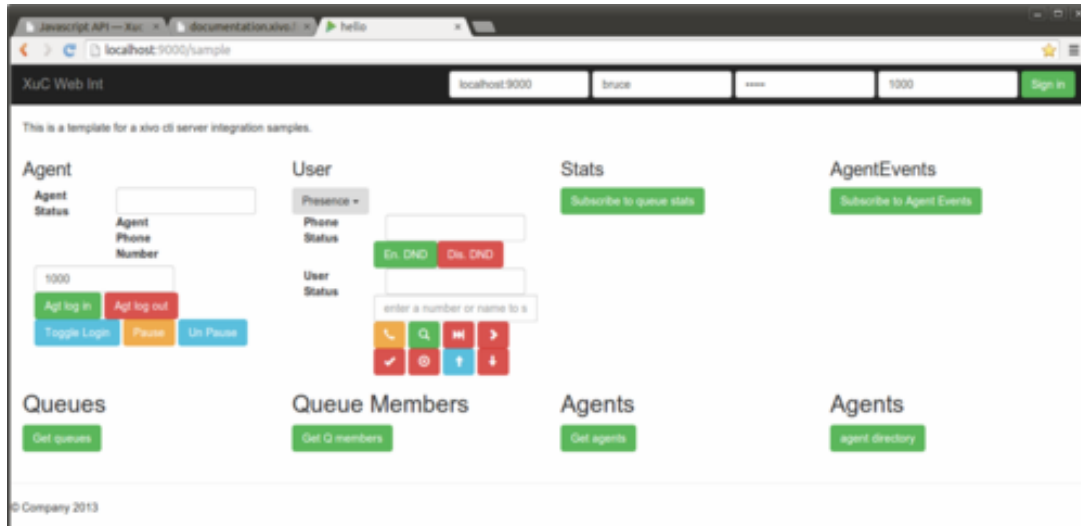
- Place a call, log an agent ....

```
...
$('#login_btn').click(function(event) {
 Cti.loginAgent($('#agentPhoneNumber').val());
});
$('#logout_btn').click(function(event) {
 Cti.logoutAgent();
});
$('#xuc_dial_btn').click(function(event) {
 Cti.dial($('#xuc_destination').val());
});
```

```
});
...
```

**Sample Application** A sample application is provided by the XuC server. This application allows to display events and using different methods exposed by the XuC

`http://<sucserver>:<xucport>/sample`



You may browse and use the `sample.js` javascript file as an example

- Calling Cti methods :

```
$('#xuc_login_btn').click(function(event) {
 Cti.loginAgent($('#xuc_agentPhoneNumber').val());
});

$('#xuc_logout_btn').click(function(event) {
 Cti.logoutAgent();
});

$('#xuc_pause_btn').click(function(event) {
 Cti.pauseAgent();
});

$('#xuc_unpause_btn').click(function(event) {
 Cti.unpauseAgent();
});

$('#xuc_subscribe_to_queue_stats_btn').click(function(event) {
 Cti.subscribeToQueueStats();
});

$('#xuc_answer_btn').click(function(event) {
 Cti.answer();
});

$('#xuc_hangup_btn').click(function(event) {
 Cti.hangup();
});

$('#xuc_login_btn').click(function(event) {
 Cti.loginAgent($('#xuc_agentPhoneNumber').val());
});

$('#xuc_logout_btn').click(function(event) {
 Cti.logoutAgent();
});

$('#xuc_togglelogin_btn').click(function(event) {
 Cti.toggleAgentLogin();
});
```

```

$('#xuc_pause_btn').click(function(event) {
 Cti.pauseAgent();
});
$('#xuc_unpause_btn').click(function(event) {
 Cti.unpauseAgent();
});
$('#xuc_subscribe_to_queue_stats_btn').click(function(event) {
 Cti.subscribeToQueueStats();
});
$('#xuc_answer_btn').click(function(event) {
 Cti.answer();
});
$('#xuc_hangup_btn').click(function(event) {
 Cti.hangup();
});
$('#xuc_get_agent_call_history').click(function() {
 Cti.getAgentCallHistory(7);
});
$('#xuc_get_user_call_history').click(function() {
 Cti.getUserCallHistory(7);
});
.....

```

- Declaring events handlers :

```

Cti.setHandler(Cti.MessageType.USERSTATUSES, usersStatusesHandler);
Cti.setHandler(Cti.MessageType.USERSTATUSUPDATE, userStatusHandler);
Cti.setHandler(Cti.MessageType.USERCONFIGUPDATE, userConfigHandler);
Cti.setHandler(Cti.MessageType.LOGGEDON, loggedOnHandler);
Cti.setHandler(Cti.MessageType.PHONESTATUSUPDATE, phoneStatusHandler);
Cti.setHandler(Cti.MessageType.VOICEMAILSTATUSUPDATE, voiceMailStatusHandler);
Cti.setHandler(Cti.MessageType.LINKSTATUSUPDATE, linkStatusHandler);
Cti.setHandler(Cti.MessageType.QUEUESTATISTICS, queueStatisticsHandler);
Cti.setHandler(Cti.MessageType.QUEUECONFIG, queueConfigHandler);
Cti.setHandler(Cti.MessageType.QUEUELIST, queueConfigHandler);
Cti.setHandler(Cti.MessageType.QUEUEMEMBER, queueMemberHandler);
Cti.setHandler(Cti.MessageType.QUEUEMEMBERLIST, queueMemberHandler);
Cti.setHandler(Cti.MessageType.DIRECTORYRESULT, directoryResultHandler);

Cti.setHandler(Cti.MessageType.AGENTCONFIG, agentConfigHandler);
Cti.setHandler(Cti.MessageType.AGENTLIST, agentConfigHandler);
Cti.setHandler(Cti.MessageType.AGENTGROUPLIST, agentGroupConfigHandler);
Cti.setHandler(Cti.MessageType.AGENTSTATEEVENT, agentStateEventHandler);
Cti.setHandler(Cti.MessageType.AGENTERROR, agentErrorHandler);
Cti.setHandler(Cti.MessageType.ERROR, errorHandler);
Cti.setHandler(Cti.MessageType.AGENTDIRECTORY, agentDirectoryHandler);

Cti.setHandler(Cti.MessageType.CONFERENCES, conferencesHandler);
Cti.setHandler(Cti.MessageType.CALLHISTORY, callHistoryHandler);

xc_webrtc.setHandler(xc_webrtc.MessageType.GENERAL, webRtcGeneralEventHandler);
xc_webrtc.setHandler(xc_webrtc.MessageType.REGISTRATION, webRtcRegistrationEventHandler);
xc_webrtc.setHandler(xc_webrtc.MessageType.INCOMING, webRtcIncomingEventHandler);
xc_webrtc.setHandler(xc_webrtc.MessageType.OUTGOING, webRtcOutgoingEventHandler);

```

## Debugging

**Cti features** Cti messages can be logged in the console if the `Cti.debugMsg` variable is set to `true`, you can do it directly in the developer tools console:

```
Cti.debugMsg=true;
```

You'll then get send and received messages in the console log (prefixed by S>>> and R<<< respectively):

```
2016-11-23 14:48:59.180 S>>> {"claz":"web","command":"dial","destination":"111","variables":{}}
2016-11-23 14:48:59.557 R<<< {"msgType":"PhoneStatusUpdate","ctiMessage":{"status":"CALLING"}}
```

**WebRTC features** The WebRTC debug can be activated separately by the following method:

```
xc_webrtc.setDebug(sipml5level, event, handler)
```

Where:

- sipml5level refers to the SIPml5 library log level string as described on [SIPml5 log level documentation](#),
- event is a boolean value activating event logging (each event is prefixed by RE<<<),
- handler is a boolean value activating logging of message handler subscription/unsubscription.

**WebRTC on sample page** Once logged on the sample page, you can init the webRTC through the init button, follow events shown in the webRTC section and send and receive calls. You can terminate a call by the terminate button in the phone section. Direct and attended transfer can be performed using phone section methods. Hold and DTMF features are available via the webRTC API. Current implementation support just one simultaneous call.

Current browsers doesn't allow media sharing without secure connections - https and wss. The xivoxc\_nginx docker image contains the configuration required for loading the sample page over a secure connection using an auto-signed certificate. This certificate is automatically generated by the installation script. It is meant to be used only for test purposes, you should replace it by a signed certificate before switching to production. The sample page is available on the following address: [https://MACHINE\\_IP:8443/sample](https://MACHINE_IP:8443/sample)

## Login

**User login** Users can connect using login, password and phone number:

```
var wsurl = "ws://" + server + "/ctichannel?username="+username+"&agentNumber="+phoneNumber+"&";
Cti.WebSocket.init(wsurl, username, phoneNumber);
```

**Agent login** An agent can be logged in using *Cti.loginAgent(agentPhoneNumber, agentId)*. For the moment, the phone number used for agent login should be the same as the one used for user login, otherwise you will get many error messages "LoggedInOnAnotherPhone".

Following cases are handled:

- agent is not logged and requests a login to a known line: the agent is logged in
- agent is not logged and requests a login to an unknown line: an error is raised:

```
{"Error": "PhoneNumberUnknown"}
```

- agent is already logged on the requested line: the agent stays logged
- agent is already logged on another line: an error is raised and the agent stays logged (on the number where he was logged before the new request). It's up to the implementation to handle this case.

```
{"Error": "LoggedInOnAnotherPhone", "phoneNb": "1002", "RequestedNb": "1001"}
```

- agent is not logged and requests a login to a line already used by another agent: the agent takes over the line and the agent previously logged on the line is unlogged

## Generic CTI Messages



**Error**

- Cti.MessageType.ERROR

**LoggedOn**

- Cti.MessageType.LOGGEDON

**Sheet**

- Cti.MessageType.SHEET

```
{ "msgType": "Sheet", "ctiMessage": { "timenow": 1425055334, "compressed": true, "serial": "xml",
 "payload": { "profile": { "user": { "internal": [{ "content": "xivo", "name": "ipbxid",
 { "content": "link", "name": "where", { "content": "1425055330.23", "name": "uid",
 { "content": "no", "name": "focus", { "content": "1", "name": "zip" }] },
 "sheetQtui": null, "sheetInfo": [{ "value": "http://www.google.fr/", "name": "popupUrl", "order": 1,
 { "value": "&folder=1234", "name": "folderNumber", "order": 30, "type": "text",
 { "value": "http://www.google.fr/", "name": "popupUrl1", "order": 20, "type": "url" }] }, "systrayInfo": {
```

**Directory And Favorites**

**Cti.directoryLookUp:** **function(term)** This command deprecates previously used *Cti.searchDirectory(pattern)*.

**Associated Handler**

- Cti.MessageType.DIRECTORYRESULT

Triggered by command *Cti.directoryLookUp(pattern)*. This command deprecates previously used *Cti.searchDirectory(pattern)*.

```
{ "msgType": "DirectoryResult",
 "ctiMessage": {
 "entries": [
 { "status": 0, "entry": ["hawkeye", "pierce", "1002", "0761187406", "false"] },
 { "status": -2, "entry": ["peter", "pan", "1004", "", "false"] },
 "headers":
 ["Firstname", "Lastname", "Number", "Mobile", "Favorite"] }
```

**Cti.getFavorites: function()**

**Cti.addFavorite: function(contactId, source)**

**Cti.removeFavorite: function(contactId, source)**

**User Statuses**

- Cti.MessageType.USERSTATUSES : “UsersStatuses”

**User Status Update**

- Cti.MessageType.USERSTATUSUPDATE : “UserStatusUpdate”,

## User Config Update

- Cti.MessageType.USERCONFIGUPDATE : “UserConfigUpdate”,

```
{ "msgType": "UserConfigUpdate",
 "ctiMessage": { "userId": 9, "dndEnabled": false, "naFwdEnabled": false, "naFwdDestination": "", "uncFwdDestination": "",
 "firstName": "Alice", "lastName": "Johnson", "fullName": "Alice Johnson", "mobileNumber": "064574512"
```

## Phone Status Update

- Cti.MessageType.PHONESTATUSUPDATE

## Phone Events

- Cti.MessageType.PHONEEVENT

Phone events are automatically sent when application is connected

Format

```
{
 "msgType": "PhoneEvent",
 "ctiMessage": {
 "eventType": "EventRinging",
 "DN": "1118",
 "otherDN": "1058",
 "linkedId": "1447670380.34",
 "uniqueId": "1447670382.37",
 "queueName": "blue",
 "userData": {
 "XIVO_CONTEXT": "default", "XIVO_USERID": "9", "XIVO_SRCNUM": "1058", "XIVO_DSTNUM": "3000"
 }
 }
}
```

fields	Description
Event types	<ul style="list-style-type: none"> <li>• EventReleased</li> <li>• EventDialing</li> <li>• EventRinging</li> <li>• EventEstablished</li> </ul>
DN	The directory number of the event
otherDN	Can be calling number of called number
queueName	Optional, the queue name for inbound acd calls
UserData	Contains a list of attached data, system data XIVO_ or data attached to the call key beginning by USR_

If you use the following preprocess subroutine

```
[user_data_test]
exten = s,1,Log(DEBUG,**** set user data ****)
same = n,SET(USR_DATA1=hello)
same = n,SET(USR_DATA2=24)
same = n,SET(USR_DATA3=with space)
same = n,Return()
```

you will get these data in the events. Data can also be attached using the *Cti.dial* command.

## Voice Mail Status Update

- VOICEMAILSTATUSUPDATE : “VoiceMailStatusUpdate”,

```
{ "msgType": "VoiceMailStatusUpdate", "ctiMessage": { "voiceMailId": 58, "newMessages": 2, "waitingMessages": 1 }
```

## Link Status Update

- Cti.MessageType.LINKSTATUSUPDATE

## Queue Statistics

- Handler on : Cti.MessageType.QUEUESTATISTICS

The handler is executed when a notification of new statistic values is received. Each message contains one or more counters for one queue. The queue is identified by its queueId. See example below for reference. The queue's id can be used to retrieve queue's configuration, see [Queue Configuration](#).

Following counters are available:

- TotalNumberCallsEntered
- TotalNumberCallsAnswered
- PercentageAnsweredBefore15
- TotalNumberCallsAbandoned
- TotalNumberCallsAbandonedAfter15
- PercentageAbandonedAfter15
- WaitingCalls
- LongestWaitingTime
- EWT
- AvailableAgents
- TalkingAgents

```
{
 "msgType": "QueueStatistics",
 "ctiMessage": {
 "queueId": 11, "counters": [{ "statName": "AvailableAgents", "value": 0 }, { "statName": "LoggedAgents", "value": 1 }]
 }
}
```

Some messages contain a queueRef with a queue's name instead of the queueId. This issue should be eliminated in future versions.

```
{ "queueRef": "travels", "counters": [{ "statName": "TotalNumberCallsAbandoned", "value": 19 }] }
```

## Queue Calls

- Handler on: Cti.MessageType.QUEUECALLS

Awaiting calls in a queue. Subscription to the events with : *Cti.subscribeToQueueCalls(9)* (9 being the queueId). Unsubscription with: *Cti.unsubscribeToQueueCalls(9)*.

```
{ "queueId": 9, "calls": [{ "position": 1, "name": "John Doe", "number": "33356782212", "queueTime": "2015-07-14 14:14:14" }] }
```

## Queue Configuration

- QUEUECONFIG : “QueueConfig”,

```
{ "id": 8, "context": "default", "name": "blue", "displayName": "blue sky", "number": "3506" }
```

## Queue List

- QUEUelist : “QueueList”,

```
{
 "msgType": "QueueList",
 "ctiMessage": [
 { "id": 170, "context": "default", "name": "bluesky", "displayName": "Bl Record", "number": "3012" },
 { "id": 5, "context": "default", "name": "noagent", "displayName": "noagent", "number": "3050" },
 { "id": 6, "context": "default", "name": "__switchboard_hold", "displayName": "Switchboard hold", "number": "3099" },
 { "id": 173, "context": "default", "name": "outbound", "displayName": "outbound", "number": "3099" },
 { "id": 2, "context": "default", "name": "yellow", "displayName": "yellow stone", "number": "3001" },
 { "id": 7, "context": "default", "name": "green", "displayName": "green openerp", "number": "3006" },
 { "id": 3, "context": "default", "name": "red", "displayName": "red auto polycom", "number": "3002" },
 { "id": 11, "context": "default", "name": "pool", "displayName": "Ugips Pool", "number": "3100" },
 { "id": 4, "context": "default", "name": "__switchboard", "displayName": "Switchboard", "number": "3099" },
]
}
```

## Queue Member

- Handler on : Cti.MessageType.QUEUEMEMBER

Received when an agent is associated to a queue or a penalty is updated. Penalty is -1 when agent is removed from a queue

```
{ "agentId": 19, "queueId": 3, "penalty": 12 }
```

## Queue Member List

- Handler on : Cti.MessageType.QUEUEMEMBERLIST

```
{
 "msgType": "QueueMemberList",
 "ctiMessage": [
 { "agentId": 129, "queueId": 8, "penalty": 2 },
 { "agentId": 139, "queueId": 168, "penalty": 2 },
 { "agentId": 129, "queueId": 10, "penalty": 0 },
 { "agentId": 129, "queueId": 11, "penalty": 0 }
]
}
```

## Agent State Event

- Cti.MessageType.AGENTSTATEEVENT

- AgentLogin

```
{ "name": "AgentLogin", "agentId": 19, "phoneNb": "1000", "since": 1423839787, "queues": [8, 14, 170, 4, 1] }
```

- AgentReady

```
{ "name": "AgentReady", "agentId": 19, "phoneNb": "1000", "since": 0, "queues": [8, 14, 170, 4, 1], "cause": "AgentReady" }
```

- AgentOnPause

```
{ "name": "AgentOnPause", "agentId": 19, "phoneNb": "1000", "since": 0, "queues": [8, 14, 170, 4, 1], "cause": "AgentOnPause" }
```

- AgentOnWrapup

```
{ "name": "AgentOnWrapup", "agentId": 19, "phoneNb": "1000", "since": 2, "queues": [8, 14, 170, 4, 1], "cause": "AgentOnWrapup" }
```

- AgentRinging

```
{ "name": "AgentRinging", "agentId": 19, "phoneNb": "1000", "since": 0, "queues": [8, 14, 170, 4, 1], "c
```

#### – AgentDialing

```
{ "name": "AgentDialing", "agentId": 19, "phoneNb": "1000", "since": 0, "queues": [8, 14, 170, 4, 1], "c
```

#### – AgentOnCall

```
{ "msgType": "AgentStateEvent", "ctiMessage":
 { "name": "AgentOnCall", "agentId": 19, "phoneNb": "1000", "since": 0, "queues": [8, 14, 170, 4, 1]
 "cause": "available", "acd": false, "direction": "Incoming", "callType": "External", "mor
```

#### – AgentLoggedOut

```
{ "name": "AgentLoggedOut", "agentId": 19, "phoneNb": "1000", "since": 0, "queues": [8, 14, 170, 4, 1], "
```

### Agent Error

- Cti.MessageType.AGENTERROR

### Agent Directory

- Cti.MessageType.AGENTDIRECTORY

Triggered by command *Cti.getAgentDirectory*

```
{ "directory": [
 { "agent":
 { "context": "default", "firstName": "bj", "groupId": 1, "id": 8, "lastName": "agent", "num
 "agentState": { "agentId": 8, "cause": "", "name": "AgentReady", "phoneNb": "1001", "queue:
```

### Agent Configuration

- Cti.MessageType.AGENTCONFIG

Triggered when agent configuration changes

```
{ "id": 23, "firstName": "Jack", "lastName": "Flash", "number": "2501", "context": "default" }
```

### Agent List

- Cti.MessageType.AGENTLIST

Receives agent configuration list in a javascript Array : Command *Cti.getList("agent")*;

```
[
 { "id": 24, "firstName": "John", "lastName": "Waynes", "number": "2601", "context": "default", "groupId":
 { "id": 20, "firstName": "Maricé", "lastName": "Sapritchà", "number": "2602", "context": "default", "gro
 { "id": 147, "firstName": "Etienne", "lastName": "Burgad", "number": "30000", "context": "default", "gro
 { "id": 148, "firstName": "Caroline", "lastName": "HERONDE", "number": "29000", "context": "default", "g
 { "id": 149, "firstName": "Eude", "lastName": "GARTEL", "number": "75000", "context": "default", "groupI
 { "id": 22, "firstName": "Alice", "lastName": "Johnson", "number": "2058", "context": "default", "groupI
]
```

### Agent Listen

- AGENTLISTEN: “AgentListen”,

Receives agent listen stop / start event, received automatically if user is an agent, no needs to subscribe.

```
{ "started": false, "phoneNumber": "1058", "agentId": 22 }
```

## Agent Group List

- AGENTGROUPLIST : “AgentGroupList”

Agent group list triggered by command : *Cti.getList(“agentgroup”)*

```
[
 { "id":1, "name": "default" },
 { "id":2, "name": "boats" },
 { "id":3, "name": "broum" },
 { "id":4, "name": "bingba3nguh" },
 { "id":5, "name": "salesexpert" },
 { "id":6, "name": "a_very_long_group_name" }
]
```

**Agent Statistics** Received on subscribe to agent statistics with method *Cti.subscribeToAgentStats()*, current statistics are received automatically on subscribe.

- AGENTSTATISTICS : “AgentStatistics”

```
{ "id":22,
 "statistics": [
 { "name": "AgentPausedTotalTime", "value":0 },
 { "name": "AgentWrapupTotalTime", "value":0 },
 { "name": "AgentReadyTotalTime", "value":434 },
 { "name": "LoginDateTime", "value": "2015-04-27T08:15:01.081+02:00" },
 { "name": "LogoutDateTime", "value": "2015-04-27T08:14:49.427+02:00" }
]
}
```

## Call History

**Cti.getUserCallHistory(size)** Get the call history of the logged in user, limited to the last *size* calls.

**Cti.getAgentCallHistory(size)** Get the call history of the logged in agent, limited to the last *size* calls.

**Cti.getQueueCallHistory(queue, size)** Get a call history for a queue or a set of queues. You may pass part of a queue name (not display name).

i.e. pass bl if you want to match queue name blue, black and blow

**Associated Handler CALLHISTORY** Received when calling the above methods *Cti.getAgentCallHistory(size)* or *Cti.getUserCallHistory(size)*.

- CALLHISTORY : “CallHistory”

```
{
 "start": "2014-01-01 08:00:00",
 "duration": "00:21:35",
 "srcNum": "0115878",
 "dstNum": "2547892",
 "status": "answered"
}
```

For queue calls status can be :

- full - full queue
- closed - closed queue
- joinempty - call arrived on empty queue

- leaveempty - exit when queue becomes empty
- divert\_ca\_ratio - call redirected because the ratio waiting calls/agents was exceeded
- divert\_waittime - call redirected because estimated waiting time was exceeded;
- answered - call answered
- abandoned - call abandoned
- timeout - maximum waiting time exceeded

For other calls

- emitted
- missed
- ongoing

## Callback Messages

**Callback lists** Received when calling *Callback.getCallbackLists()*.

- CALLBACKLISTS : “CallbackLists”

```
{ "uuid": "b0849ac0-4f4a-4ed0-9386-53ab2afd94b1",
 "name": "Liste de test",
 "queueId": 1,
 "callbacks": [
 { "uuid": "a967da84-bc41-4bf4-a4fc-2bcc54e11606",
 "listUuid": "b0849ac0-4f4a-4ed0-9386-53ab2afd94b1",
 "phoneNumber": "0230210082",
 "mobilePhoneNumber": "0789654123",
 "firstName": "Alice",
 "lastName": "O'Neill",
 "company": "YourSociety",
 "description": null,
 "agentId": null,
 "dueDate": "2016-08-01",
 "preferredPeriod": {
 "default": false,
 "name": "Afternoon",
 "periodStart": "14:00:00",
 "periodEnd": "17:00:00",
 "uuid": "d3270038-e20e-498a-af71-3cf69b5cc792"
 }
 }
]
}
```

**Callback Taken** Received after taking a callback with *Callback.takeCallback(uuid)*.

- CALLBACKTAKEN : “CallbackTaken”

```
{ "uuid": "a967da84-bc41-4bf4-a4fc-2bcc54e11606",
 "agentId": 2 }
```

**Callback Started** Received after starting a callback with *Callback.startCallback(uuid, phoneNumber)*.

- CALLBACKSTARTED : “CallbackStarted”

```
{ "requestUuid": "a967da84-bc41-4bf4-a4fc-2bcc54e11606",
 "ticketUuid": "8e82de0f-847a-4606-97bf-bef5a18ea8b0" }
```

**Callback Clotured** Received after giving to a callback a status different of `Callback`.

- `CALLBACKCLOTURED` : “CallbackClotured”

```
{ "uuid": "a967da84-bc41-4bf4-a4fc-2bcc54e11606" }
```

**Callback Released** Received after releasing a callback with `Callback.releaseCallback(uuid)`.

- `CALLBACKRELEASED` : “CallbackReleased”

```
{ "uuid": "a967da84-bc41-4bf4-a4fc-2bcc54e11606" }
```

**Callback Updated** Received when calling `Callback.updateCallbackTicket(uuid, status, description, dueDate, periodUuid)` with a new due date or period.

- `CALLBACKREQUESTUPDATED` : “CallbackRequestUpdated”

```
{ "request": {
 "uuid": "a967da84-bc41-4bf4-a4fc-2bcc54e11606",
 "listUuid": "b0849ac0-4f4a-4ed0-9386-53ab2afd94b1",
 "phoneNumber": "0230210082",
 "mobilePhoneNumber": "0789654123",
 "firstName": "Alice",
 "lastName": "O'Neill",
 "company": "YourSociety",
 "description": null,
 "agentId": null,
 "dueDate": "2016-08-01",
 "preferredPeriod": {
 "default": false,
 "name": "Afternoon",
 "periodStart": "14:00:00",
 "periodEnd": "17:00:00",
 "uuid": "d3270038-e20e-498a-af71-3cf69b5cc792"
 }
}
}
```

## Membership Messages

**User default membership** Received when calling `Membership.getUserDefaultMembership(userId)`.

- `USERQUEUEDEFAULTMEMBERSHIP`: “UserQueueDefaultMembership”

```
{
 "userId": 186,
 "membership": [
 { "queueId": 8, "penalty": 1 },
 { "queueId": 17, "penalty": 0 },
 { "queueId": 18, "penalty": 0 },
 { "queueId": 23, "penalty": 0 }
]
}
```

## Methods

**Cti.changeUserStatus()** Update user status using a cti server configured status name

**Cti.loginAgent(agentPhoneNumber, agentId)** Log an agent



**Cti.logoutAgent(agentId)** Un log an agent

**Cti.pauseAgent(agentId)** Change agent state to pause

**Cti.unpauseAgent(agentId)** Change agent state to ready

**Cti.listenAgent(agentId)** Listen to an agent

**Cti.dnd(state)** Set or unset do not disturb, state true or false

**Cti.dial(destination, variables)** Place a call to destination with the provided variables. Variables must take the following form:

```
{
 var1: "value 1",
 var2: "value 2"
}
```

USR\_var1 and USR\_var2 will be attached to the call and propagated to *Phone Events*

**Cti.dialFromMobile(destination, variables)** Place a call from logged user's mobile number to destination with the provided variables. Variables must take the following form:

```
{
 var1: "value 1",
 var2: "value 2"
}
```

USR\_var1 and USR\_var2 will be attached to the call and propagated to *Phone Events*

**Cti.originate(destination)** Originate a call

**Cti.hangup()** Hangup a call

**Cti.answer()** Answer a call

**Cti.hold()** Put current call on hold

**Cti.directTransfer(destination)** Tranfert to destination

**Cti.attendedTransfer(destination)** Start a transfer to a destination

**Cti.completeTransfer()** Complete previously started transfer

**Cti.cancelTransfer()** Cancel a transfer

**Cti.conference()** Start a conference using phone set capabilities

**Cti.monitorPause(agentId)** Pause call recording

---

**Note:** You can only pause the recording of a call answered by an agent (i.e. a call sent via a Queue towards an Agent).

---

**Cti.monitorUnpause(agentId)** Unpause call recording

---

**Note:** You can only pause the recording of a call answered by an agent (i.e. a call sent via a Queue towards an Agent).

---

**Cti.getList(objectType)** Request a list of configuration objects, objectType can be :

- queue
- agent
- queuemember

Triggers handlers QUEUelist, AGENTlist, QUEUEMEMBERlist. Subscribes to configuration modification changes, handlers QUEUECONFIG, AGENTCONFIG, QUEUEMEMBER can also be called

**Cti.setAgentQueue(agentId, queueId, penalty)**

- agentId (Integer) : id of agent, returned in message *Agent Configuration*
- queueId (Integer) : id of queue, returned in message *Queue Configuration*
- penalty (Integer) : positive integer

If agent is not associated to the queue, associates it, otherwise changes the penalty

On success triggers a *Queue Member* event, does not send anything in case of failure :

```
{ "agentId": <agentId>, "queueId": <queueId>, "penalty": <penalty> }
```

**Cti.removeAgentFromQueue(agentId, queueId)**

- agentId (Integer) : id of agent, returned in message *Agent Configuration*
- queueId (Integer) : id of queue, returned in message *Queue Configuration*

On success triggers a queue member event with penalty equals to -1, does not send anything in case of failure :

```
{ "agentId": <agentId>, "queueId": <queueId>, "penalty": -1 }
```

**Cti.subscribeToAgentStats()** Subscribe to agent statistics notification. When called all current statistics are receive, and a notification is received for each updates. Both initial values and updates are transmitted by the *Agent Statistics* messages.

**Cti.subscribeToQueueStats()** This command subscribes to the queue statistics notifications. First, all actual statistics values are sent for initialisation and then a notification is sent on each update. Both initial values and updates are transmitted by the QUEUESTATISTICS messages.

**Cti.naFwd(destination,state)** Forward on non answer

**Cti.uncFwd(destination,state)** Unconditionnal forward

**Cti.busyFwd(destination,state)** Forward on busy

## Callback Commands

**Callback.getCallbackLists()** Retrieve the lists of callbacks with their associated callback requests, and subscribe to callback events.

**Callback.takeCallback(uuid)** Take the callback with the given uuid with the logged-in agent.

**Callback.releaseCallback(uuid)** Release the callback which was previously taken

**Callback.startCallback(uuid, phoneNumber)** Launch the previously taken callback with the provided phone number.

**Callback.updateCallbackTicket(uuid, status, description, dueDate, periodUuid)** Update a callback ticket with the provided description and status. Allowed values for status are:

- NoAnswer
- Answered
- Fax
- Callback

dueDate is an optional parameter specifying the new due date using ISO format (“YYYY-MM-DD”).

periodUuid is an optional parameter specifying the new preferred period for the callback.

## Membership Commands

**Membership.init(cti)** Initialize the Membership library using the given Cti object.

**Membership.getUserDefaultMembership(userId)** Request the default membership for the given user id. Warning, the userId is not the same as the agentId.

**Membership.setUserDefaultMembership(userId, membership)** Set the default membership for the given user id. Warning, the userId is not the same as the agentId. ‘membership’ should be an array of Queue membership like:

```
[
 { "queueId": 8, "penalty": 1 },
 { "queueId": 17, "penalty": 0 },
 { "queueId": 18, "penalty": 0 },
 { "queueId": 23, "penalty": 0 }
]
```

**Membership.setUsersDefaultMembership(userIds, membership)** Set the default membership for the given array of user id. Warning, the userId is not the same as the agentId. ‘userIds’ should be an array of user id like :

```
[1, 2, 3]
```

‘membership’ should be an array of Queue membership like:

```
[
 {"queueId":8,"penalty":1},
 {"queueId":17,"penalty":0},
 {"queueId":18,"penalty":0},
 {"queueId":23,"penalty":0}
]
```

**Membership.applyUsersDefaultMembership(userIds)** Apply the existing default configuration to a set of users. Warning, the `userId` is not the same as the `agentId`. ‘usersIds’ should be an array of `userId` like:

:: [1, 2, 7, 9]

## webRTC integration

**Methods** Once the `cti` login done, you can init the webRTC component by calling the `xc_webrtc.init` method.

**xc\_webrtc.init(name, ssl, websocketPort, remoteAudio, ip)** Init the webRTC connection and register the user’s line.

- `name` - user’s login to get the line details,
- `ssl` - if set to true the wss is used,
- `websocketPort`, `ip` - port and address for the webRTC websocket connection, when `ip` is not passed the xivo `ip` is used,
- `remoteAudio` - id of the HTML5 audio element for remote audio player, if not passed ‘audio\_remote’ is used. The element should look like:

```
<audio id="audio_remote" autoplay="autoplay"></audio>
```

**xc\_webrtc.dial(destination)** Start a webRTC call.

**xc\_webrtc.answer()** Answer an incoming webRTC call.

**xc\_webrtc.hold()** Toggle hold on a webrtc call.

**xc\_webrtc.dtmf(key)** Send a DTMF.

**xc\_webrtc.setHandler(eventName, handler)** Set a handler for `eventName` from `xc_webrtc.MessageType`.

**xc\_webrtc.disableICE()** Disable ICE server use, only LAN addresses will be used in the SDP.

**xc\_webrtc.setIceUrls(urls)** Set a list of STUN/TURN servers, for example:

```
[{ url: 'stun:stun.l.google.com:19302'}, { url:'turn:turn.server.org', username: 'user', credenti
```

**Events** There are for groups of events:

- general,
- register,
- incoming,
- outgoing.

List of associated events is defined in the *xc\_webrtc.General*, *xc\_webrtc.Registration*, *xc\_webrtc.Incoming*, *xc\_webrtc.Outgoing*. See the *xc\_webrtc.js* on [https://gitlab.com/xivoxc/xucserver/blob/master/app/assets/javascripts/xc\\_webrtc.js](https://gitlab.com/xivoxc/xucserver/blob/master/app/assets/javascripts/xc_webrtc.js). The error state events contains a description in the reason field. Call establishment event contains *caller* or *callee* detail. Use the sample page to see some examples.

## Rest API

### General form

`http://localhost:$xucport/xuc/api/1.0/$method/$domain/$username/`

withHeaders(("Content-Type", "application/json"))

- `$xucport` : Xuc port number (default 8090)
- `$method` : See available methods below
- `$domain` : Represents a connection site, can be anything
- `$username` : XiVO client user username

### Events

Xuc post JSON formatted events on URL `eventUrl = "http://localhost:8090/xivo/1.0/event/avencall"` configured in `/usr/share/xuc/application.conf`

**Phone Event Notification** Related to a username, phone event is in message payload same structure as javascript *Phone Events*

```
{
 "username": "alicej",
 "message": {
 "msgType": "PhoneEvent",
 "ctiMessage": { "eventType": "EventDialing", "DN": "1058", "otherDN": "3000", "linkedId": "1447670380"
 }
}
```

### Connection

POST `http://localhost:$xucport/xuc/api/1.0/connect/$domain/$username/`

```
{ "password" : "password" }

curl -XPOST -d '{"password": "<password>"}' -H "Content-Type: application/json" http://localhost
```

### DND

POST `http://localhost:$xucport/xuc/api/1.0/dnd/$domain/$username/`

```
{ "state" : [false|true] }

curl -XPOST -d '{"state": false}' -H "Content-Type: application/json" http://localhost:8090/xuc/ap
```

### Dial

POST [http://localhost:\\$xucport/xuc/api/1.0/dial/\\$domain/\\$username/](http://localhost:$xucport/xuc/api/1.0/dial/$domain/$username/)

```
{ "number" : "1101" }
```

```
curl -XPOST -d '{"number":"<number>"}' -H "Content-Type: application/json" http://localhost:8090
```

**Phone number sanitization** Dial command automatically applies filters to the phone number provided to make it valid for Xivo. Especially, it removes invalid characters and handles properly different notations of international country code.

Some countries don't follow the international standard and actually keep the leading zero after the country code (e.g. Italy). Because of this, if the zero isn't surrounded by parenthesis, the filter keeps it <sup>1</sup>.

**Forward** All forward commands use the above payload

```
{ "state" : [true|false],
 "destination" : "1102" }
```

### Unconditional

POST [http://localhost:\\$xucport/xuc/api/1.0/uncForward/\\$domain/\\$username/](http://localhost:$xucport/xuc/api/1.0/uncForward/$domain/$username/)

```
curl -XPOST -d '{"state":true,"destination":"<destnb>"}' -H "Content-Type: application/json" http://localhost:8090
```

### On No Answer

POST [http://localhost:\\$xucport/xuc/api/1.0/naForward/\\$domain/\\$username/](http://localhost:$xucport/xuc/api/1.0/naForward/$domain/$username/)

```
curl -XPOST -d '{"state":true,"destination":"<destnb>"}' -H "Content-Type: application/json" http://localhost:8090
```

### On Busy

POST [http://localhost:\\$xucport/xuc/api/1.0/busyForward/\\$domain/\\$username/](http://localhost:$xucport/xuc/api/1.0/busyForward/$domain/$username/)

```
curl -XPOST -d '{"state":true,"destination":"<destnb>"}' -H "Content-Type: application/json" http://localhost:8090
```

**Handshake** Will repost all events on the configured URL

POST [http://localhost:\\$xucport/xuc/api/1.0/handshake/\\$domain/](http://localhost:$xucport/xuc/api/1.0/handshake/$domain/)

**AgentLogout** Logout un agent

POST [http://\\$xuchost:\\$xucport/xuc/api/1.0/agentLogout/](http://$xuchost:$xucport/xuc/api/1.0/agentLogout/)

```
curl -XPOST -d '{"phoneNumber":"<phoneNumber>"}' -H "Content-Type: application/json" http://localhost:8090
```

**TogglePause** Change state of an agent, pause if ready, ready if on pause

POST [http://\\$xuchost:\\$xucport/xuc/api/1.0/togglePause/](http://$xuchost:$xucport/xuc/api/1.0/togglePause/)

```
curl -XPOST -d '{"phoneNumber":"<phoneNumber>"}' -H "Content-Type: application/json" http://localhost:8090
```

---

<sup>1</sup> See Redmine ticket #150

## Statistics

### Exposed by xuc

**Queue statistics** These real time statistics are calculated nearly in real time from the queue\_log table. Statistics are reset to 0 at midnight (24h00) and can be changed by configuration.

#### Real time calculated Queue statistic

name	Description
TotalNumberCallsEntered	Total number of calls entered in a queue
TotalNumberCallsAbandoned	Total number of calls abandoned in a queue (not answered)
TotalNumberCallsAbandonedAfter15	Total number of calls abandoned after 15 seconds
TotalNumberCallsAnswered	Total number of calls answered
TotalNumberCallsAnsweredBefore15	Total number of calls answered before 15 seconds
PercentageAnsweredBefore15	Percentage of calls answered before 15 seconds over calls entered
PercentageAbandonedAfter15	Percentage of calls abandoned after 15 seconds over calls entered
TotalNumberCallsClosed	Total number of calls received when queue is closed
TotalNumberCallsTimeout	Total number of calls diverted on queue timeout

All queue statistics counters are also available for the sliding last hour by adding LastHour to the name .i.e. TotalNumberCallsAbandonedLastHour

For percentage, it is the mean of the sliding last hour value

**Other queue statistics** Other queue statistics are calculated by xivo cti server

- AvailableAgents
- TalkingAgents
- LongestWaitTime
- WaitingCalls
- EWT

Definition in xivo documentation [xivo documentation](#)

#### Calculated Agent statistics

name	Description
PausedTime	Total time agent in pause
WrapupTime	Total time agent in wrapup
ReadyTime	Total time agent ready
InbCalls	Total number of inbound calls received internal and external
InbCallTime	Total time for inbound calls received internal and external
InbAnsCalls	Answered inbound calls received internal and external
InbUnansCalls	Unanswered inbound calls received internal and external
InbPercUnansCalls	Percentage of unanswered inbound calls received internal and external
InbAverCallTime	Average time for inbound calls received internal and external
OutCalls	Total number of outbound calls received internal and external
LoginDateTime	Last login date time
LogoutDateTime	Last logout date time

Inbound calls, are all calls received by an agent, internal, external or acd calls. Outbound calls are all calls dialed by an agent, internal or external calls.

Agent statistics are calculated internally on a daily basis and reset to 0 at midnight (default configuration). see javascript api

If some status are configured in xivo cti server with activate pause to all queue = true, additionnal statistics computing the total time in not ready with this status are calculated. This statistics name is equal to the presence name configuration in XiVO.

### Technical structure of XiVO-CC

**Reporting** The reporting is composed of four packages: pack-reporting, xivo-full-stats, xivo-reporting-db and xivo-db replication.

These packages will feed the tables of the xivo\_stats database:

- xivo-db-replication feeds the tables cel and queue\_log in real time, and the configuration tables (dialaction, linefeatures, etc...) every 5 minutes
- xivo-full-stats feeds in real time the tables call\_on\_queue, call\_data, stat\_queue\_periodic, stat\_agent\_periodic and agent\_position
- xivo-reporting-db and pack-reporting work together to feed the tables stat\_queue\_specific, stat\_agent\_queue\_specific and stat\_agent\_specific every 15 minutes

### Troubleshooting

In this section, we give some troubleshooting hints. Continue by choosing the component.

#### Check installation

In order for the XivoCC components to be fully functional, some customizations need to be done on the XiVO CC and the XiVO PBX.

This page can help to check that all the correct customization have been done by the installation package.

For the rest of this page we will make the following assumptions: - XiVO PBX has the IP 192.168.0.1 - XiVO CC has the IP 192.168.0.2

#### Check the XiVO CC configuration

##### Check the prerequisites

- the OS must be Debian 8 (jessie), 64 bit,
- Docker must be installed,
- Docker-compose must be installed,
- the XiVO PBX must be reachable on the network.

**Check ntp installation** The XiVO CC server and the XiVO server must be synchronized to the same source NTP source.

**Check Logrotate configuration** A file /etc/logrotate.d/docker-container must be present which should log rotate files /var/lib/docker/containers/\*/\*.log

You can test it with logrotate -fv /etc/logrotate.d/docker-container. You should get some output and a new log file with suffix [CONTAINER ID]-json.log.1 should be created. This file is compressed in next rotation cycle.



## Check Docker compose

- An alias for docker-compose must be present like:

```
alias dcomp='docker-compose -p xivocc -f /etc/docker/compose/docker-xivocc.yml'
```

- The version of the docker images in the file /etc/docker/compose/docker-xivocc.yml must be 2016.03.latest:

```
...
xivo_stats:
 image: xivocc/xivo-full-stats:2016.03.latest
...

xuc:
 image: xivocc/xuc:2016.03.latest
...
```

## Check the services The list of the services launched should look like :

# dcomp ps	Name	Command	State	Ports
xivocc_config_mgt_1	bin/config-mgt-docker	Up	0.0.0.0:9100->9000/tcp	
xivocc_elasticsearch_1	/docker-entrypoint.sh elas...	Up	0.0.0.0:9200->9200/tcp, 0.0.0.0:9300->9300/tcp	
xivocc_fingerboard_1	/bin/sh -c /usr/bin/tail - ...	Up		
xivocc_kibana_volumes_1	/bin/sh -c /usr/bin/tail - ...	Up		
xivocc_nginx_1	nginx -g daemon off;	Up	443/tcp, 0.0.0.0:80->80/tcp	
xivocc_pack_reporting_1	/bin/sh -c echo ...	Up		
xivocc_pgxivocc_1	/docker-entrypoint.sh postgres	Up	0.0.0.0:5443->5432/tcp	
xivocc_postgresvols_1	/bin/bash	Exit 0		
xivocc_recording_server_1	bin/recording-server-docker	Up	0.0.0.0:9400->9000/tcp	
xivocc_reporting_rsync_1	/usr/local/sbin/run-rsync.sh	Up	0.0.0.0:873->873/tcp	
xivocc_spagobi_1	/bin/sh -c /root/start.sh	Up	0.0.0.0:9500->8080/tcp	
xivocc_timezone_1	/bin/bash	Exit 0		
xivocc_xivo_replic_1	/usr/local/bin/start.sh /o ...	Up		
xivocc_xivo_stats_1	/usr/local/bin/start.sh /o ...	Up		
xivocc_xivocclogs_1	/bin/bash	Exit 0		
xivocc_xuc_1	bin/xuc-docker	Up	0.0.0.0:8090->9000/tcp	
xivocc_xucmgt_1	bin/xucmgt-docker	Up	0.0.0.0:8070->9000/tcp	

## Check the XiVO PBX

### Check PostgreSQL configuration

- Postgresql has to be configured to listen on all interfaces. See *listen\_addresses* in file /etc/postgresql/9.4/main/postgresql.conf.
- Connection from the XiVO CC for user asterisk must be authorized. See file /etc/postgresql/9.1/main/pg\_hba.conf which must contain a line:

```
host asterisk all 192.168.0.2/32 md5
```

- A user *stats* must exists. Use command \dg in psql.

### Check AMI configuration

- A *xuc* user must be configured in the file /etc/asterisk/manager.d/02-xivocc.conf
- The command:

```
asterisk -rx "manager show user xuc"
```

must show the user.

**CEL Configuration** The correct events must be activated in the file `/etc/asterisk/cel.conf`:

```
[general]
enable = yes
apps = dial,park,queue
events = APP_START,CHAN_START,CHAN_END,ANSWER,HANGUP,BRIDGE_ENTER,BRIDGE_EXIT,USER_DEFINED,LINKED

[manager]
enabled = yes
```

**Check CTI configuration** In *Services* → *IPBX* → *Users* a user the must be created with the following parameters:

- CTI login : xuc
- CTI password : 0000
- Profile supervisor

**Check WS configuration** In *Configuration* → *Web Services Access* a user must be created with the following parameters :

- Login : xivows
- Password : xivows
- Host : 192.168.0.2

**Check ACD configuration** In *Services* → *Ipbx* → *Advanced configuration* make sure `Multiqueues` `call stats` `sharing` is checked.

**Check the phone integration** Verify that the phone configuration where customized as detailed in [Required configuration for phone integration](#).

**Check the recording** The package `xivo-recording` must be installed.

If you want to use call recording filtering, the package `call-recording-filtering` must be installed too.

## Xuc et Xuc\_mgt - applications web ccmanager, agent et assistant

### Basic checks

**XUC overview page** XUC overview page available at `@XUC_IP:PORT`, usually `@SERVER_IP:8090`. You have to check if the “Internal configuration cache database” contains agents, queues etc.

**XUC sample page** XUC sample page available at `@XUC_IP:PORT/sample`, usually `@SERVER_IP:8090/sample`. You can use this page to check user login and other API functions. CCManager, agent and assistant web use functions available on the sample page.

## Desktop Assistant

**Running with debug enabled** If needed, Desktop Assistant can be started with `-d` option to enable debug menu.

## Application Configuration (xuc\_rigths)

## Recording

## SpagoBI

## Kibana

## NGINX - proxy web

**Basic check** On the standard HTTP port of the machine (80) you have the fingerboard page.

## Docker says nginx is restarting

- Check logs for missing files or links, nginx refuses to start if one of servers is not accessible, e.g. xuc is down.

## XiVO Centralized Interface

The XiVO Centralized Interface (XCI) allows to manage several XiVO servers through a unique web interface. Thanks to this interface, it becomes possible to quickly add users that are automatically routed across servers. This documentation will describe the installation process of the interface, how to use the web interface and the REST API it exposes.

## Installation

**Requirements** The XiVO Centralized Interface (XCI) requires :

- A Linux server with PostgreSQL, Docker and Docker-Compose installed
- Some XiVOs to manage !

**Automated installation** An installation script is provided to execute all the installations tasks. To run it, execute the following command :

```
curl https://gitlab.com/xivo-utils/icdu-packaging/raw/master/install-icdu.sh | sudo bash
```

It will ask you a passphrase for generating an SSH key.

The configuration files are located in `/etc/docker`.

**Run the application** Optionally, you can set a bash alias for conveniently run XCI :

```
alias dcomp='docker-compose -p icdu -f /etc/docker/compose/icdu.yml'
```

Then simply :

```
dcomp up -d
```

XCI should now be accessible through <http://my-server-ip:9001>

**Manual installation** The configuration files and the Docker-Compose files are available in a specific [Git repository](#).

**Database setup** XCI stores some data in a PostgreSQL database. By default, `application.conf` is configured to connect to a local database named `icx` with the username `icx` and password `icx`. You can change these parameters if you wish. We will use the default parameters in this documentation.

First, we need to install PostgreSQL extensions to use UUID functions :

```
sudo apt-get install postgresql-contrib
```

We can now create the user and the database associated :

```
sudo -u postgres psql -c "CREATE USER icx WITH PASSWORD 'icx'"
```

```
sudo -u postgres psql -c "CREATE DATABASE icx WITH OWNER icx"
```

We then have to enable UUID extension on the `icx` database. Connect as `root` on the `icx` database :

```
sudo -u postgres psql icx -c 'CREATE EXTENSION IF NOT EXISTS "uuid-osspl";'
```

**I can't connect to PostgreSQL** It is possible that PostgreSQL complains when you're trying to connect. The solution is to modify the `pg_hba.conf` (in Debian, located in `/etc/postgresql/X.X/main`) and add the following line at the end :

```
local all all trust
```

**Generate SSH key** In order to let XCI communicate with the various XiVOs, an SSH key is used. Generate one using the following command :

```
ssh-keygen -t rsa -f /etc/docker/interface-centralisee/ssh_key
```

## Web interface

The XiVO Centralized Interface (XCI) is managed through a web interface. In the following sections, we will highlight the main features of the system.

**Definitions** XCI uses a few concepts that are important to understand in order to use the interface correctly.

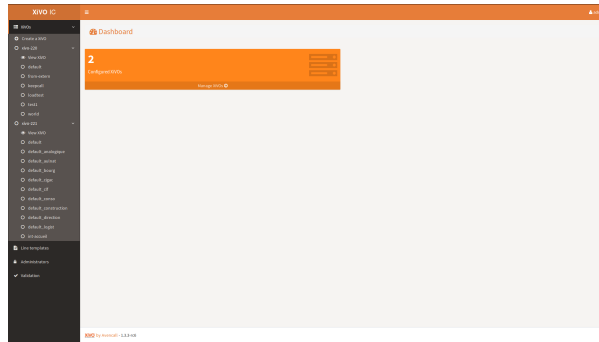
**XiVO** The XiVOs servers that are managed by XCI. XCI will automatically retrieve the entities and the users from them and apply the configuration to them.

**Entity** Entities, also called Contexts, are the parts of the dialplan. Users are attached to them.

**Line template** Line templates are used to quickly create users : they define a few default options (ringing time, voice mail, etc.) that will be applied to the new user. **A line template is required to create a user.**

**User** Actual users that are associated with a phone number

**Administrators** Users that are able to connect to the XCI and manage the XiVOs.

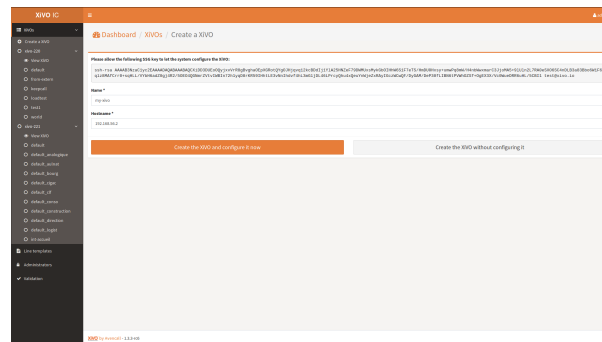


## Dashboard

The dashboard provides you some insights about your XiVO systems.

The left sidebar, displayed in every page of the application, gives you access to the various actions you can perform. The list of the configured XiVOs and their entities is shown to give a quick access to the one you want to manage.

## XiVO



## Create XiVO

This page allows you to add a new XiVO that will be managed by XCI. The first step is to add the displayed SSH key to the authorized keys of your XiVO server. This will allow XCI to connect and configure the XiVO server. You could do this kind of command :

```
echo 'ssh-rsa TheVeryLongSSHKeyYouCopied toto@someServer' | ssh root@xivoIp 'cat >> .ssh/authorized_keys'
```

Then, you have to provide the following informations :

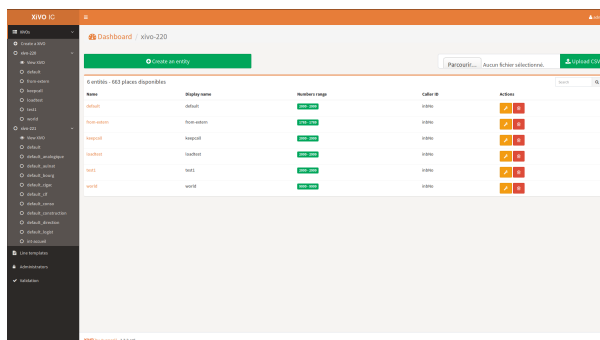
- **Name** : name of the XiVO server that will be displayed in XCI
- **Hostname** : hostname or IP address of the XiVO server

You then have two options :

- **Create the XiVO and configure it now** : XCI will save the informations, try to connect to the XiVO server and perform the configuration. XiVO services will be unavailable during the operation.

**Warning:** The configuration takes a while. Relax, go drink a coffee, XCI is doing the legwork for you :)

- **Create the XiVO without configuring it** : XCI will only save the informations.

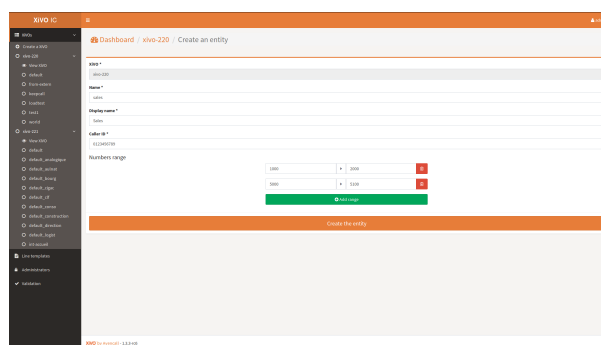


## View XiVO

On the sidebar, each XiVO has its own **View XiVO** link. This page allows you to :

- **Add a new entity** to this XiVO by clicking on the green button
- **See the entities associated to this XiVO and perform some operations to them :**
  - **Edit one** by clicking on the yellow button with the wrench icon
  - **Delete one** by clicking on the red button with the trash icon

## Entity

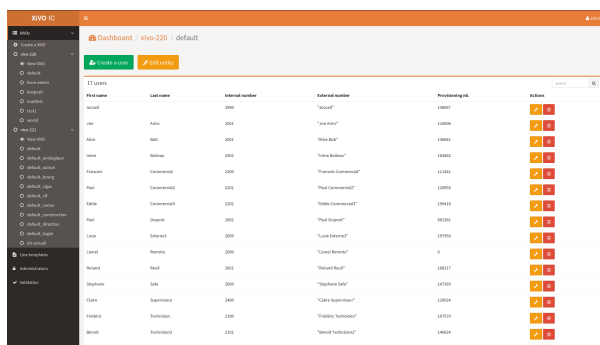


## Create entity

This page allows you to add a new entity to a XiVO. You have to provide the following informations :

- **Name** : name that will be used by the XiVO server
- **Display name** : name that will be displayed on XCI
- **Caller ID** : phone number that will be displayed on outgoing call from this entity
- **Intervals** : ranges of phone numbers that will be available to this entity. For each one, provide :
  - **Start**
  - **End**

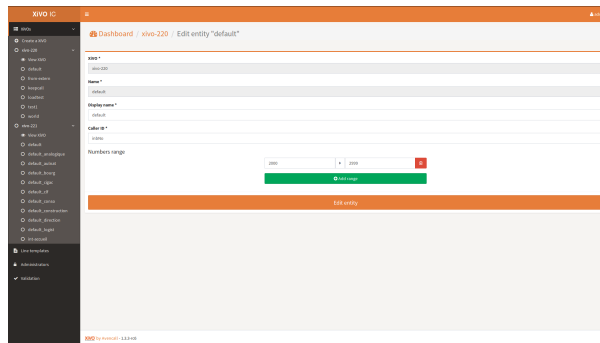
*The system will return an error if the intervals overlap with other entities*



## View entity

On the sidebar, each entity has its own link. This page allows you to :

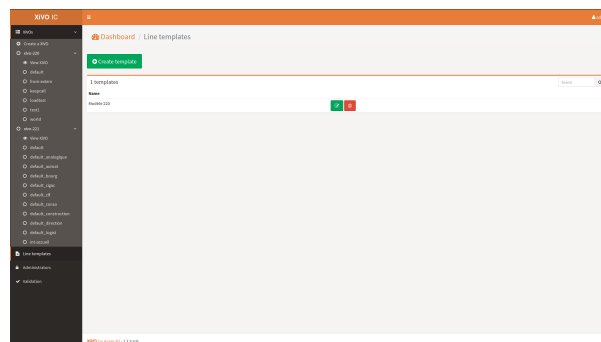
- **Add a new user** to this entity by clicking on the green button
- **Edit the entity** by clicking on the yellow button with the wrench icon
- **See the users associated to this entity and perform some operations to them :**
  - **Edit one** by clicking on the yellow button with the wrench icon
  - **Delete one** by clicking on the red button with the trash icon. *At first click, the icon turns into a question mark. You have 5 seconds to click again to launch user deletion. This process prevents you from accidentally delete users.*



## Edit entity

This page allow you to modify an entity. Please refer to the [Create entity](#) section for fields details.

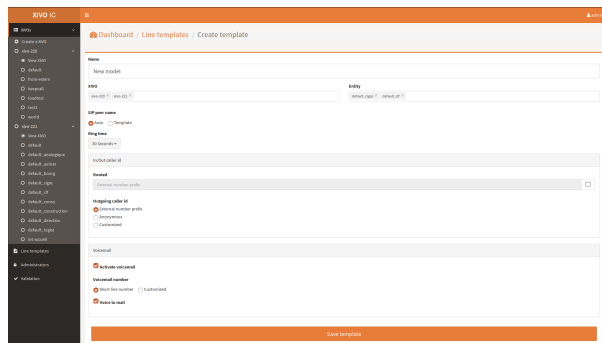
## Line templates



## List templates

On the sidebar, **Line template** has its own link. This page allows you to :

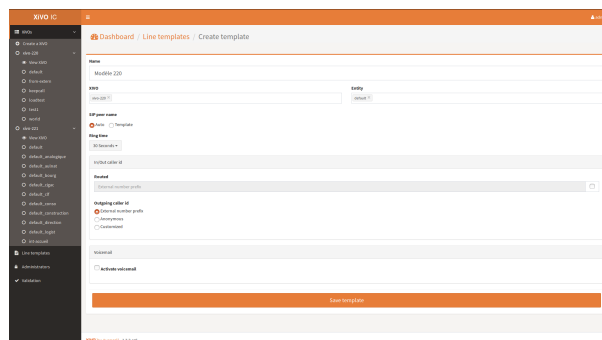
- **Add a new line template** by clicking on the green button
- **See all the line templates and perform some operations to them :**
  - **Edit one** by clicking on the yellow button with the wrench icon
  - **Delete one** by clicking on the red button with the trash icon



## Create template

This page allows you to add a new line template. You have to provide the following informations :

- **Name** : name that will be displayed on XCI
- **XiVO** : select the XiVOs for which this template will be available
- **Entity** : select the entities for which this template will be available. *Only entities of the selected XiVOs are displayed*
- **SIP peer name** : *Auto* or *Model*
- **Ring time** : number of seconds before incoming call is rejected
- **Routed** :
  - The text field allows you to provide the SDA prefix to call the phone
  - Uncheck the checkbox if you don't want the phone to be called from the outside
- **Outgoing caller id** : specify what number is displayed on outgoing call. Possible values are :
  - External number prefix
  - Anonymous
  - Customized : a text field appears to provide the custom number
- **Voicemail** :
  - **Activate voicemail** : enable or not the voicemail
  - **Voicemail number** : specify what number is used to call the voice mail. Possible values are :
    - \* **Short line number** : use the default short number
    - \* **Customized** : a text field appear to provide the custom number
  - **Voice to mail** : whether or not to send an email when a new message is left

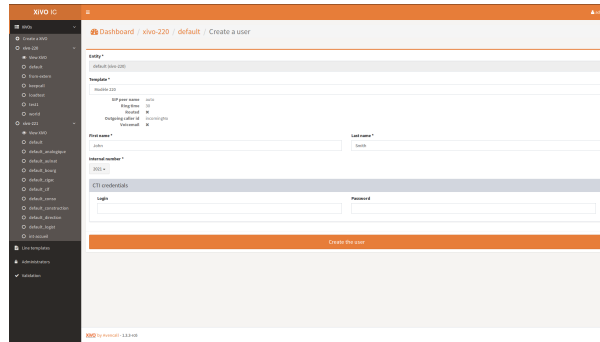


## Edit template

This page allows you to modify a template. Please refer to the [Create template](#) section for fields details.

## User

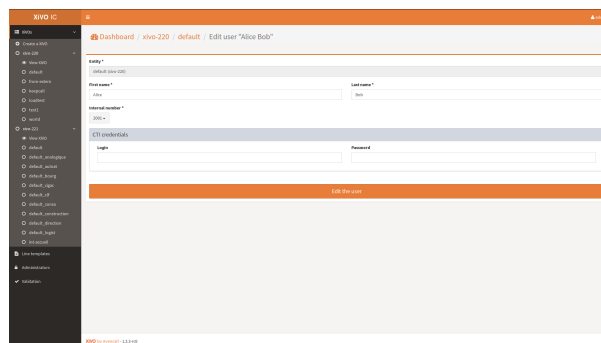




## Create user

This page allows you to add a new user to an entity. You have to provide the following informations :

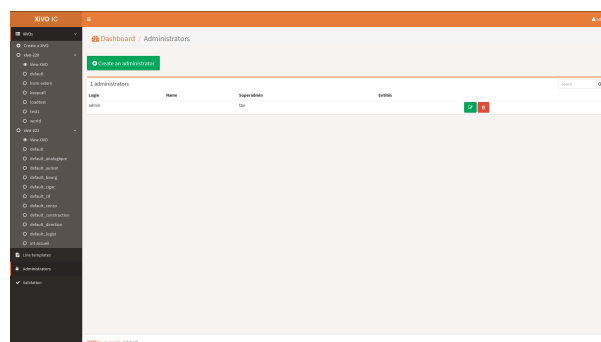
- **Template** : line template to use as a template to create the user. *The main options of the template are displayed below*
- **First name**
- **Last name**
- **Internal number** : number that will be used to internally call the user. *Only the available numbers are displayed*
- **CTI credentials** : provide a login and a password to allow the user to connect through CTI interfaces



## Edit user

This page allows you to modify a user. Please refer to the [Create user](#) section for fields details.

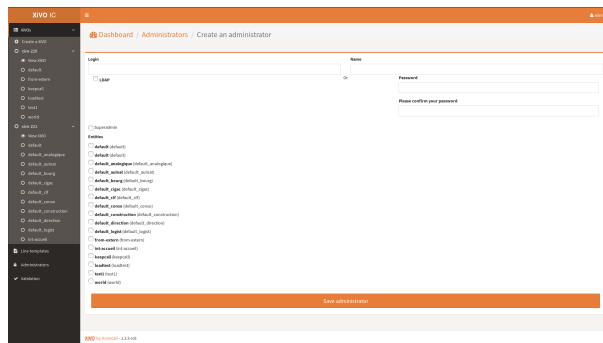
## Administrators



## List administrators

On the sidebar, **Administrators** has its own link. This page allows you to :

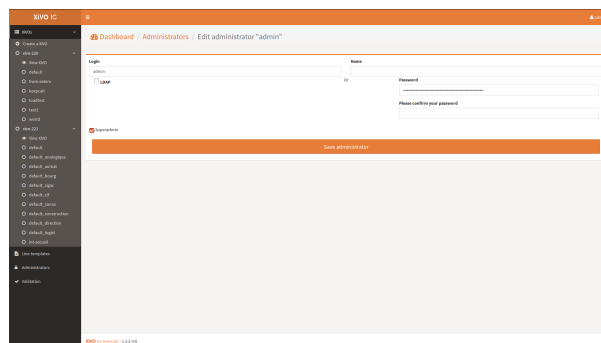
- **Add a new administrator** by clicking on the green button
- **See all the administrators and perform some operations to them** :
  - **Edit one** by clicking on the yellow button with the wrench icon
  - **Delete one** by clicking on the red button with the trash icon



## Create administrator

This page allows you to add a new administrator. You have to provide the following informations :

- **Login** : login used by the administrator to connect to XCI
- **Name** : name that will be displayed on XCI
- **LDAP** : if checked, the LDAP authentication configured in `application.conf` will be used
- **Password** : password used by the administrator to connect to XCI. *Shown only if LDAP disabled*
- **Superadmin** : whether or not this administrator is a super-administrator. Super-administrators can manage everything in XCI
- **Entities** : select the entities this administrator will be able to manage *Shown only if Superadmin disabled*



## Edit administrator

This page allows you to modify an administrator. Please refer to the [Create administrator](#) section for fields details.

## REST API

The XiVO Centralized Interface (XCI) exposes some REST API that you can use to integrate with your tools.

## General form

http://\$my-server-ip:\$xcipport/api/1.0/\$method

```
withHeaders(("Content-Type", "application/json"))
```

- \$xciport : XCI port number (default 9001)
- \$method : See available methods below

**Login** A login request is required before subsequent API calls in order to get a session cookie.

POST /api/1.0/login

Payload parameters :

**login (String)** Login to connect with

**password (String)** Password corresponding to the login

The server will return a cookie and you will be able to do other API calls. Example with CURL :

```
curl 'http://localhost:9000/api/1.0/login' -H 'Content-Type: application/json' -c 'xci-cookie' --
curl 'http://localhost:9000/api/1.0/xivo' -H 'Content-Type: application/json' -b 'xci-cookie'
```

**XiVO** The following methods allow you to operate on the XiVOs managed by XCI.

**List** List all the XiVOs configured on XCI.

GET /api/1.0/xivo

```
{
 "items": [
 {
 "id": 1,
 "uuid": "8f159082-4b25-48b3-afec-1873491a60be",
 "name": "xivo-220",
 "host": "192.168.29.220",
 "remainingSlots": 664
 },
 {
 "id": 2,
 "uuid": "15585b75-1d75-45b1-8678-520d1210ec59",
 "name": "xivo-221",
 "host": "192.168.29.221",
 "remainingSlots": 280
 }
]
}
```

**Get** Get a XiVO by its id.

GET /api/1.0/xivo/\$id

```
{
 "id": 1,
 "uuid": "8f159082-4b25-48b3-afec-1873491a60be",
 "name": "xivo-220",
 "host": "192.168.29.220",
 "remainingSlots": 664
}
```

**Create** Create a new XiVO.

POST /api/1.0/xivo

Payload parameters :

**name (String)** Display name of the XiVO

**host (String)** Hostname or IP address of the XiVO

**configure (Boolean)** If set to `true`, XCI will immediately make the necessary configurations on the XiVO.  
If set to `false`, it will only be added to XCI but not configured.

**Synchronize configuration files**

GET /api/1.0/xivo/synchronize\_config\_files

**Entities** The following methods allow you to operate on the entities made available by the XiVOS.

**List** List all the entities available.

GET /api/1.0/entities

```
{
 "items": [
 {
 "id": 17,
 "combinedId": "default@15585b75-1d75-45b1-8678-520d1210ec59",
 "name": "default",
 "displayName": "default",
 "xivo": {
 "id": 2,
 "uuid": "15585b75-1d75-45b1-8678-520d1210ec59",
 "name": "xivo-221",
 "host": "192.168.29.221",
 "remainingSlots": 280
 },
 "intervals": [
 {
 "start": "1700",
 "end": "1799"
 },
 {
 "start": "1961",
 "end": ""
 },
 {
 "start": "2600",
 "end": "2799"
 }
],
 "presentedNumber": "inbNo"
 },
 {
 "id": 22,
 "combinedId": "default_analogique@15585b75-1d75-45b1-8678-520d1210ec59",
 "name": "default_analogique",
 "displayName": "default_analogique",
 "xivo": {
 "id": 2,
 "uuid": "15585b75-1d75-45b1-8678-520d1210ec59",
 "name": "xivo-221",
 "host": "192.168.29.221",
 "remainingSlots": 280
 },
 "intervals": [
 {
 "start": "3990000",
 "end": "3999999"
 },
 {
 "start": "39990000",
 "end": "39999999"
 }
],
 "presentedNumber": "inbNo"
 }
]
}
```

**Get** Get an entity by its combinedId.

GET /api/1.0/entities/\$combinedId

```
{
 "id": 22,
 "combinedId": "default_analogique@15585b75-1d75-45b1-8678-520d1210ec59",
 "name": "default_analogique",
 "displayName": "default_analogique",
 "xivo": {
 "id": 2,
 "uuid": "15585b75-1d75-45b1-8678-520d1210ec59",
 "name": "xivo-221",
 "host": "192.168.29.221",
 "remainingSlots": 280
 },
 "intervals": [
 {
 "start": "3990000",
 "end": "3999999"
 },
 {
 "start": "39990000",
 "end": "39999999"
 }
],
 "presentedNumber": "inbNo"
}
```

**Create** Create a new entity.

POST /api/1.0/entities

Payload parameters :

**name (String)** Name of the entity

**displayName (String)** Displayed name of the entity

**xivoId (Integer)** Id of the XiVO the entity will be attached to

**intervals (Array)** Intervals of numbers this entity will support

**start (String)** Starting number of the interval

**end (String)** Ending number of the interval

**presentedNumber (String)** Number to show on outgoing calls

**Delete** Delete an entity.

DELETE /api/1.0/entities/\$combinedId

**Edit** Edit an entity. See [Create entity](#) for fields details.

PUT /api/1.0/entities/\$combinedId

**List users** List users attached to an entity.

GET /api/1.0/entities/\$combinedId/users

```
{
 "items": [
 {
 "id": 559,
 "entity": {
```

```

 "id": 22,
 "combinedId": "default_analogique@15585b75-1d75-45b1-8678-520d1210ec59",
 "name": "default_analogique",
 "displayName": "default_analogique",
 "xivo": {
 "id": 2,
 "uuid": "15585b75-1d75-45b1-8678-520d1210ec59",
 "name": "xivo-221",
 "host": "192.168.29.221",
 "remainingSlots": 280
 },
 "intervals": [
 {
 "start": "3990000",
 "end": "3999999"
 },
 {
 "start": "39990000",
 "end": "39999999"
 }
],
 "presentedNumber": "inbNo"
 },
 "firstName": "Sous sol Logistique",
 "lastName": "CLF 88:40 P3",
 "internalNumber": "6260",
 "externalNumber": "\"Sous sol Logistique CLF 88:40 P3\"",
 "mail": null,
 "ctiLogin": null,
 "ctiPassword": null,
 "provisioningNumber": "114133"
}
]
}

```

**List available numbers** List available numbers for an entity

GET /api/1.0/entities/\$combinedId/available\_numbers

```

{
 "items": [
 "3990000",
 "3990001",
 "3990002",
 "3990003",
 "3990004"
]
}

```

**Users** The following methods allow you to operate on the users made available by the XiVOS.

**Get** Get a user by its id.

GET /api/1.0/users/\$id

```

{
 "id": 559,
 "entity": {
 "id": 22,
 "combinedId": "default_analogique@15585b75-1d75-45b1-8678-520d1210ec59",
 "name": "default_analogique",

```

```

"displayName": "default_analogique",
"xivo": {
 "id": 2,
 "uuid": "15585b75-1d75-45b1-8678-520d1210ec59",
 "name": "xivo-221",
 "host": "192.168.29.221",
 "remainingSlots": 280
},
"intervals": [
 {
 "start": "3990000",
 "end": "3999999"
 },
 {
 "start": "39990000",
 "end": "39999999"
 }
],
"presentedNumber": "inbNo"
},
"firstName": "Sous sol Logistique",
"lastName": "CLF 88:40 P3",
"internalNumber": "6260",
"externalNumber": null,
"mail": null,
"ctiLogin": null,
"ctiPassword": null,
"provisioningNumber": "114133"
}

```

**Create** Create a new user.

POST /api/1.0/users

Payload parameters :

**entityCid (String)** Entity combinedId the user will be attached to

**templateId (Integer)** Line template to apply to the user

**firstName (String)** First name of the user

**lastName (String)** Last name of the user

**internalNumber (String)** Internal phone number of the user

**ctiLogin (String) *Optional*** CTI login of the user

**ctiPassword (String) *Optional*** CTI password of the user

**Delete** Delete a user.

DELETE /api/1.0/users/\$id

**Edit** Edit a user. See [Create user](#) for fields details.

PUT /api/1.0/users/\$id

**Templates** The following methods allow you to operate on the line templates used to create users.

**List** List all the templates available.

GET /api/1.0/templates

```
[
 {
 "id": 1,
 "name": "Modèle 220",
 "peerSipName": "auto",
 "routedInbound": false,
 "callerIdMode": "incomingNo",
 "ringingTime": 30,
 "voiceMailEnabled": false,
 "voiceMailNumberMode": "short_number",
 "xivos": [
 1
],
 "entities": [
 "default@8f159082-4b25-48b3-afec-1873491a60be"
]
 }
]
```

**Get** Get a template by its id.

GET /api/1.0/templates/\$id

```
{
 "id": 1,
 "name": "Modèle 220",
 "peerSipName": "auto",
 "routedInbound": false,
 "callerIdMode": "incomingNo",
 "ringingTime": 30,
 "voiceMailEnabled": false,
 "voiceMailNumberMode": "short_number",
 "xivos": [
 1
],
 "entities": [
 "default@8f159082-4b25-48b3-afec-1873491a60be"
]
}
```

**Create** Create a new template.

POST /api/1.0/templates

Payload parameters :

**name (String)** Name of the template

**xivos (Array of Integer)** List of XiVOs ids the template will be available to

**entities (Array of String)** List of entities combinedIds the template will be available to

**peerSipName (String)** Possible values are `auto` or `model`

**ringingTime (Integer)** Number of seconds before incoming call is rejected

**routedInbound (Boolean)** Whether or not the phone can be called from the outside

**routedInboundPrefix (String)** *Compulsory if routedInbound is true* SDA prefix to call the phone

**callerIdMode (String)**



**Option specifying what number is displayed on outgoing call. Possible values are :**

- `incomingNo` : use the SDA prefix
- `anonymous` : masked call
- `custom` : a custom number

**customCallerId (String)** *Compulsory if callerIdMode is custom* Custom number to display on outgoing call

**voiceMailEnabled (Boolean)** Whether or not to enable the voice mail

**voiceMailNumberMode (Boolean)**

**Option specifying what number is used to call the voice mail. Possible values are :**

- `short_number` : use the default short number
- `custom` : a custom number

**voiceMailCustomNumber (String)** *Compulsory if voiceMailNumberMode is custom* Custom number to call the voice mail

**voiceMailSendEmail (Boolean)** Whether or not to send an email when a new message is left

**Delete** Delete a template.

DELETE /api/1.0/templates/\$id

**Edit** Edit a template. See [Create template](#) for fields details.

PUT /api/1.0/templates/\$id

**Administrators** The following methods allow you to operate on the administrators of the XCI.

**List** List all the administrators present.

GET /api/1.0/administrators

```
{
 "items": [
 {
 "id": 1,
 "login": "admin",
 "name": "",
 "password": "+\\\\/rIncoyp\\Ai\\813xSEeSY+P+x4uNle7cHkL6rpPS3ucgr2EAJIqnQbsIpSGwHj",
 "superAdmin": true,
 "ldap": false,
 "entities": [
]
 }
]
}
```

**Get** Get an administrator by its id.

GET /api/1.0/administrators/\$id

```
{
 "id": 1,
 "login": "admin",
 "name": "",
```

```
"password": "+\\\\"/rIncoyp\\Ai\\"/8l3xSEeSY+P+x4uNle7cHkL6rpPS3ucgr2EAJIqnQbsIpSGwHj",
"superAdmin": true,
"ldap": false,
"entities": [

]
}
```

**Create** Create a new administrator.

POST /api/1.0/administrators

Payload parameters :

**login (String)** Login of the administrator

**name (String)** Displayed name of the administrator

**ldap (Boolean)** Whether or not to use the LDAP authentication configured in `application.conf`

**password (String)** *Compulsory if ldap is false* Password used by the administrator to login

**superAdmin (Boolean)** Whether or not this administrator is a super-administrator. Super-administrators can manage everything in XCI.

**entityIds (Array of Integer)** List of entities this administrator has the rights to manage

**Delete** Delete an administrator.

DELETE /api/1.0/administrators/\$id

**Edit** Edit an administrator. See [Create administrator](#) for fields details.

PUT /api/1.0/administrators/\$id

### Example (Python 3)

```
#!/usr/bin/env python3
-*- coding: utf-8 -*-

from urllib.parse import urlencode
from urllib.request import Request, urlopen
import json, sys

class XCIApiExample:
 base_url = None
 cookie = None

 def __init__(self, base_url, login, password):
 self.base_url = base_url
 self.make_login(login, password)

 def make_login(self, login, password):
 data = {"login": login, "password": password}
 response = self.make_post_request("/login", data)
 self.cookie = response.info()["Set-Cookie"]

 def get_entities(self):
 response = self.make_get_request("/entities")
 return self.handle_json_response(response)

 def get_available_numbers(self, entity):
 response = self.make_get_request("/entities/" + entity["combinedId"] + "/available"
```

```

 return self.handle_json_response(response)

 def create_line_template(self, data):
 self.make_post_request("/templates", data)

 def get_line_templates(self):
 response = self.make_get_request("/templates")
 return self.handle_json_response(response)

 def create_user(self, data):
 self.make_post_request("/users", data)

 def make_get_request(self, method):
 request = Request(self.base_url + method, headers = {"Cookie": self.cookie})
 response = urlopen(request)
 return response

 def make_post_request(self, method, data):
 header = {"Content-Type": "application/json", "Cookie": self.cookie}
 request = Request(self.base_url + method, json.dumps(data).encode(), header)
 response = urlopen(request)
 return response

 def handle_json_response(self, response):
 return json.loads(response.read().decode())

Initialize API
api_example = XCIApiExample("http://192.168.29.103:9001/api/1.0", "admin", "superpass")

Get an entity and its XiVO
entities = api_example.get_entities()["items"]
if (len(entities) == 0):
 sys.exit("There isn't any XiVO configured yet or they don't have any entity")
else:
 entity = entities[1]
 xivo = entity["xivo"]
 print("Selected entity \"%s\" in XiVO \"%s\"" % (entity["name"], xivo["name"]))

Create a line template
template_data = {
 "name": "My line template",
 "xivos": [xivo["id"]],
 "entities": [entity["combinedId"]],
 "peerSipName": "auto",
 "ringingTime": 30,
 "routedInbound": False,
 "callerIdMode": "anonymous",
 "voiceMailEnabled": False
}
api_example.create_line_template(template_data)
line_template = api_example.get_line_templates()[0]
print("New line template created")

Create a user
user_data = {
 "entityCId": entity["combinedId"],
 "templateId": line_template["id"],
 "firstName": "Alice",
 "lastName": "In Wonderland",
 "internalNumber": api_example.get_available_numbers(entity["items"][0])
}
api_example.create_user(user_data)

```

```
print("New user created")
```

## 1.2.2 Indices and tables

- [genindex](#)
- [search](#)

---

## Indices and tables

---

- `genindex`
- `search`



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