IvozProvider 1.5 Documentation

Release Artemis

Irontec

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Basic Concepts

CHAPTER 1

Introduction to IvozProvider

The following sections will serve as general introduction to IvozProvider:

1.1 About this document

This document describes the process of installation and usage of IvozProvider, the multi-tenant telephony platform for providers developed by Irontec.

This should be the starting point for anyone interested in this solution, both from the technical point of view and the user one and it's divided in four blocks:

- The first block is about *Basic Concepts* where each element of the product and its main function is decscribed.
- The second block describes the *Getting Started* process with a fresh installed platform, leaving the deep configuration details for the next block.
- The third block goes deeper into the most *Administation portal* features like **tarification, billing, PBX advanced options and every call details** that were omitted in the previous block.
- The fourth and last block describes the *Security and Maintenance* measures that implements the solution.

1.2 Getting help

IvozProvider is an alive and highly developed project. There are multiple channels to get information or report bugs:

- GitHub: https://github.com/irontec/ivozprovider
- email: vozip@irontec.com
- Twitter: @irontec
- IRC Channel #ivozprovider at irc.freenode.net

Don't hesitate to contact us for any kind of feedback :)

1.3 What is IvozProvider?

IvozProvider is a *provider oriented multilevel IP telephony* solution *exposed to the public network*.

1.3.1 IP Telephony

IvozProvider supports telephony systems that use *Session Initialitation Protocol*, **SIP**, described in RFC 3261 and any related RFCs independent of manufacturers.

This allows total freedom to choose *softphones*, *hardphones* and the rest of elements that interact with IvozProvider, without any kind of binding with a manufacturer.

Right now, IvozProvider supports the following **transport protocols** for SIP:

- UDP
- TCP
- TLS
- Websockets

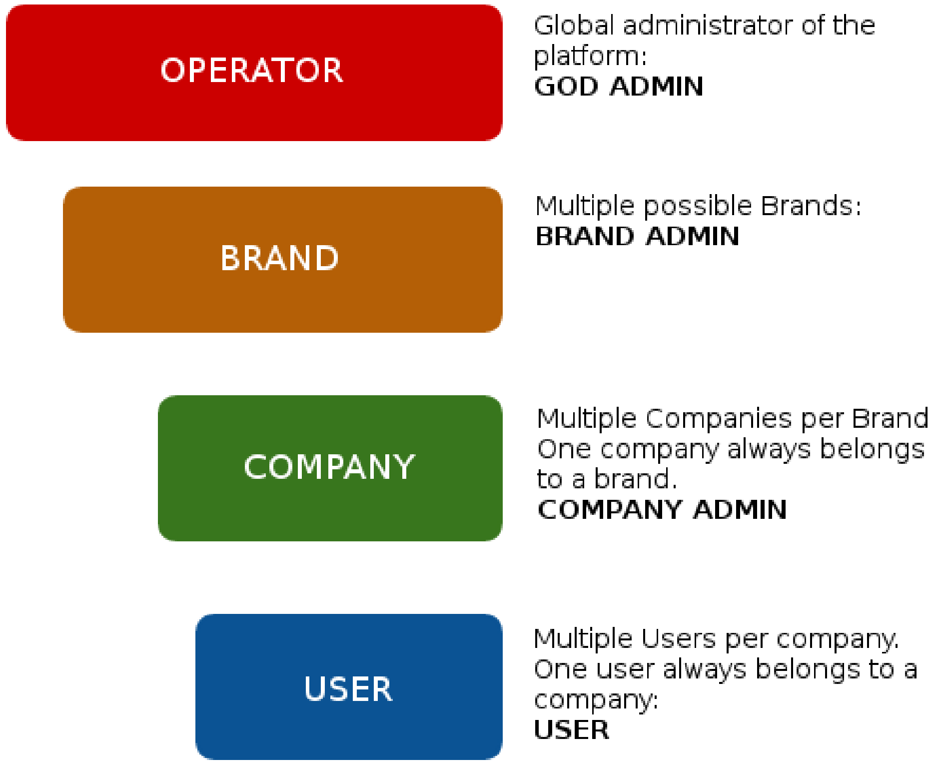
This last transport protocol described in RFC 7118 supports web intregrated softphones, using the WebRTC standard allowing browsers to establish real-time *peer-to-peer* connections.

The **supported audio codec** list is:

- PCMA (*alaw*)
- PCMU (*ulaw*)
- GSM
- SpeeX
- G.722
- G.726
- G.729
- iLBC
- OPUS

1.3.2 Multilevel

The web portal design of IvozProvider allows **multiple actors within the same infrastructure**:



In *Platform roles* section, the different roles are deeply described, but to sum up:

- **God Admin**: The administrator and maintainer of the solution. Provides access to multiple brand operators.
- **Brand Operator**: Responsible of giving access, tarificate and bill to multiple company operators.
- **Company Operator**: Responsible of its own PBX configuration and to manage the final platform users.
- **Users**: The last link of the chain, has SIP credentials and can access its own portal for custom configurations.

Each one of this roles **has its own portal** that allows them to fullfill their tasks. Each portal can be customized in the following ways:

- Themes and *skins* for corporative colours.
- Company Logos.
- Customized URLs with the Brand or Company domain.

1.3.3 Provider oriented

IvozProvider is a telephony solution **designed with horizontal scaling in mind**, what allows handling a great amount of **traffic and users** only by increasing the machines and resources of them.

This are the main ideas that makes this product provider oriented:

- Despite the fact that all machine profiles can run in the same host, whatmakes it easier for the initial testing, each profile of IvozProvider can be splitted from the rest to make it run in its own machine.
- A **distributed installation** allows to distribute the correct amount of resources to each task, but also:
  - Geographic distribution of elements to warranty high availability in caseof CPD failure.

  - Setup of key elements near the final users, to minimize the communication latencies.
  - Horizontal scaling of key profiles to handle hundred of thousands concurrent calls.

The resource consuming elements that limit the service of VoIP solutions use to be:

- Already established calls audio management.
- Managing configuration for each company administrator (IVRs, conference rooms, external call filters, etc.)
- Databases of configuration and records.

IvozProvider was designed always keeping in mind the **horizontal scaling** of each of its elements, so it **can handle hundred of thousands concurrent calls** and what is more important, **adapt the platform resources to the expected service quality**:

- **Media-relay** servers handle audio frames for the already established calls:
  - You can use as many media-relays as you need.
  - You can join media-relay in groups, and force some companies to use a group if you want.
  - You can setup media-relays near the final users, to minimize network latencies in the calls.
- **Application servers** are in charge of processing the configurad logics:
  - They scale horizontally: new Application Serves can be installed and added to the pool if you feel the need.
  - Every call is handled by the least busy Appliction Server
  - By default, there is no static assigment * between Companies and Application Servers. This way failure of any Application Server is not critical: the platform will ignore the faulty Application Server while distributing calls.

1.3.4 Exposed to the public network

As showed in the installation proces, **IvozProvider is designed to serve users directly from Internet**. Although it can be used in local enviroments, IvozProvider is designed to use public IP addresses for its services, removing the need of VPN or IPSec tunnels that connect the infrastructure with the final users

Highlights:

- Only the required services will be exposed to Internet.
- The unstrusted origins access can be filtered out by integrated firewall
- Access from IP addresses or networks can be filtered to avoid any kind of phishing.
- There is also an anti-flood mechanism to avoid short-life Denial of Service attacks.
- Each company concurrent calls can be limited to a fixed amount.
- IvozProvider supports connection from terminals behind NAT.
- IvozProvider keep track of those NAT windows and keep them alive with *nat-piercing* mechanisms.

1.4 What is inside IvozProvider?

IvozProvider uses well-known and stable Free Software projects to fullfill the different required task of the platform.

Nothing better than an image to show all the software that its integrated into IvozProvider:

Note: We can not stress enough our gratitude to the developers and communities of this projects.

The task of each of this software will be deeply detailed in the block *Platform general architecture*.

1.5 Who should use IvozProvider?

IvozProvider is a good option for those insterested in having a telephony platform that can provide service to **thousands concurrent calls**.

The greatest strengths of IvozProvide can help to decide if the solution feeds your needs:

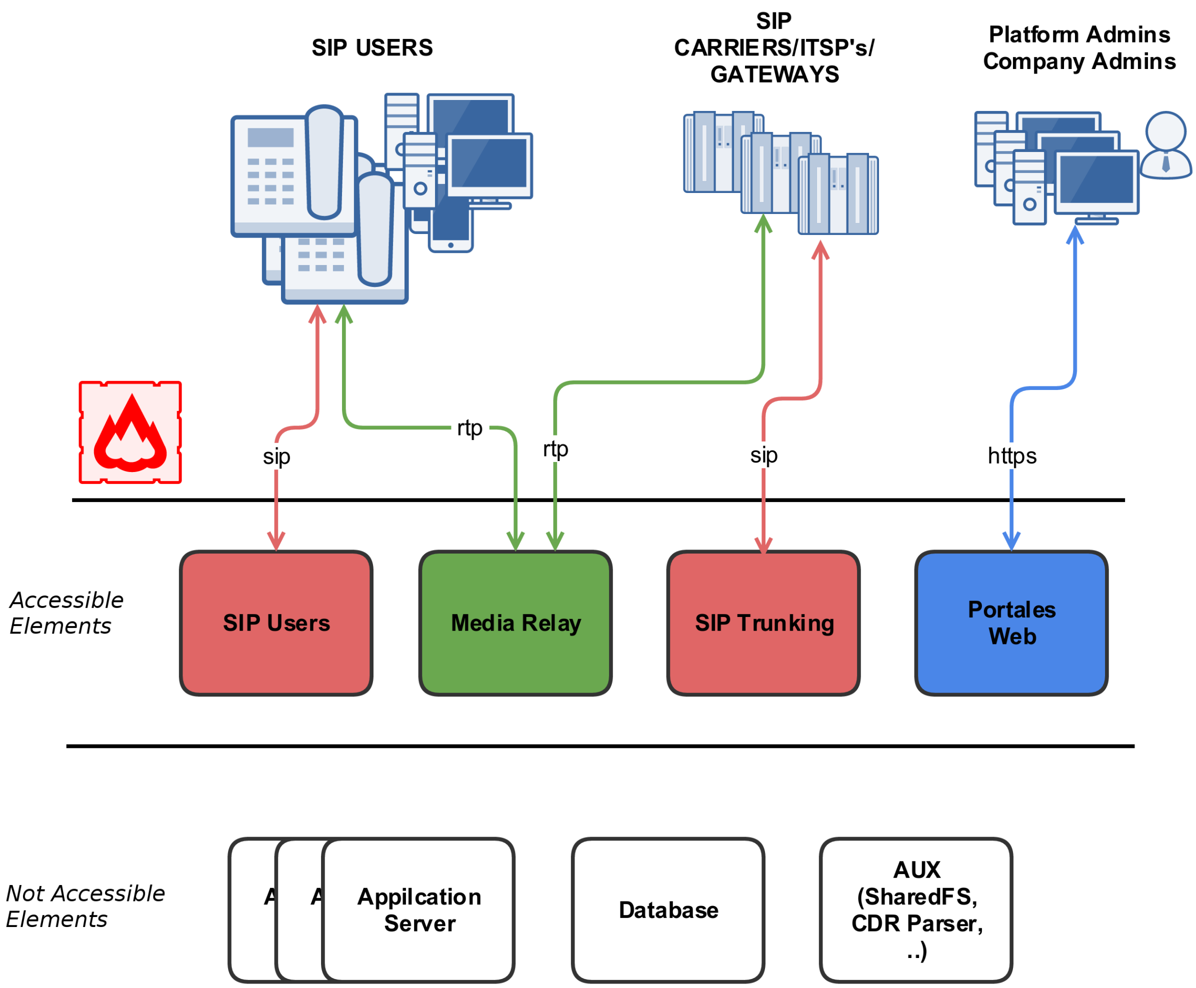
- VoIP: SIP
- Multilevel, multitenant
- Horizontal scaling
- PseudoSBC: open to Internet
- PBX Features

The installation process is so simple, that the best way to test if IvozProvider fullfills yout needs is to test it!

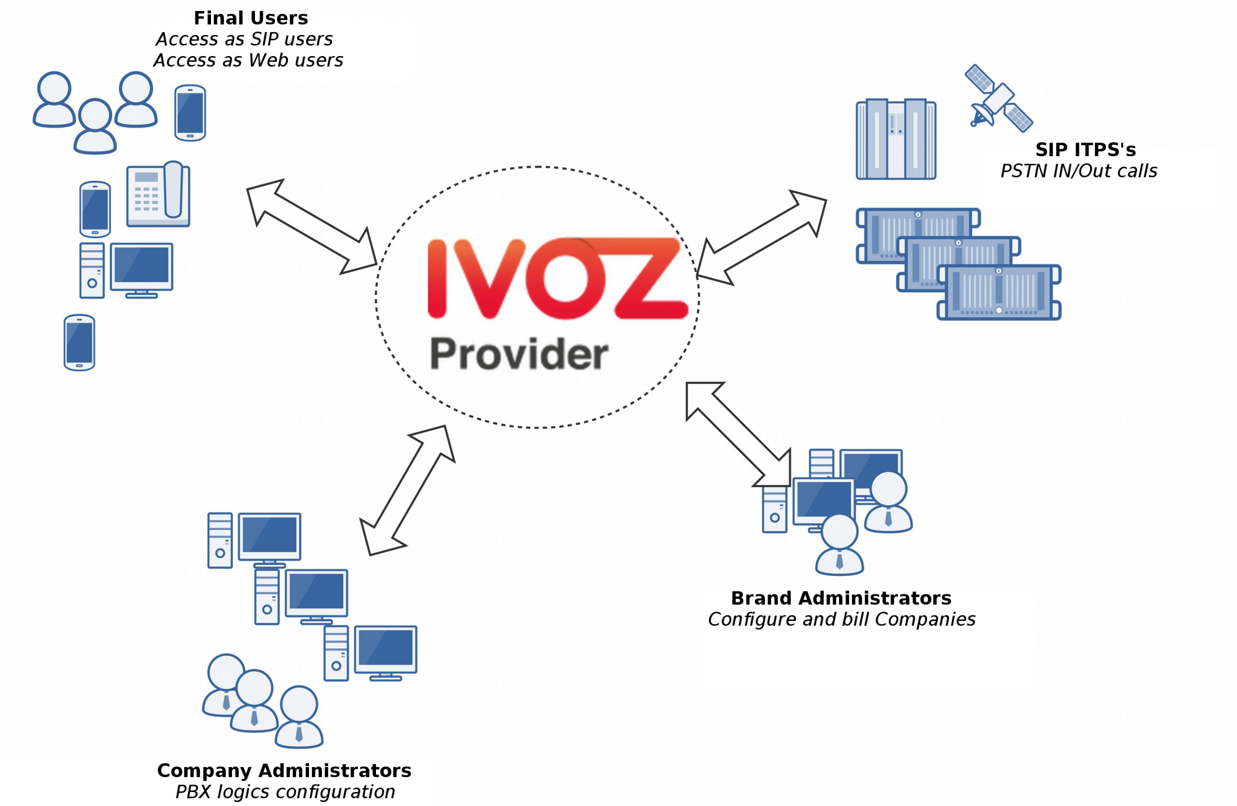
Platform general architecture

2.1 General diagram

Following diagram shows the global architecture of IvozProvider solution, with all its components:



This is a more conceptual diagram:



2.2 SIP signalling flow

The first diagram shows the SIP signalling traffic involved in the establishment, modification and termination of sessions following the SIP RFC 3261 and any related RFCs.

These are the **external SIP entities** involved:

- UACs: users hardphones, softphones, SIP-capable gadget.
- SIP carriers: carriers used to interconnect IvozProvider with external SIP networks (and, probably, with PSTN).

All the SIP traffic (in any of the supported transports: TCP, UDP, TLS, WSS) they send/receive is to/from this two **internal SIP entities** of IvozProvider:

- Users SIP Proxy (running Kamailio).
- Trunks SIP Proxy (running Kamailio).

In fact, users UACs only talks to *Users SIP Proxy* and 'SIP carriers' only talks to *Trunks SIP Proxy*.

Inside IvozProvider these two proxies talk to *Application Servers* running Asterisk, but **no external element is allowed to talk to Application Servers directly**.

2.3 RTP audio flow

Sessions initiated by SIP signalling protocol imply media streams shared by involved entities.

This media streams use RTP to send and receive the media itself, usually using UDP as a transport protocol.

External entities involved in RTP sessions can be divided in:

- Users.
- Carriers.

Both entities exchanges RTP with the same IvozProvider entity: *media-relays*.

IvozProvider implements *media-relays* using RTPengine.

Similar to SIP, these *media-relays* exchanges RTP when is needed with *Application Servers*, but **external entities never talk directly to them**.

2.4 HTTPS traffic

HTTPS is the third traffic type exchanged between IvozProvider and *external world*.

HTTPS traffic is used for:

- **Terminal provisioning**: several hardphones ask for their configuration when they wake up and this configuration files can be served through HTTPS.
- **Web portals**: IvozProvider has 4-level web portals for all the *platform roles*.

  Both of these traffics are handled by *Web portals* IvozProvider entity.

2.5 Additional elements

IvozProvider has multiple elements that are not exposed to the *external world* but play a crucial task.

The most remarkable profile is **database profile** that gathers all the information of the platform and shares it between the majority of software packaged. IvozProvider uses MySQL database engine for this task.

Another remarkable task is **asychronous tasks handler** in charge of encoding recordings, generating invoices, reloading services, importing data, etc.

2.6 Auxiliar elements

Aux profile runs software that, even though is not vital for calls placing, makes IvozProvider mantainer's life much more easier.

In fact, without them, debugging problems would be much harder and the quality of given service would be damaged.

IvozProvider ships:

- **Homer SIP capture**: This amazing software lets us capture all the SIP traffic for later analysis, for obtaining statistics, call quality measuring, etc. Visit SIP Capture website for more information.
- **Graylog log viewer**: All logs of all IvozProvider profiles are stored and shown with Graylog and divied in brands.
- **Grafana graph dashboard**: Grafana lets us graph everything. Literally.

CHAPTER 3

Initial Installation

3.1 Installation Types

3.1.1 Distributed Install

IvozProvider software is designed to run distributed between multiple systems in what we call profiles:

Each profile is in charge of performing one of the platform functionalities:

- Data storage
- SIP Proxy
- Application Server
- Web portal

For each of this profiles, there's a virtual package that will install all the required dependencies (see *Installing profile package*).

You can install as many instances as you want for each profile, but take into account, that while some of them are designed to scale horizontally (for example: asterisk or media-relays) others will require aditional software so the systems that have the same profile are syncronized (for example: database replication or http request balancing).

3.1.2 StandAlone Install

If you want a small installation to make a couple of tests or give a basic service, we have designed all this configuration so they can work in a single machine.

We have called this kind of installations **StandAlone** and we have also created *Automatic ISO CD image* so you can install in a couple of minutes.

3.2 Minimum requirements

3.2.1 System requirements

IvozProvider is designed to be installed using Debian GNU/Linux APT package system.

Important: It's recommended to install IvozProvider in a dedicated server for the platform. Many of the installed software may not work properly with other pre-installed services (like MySQL or DNS servers).

For a StandAlone installation, we recommend at least:

- 4 CPUs (x86_64 or i386)
- 4 Gb memory
- 30GB HDD
- 1/2 public IP Addresses (read note behind)

Note: Since version 1.2 it is possible to make both KamUsers and KamTrunks share a unique public IP address. If so, **KamTrunks ports will be changed from 5060 (TCP/UDP) to 7060 (TCP/UDP) and from 5061 (TCP) to 7061 (TCP)**.

If you're not using a *Automatic ISO CD image* you will also need:

- Debian Stretch 9.0 base install
- Internet access

3.3 Debian packages install

IvozProvider is designed to be installed and updated using Debian packages. More exactly, the current release is ready to be installed on Debian Stretch 9.

It's recommended to use one of the official installation guides to install the minimum base system. The rest of required dependencies will be installed automatically with IvozProvider meta packages.

No matter if you are installing a *StandAlone Install* or a *Distributed Install*, it's required to configure Irontec debian repositories.

3.3.1 APT Repository configuration

Right now, two different repositories are used for the latest IvozProvider release (called oasis) and it's frontend Klear release (called chloe).

```
cd /etc/apt/sources.list.d
echo deb http://packages.irontec.com/debian artemis main extra > ivozprovider.list
echo deb http://packages.irontec.com/debian tayler main > klear.list
```

Optionally, we can add the repository key to check signed packages:

```
wget http://packages.irontec.com/public.key -q -O - | apt-key add -
```

3.3.2 Installing profile package

Once the repositories are configured, it will be required to select the proper metapackage depending on the type of installation.

- **For a *StandAlone Install*:**

    – ivozprovider

- For a *Distributed Install*: one of the profile packages depending on the role the machine will perform.
  - ivozprovider-profile-data
  - ivozprovider-profile-proxy
  - ivozprovider-profile-portal
  - ivozprovider-profile-as

```
apt-get update
apt-get install ivozprovider
```

3.3.3 Finish the installation

Distributed installation require a couple manual configuration based on the roles that are performing. Check finishing role configuration for more information.

Standalone installation have a menu that can be used to configure the basic services used in IvozProvider. Most of the services are automatically configured to work in the same machine with the default values.

This menu allows:

- Configure IP address(es) for SIP proxies
- Default platform language
- Administrator MySQL database password

It's possible to change any of this values anytime by running:

```
dpkg-reconfigure ivozprovider
```

Important: Any of the public IP addresses configured during the installation will work to acces the web portal. Default credentials are **admin / changeme**.

3.4 Automatic ISO CD image

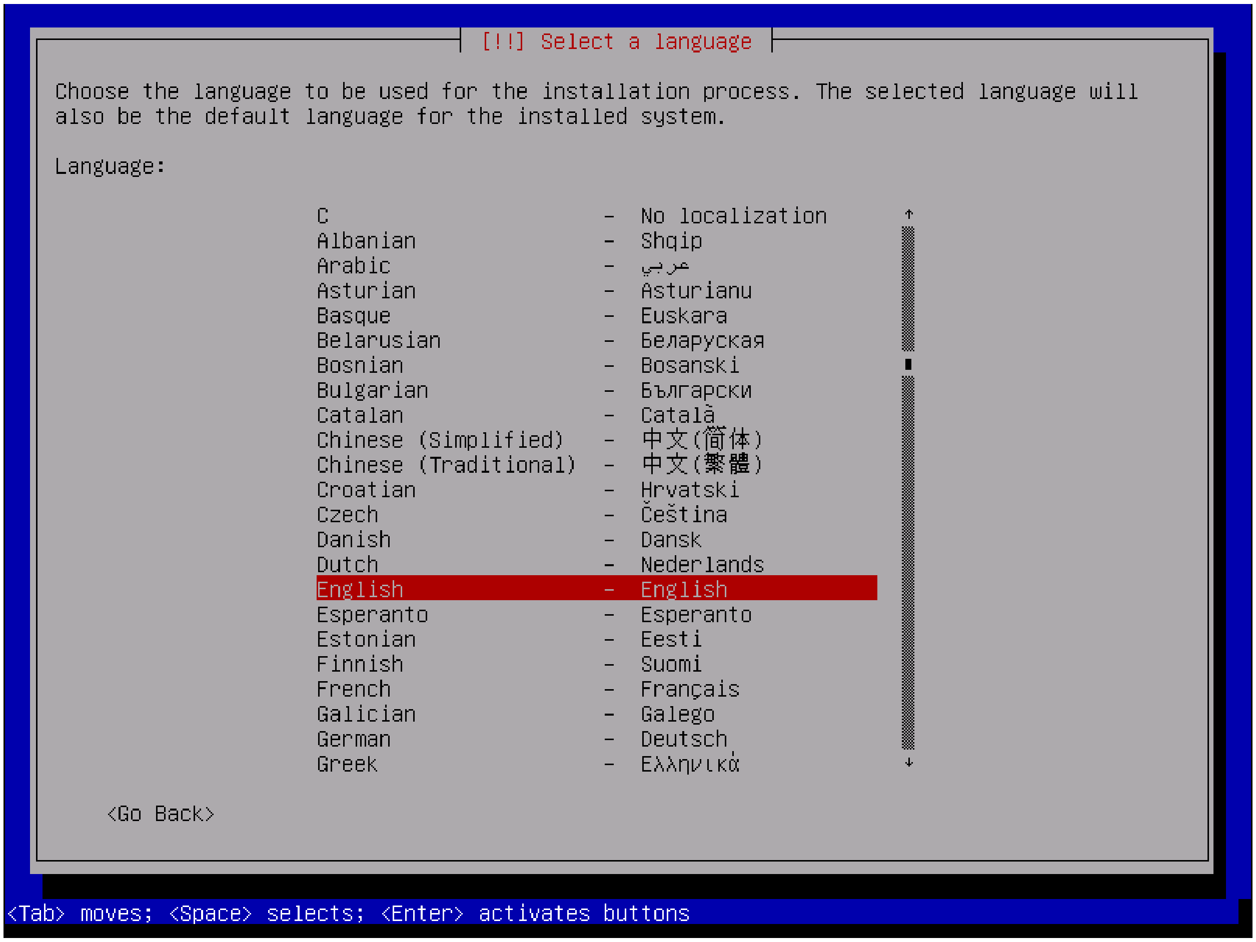
You can download one of the IvozProvider Automatic ISO CD images (generated using simplecdd) in stable or nighly versions:

Important: IMPORTANT: Automatic install CDs will format target machine disk!

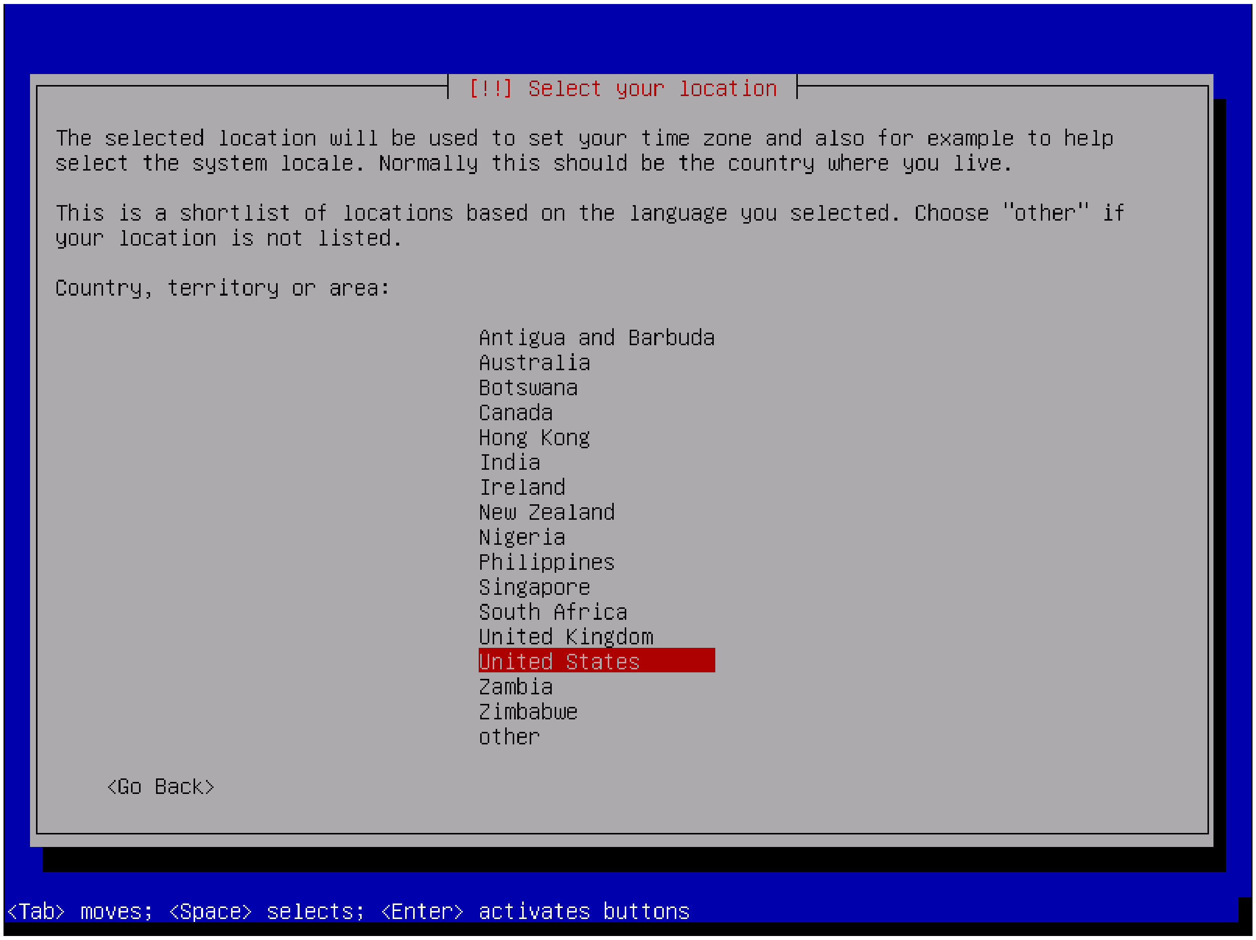
- Configure the target machine to boot from CD. It will display the Debian GNU/Linux installation menu.

Note: You can use graphic installation if you prefer, but the following screenshots show the standard installation.

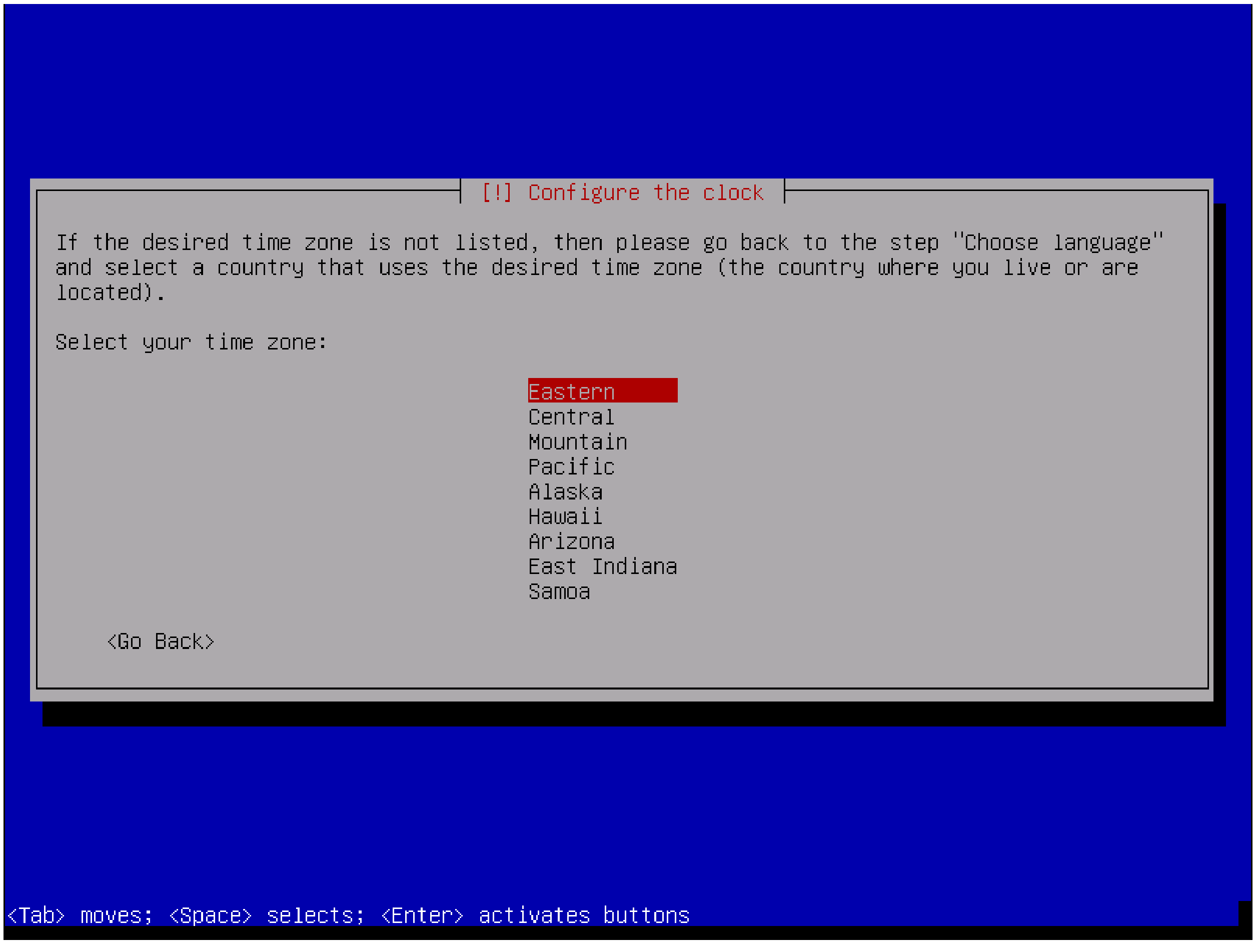
- Choose installation language:



- Choose location:

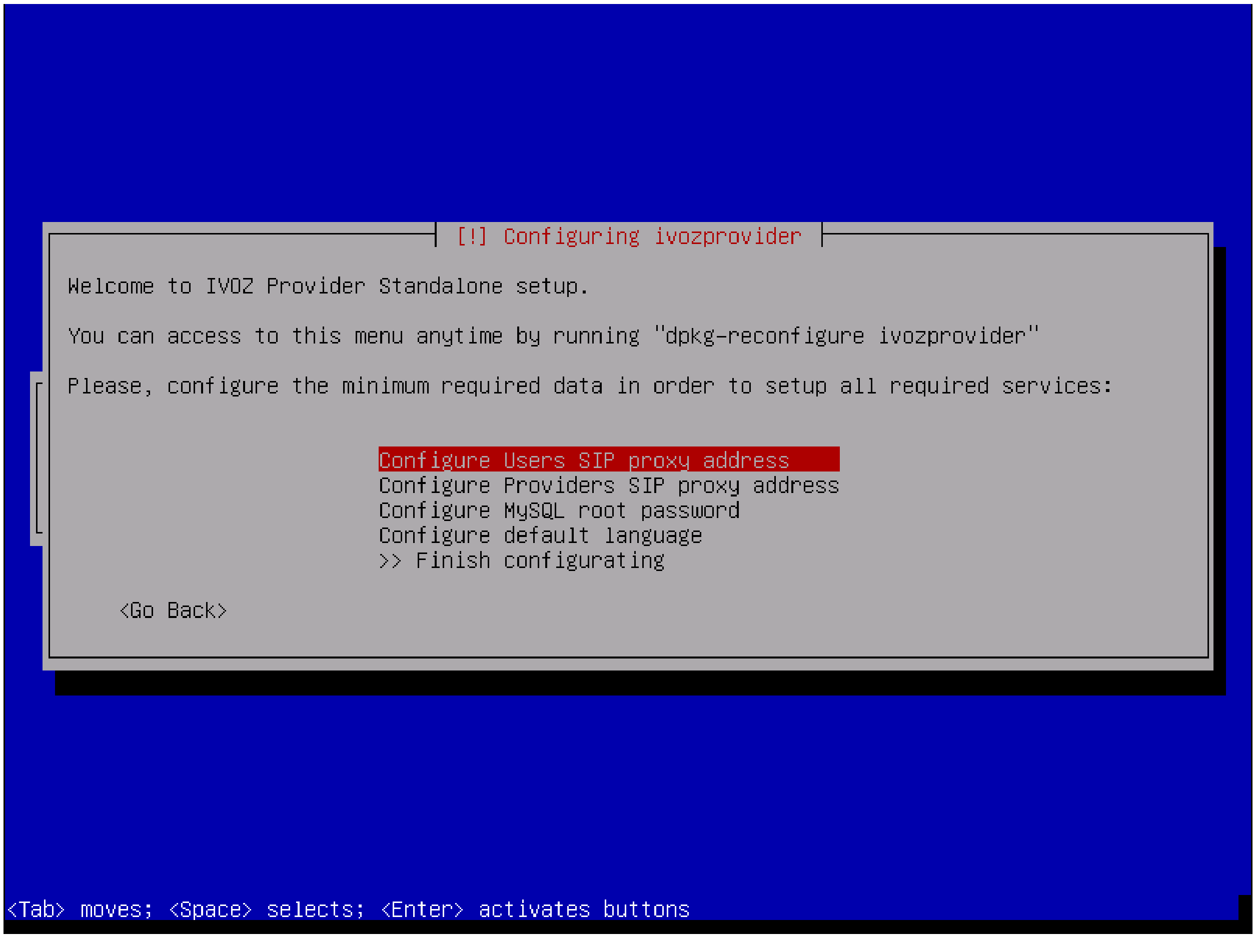


- Choose date and time configuration:



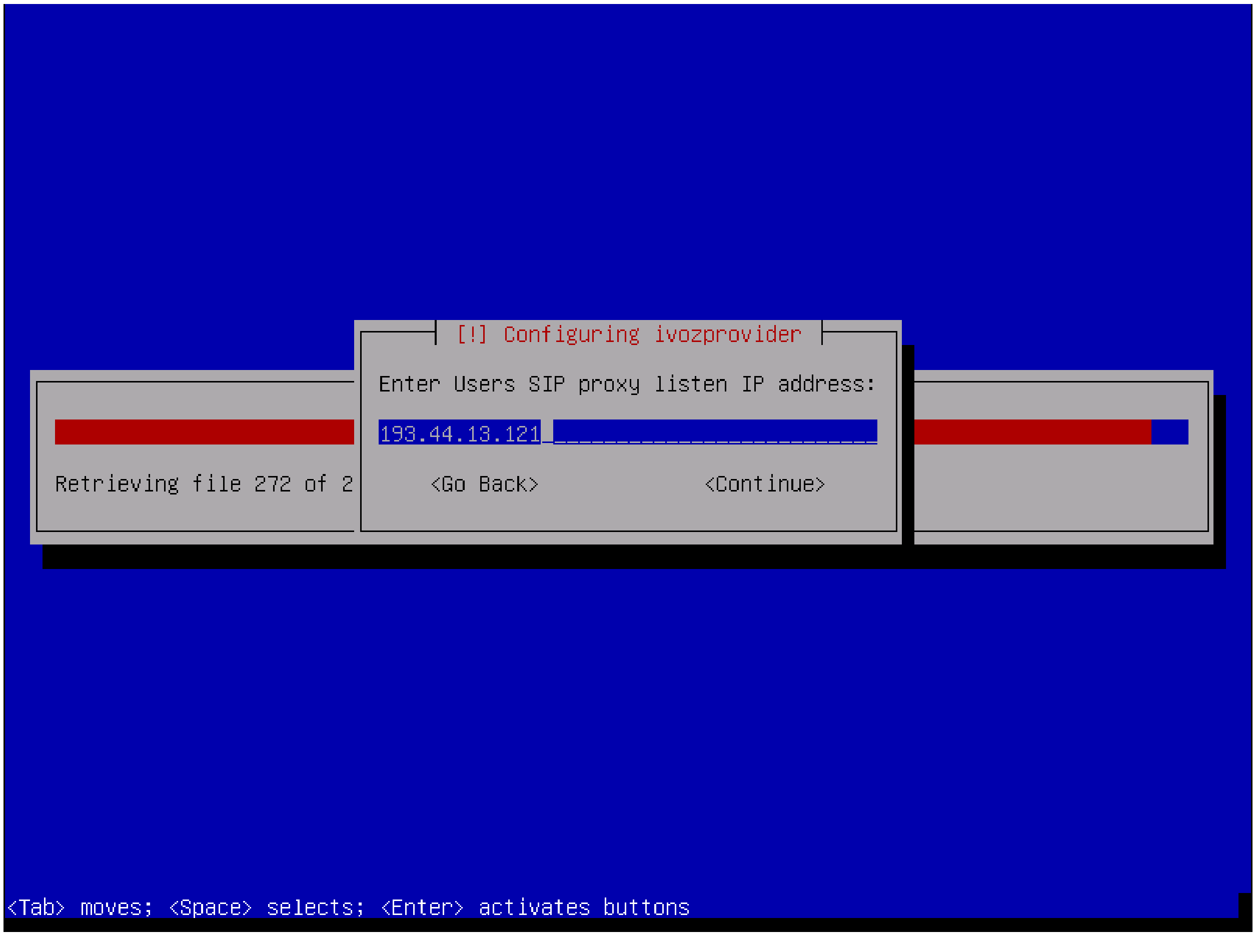
Note: At this point, a generic network configuration and disk partitioning will be performed, and also a installation of base system.

- Configure IvozProvider:



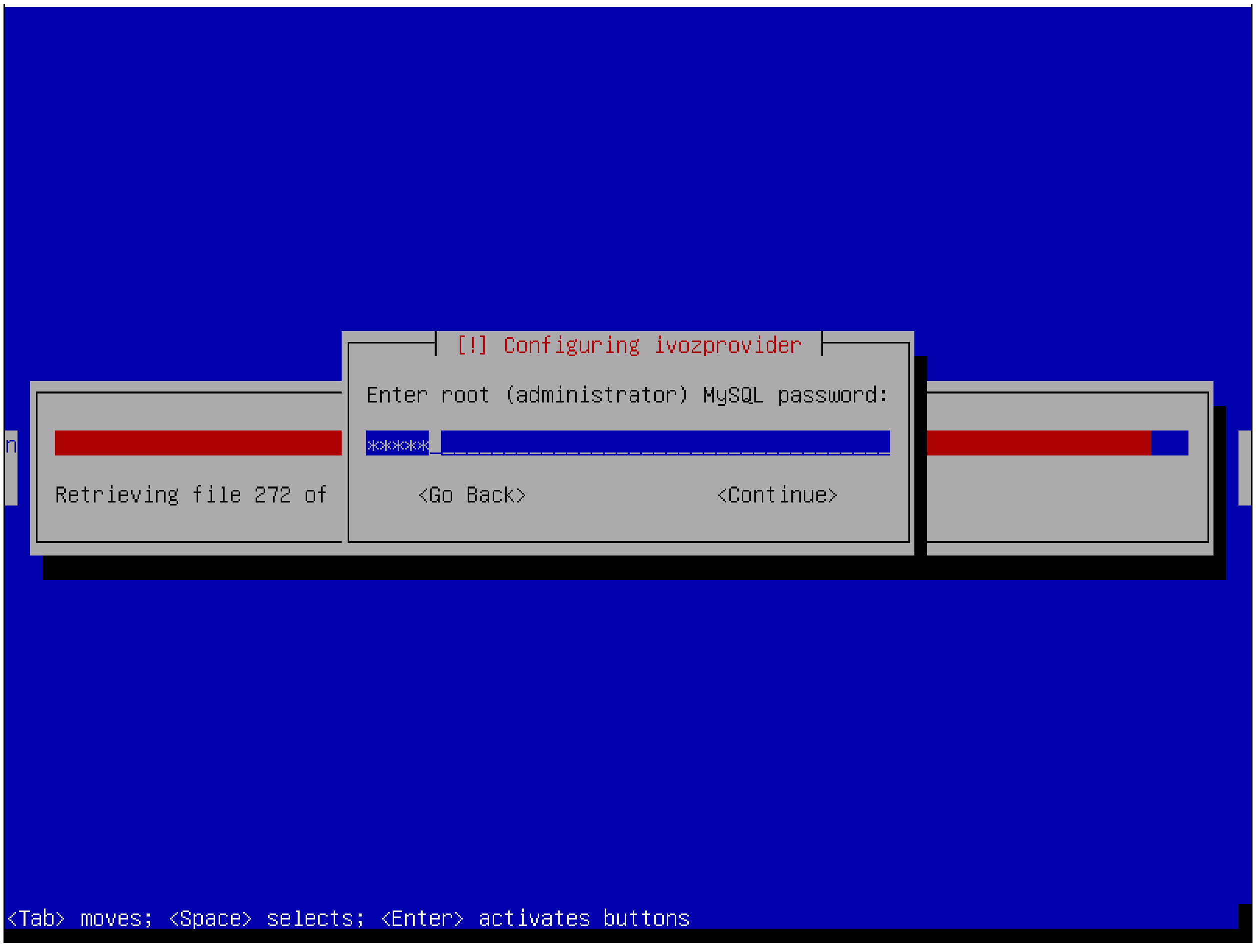
As mentioned in *Minimum requirements* is required at least one public IP address for User and Trunk SIP proxies. Remember that if you use only one, KamTrunks will use different SIP ports to avoid collision.

You can set its addresses right now and configure the interfaces properly when the system is fully installed. This menu can be displayed anytime after the installation.

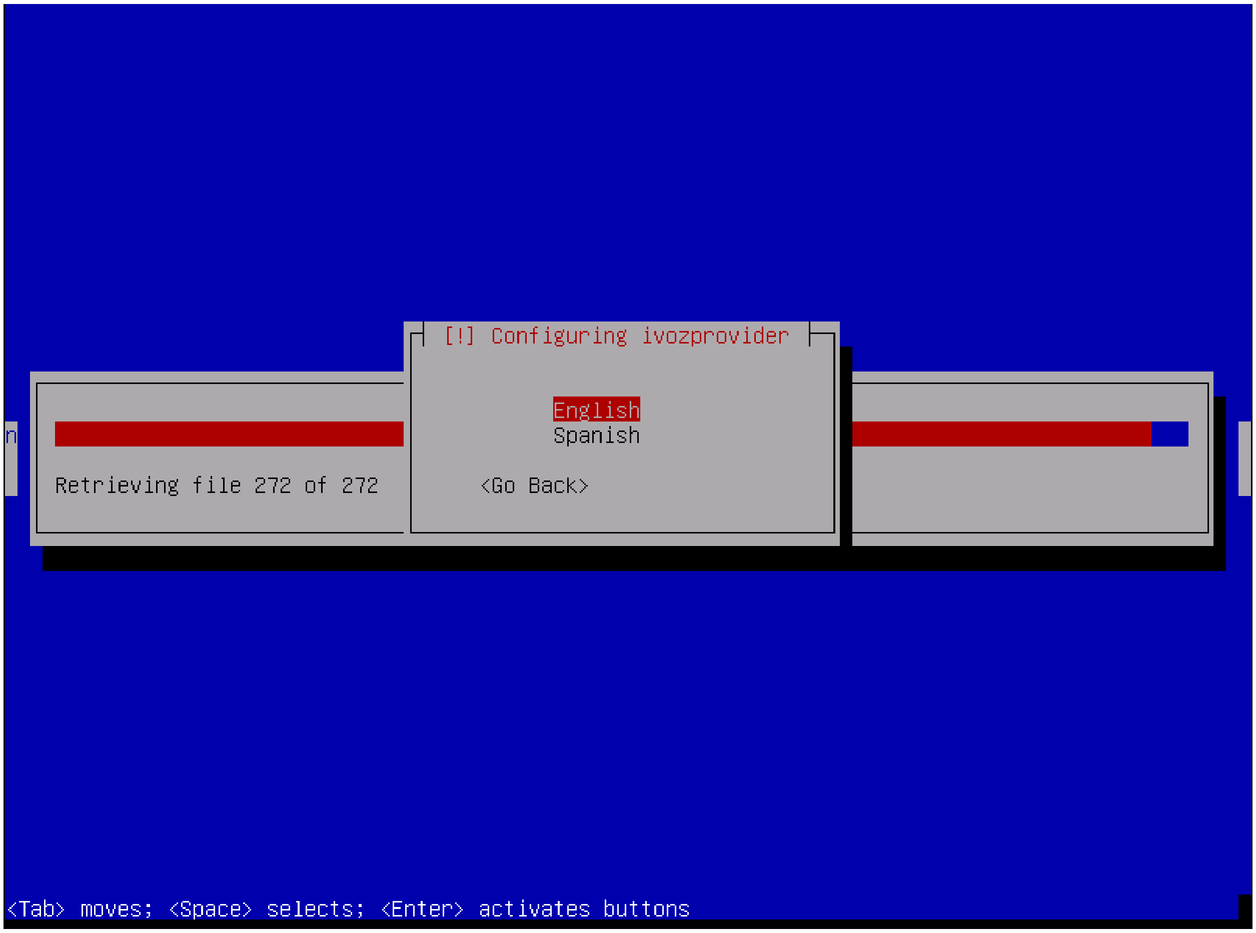


You can also configure default root MySQL password right now.

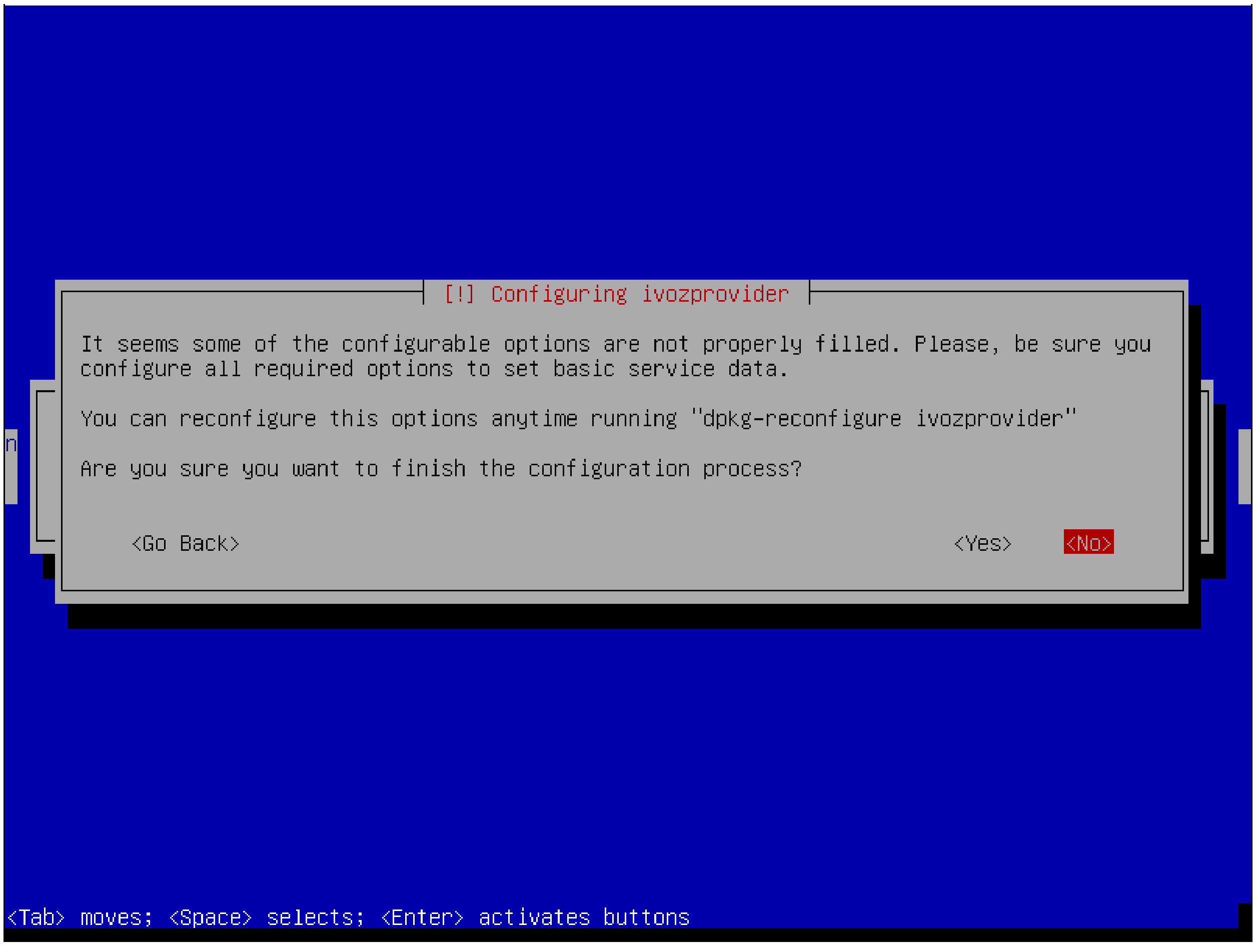
Note: If you don't configure MySQL password, default password will be used (changeme). You can still change it later.



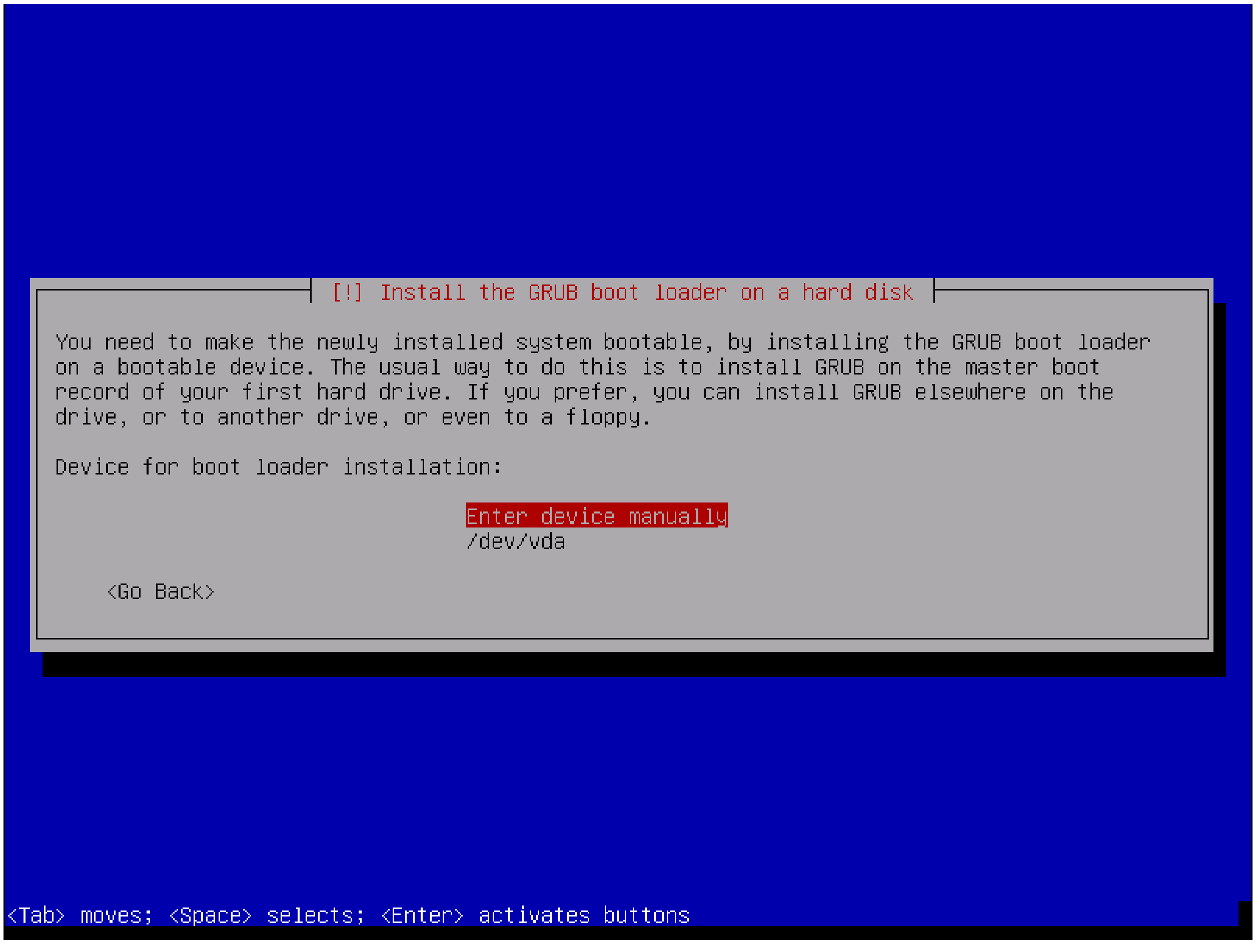
And default language for portals:



Note: It is not require to configure all settings during initial installation. In case any setting has been left without configuration a warning dialog will be displayed.



At last, select where the GRUB boot loader will be installed.



After the reboot, you are ready to access using the web portals!

Important: Any of the public IP addresses configured during the installation will work to acces the web portal. Default credentials are **admin / changeme**.

3.5 Extra components

3.5.1 G.729

Important: In some countries, you might have to pay royalty fees in order to use G.729 codec to their patent holders. We're not legal advisors regarding active or withdrawn world patents.

You can use G.729 with IvozProvider, but installation must be done manually. G.729 codec is optimized for each CPU type and version of asterisk, so each installation may require a different codec module.

You can download codec from here under the section Asterisk 13.

Once downloaded, move the *.so* file to **/usr/lib/asterisk/modules/** and rename it to **codec_g729.so**

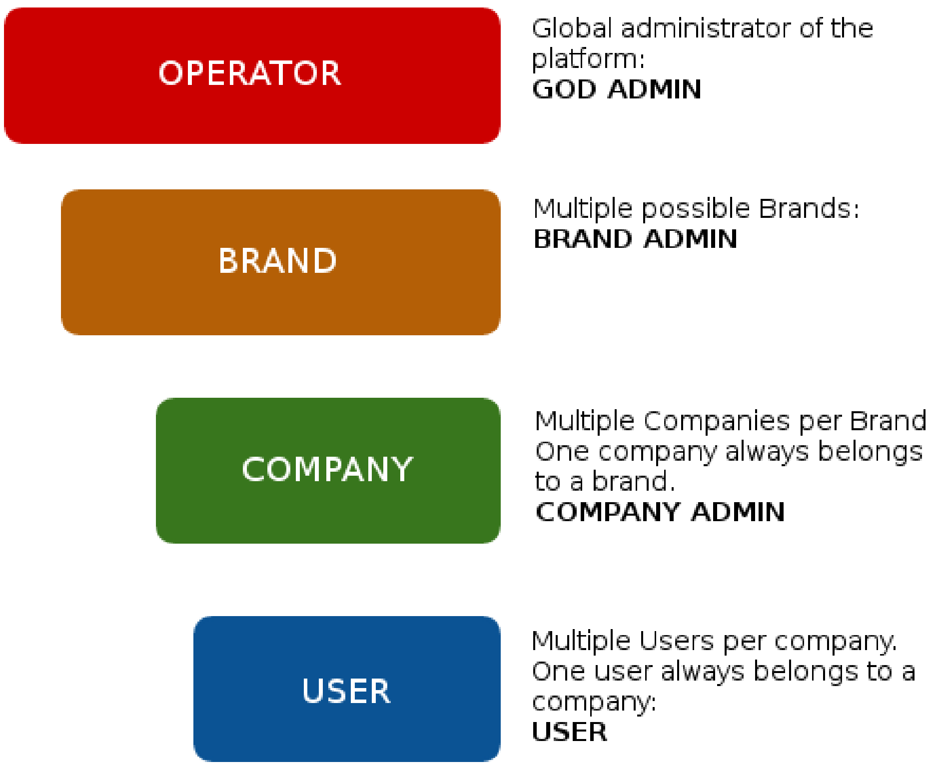
You can check the codec is valid by restarting asterisk and printing the available codec translations using:

```
systemctl restart asterisk # Restart asterisk
asterisk -rx 'core show translation' | grep 729
```

Platform roles

IvozProvider is a multilevel role provider solution.

The following images shows the different available levels and the relation between them:



This section will explain each of the available roles, describing their responsibilities and more important tasks.

4.1 Global administator role

The global administator role (operator in the image) is usually done by the installation responsible.

All options and platform features are visible to this role and usually is in charge of its maintenance.

Their most important task is to **create Brands** and configure them so they have the enough autonomy to properly use the platform:

- Configure their web access.
- Configure their brand portal look and feel: themes, colors, etc.

Appart from their main task, their global visibility and total access makes them responsible of:

- Monitor the platform so it keeps always UP & RUNNING
- Analyze platform logs to track possible errors.
- Polish the security mechanisms to avoid external attacks.
- Obtain global statistics of calls audio quality.
- Increase the available resources of the platform as long as is needed:
  - Increasing resources available in a standalone installation
  - Migrating, whenever required, to a distributed installation with multiple AS, media relays, etc.

To sum up, **this role is the only one that has no limits within the platform**, thats why *God* is a term used in multiple places along this documentation.

Important: *This role is responsible of maintain the platform**, configuring it for the correct behaviour. This role **doesn't have any kind of limit** and **grants access** to the **brand operators**.

4.2 Brand andministator role

Brand operator can access a portal with less sections available compared to the previous role. The general (God) administrator is in charge of providing an URL with credentials for its brand portal.

The most important task for brand operator can be managed through this portal: **create and configure companies so they can work properly**.

Due to brand operators are also resposible of billing their companies and make sure the external cals are properly setup, it must also manage:

- Peering contracts with other IP providers for PSTN interconectivity.
- Include all required company information for the billing process.
- Pricing plans that will offer to their companies, that will determine how match they pay for each call.
- Setup the routes for each outgoing call types based on their final destination
- Create the invoices for each billing period and send them to their clients.

As you can see, the task of brand operator has little in common with the global operator, but their importance is vital so the final users can use all the features includes in IvozProvider

Important: **To sum up**, the brand operators **grant access** to the **company** administrators they serve and **configure the platform to route, tarificate and bill their calls**.

4.3 Company administator role

The company administrator has access to the portal supplied by the brand operator.

From its point of view, it has a virtual pbx in the cloud that must configure for its users.

To accomplish that, it's required:

- Configure terminals, extensions and users.
- Configure the DDI incoming process with the proper logic:
  - Directly to an user
  - IVRs
  - Hunt groups
  - Faxes
- Give access to the final users to their web portal, so they can configure their profile options:
  - Call forward
  - Do not disturb
  - Call waiting

Important: To sum up, the company administrators are resposible for **configuring the telephony system and make use of all the features available in IvozProvider**.

4.4 Final user role

The final user has two different kinds of credentials, both supplied by its company administator:

- User portal access credetentials
- SIP credentials used to register its terminal (or terminals) to IvozProvider

Through the user portal, it can browse their call registry and configure:

- Call forward
- Do not disturb
- Call waiting

On the other hand, the SIP creadentials allow the users to configure their terminal (or terminals) to place and receive calls.

Note: The same SIP credentials can be used in multiple devices at the same time,generating what is known as *parallel-forking*: whenever a call is placed to an user, all the active devices will ring so the user can answer the call from any of them.

Important: Final users are the ones that use and enjoy all the feature of IvozProvider

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CHAPTER 5

Making internal calls

The goal of this block will be to configure IvozProvider in order to make internal calls, using as the starting point the base installation described in the previous step.

In order to archive a call between Alice and Bob, we have to do some task in the three configuration levels described in *Platform roles*, and that is why we have ordered the index in this blocks:

5.1 Global Configuration

Important: Any of the 2 Public IP addresses configured during the installation will work to acces the web portal. Default credentials are **admin / changeme**.

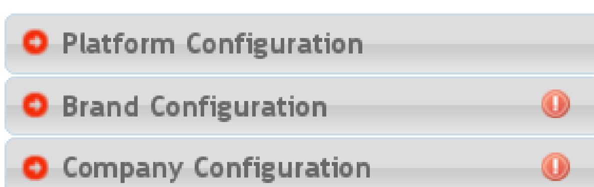
In this section will reference global administrator configuration options, avaible in the menu (**Main management**) of the web portal (only visible to God Admins):

5.1.1 Emulate the Demo brand

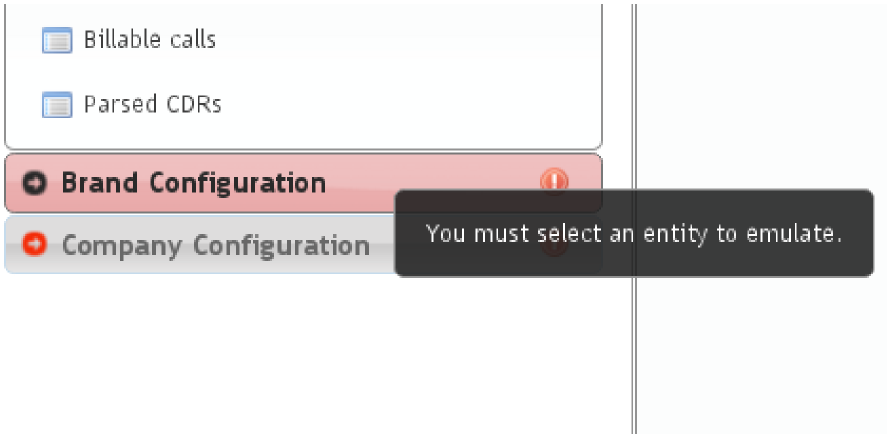
As mentioned above, the initial installation will have an already created brand called DemoBrand, that will be used for our goal: to have 2 telephones registered that can call each other.

Before going to the next section, is quite important to understand how the **emulation** works.

- As global operator, you have access to the menu **Main management** only visible to *God* administators.
- Apart from that menu, you will also have access to the **Brand configuration** and **Company configuration** that will look more or less like this:

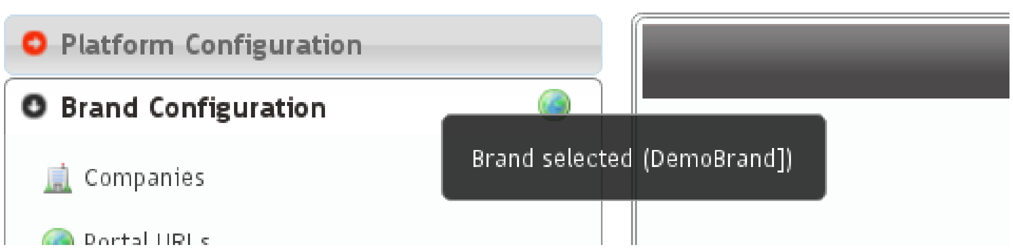


- Check following button

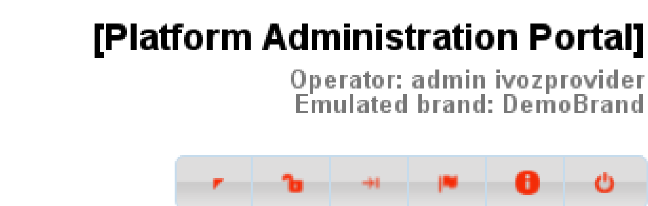


- When pressed, a popup will be displayed:

- After selecting the DemoBrand brand, the icon will change and shows the emulted brand:



- The upper right corner of the portal will also display the brand that is being emulted:



5.1.2 What emulation means

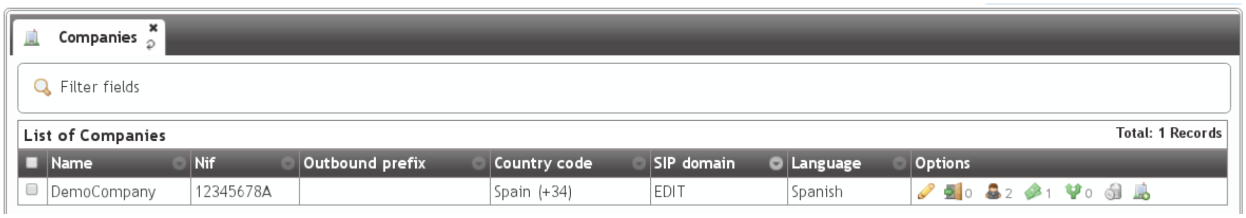
Basically, that **everything in the menu 'Brand configuration' will be relative to the chosen brand** and is **exactly** the same menu entries that the brand operator will see using its brand portal.

Tip: Ok, ok. maybe exactly is not totally accurate. The global operator is able to see some fields in some screens that other admins cann't (i.e. On Company edit screen, fields like 'Media relays' or 'Application server' are only configurable by the global operator.

5.2 DemoBrand Configuration

We need that the default DemoBrand have a client with at least 2 users. In order to archive this we will require little configuration in this section.

In fact, if we check **Virtual PBXs** in the brand menu, we'll discover that there is already an existing *DemoCompany* that we can use to fulfill our desired goal :)



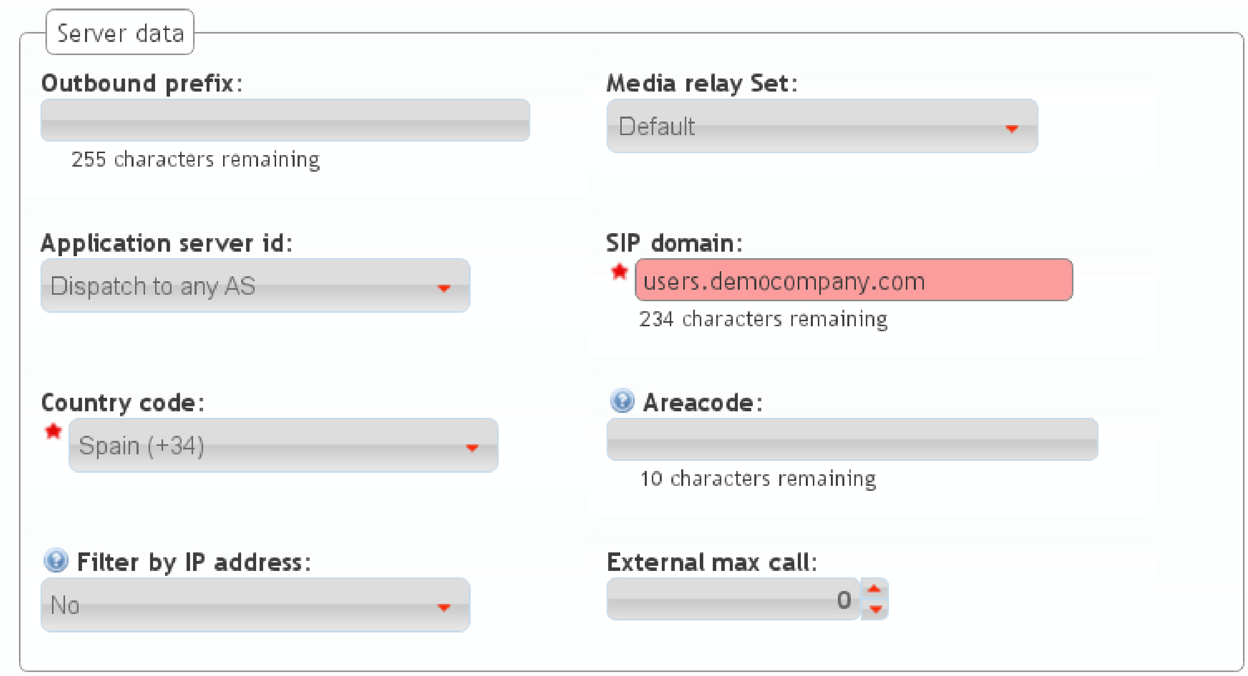
Only a thing is required to configure for this company, marked as edit in the previous image.

5.2.1 Company SIP Domain

As mentioned in the previous section, is **required** that each of the companies have a public domain that resolves to the configured IP address for *Users SIP Proxy*.

Note: DNS register can be type A (supported by all the hardphones/softphones) or even NAPTR+SRV.

Once the domain has been configured (by means that are out of scope of this document), it will be enought to write it in our company configuration:



Once the company has been saved, the domain will be also displayed in the list *previously mentioned*:

List of Domains — Total: 3 Records

Domain	Scope	Points to	description	Options
users.ivozprovider.local	Global	proxyusers	Minimal proxyusers global domain	
trunks.ivozprovider.local	Global	proxytrunks	Minimal proxytrunks global domain	
users.democompany.com	DemoCompany (company)	proxyusers	DemoCompany proxyusers domain	

> **Attention:** It's important to understand this block. *Unless we've a single company registered*, wihout a DNS domain pointing to our users proxy IP address, everything will fail.

This is a good sign for the domain we have configured right now, replacing the 10.10.3.10 with the public address we have used to configure *Users SIP Proxy*.

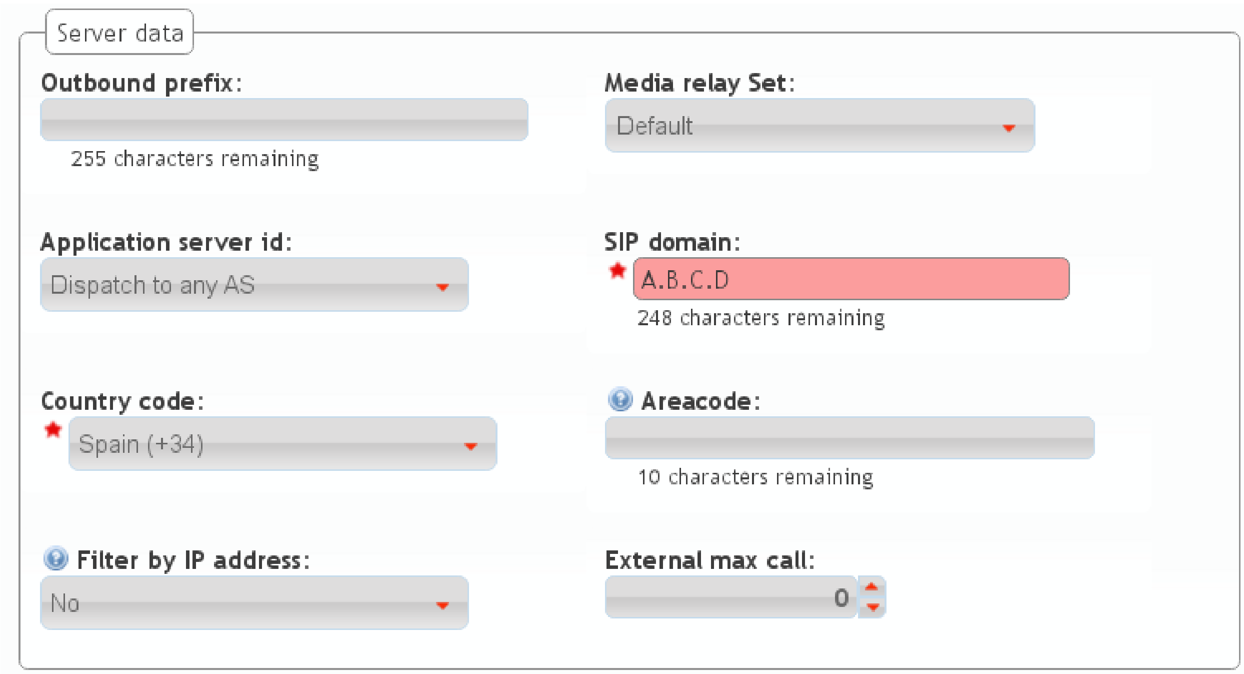
```
[~]$ ping users.democompany.com
PING users.democompany.com (10.10.3.10) 56(84) bytes of data.
64 bytes from oasis-dev (10.10.3.10): icmp_seq=1 ttl=63 time=0.476 ms
64 bytes from oasis-dev (10.10.3.10): icmp_seq=2 ttl=63 time=0.467 ms
64 bytes from oasis-dev (10.10.3.10): icmp_seq=3 ttl=63 time=0.548 ms
^C
--- users.democompany.com ping statistics ---
3 packets transmitted, 3 received, 0% packet loss, time 1998ms
rtt min/avg/max/mdev = 0.467/0.497/0.548/0.036 ms
```

> **Danger:** Have we stressed enough that without a properly configured DNS pointing to the Users proxy IP address nothing will work?

I have no time for a DNS registry

Everything we have said is true: as we create new brands and brands create new companies, each of them will need a DNS registry.

But the first company of the platform is quite special and can take over the IP address of the proxy to use it as a domain:



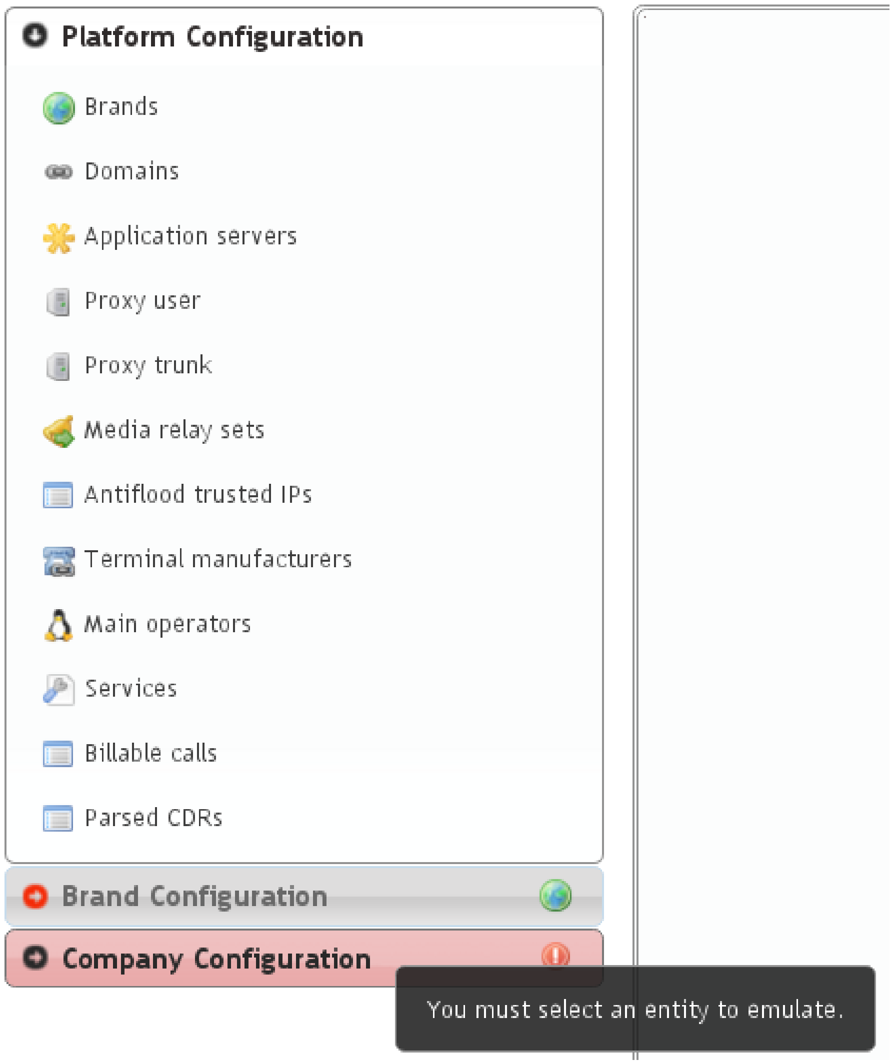
Although it is not a domain, but being used like it was, it will be displayed in Domain section:

Domain	Scope	Points to	description
A.B.C.D	DemoCompany (company)	proxyusers	DemoCompany proxyusers domain

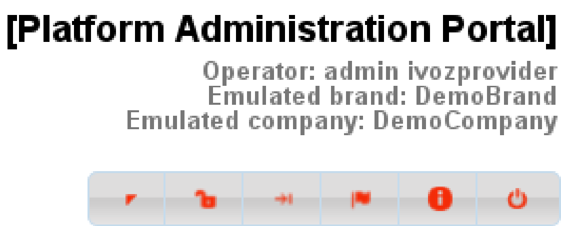
Tip: It's important to understand the this trick is only valid for the first company of the platform ;)

5.2.2 Emulate Demo company

The company emulation process is the same as the brand emulation, with the difference that it filters the block 'Company Configuration' instead of 'Brand Configuration'.



Once the company has been emulated, the top right corner of the portal will show that we are in the right path :)



5.3 DemoCompany Configuration

We're close to make our fist call in our fresh installed IvozProvider, there are only 6 steps to configure in our Demo-Company virtual pbx.

- 2 terminals
- 2 extensions
- 2 users

5.3.1 Creating Terminals

Go to the terminal section and... voilà! We already have 2 terminals created:

List of Terminals — Total: 2 Records

	Name	Domain	Terminal Model	Mac	Allowed codecs	Status	Options
	alice	asd	Generic		alaw - G.711 a-law		
	bob	asd	Generic		alaw - G.711 a-law		

5.3.2 Creating Extensions

Then we go to extensions, just to check that we have 2 extensions already created for us:

List of Extensions — Total: 2 Records

	Number	Route type	Target	Options
	101	User	Alice Alison	
	102	User	Bob Bobson	

Nothing more to do in this section, let's go the next one!

5.3.3 Creating Users

As expected, we also have 2 created users:

List of Users — Total: 2 Records

	Name	Lastname	Screen Extension	Terminal	Outgoing DDI	Options
	Alice	Alison	101	alice	Unassigned	0 0
	Bob	Bobson	102	bob	Unassigned	0 0

At this point, we have everthing ready make a call between this two users: Alice and Bob.

5.4 SIP Terminal configuration

The last thing we need is 2 SIP terminals (hardphones, softphones or even mobile applications) and configure them as follows:

ALICE

- **User**: alice
- **Password**: alice
- **Domain**: users.democompany.com (or the IP if we are using *the DNS trick*)

BOB

- **User**: bob
- **Password**: bob
- **Domain**: users.democompany.com (or the IP if we are using *the DNS trick*)

Tip: Sometimes the user and domain is configured in a single option. In this case we should enter alice@users.democompany.com and bob@users.democompany.com (or the IP if we are using *the DNS trick*)

After configuring the terminals, Alices should be able to call Bob only by dialing 102 in her terminal.

CHAPTER 6

Receive external calls

The goal of this block will be configure IvozProvider to receive incoming external calls.

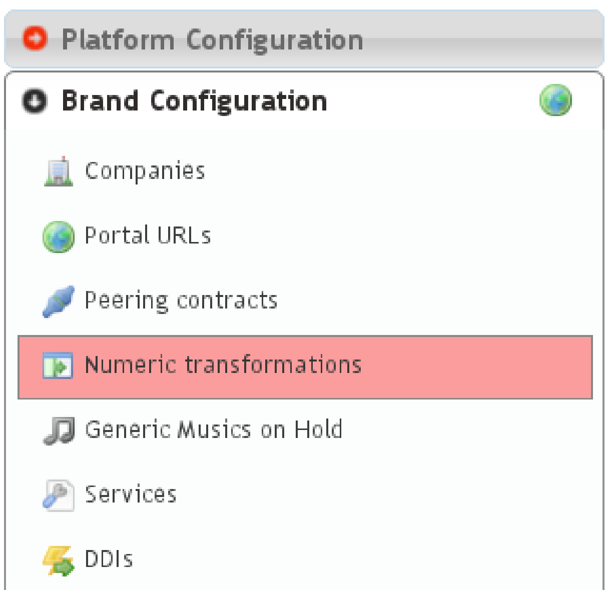
In order tho archive this, this steps will be followed:

6.1 Transformations configuration

IvozProvider is designed to provide service **anywhere in the planet**, not only the original country where the platform is installed.

A very important contept to archive this goal is the numeric transformation, that **adapts the different number format systems of the countries of the world** defined in E.164 **to a neutral format**.

The section that allows the brand operator to configure all the **numeric transformations** is:

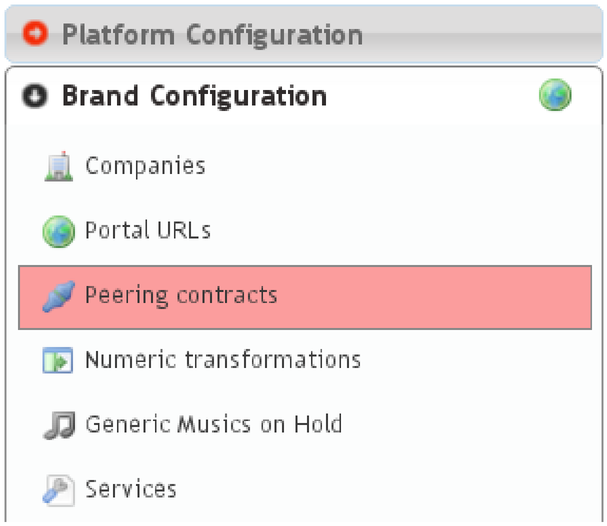


You can find more information about transformations in *Numeric transformations* section.

6.2 Peering configuartion

We understand a **Peering contract** the agreeming between a **Brand Operator** and a VoIP Provider to make and receive calls.

IvozProvider is ready to integrate with IP providers created on the secteion **Perring contracts**:



6.2.1 DDI Providers

TODO https://github.com/irontec/ivozprovider/issues/442

6.2.2 DDI Provider Registrations

Some providers require a SIP Register active in order to receive incoming calls to our DDIs. Some of them, even require this register in order to process our outgoing calls through their services.

Note: IvozProvider supports any kind of *peering*, but we highly recomend *peer to peer peerings*: without authentication, without registry and validated by IP. This will avoid unnecessary traffic (authentication in each session and preriodic registers) and simplifies its configuration, just by leaving most of the fields by default.

Username Account number or similar provider by the provider that requires SIP register.

Domain Domain or IP of the registar server. Usually the same as the SIP proxy of the Peer server.

DDI This will be sent in the SIP Contact header and must be unique in all the platform. For DDI Providers with an associated DDI, it is recommended to enter that DDI. In case of multiples DDI for the same DDI Provider, use any of them. If no DDI is associated with this DDI Provider just enter an unique numeric value.

User Authentication user. Most of the time it's the same as username, so it's recommended to leave empty.

Register server URI Usually this can be left empty, as it can be obtained from the Domain. If it is not the case, enter the IP address with the 'sip:' prefix.

Realm Leave empty to accept the authentication realm proposed by the provider. Define only if you are familiar to the authentication mechanism used in SIP.

Expire Default suggested register expire time.

Tip: Similar to the Peer Servers, there are lots of fields in the screen. You must have into account that most of the provider doesn't require register , and those who does, will only use user, domain and password.

Once we have an agreement with a VoIP provider and we have configured it in the *peering* section, only two task are pending:

6.3 Configuring an external DDI

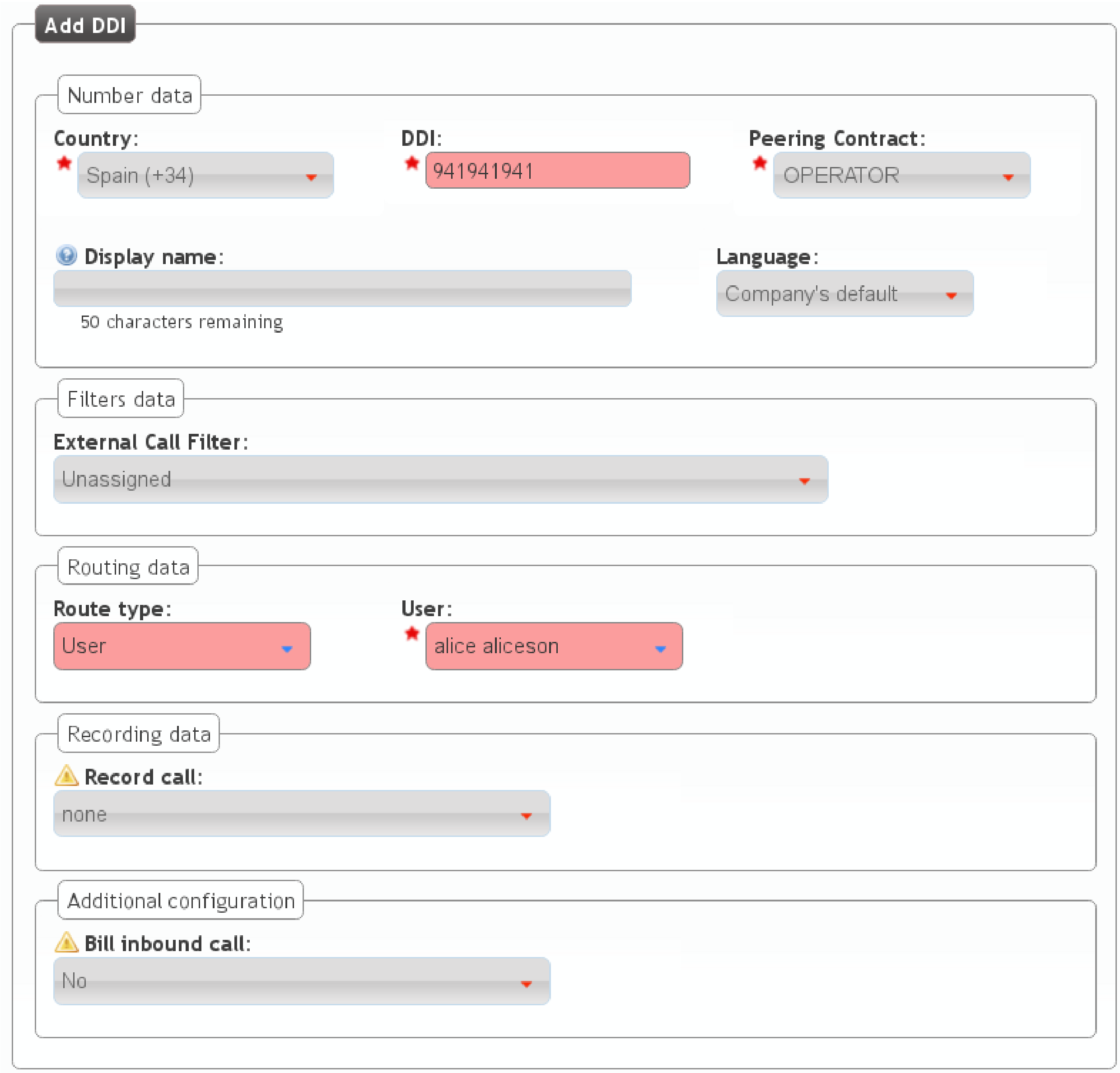
The brand operator, responsible of this *peering* agreements with VoIP providers , has the task to create the DDIs for each provider.

Notice that in order to access this section, the brand operator (or *god*) must have emulated the proper company and access the menu section **Company Configuration**.

Attention: Section **Company configuration > DDIs** is different when the company administrator access than the displayed data when a global or brand administrator does. Company administrator are unable to create or delete DDIs, just edit the one created by the brand or god administrator.

The section **Brand configuration > DDIs** is a *read-only* display of all the DDIs of the brand, associated with the different companies.

Taking into account this concepts, we create a new DDI and fill the required fields:



For detailed information about configuration fields, check *DDIs* section.

6.4 Configure incoming routes

In the previous section, we have created the DDI and configure it, but **the most common procedure** is that the brand operator just create it while the **company administator**, using the same saction **will configure** it choosing the correct route (user, huntgroup, etc.), its filters with calendars and so on.

Note: At this point, calling the number of the configured DDI will make the *Alice* phone ring.

CHAPTER 7

Making external calls

The goal of this section is configuring IvozProvider to make external outgoing calls, taking previous section configuration as a starting point.

We will follow this steps:

7.1 Where do I call?

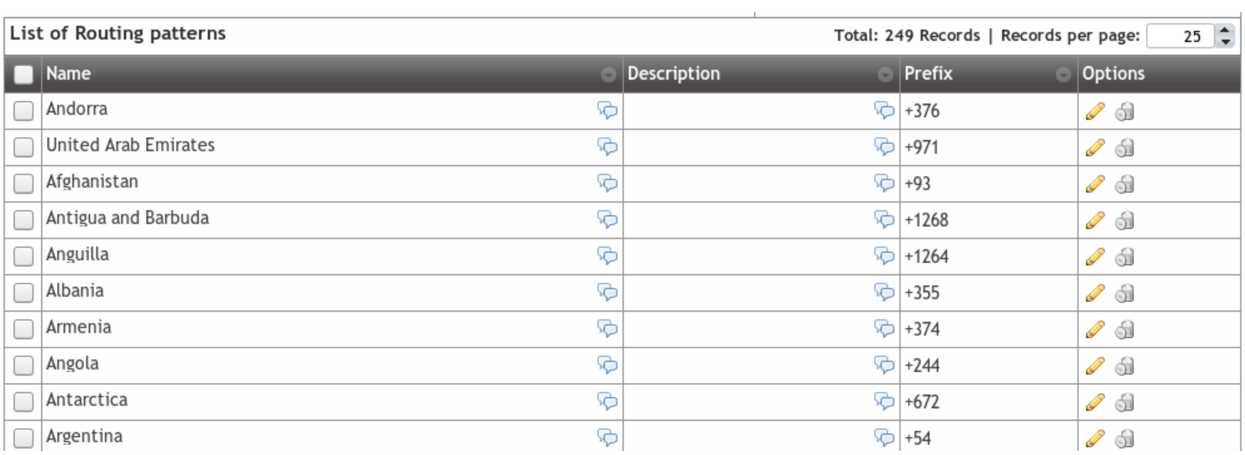
At this point of the configuration, we have to configure IvozProvider to use the already configured *Contract Peering* to place the external calls we are making.

To achieve this, in first place, we need that the dialed external numbers fall in an existing **target pattern**.

7.1.1 Routing patterns

When a user dials an external phone number, IvozProvider tries to categorize this call into a one of the routing patterns defined in this section:

Usually, it will we useful to have one routing pattern for the countries defined in the ISO 3166. That's why IvozProvider automatically includes all this countries and their prefixes:



List of Routing patterns — Total: 249 Records | Records per page: 25

Name	Description	Prefix	Options
Andorra		+376	
United Arab Emirates		+971	
Afghanistan		+93	
Antigua and Barbuda		+1268	
Anguilla		+1264	
Albania		+355	
Armenia		+374	
Angola		+244	
Antarctica		+672	
Argentina		+54	

Warning: Brand operator can choose between keeping this routing pattern if finds them useful or deleting them an creating the ones that meet his needs.

7.1.2 Routing pattern groups

As we will see in *Outgoing Routing* section, every routing pattern will be linked to a Carrier.

That's why it can be useful to group the routing patterns in **routing pattern group** so that we can link a whole group to a Carrier more easily.

By default we can see the countries grouped in the continents defined in ISO 3166:

List of Routing pattern groups — Total: 7 Records

	Name	Description	Destination patterns	Options
	Europe		Andorra (376), Albania (355), Austria (43), Åland Islands (358), Bosnia and Herzegovina (387), Belgium (32), Bulgaria (359), Belarus (375), Switzerland (41), Czech Republic (420), Germany (49), Denmark (45), Estonia (372), Spain (34), Finland (358), Faroe Islands (298), France (33), United Kingdom (44), Guernsey (44), Gibraltar (350), Greece (30), Croatia (385), Hungary (36), Ireland (353), Isle of Man (44), Iceland (354), Italy (39), Jersey (44), Liechtenstein (423), Lithuania (370), Luxembourg (352), Latvia (371), Monaco (377), Moldova (373), Montenegro (382), Macedonia (389), Malta (356), Netherlands (31), Norway (47), Poland (48), Portugal (351), Romania (40), Serbia (381), Russia (7), Sweden (46), Slovenia (386), Svalbard and Jan Mayen (47), Slovakia (421), San Marino (378), Turkey (90), Ukraine (380), Vatican City State (39)	
	Asia		United Arab Emirates (971), Afghanistan (93), Armenia (374), Azerbaijan (994), Bangladesh (880), Bahrain (973), Brunei (673), Bhutan (975), Cocos (Keeling) Islands (61), China (86), Christmas Island (61), Cyprus (357), Georgia (995), Hong Kong (852), Indonesia (62), Israel (972), India (91), British Indian Ocean Territory (246), Iraq (964), Iran (98), Jordan (962), Japan (81), Kyrgyzstan (996), Cambodia (855), North Korea (850), South Korea (82), Kuwait (965), Kazakhstan (7), Laos (856), Lebanon (961), Sri Lanka (94), Myanmar (95), Mongolia (976), Macao (853), Maldives (960), Malaysia (60), Nepal (977), Oman (968), Philippines (63), Pakistan (92), Palestine (970), Qatar (974), Saudi Arabia (966), Singapore (65), Syria (963), Thailand (66), Tajikistan (992), East Timor (670), Turkmenistan (993), Taiwan (886), Uzbekistan (998), Vietnam (84), Yemen (967)	

Important: To sum up, when a user dials an external number, IvozProvider looks up a matching routing pattern to decide which Carrier must be used to place this call.

To achieve our goal of making an external call to a spanish number, we didn't have to modify the initial contents of this two sections :)

7.2 Outgoing Routing configuration

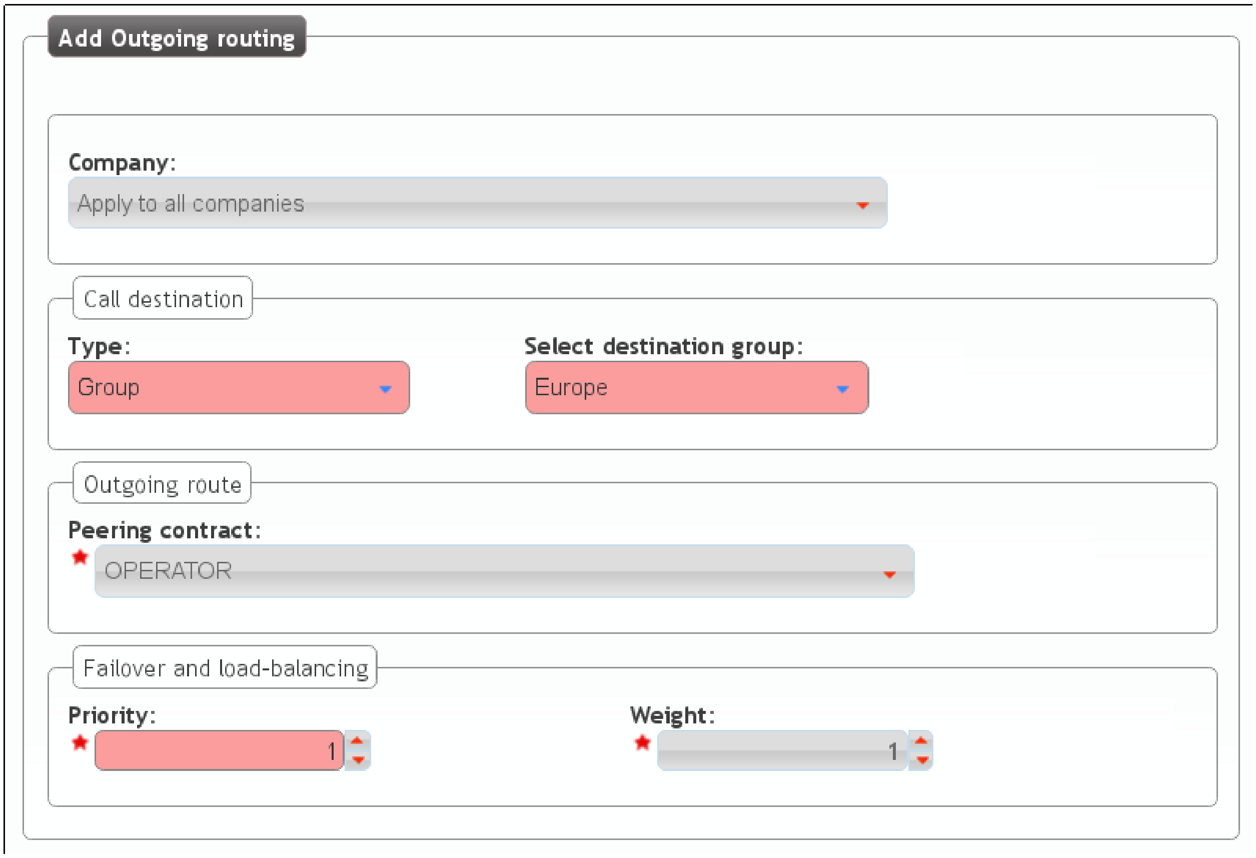
We already have our test call categorized as a call within the **Target pattern** 'Spain'. In addition, we also have a **Target pattern group** including 'Spain', called 'Europe'.

Now we have to tell IvozProvider that calls to 'Spain' or 'Europe' should be established through our **Contract Peering**.

To make this assignment, we use the section **Outgoing routing**:

If we choose routing 'Spain' calls only through our *Peering contract*, we will make this configuration:

On the other hand, if we are more generous and we decide to place calls to all european countries, we would make this configuration:

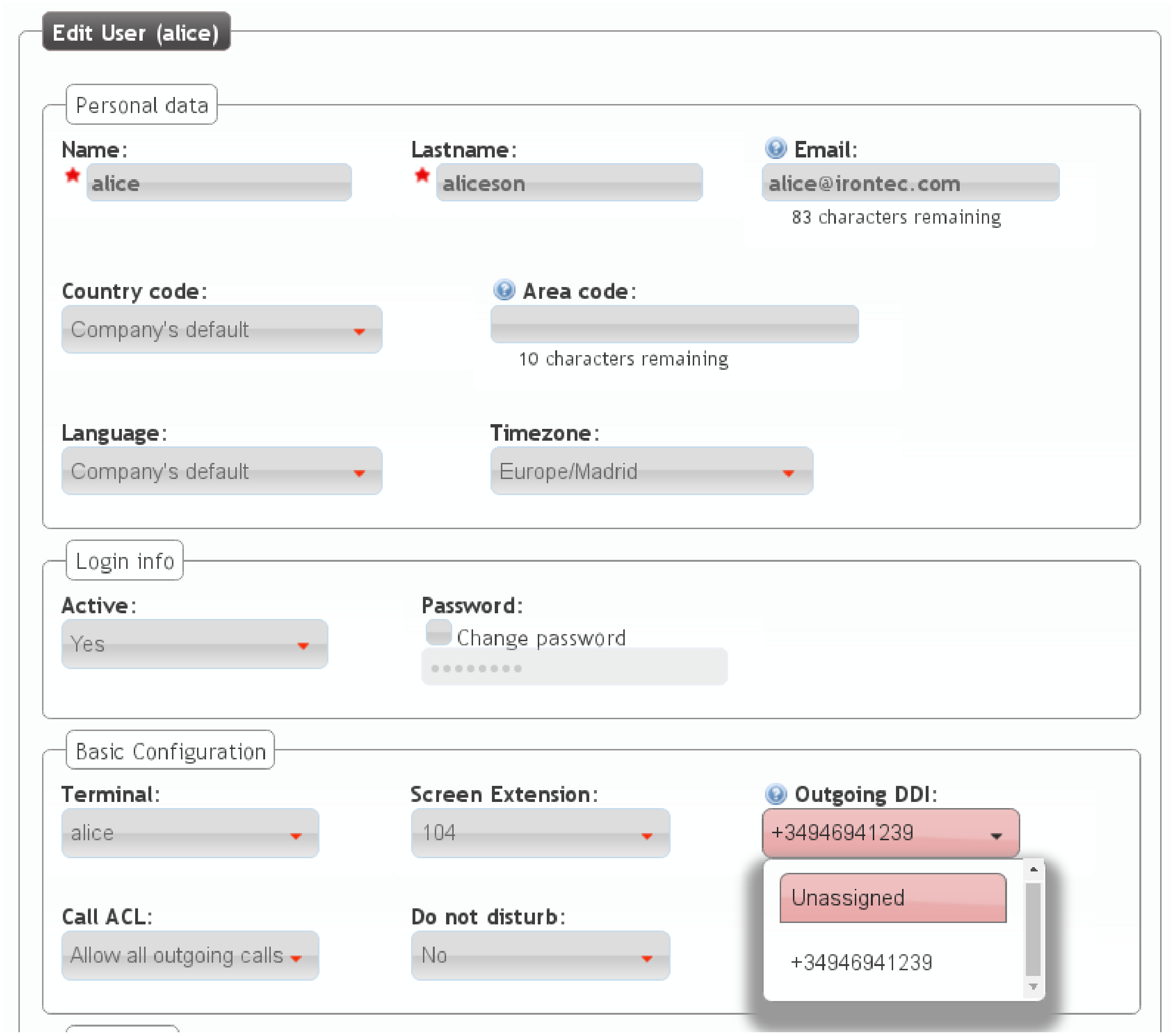


For more information about routing and load balancing check *Outgoing Routing* section:

7.3 Outgoing DDI configuration

Before placing our first outgoing call, it would be desirable to choose the number that the callee will see when the phone rings, so that he can return the call easily.

To achieve this goal, we have to configure our DDI as *Alice's* **outbound DDI**, because she will be the chosen one to place our first outgoing call:



We can set this up editing *Alice* in **Company Configuration > Users**. If this change is made by brand operator or global operator, he must *emulate the corresponding company* previously.

> **Warning:** Calls from users without an outgoing DDI will be rejected by IvozProvider.

At this point, we are looking forward to make our first outgoing call with our new IvozProvider, we may have even tried to call with current configuration but...

7.4 No pricing plan, no call

Just the way we warned *when we described the duties of the brand operator*, the brand operator is **responsible for making all the needed setup so that IvozProvider is able to bill all external calls**.

> **Note:** **Billing a call** is the action of **assigning price** to a call that implies cost.

IvozProvider checks live that a call can be billed when it is established to avoid placing calls that imply cost but won't be billed because Brand Operator, due to a mistake, hasn't assigned a price.

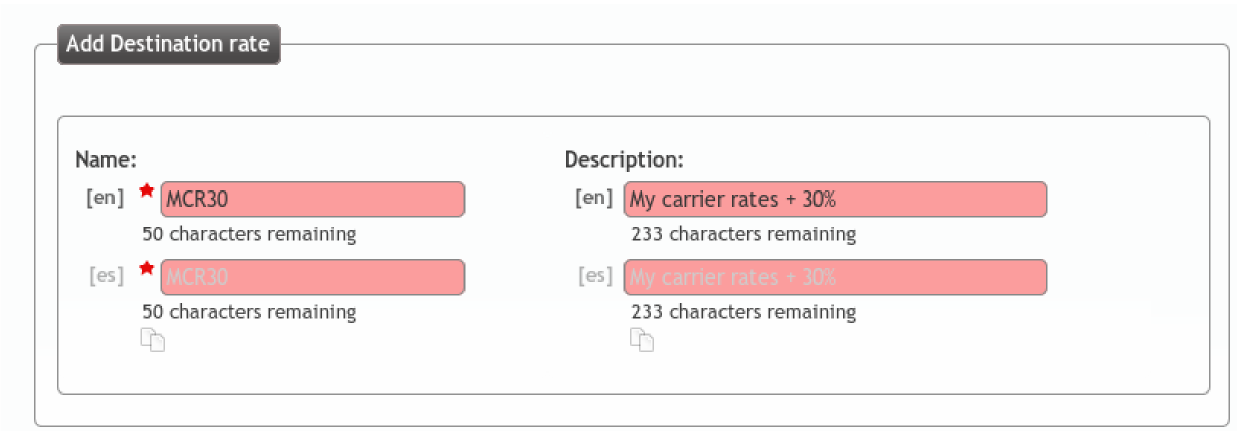
> **Error:** If a call can't be billed, IvozProvider won't allow its establishment.

7.5 Creating a destination rate

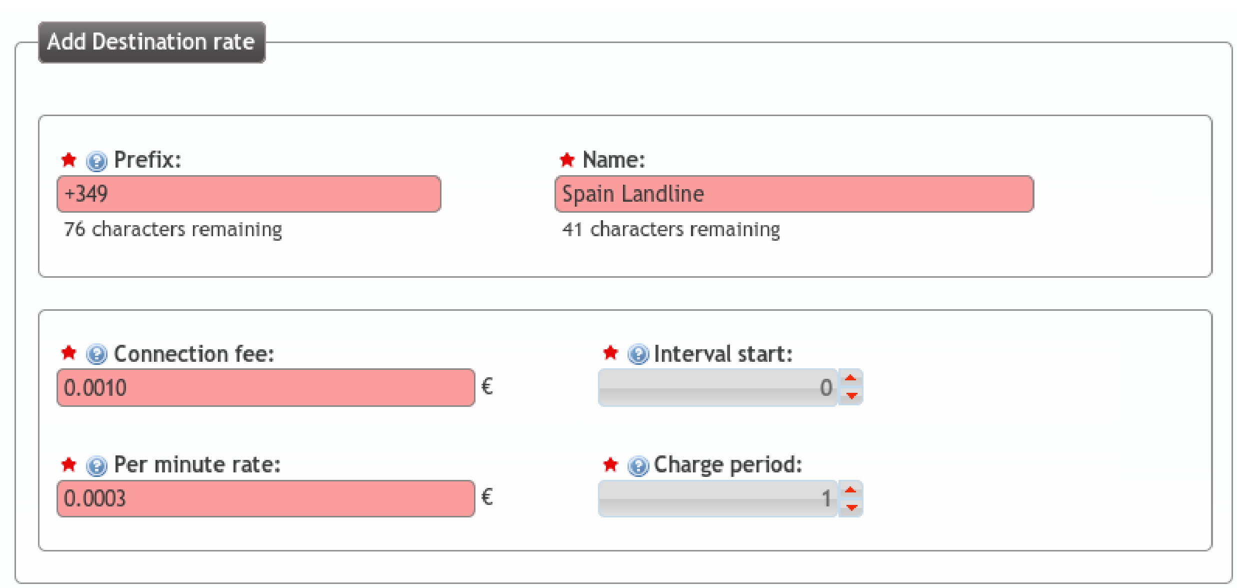
Destination rates section is empty by default, as opposed to target patterns section, that has all the 254 countries of the world. The reason is that one destination rate will usually imply lots of pattern per country (GSM networks, especial numbers, mobile numbers, fixed lines, etc.).

In most of the cases, this section data will be imported from CSV provided by your Peering Contracts, but for our test we will create it manually. Check section *Destination Rates* for more information.

Create a **destination rate**:



And we add a price for a given prefix in E.164 (with + sign)



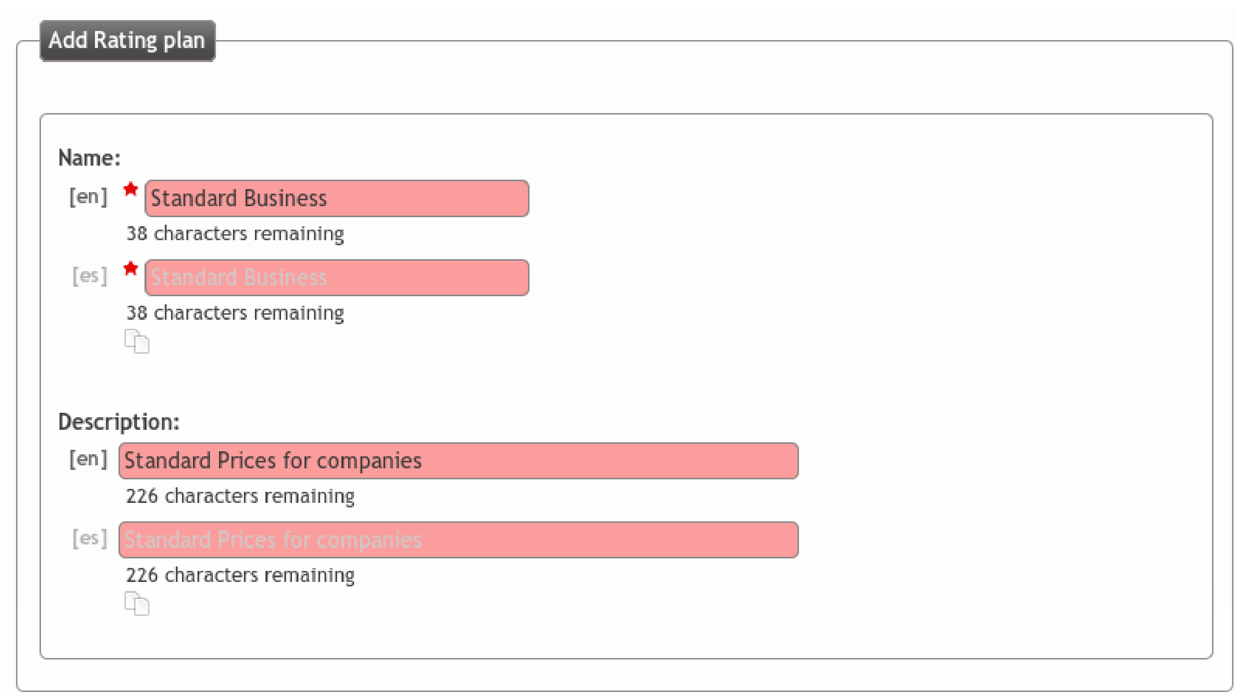
Note: Floating number must use the "." as decimal separator (e.g. 0.02)

7.6 Rating plans

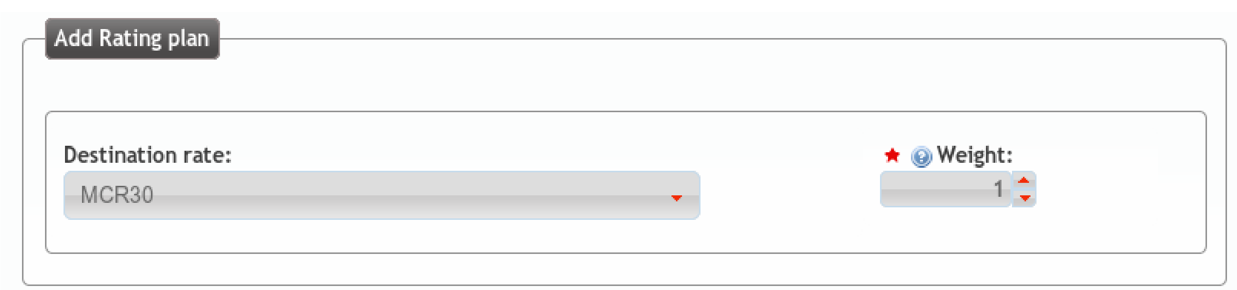
7.6.1 Creating a rating plan

Destination rates are grouped using Rating plans. This offers the possibility to have base pricing data and customize some destination with different prices.

Create a **rating plan**:



And then we can add our destination rate:



The **metric** of the link lets you assign more than one *destination rate* for a plan, even though some destinations are included in more than one of those destination rates.

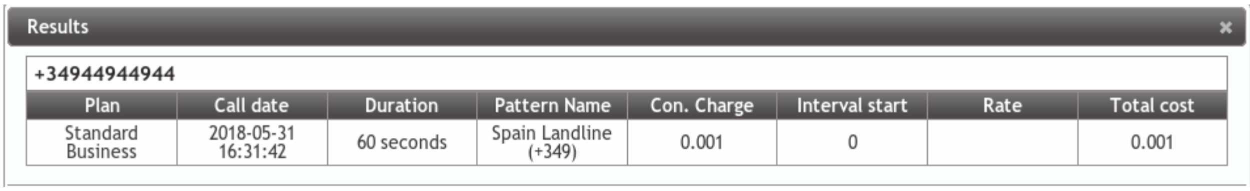
Attention: If a given call can be billed with more than one destination rate, it will be billed using the one with lowest metric.

Tip: This allows having a general *Destination rate* and concrete the price of a specific destination in another *destination rate* with lower metric (free cell phone calls, for example).

Checking Rating plans

To check the configuration so far we can **Simulate a call** from the rating plans list.

We introduce the destination number in *E.164 format*, and we can check that it matches the **rating plan** we have just created:


Results

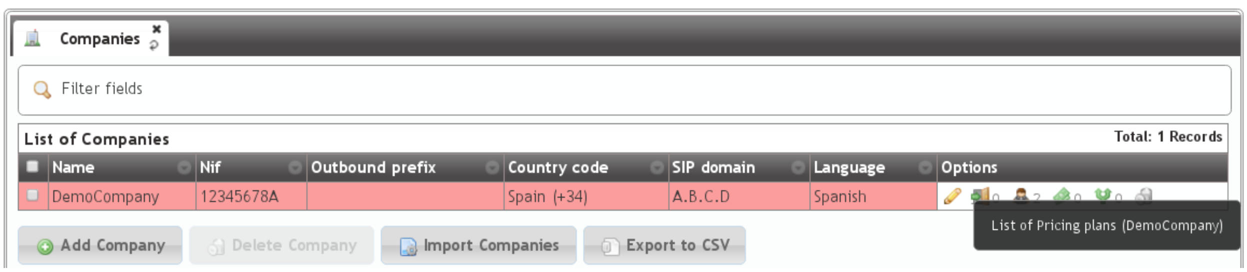
+34944944944

Plan	Call date	Duration	Pattern Name	Con. Charge	Interval start	Rate	Total cost
Standard Business	2018-05-31 16:31:42	60 seconds	Spain Landline (+349)	0.001	0		0.001

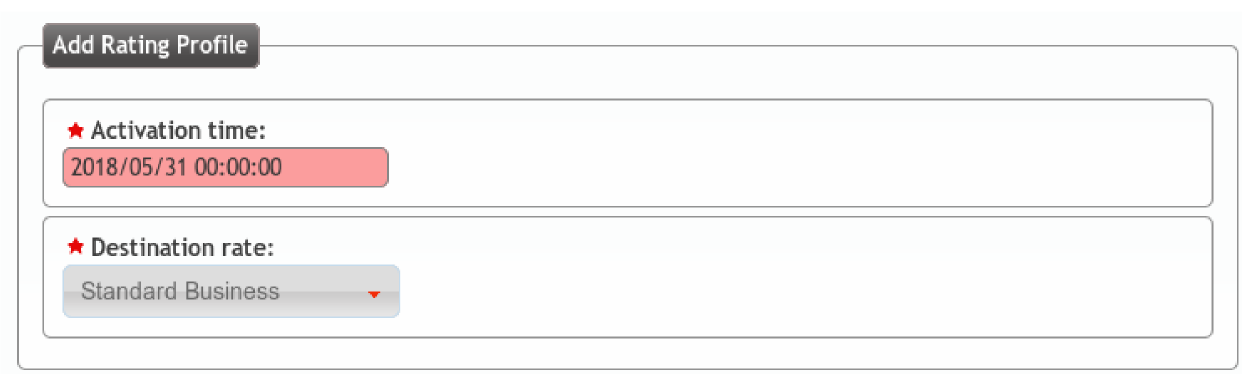
7.6.2 Assigning a rating plan to a company

A specific **rating plan** can be linked to multiple companies.

In the section **Brand configuration** > **Virtual PBXs** we select the *demo* company:



The **Rating plan** have an activation time, and only one can be active for each company.



Simulating a call of a specific company

In this list we can also simulate a call for a given company like we did previously in the rating plan list and check the price it will imply. This way, we can be sure that the configuration is ok.

7.7 Outgoing configuration complete!

That's it!

At this point, *Alice* should be able to make outgoing calls to spanish destinations and this calls should be billed accordingly.

CHAPTER 8

Platform Configuration

8.1 Brands

Name Sets the name for this brand.

TIN Number used in this brand's invoices.

Logo Used as default logo in invoices and in portals (if they don't specify another logo).

Invoice data Data included in invoices created by this brand.

SIP domain Introduced in 1.4. Domain pointing to Users SIP proxy used by all the Retail Accounts and Residential Devices of this brand.

Recordings Configures a limit for the size of recordings of this brand. A notification is sent to configured address when 80% is reached and older recordings are rotated when configured size is reached.

Features Introduced in 1.3, lets god operator choose the features of the created brand. An equivalent configuration is available in Companies, to choose between the ones that god operator gave to your Brand. Related sections are hidden consequently.

Max calls Limits both user generated and **external** received calls to this value (0 for unlimited).

Locales Define default Timezone and Language for clients of this brand.

Hint: Some features are related to brand and cannot be assigned to companies. Other ones are also related to companies and lets the brand operator to assign them to its companies.

Warning: Disabling billing hides all related sections and assumes that an external element will set a price for calls (external tarification module is needed, ask for it!).

Note: Disabling invoices hides related sections, assuming you will use an external tool to generate them.

Note: SIP domain is only visible for Brands with Retail or Residential features enabled.

8.2 SIP Domains

The section **Domains** will display the SIP domains that points to our two public IP addresses.

- Users SIP Proxy IP address
- Trunks SIP Proxy IP address

After the initial installation, there will be two domains, one for each address:

Domain	Scope	Points to	description	Options
users.ivozprovider.local	Global	proxyusers	Minimal proxyusers global domain	
trunks.ivozprovider.local	Global	proxytrunks	Minimal proxytrunks global domain	

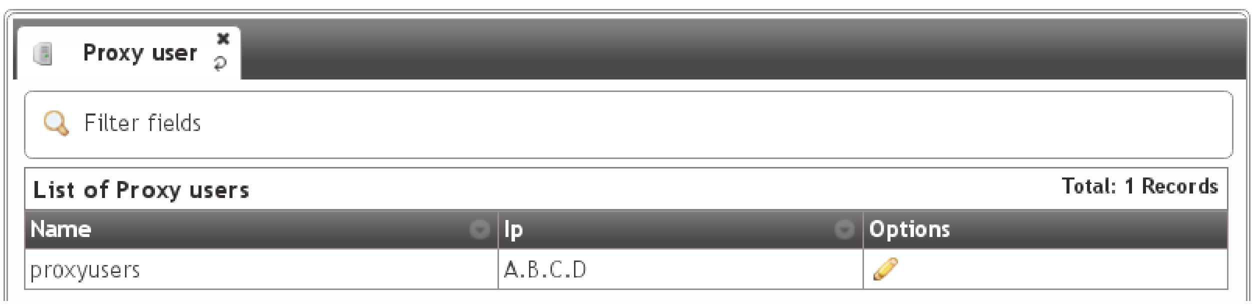
This domains will be used internally by a builtin DNS server included in the solution.

Attention: As mentioned in the section *Company SIP Domain*, each company will require a DNS pointing to the users SIP proxy. Once configured, the domain will be displayed in this list so global administrator can check what domains are registered for each company.

8.3 Users SIP Proxy

This is the SIP proxy exposed to the external world where users register their terminals.

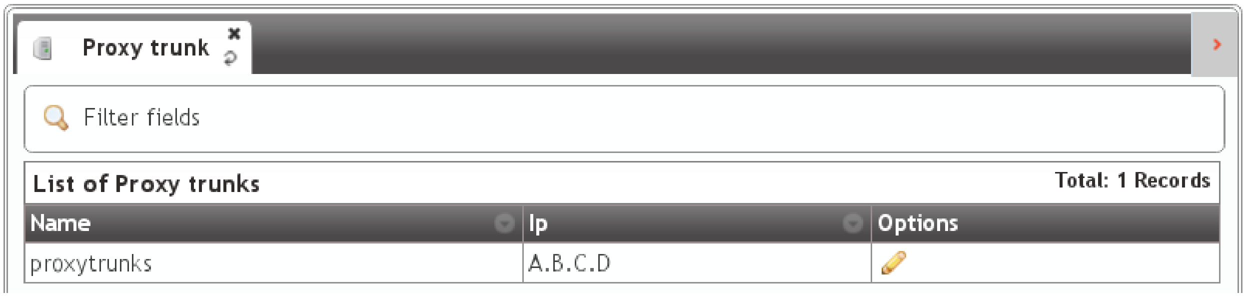
The value displayed in the section **Proxy users** will show the IP address entered during the installation process.



8.4 Providers SIP proxy

This is the SIP proxy exposed to the external world in charge of connecting the provider that brand aministrators will configure for *peering*.

The value displayed in the section **Proxy trunk** will show the IP address entered during the installation process.



Note: Only the IP address will be entered as the port will be always 5060 (5061 for SIP over TLS).

Danger: This 2 values can be changed from the portal, but they must always have the same IP address that proxy process listen to requests.

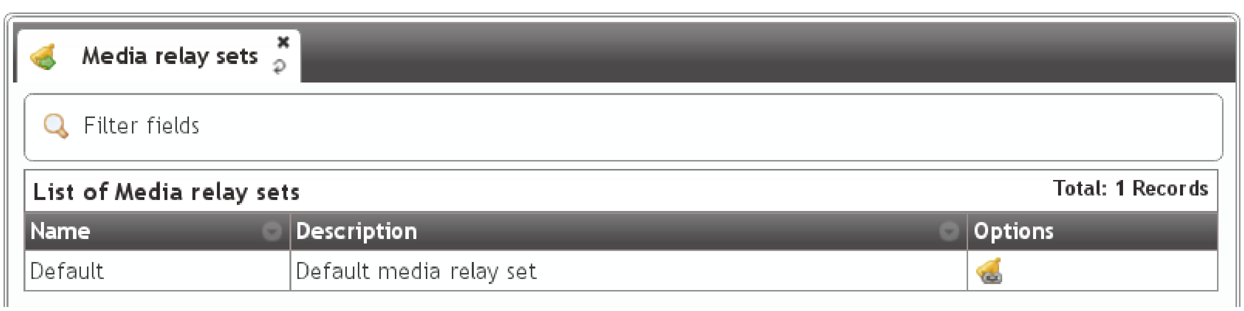
8.5 Media relay

Media relays are in charge of bridging RTP traffic of established calls. Like the Application Servers, they can scale horizontally as much as required.

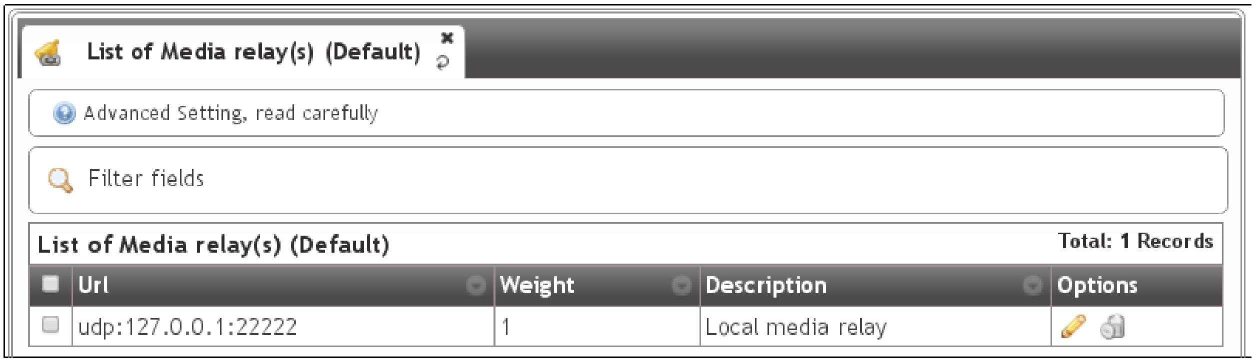
Media relays are organized in groups so they can be assigned to a company. Each element of the group has a **metric** that allows non-equal load balancing within the same group (i.e. media-relay1 metric 1; media-relay2 metric 2: the second media relay will handle two times the calls than the first one).

Hint: The static assigment of media relay groups is not the common practice but allow us to assign strategic resources to companies that need a warranted service. The most common usage of this **groups of media relays** is to place them near the geographic area of the company (usually far from the rest of the platform systems) in order to reduce **latencies** in their conversations.

In a standalone installation, only one media relay group will be exist:



By default this group only has a media server:

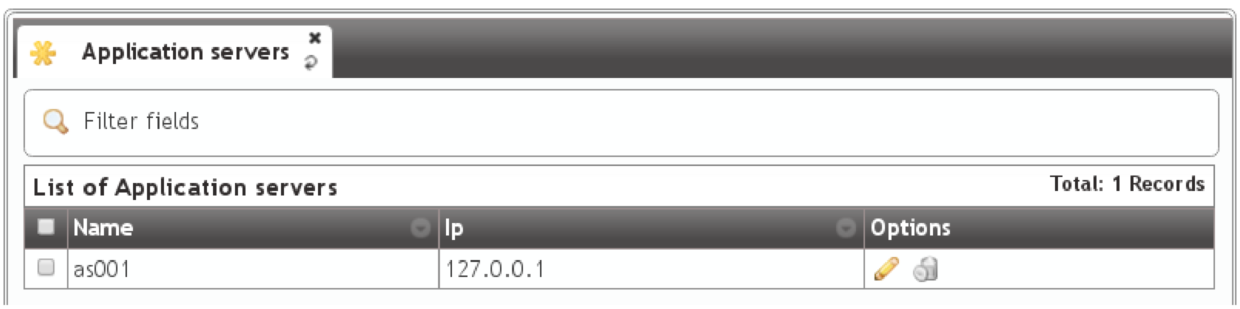


Note: The address displayed is the control socket, not the SDP address that will be included during SIP negociation. By default this alone media-relay will share the same IP address that the User's SIP proxy.

8.6 Application Servers

The section **Application Servers** will list the IP address where the existing Asterisk processes will listen for request, and like previously mentioned, can scale horizontaly to adapt the platform for the required load.

Contrary to the Proxies, Asterisk is not exposed to the external world, so for a standalone installation there will only be one listening at 127.0.0.1.



Note: The listening port will not be displayed in the field because it will always be 6060 (UDP).

Important: As soon as another Application Server is added, the proxies will try to balance load using it. If no response is received from added Application server, it will be disabled automatically.

8.7 Antiflood trusted IPs

IvozProvider comes with an *anti-flooding* mechanism to avoid that a single sender can deny the platform service by sending lots of requests. Both *proxies* (users and trunks) use this mechanism, that **limits the number of requests from an origin address in a time lapse**.

> **Warning:** When an origin reaches this limit, the proxy will stop sending responses for a period of time. After this time, the requests will be again handled normally.

Some origins are automatically excluded from this *anti-flooding* mechanism:

- Application Servers from the platform.
- Company authorized IP addresses or ranges (see previous section).

Global operator of the platform can also add exceptions to this mechanism in the section **Global configuration > Antiflood trusted IPs**.

Listado de Direcciones IP de confianza		Total:1 Registros
IP	Descripción	Opciones
5.196.32.101	I know this is safe	

8.8 Terminal provisioning

8.8.1 Overview

IvozProvider supports provisioning of terminals via HTTP/HTTPS that fulfill the following requirements:

- Assuming a just unboxed terminal, just plugged and connected to the network:
  - Ask IP address via DHCP.
  - DCHP has enabled the option 66 that points to the platform portal
  - The first requested provisioning file is a static file (different for each model) prefixed with the previous step URL.
  - The served file can redefine the URL for further requests

Any terminal model that can adapt to this provisioning way can be added into the section **Platform Configuration > Terminal manufacturers**.

Example Cisco SPA504G

- Cisco SPA504G is turned on and requests an IP address to DHCP
- Receives "http://provision.example.com/provision" as DHCP option 66
- Request HTTP configuration from http://provision.example.com/provision/spa504g.cfg
- All 504G request the same file (spa504.cfg), prefixed with the given URL
- This file only contain basic configuration settings for the model and the URL for the next request (p.e. https://provision.example.com/provision/\protect\T1\textdollarMAC.cfg)
- This way, each terminal (MAC should be unique) request a specific file (and different) after the generic one has been served.
- This file will contain the specific configuration for the terminal:
  - User
  - Password

- SIP Domain

Note: IvozProvider provisioning system, right now, only has one goal: provide credentials and language settings for the terminals.

8.8.2 Configuration of supported models

IvozProvider uses a template system that allows global operator (God) to define new models and configure what files will be served.

The help section of **Terminal manufacturers** has examples for some models that work (in the moment of writting this) with IvozProvider provisioning system.

Hint: These models will be available after the initial installation, but you must edit them and load the default configuration before you can use the provisioning system (option **Restore default template**).

Error: UACs firmware changes may cause that given examples stop working. We will try to keep templates updated, but we can't guarantee this point.

Analyzing the suggested templates you can have a basic idea of the flexibility of the system to configure any existing terminal model in the market and to adapt them to eventual changes in given examples.

8.8.3 Getting technical

Imagine an environment with this configuration:

- Provisioning URLs:
  - Generic file: http://PROV_IP/provision
  - Specific file: https://PROV_IP:PROV_PORT/provision
- TerminalModels.genericUrlPattern: y000000000044.cfg

Which requested URLs will be valid?

For generic file, just one: http://PROV_IP/provision/y000000000044.cfg

For specific file, requests are right as long as this rules are fulfilled:

- All HTTP requests are wrong.
- HTTPS requests to 443 are wrong (PROV_PORT must be used).
- Subpaths after provisioning URL are ignored, both in request and in specificUrlPattern.
- On specific file request, extension must match as long as extension is used in specificUrlPattern.
- On specific file request, the filename must match exactly once {mac} is replaced.
- MAC address is case insensitive and can contain colons or not (':').

Let's analyze the examples below to understand this rules better:

Example 1 - TerminalModels.specificUrlPattern: {mac}.cfg

Working requests:

```
https://PROV_IP:PROV_PORT/provision/aabbccddeeff.cfg
https://PROV_IP:PROV_PORT/provision/aa:bb:cc:dd:ee:ff.cfg
https://PROV_IP:PROV_PORT/provision/aabbccdd:ee:ff.cfg
https://PROV_IP:PROV_PORT/provision/aabbccddeeff.cfg
https://PROV_IP:PROV_PORT/provision/AABBCCDDEEFF.cfg
https://PROV_IP:PROV_PORT/provision/subpath1/aabbccddeeff.cfg
https://PROV_IP:PROV_PORT/provision/subpath1/subpath2/aabbccddeeff.cfg
```

Wrong requests:

```
https://PROV_IP:PROV_PORT/provision/aabbccddeeff.boot
https://PROV_IP:PROV_PORT/provision/subpath1/subpath2/aabbccddeeff.boot
```

This example is identical to 't23/{mac}.cfg', as subpaths are ignored.

Example 2 - TerminalModels.specificUrlPattern: {mac}

All previous examples are ok, as extension is ignored if no extension is found in specificUrlPattern.

This example is identical to 't23/{mac}', as subpaths are ignored.

Example 3 - TerminalModels.specificUrlPattern: yea-{mac}.cfg

All previous examples are wrong, as no 'yea-' is found ('yea' match is case sensitive).

Working requests:

```
https://PROV_IP:PROV_PORT/provision/subpath1/yea-aabbccdd:ee:ff.cfg
```

Wrong requests:

```
https://PROV_IP:PROV_PORT/provision/subpath1/yea-aabbccdd:ee:ff.boot
https://PROV_IP:PROV_PORT/provision/subpath1/YEA-aabbccdd:ee:ff.cfg
```

This example is identical to 't23/yea-{mac}.cfg', as subpaths are ignored.

Example 4 - TerminalModels.specificUrlPattern: yea-{mac}

As no extension is given:

```
https://PROV_IP:PROV_PORT/provision/subpath1/yea-aabbccdd:ee:ff.cfg
https://PROV_IP:PROV_PORT/provision/subpath1/yea-aabbccdd:ee:ff.boot
```

Wrong requests:

```
https://PROV_IP:PROV_PORT/provision/subpath1/YEA-aabbccdd:ee:ff.cfg
```

This example is identical to 't23/yea-{mac}', as subpaths are ignored.

8.9 Services

There are **special services** that can be accessed by calling to some codes **from the terminal**.

> **Danger:** Services defined in this section **are not accessible during a conversation**. They are activated by **calling the codes**, not using DTMF codes while talking.

There are the folowing **special services** avaialble in the section **Global configuration** > **Services**:

List of Services				Total: 4 Records
Iden	Name	Description	Code	Options
RecordLocution	Record Locution	Add the locution code after the service code	* 00	
Voicemail	Check Voicemail	Check and configure the voicemail of the user	* 93	
DirectPickUp	Direct Pickup	Add the capture extension after the service code	* 94	
GroupPickUp	Group Pickup	Captura la llamada de un miembro de los grupos de captura del usuario	* 95	

Direct pickup This service allows capturing a ringing call from another terminal by calling the code followed by the extension from the target user.

Group pickup This service allows capturing a ringing call for any terminal whose user is part of one of the capturer pickup groups.

Check voicemail This service allows checking the user's voicemail using an interactive menu from which new voicemails can be listen, deleted, etc. This is an active alternative to receive voicemails via the email. Since 1.4, this service allows optional extension after the service code to check another users voicemails. Users can protect their voicemail using the internal menu options.

Record locution This service allows any user to record their company's locutions by dialing an special code. Voice instructions will be provided in the user's language.

Open Lock Calling this service code will set route lock status to 'Opened' (see *Route locks*).

Close Lock Calling this service code will set route lock status to 'Closed' (see *Route locks*).

Toggle Lock Calling this service code will change the current status of the lock (see *Route locks*).

As soon as new services are implemented into IvozProvider, they will be listed in this section.

> **Attention:** This section lists the available services and the default codes when a **new brand** is created.

Hint: Changing the default code in this section will only affect new created brands.

CHAPTER 9

Brand Configuration

9.1 Virtual PBX

Virtual PBX clients are designed to provide service to clients with multiple terminals that require featureful call flows.

Hint: Some fields described below may not be visible depending on enabled features.

Name Sets the name for this client.

SIP domain DNS for this client. See *Company SIP Domain* section.

Features Allow configuration of avaialble features for this client. Related sections are hidden consequently and the client cannot use them.

Billing method When billing feature is enabled determines when calls will be priced. See *Billing* section.

Geographic Configuration General client configuration for language and timezones. Most of the settings in the section can be configured per user if required.

Security Limits the external concurrent calls and source of calls for this client.

Invoice data Data included in invoices created by this brand.

Externally rated options For *Carriers* with externally rated enabled, this field can be used to store specific information for this client.

Notifications Configure the email *Notification Templates* to use for this company.

Outgoing DDI Selects a DDI for outgoing calls of this client, if it is no overridden in a lower level.

Media relay set As mentioned above, media-relay can be grouped in sets to reserve capacities or on a geographical purpose. This section lets you assign them to companies.

Distribute Method 'Hash based' distributes calls hashing a parameter that is unique per client, 'Round robin' distributes calls equally between AS-es and 'static' is used for debugging purposes.

Application Server If 'static' *distribute method* is used, select an application server here.

Recordings Configures a limit for the size of recordings of this client. A notification is sent to configured address when 80% is reached and older recordings are rotated when configured size is reached.

Most of the features are self-explanatory, but **voice notification** deserves an explanation: if you enable them, when a call fails, the user will listen a locution explaining what occurred ("you have no permissions to place this call", "the call cannot be billed", etc.)

Warning: Recordings rotation happens at two levels: brand and client. This means that **a client's recordings can be rotated even though its limit has not arrived (or even it has no limit) if brand's limit applies first**.

Error: Again: recordings rotation happens at two levels: brand and client. This means that **a client's recordings can be rotated even though its limit has not arrived (or even it has no limit) if brand's limit applies first**.

Hint: To avoid this, make sure that the sum of all companies does not exceed the size assigned to your brand and make sure that all companies has a size configured (if 0, it has unlimited size).

Both **Distribute method** and **Application Server** are only visible for God Administrator.

Warning: 'Round-robin' distribute method is reserved for huge companies whose calls cannot be handled in a single AS. **Use 'Hash based' for remaining ones**, as 'Round-robin' imposes some limitations to client features (no queues, no conferences).

9.2 Retail

Retail are designed to provider DDI routing service to clients with self hosted PBXs..

Hint: Some fields described below may not be visible depending on enabled features.

Name Sets the name for this client.

Features Allow configuration of avaialble features for this client. Related sections are hidden consequently and the client cannot use them.

Billing method When billing feature is enabled determines when calls will be priced. See *Billing* section.

Geographic Configuration General client configuration for language and timezones. Most of the settings in the section can be configured per user if required.

Security Limits the external concurrent calls and source of calls for this client.

Invoice data Data included in invoices created by this brand.

Externally rated options For *Carriers* with externally rated enabled, this field can be used to store specific information for this client.

Notifications Configure the email *Notification Templates* to use for this company.

Outgoing DDI Selects a DDI for outgoing calls of this client, if it is no overridden in a lower level.

Media relay set As mentioned above, media-relay can be grouped in sets to reserve capacities or on a geographical purpose. This section lets you assign them to companies.

Distribute Method 'Hash based' distributes calls hashing a parameter that is unique per client, 'Round robin' distributes calls equally between AS-es and 'static' is used for debugging purposes.

Application Server If 'static' *distribute method* is used, select an application server here.

Recordings Configures a limit for the size of recordings of this client. A notification is sent to configured address when 80% is reached and older recordings are rotated when configured size is reached.

Hint: Retail clients and their accounts use Brand's domain for authentication.

9.3 Wholesale

Wholesale allows trunking services with Carriers without any application server features, focusing on concurrency and quality rather on having lots of services.

- Just make outgoing calls.
- IP authentication only (no register, no SIP auth).
- Calls go directly from users to trunks, without any application server involved.
- Support for routing tags (client can choose the outgoing route to use)

Hint: Some fields described below may not be visible depending on enabled features.

Name Sets the name for this client.

Features Allow configuration of avaialble features for this client. Related sections are hidden consequently and the client cannot use them.

Billing method When billing feature is enabled determines when calls will be priced. See *Billing* section.

Geographic Configuration General client configuration for language and timezones. Most of the settings in the section can be configured per user if required.

Security Limits the external concurrent calls.

Routing tags Select the *Routing tags* this wholesale will be able to use.

Invoice data Data included in invoices created by this brand.

Notifications Configure the email *Notification Templates* to use for this company.

Media relay set As mentioned above, media-relay can be grouped in sets to reserve capacities or on a geographical purpose. This section lets you assign them to companies.

Audio transcoding Select the codec capabilities to add to calls.

9.4 Residential

TODO https://github.com/irontec/ivozprovider/issues/442

9.5 Balances

This section displays the balance status for *Prepaid billing* and *Pseudo-prepaid billing* clients.

Following sections are available for each client:

9.5.1 Balance Movements List

Brand administators can keep track the balance movements (increase or decrease) on this account and their status after the movement.

9.5.2 Billable Calls List

List of calls that have decreased the balance, like the one described in the section *Billable calls*.

9.5.3 Balance Notifications

Brand administrators can configure email notifications when the balance is below a given threshold. See *Notification Templates* to customize the sent email.

9.6 Portals URLS

This section allows configuration of client portals:

- **Company**: Administration portal for all company types
- **User**: Special portal for Virtual PBXs users

> **Warning:**
>
> - URLs MUST be HTTPS
> - URLs MUST not end with slash /

Each URL can also configure a logo per URL, a theme and a phrase to use as the title of the portal allowing creation of corporative portals per company.

For more information about each portal options see below menu sections.

9.7 Notification Templates

Brand administators can configure the notifications sent by IvozProvider:

- Email sent when a new voicemail is received
- Email sent when a new fax is received
- Email sent when a balance is below configured threshold
- Email sent when an automatic invoice is generated
- Email sent when scheduled CDR CSVs are generated

> **Hint:** When no custom notification is configured, default ones will be used

Notifications are created in two steps: Create a notification type and add contents to the notification for each required language.

9.7.1 Creating a new notification

Brand administrators can create new notification templates in **Brand configuration > Notification templates**:

Fields are nearly self-explanatory:

Name Used to identify this notification template

Type Determine the notification type. Each notification type has its own substitution variables avaiable to replace the contents of the subject and body.

9.7.2 Adding Notification contents

Once the notification has been created, you can add different language contents. IvozProvider will automatically use the proper language based on the destination:

- For Voicemails, the user language will be used
- For Faxes, the company language will be used.

Configurable fields of each content:

Language Language of the contents.

From Name The from name used while sending emails (p.e. IvozProvider Voicemail Notifications)

From Address The from address used while sending emails (p.e. no-reply@ivozprovider.com)

Substitution variables Avaialble variables that can be used in subject and body that will be replaced before sending the email. Each notification type has its own variables.

Subject Subject of the email to be sent. You can include Substitution variables here.

Body HTML Body of the email to be sent. You can include Substitution variables here.

Hint: There is no need to create all content languages. If custom notification has some languages not defined the default contents will be used for that notification type.

9.7.3 Assigning templates to companies

Once the notification has been configured for the desired languages, Brand administrator must assign it to the Company or Retail Client that will use it. This can be done in the Notification configuration section of Company and Retail Client edit screen.

9.8 Peering

9.8.1 Carriers

Name Used to reference this Carrier.

Description Optional field with any required extra information.

Numeric Transformation Transformation that will be applied to the origin and destination of the outgoing numbers that use this Carrier (see *Numeric transformations*).

External tarification This setting requires the external tarification module and allows tarification on special numbers. This module is not standard so don't hesitate in *contact us* if you are interested.

9.8.2 Carrier Servers

A **Carrier Server** is a SIP server associated to an IP Provider. Carrier servers are used for placing outgoing calls by using *Outgoing Routing*.

Name Used to identify this Carrier Server

Description Optional field with any required extra information.

SIP Proxy IP address (or DNS registry) of the Carrier Server. You can also specify a port if it's different from 5060.

URI Scheme Supported schemes are sip and sips. Use 'sip' in case of doubt.

Transport Supported transport protocols. Use 'udp' in case of doubt.

Outbound Proxy Usually this is left empty. It can be filled with the IP address of the **SIP Proxy** domain (to avoid DNS resolution, but keeping the domain in the SIP messages). It works like a web proxy: instead of sending the SIP messages to destination **SIP Proxy**, they will be sent to the IP:PORT of this field.

Requires Authentication Some Carriers validate our platform by IP, others require each session that we want to establish. For this last case, this section allows to configure user and password for this authentication.

Call Origin Header Some Providers get origin from SIP From header. Others use the From header for accounting and need extra headers to identify the origin. In case of doubt leave **PAI** checked.

R-URI Transformations before numeric transformations This setting allow static changes to the destination of the calls before applying numeric transformation rules mentioned in *Numeric transformations*. Some digits can be stripped from the begining, add a prefix, or even, add extra parameters to the URI followinging the given format. In case of doubt, leave empty.

From header customization For those providers that show origin in other headers (PAI/RPID), it is possible that request that From User have the account code being used and from domain their SIP domain. In case of doubt, leave empty.

Tip: There are many fields to establish *peering* with multiple kind of providers, but usually with the name and SIP Proxy will be enough (for those that validate our platform by IP) and Authentication (for those that won't).

Warning: In case of defining multiple Carrier Servers for a single Carrier, IvozProvider will balance and failover using all of them. Like with Application Servers, it will disable those who doesn't respond to our requests.

9.8.3 DDI Provider Addresses

TODO https://github.com/irontec/ivozprovider/issues/442

9.9 Numeric transformations

IvozProvider is designed to provide service **anywhere in the planet**, not only the original country where the platform is installed.

A very important contept to archive this goal is the numeric transformation, that **adapts the different number format systems of the countries of the world** defined in E.164 **to a neutral format**.

There are two different transformation scenarios:

9.9.1 Incoming transformations

When a new call is received in IvozProvider matching a provider that has been configured for *peering*, we must adapt the numbers that make reference to:

- Origin of the call
- Destination of the call

Depending on the country of the provider, the international numbers will have a format or another. In this case, the spanish provider will use, for example:

- 00 + 33 + number belonging to France
- It's possible that the international numbers came without the 00 code.
- It's possible that, if the call comes from the same country that the provider, the number comes without the calling code (911234567 insted of 00 + 34 + 911234567 for Spain).

For an Ukranian provider, that doesn't use the 00 as international code:

- It will use 810 + 33 + number belonging to France.
- It's possible that even part of the internation code (00 in most of the countries of the world) the provider use specific codes as prefix.

The goal of the incoming transformation is that, no matter what numeric system the provider uses, the number will end in a general and common format.

Important: This common format is usually called E.164 and shows the numbers without internation code, but with country calling code: i.e. +34911234567

9.9.2 Outgoing transformations

In the same whay the origin and destination must adaptat incoming numbers, it will be required to adapt outgoing dialed numbers to properly work with each of the providers that will route our call.

For example, for a number with spanish number system:

- *Spanish provider*: Destination will come in E164 (+34911234567) and for this provider, we can remove the calling code (will understand it belongs to its country), so the number sent to them will be 911234567.
- *French provider*: The destination will come in E164 (+34911234567) and we must add the international code for France, so the number sent to them will be 0034911234567.

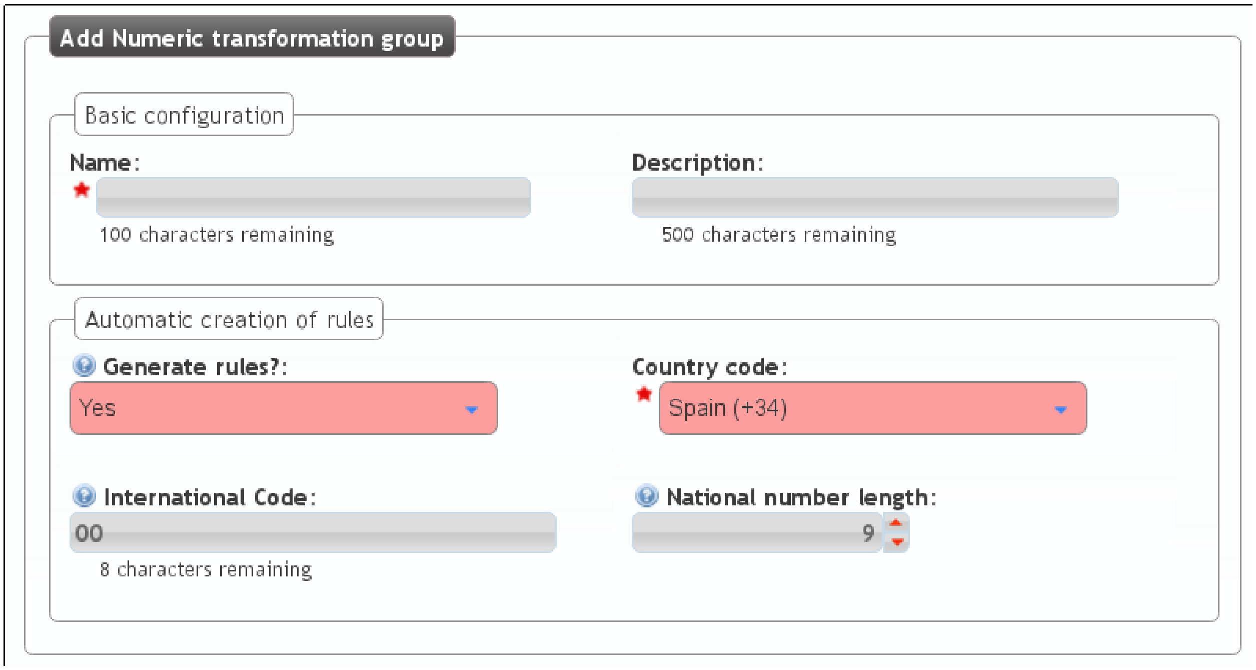
Note: To sum up, we aim to send the origin and destination in the format the provider is expecting.

Tip: Numeric transformation uses simple regular expresions to describe the changes done to the numbers. You can find multiple tutorials on net with the basic regular expression format.

'National provider' transformations

IvozProvider comes with an automatic transformation rules generator that fits with most of the countries.

In order to create a new set of transformations for spanish provider:

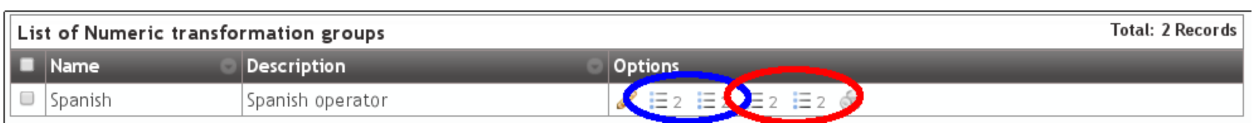


The rules that has been auto-created will transform the numbers for spanish providers that follow this rules:

- A spanish number: Neither international nor calling code (34).
- Not a spanish number: International code (00) and calling code (34).

The numeric transformation *sets* must be assigned to **Carrier**, as shown in the following section. This *set* can be shared by multiple spanish providers.

Let's check this *set* to understand what transformation rule does:



> **Attention:** The automatic rule generation will create 8 common rules based on the given parameters. This rules can be edited later to match the provider requirements.

9.9.3 Spanish incoming transformation

Displayed in blue in the previous image:

- Left called/destination
- Right callee/origin

The same rules will be applied for the origin and destination:

- The **metric** field will be used to order the rules (smaller first).

  - If a rule doesnt *match*, the next rule is evaluated.
  - If a rule *matches*, no more rules are evaluated.
  - If no rule *matches*, no change is applied.

- The **Search** field is evaluated agains the number (depending of the tranformation type it will be destination or origin).
- The **Replace** field will use the capture groups that matched the Search field (displayed between brackets, 1 for the first one, 2 for the second one, and so on) to determine how the number will end.

9.9.4 Spanish outgoing transformation

Following the same logic, this 2 rules make the change of the outgoig external destination numbers.

Attention: **To sum up**: numeric tranformation can adapt origin and destination numbers to E.164 for the platform, and to providers expected formats, based on regular expresions and metric that can be grouped in *sets* to be shared between multiple **Carriers**.

9.10 Generic Music on Hold

Music on Hold will be played when the user holds the call and the other member waits until the call is resumed.

If a company has defined a music on hold, it will be played. Otherwise, the one defined by the brand administrator in this section. If none of this is configured, a global music will be played.

Note: Multiple files can be added to be played as Music on Hold. The system will choose them randomly for each call.

9.11 Brand Services

The section **Brand configuration** > **Services** allows the brand operator to:

- Change the default service code for all the brand companies (assuming the company hasn't already customized the code)
- Delete services so it won't be available for the companies.

By default this list has all the services and codes from the global configuration:

Attention: This section lists the available services and the default codes when a **new company** is created.

Hint: Changing the default code in this section will only affect new created companies.

9.12 Generic Match Lists

Match Lists are designed to group well known numbers or patterns in order to use them in specific treatments.

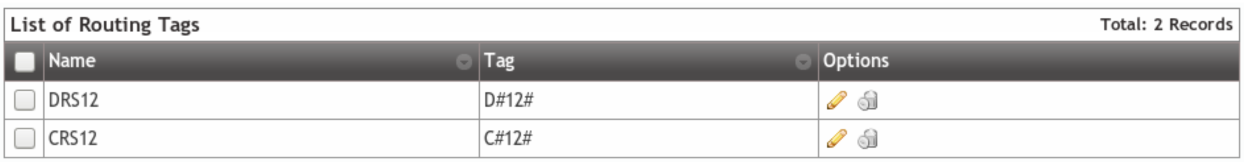
Brand administrators can create generic Match lists to share between all its companies. This can be specially handy for blacklisting numbers or configure global *Call ACL Matchlits*.

9.13 Outgoing Call Routing

9.13.1 Routing tags

In most scenarios, Brands administrators are responsible of configuring *Carriers* and *Outgoing Routing* to provide connectivity for their clients. But in some cases, clients want to choose the outgoing routing to use per call.

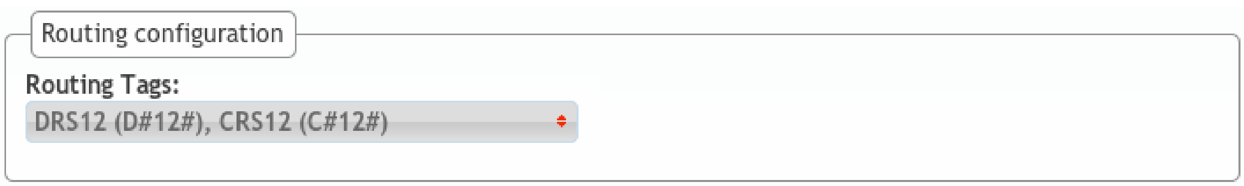
A Routing tag is an code that will prefix the destination number when placing calls to Ivoz Provider.



List of Routing Tags — Total: 2 Records

Name	Tag	Options
DRS12	D#12#	
CRS12	C#12#	

In order to use a routing tag, the client must have it enabled in their edit screen. Using a non enabled routing tag will case the call to be declined.



Important: Route tags are only available to wholesale clients at the moment.

Then, each routing tag can be used to configure their *Outgoing Routing* order.

List of Outgoing routings — Total: 2 Records

Company	Routing Tag	Type	Destination	Peering contract	Priority	Weight	Options
Apply to all companies	CRS12 (C#12#)	Group	Europe	Carrier One	1	1	
Apply to all companies	DRS12 (D#12#)	Group	Europe	Carrier Two	1	1	

9.13.2 Outgoing Routing

Priority If a call matches several routes, it will be placed using the outgoing route with lower priority, as long as it is available.

Metric If a call matches several routes with equal priority, metric will determine the proportion of calls that will use one route or another.

Note: This are the key parameters to achieve two interesting features: **load-balancing** and **failover-routes**.

Load balancing

Load-balancing lets us distribute calls matching the same pattern using several valid outgoing routes.

Example 1

- Route A: priority 1, metric 1
- Route B: priority 1, metric 1

Call matching these routes will use route A for %50 of the calls and route B for %50 of the calls.

Example 2

- Route A: priority 1, metric 1
- Route B: priority 1, metric 2

Call matching these routes will use route A for %33 of the calls and route B for %66 of the calls.

Failover routes

Failover route lets us use another route whenever the main route fails.

Example

- Route A: priority 1, metric 1
- Route B: priority 2, metric 1

All calls matching these routes will try to use route A. In case the call fails, the call will be placed using route B.

Tip: Although given examples use two routes, more routes can be chained and failover and load-balancing estrategies can be combined.

9.14 Billing

This block is essential for brand administrators as we will explain how to:

- Create pricing plans to assign a price to calls made by final users.
- Create invoices that show call details and global costs of each of their companies.

We will cover this topics:

9.14.1 Billable calls

Billable calls sections only list **calls that imply cost**.

Important: *Call Registry* sections, on the other hand, show all calls, even the ones that do not imply cost, such as internal calls, incoming calls, etc.

These lists therefore include the price of each call (once it is calculated). Since companies are notified about its call's price via invoices issued by **brand operator**, this section is only available at two levels:

- Main level (god level)
- Brand level

Each entry shows this information:

Date Date and time of the call establishment.

Brand Only visible for *god*, shows the brand of each call.

Company Visible for *god* and *brand operator*, show the company of each call.

Destination External number dialed.

Duration Shows how long the call lasted.

Price The cost of the call.

DDI Provider Shows which *DDI Provider* was used for each call.

Invoice Show if a call is already included in any *Invoices*.

Note: As soon as the call is hung up, they appear in this list. In some minutes time the asynchronous process will set *Metered* to 'yes' and will assign a **price**.

9.14.2 Destination Rates

- A Destination rate groups some prefixes with their cost details:
  - Cost per minute
  - Call establishment cost
  - Bill by seconds, by minutes, etc.

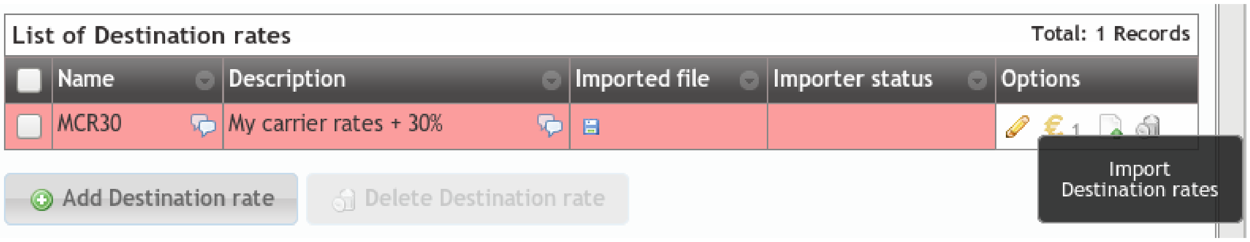
At this point, the future brand operator may have noticed that creating thousands of pricing patterns would be a really annoying and time consuming task, as there are 254 countries, each of them with their mobile networks, landline networks, special service numbers, etc.

That's why the creation of destination reate is done using a CSV file. If you still want to create data manually, check section *Creating a destination rate*

Importing a CSV file

The first step is creating an empty rating plan to import the prices in (section **Brand configuration > Destination rates**):

This is the key button for the massive pricing pattern import process:



We can select which column contains which field, in case we want to import a CSV file in a non-recommended format. We can also decide whether to import the first line or discard it as it may have titles instead of data.

Hint: The importing process is done in background, letting the brand operator continue doing other stuff while it is finished.

CSV format

Although the above window allows importing non-recommended format CSV files, we encourage you to import a file in the proposed format, as it will make this process much easier.

You can find a sample CSV for importing here.

The order of the columns should be:

- Destination name
- Destination prefix (E.164 with + sign)
- Per minute charge
- Establishment cost
- Billing period in seconds

Note: It is recommended to double quote alphanumeric entries, though it is not compulsory for single word entries (or entries without odd symbols). **If they contain any comma, they MUST be quoted**.

Error: Floating numbers **MUST use point as decimal separator**.

Note: Numeric entries can be quoted with double quotes, but it is not mandatory.

Warning: The price of the call will be increased every billing period unit.

- If *billing period* is set to 1, every second the price will be increased *price per minute* divided by 60 (bill by seconds).
- If *billing period* is set to 60, every minute the price will be increased *price per minute* (bill by minutes).

You can download the imported file of the destination rate. Take into account that while importing over existing data, the mathching values are overwritten and the not matching are kept. This allows downloading the imported file, changing some values and importing pricing back.

Once the import process is over, we only have to include this destination rate into some rating plan and bind it to the companies we want following the procedure explained in *Rating plans*.

9.14.3 Call billing

Billing a call is the **action of setting a price** to a call that implies cost.

Automatic billing

As exposed previously, billing calls depends upon an automatic process:

- When a call is about to be established, IvozProvider verifies that it will be able to bill it.
  - If with the current configuration (active and applicable rating plans for a given company and for the specific destination) it won't be possible to bill the call, IvozProvider will prevent its establishment.
- Once a call that implies cost is hung up, it is listed in *Billable calls*.

Postpaid billing

- Call tarification is done after the call ends
- No configurable limit or balances involved

Prepaid billing

- Call trarification is done during the call
- Clients with prepaid have a preconfigured balance that will be decrement during the call
- When the balance reaches zero, all established calls for the client will hang up
- Clients can not place new calls with zero or below balance
- Low balance email notifications can be configured

Pseudo-prepaid billing

- Call trarification is done during the call
- Clients with pseudo-prepaid have a preconfigured balance that will be decrement after the call ends
- Clients can not place new calls with zero or below balance
- Low balance email notifications can be configured

9.15 Invoicing

The final goal of this section is to generate invoices with the call that imply cost of a given company.

9.15.1 Invoice templates

Before generating an example invoice, it is important to understand that invoice creation process uses templates.

Hint: This way, every **brand operator** can adapt which information is shown and how this information is shown, add logos, graphs, etc..

Templates are parsed by handlebars and rendered using wkhtmltopdf library.

The helper in the section **Brand configuration** > **Invoice templates** include a summarized explanation of the creation of templates. In the official site of wkhtmltopdf there is plenty additional information. You can delve into template expressions here as well.

By default, this section provides some basic example templates:

Listado de Plantillas de facturas — Total:4 Registros

Nombre	Descripción	Opciones
Básica	Básica	
Costes fijos	Básica + costes fijos	
Llamadas entrantes	Básica + Llamadas entrantes	
Costes fijos y Llamadas entrantes	Básica + Costes fijos + Llamadas entrantes	

9.15.2 Fixed costs

Fixed cost are a constant concept that can be added to invoices that use invoice templates that take into account these fixed costs.

Take this image as an example (section **Fixed costs**):

Listado de Costes fijos — Total:3 Registros

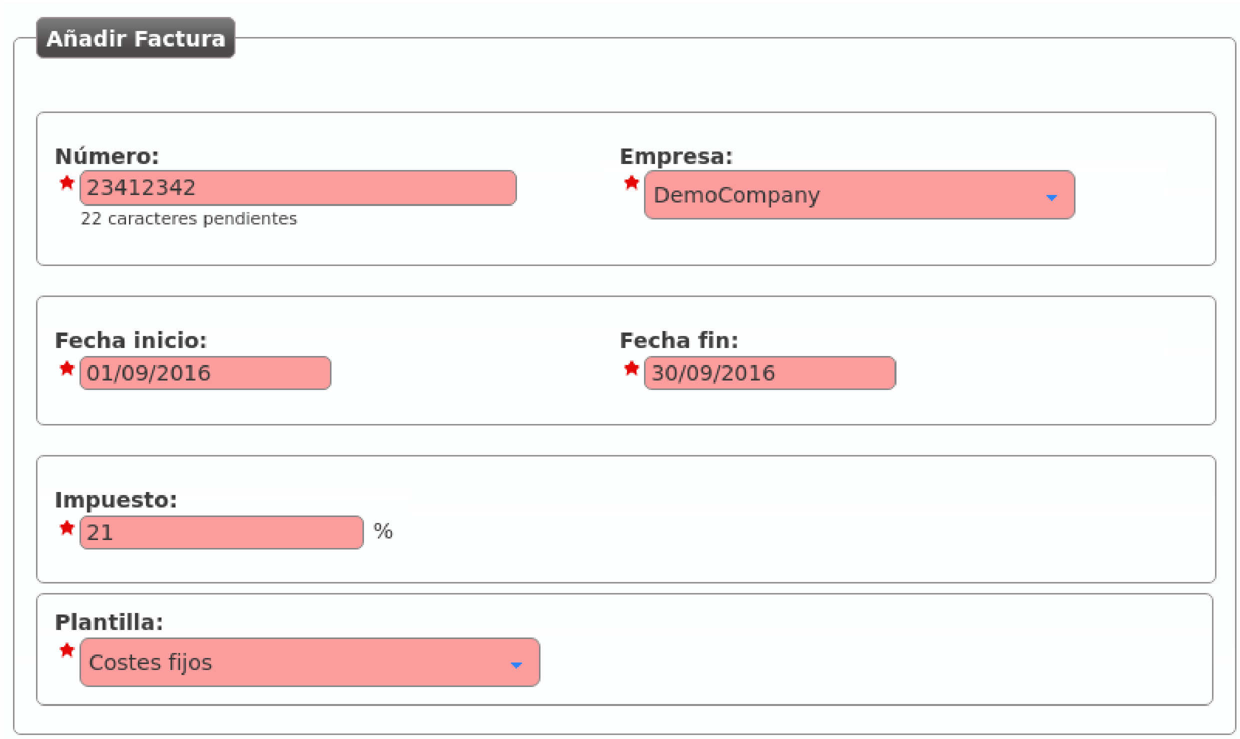
Descripción	Coste	Opciones
Alquiler de equipos	20 €	
Servicio técnico premium	7 €	
Líneas FTTH	50 €	

We will explain afterwords how these fixed costs can be added to a invoice and in what amount.

9.15.3 Invoices

Invoices section lets **brand operator** to generate invoices to issue to its clients.

This is the process to add a create a new invoice:



Number Will be included in the invoice and shows the invoice number

Company The company whose calls will be invoiced

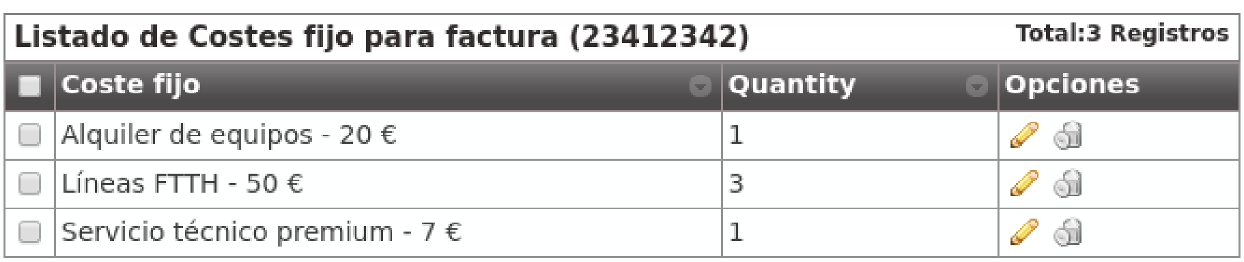
Start/End date The time period of the calls that will be invoiced

Taxes Taxes to add to the final cost (e.g. VAT)

Template Invoice template that will be used

Let's add some fixed costs to this invoice:

Select previously defined fixed costs and their amounts:



At this point, we can generate the invoice pressing this button:

Pressing this button we can see which calls have been included in the invoice:

And pressing this one we can download the invoice in PDF format:

Warning: End date must be a past date. In other words, it is not allowed to generate invoices for future dates o dates including today.

Error: All the calls of the selected period must be billed.

9.15.4 Number Sequences

TODO https://github.com/irontec/ivozprovider/issues/442

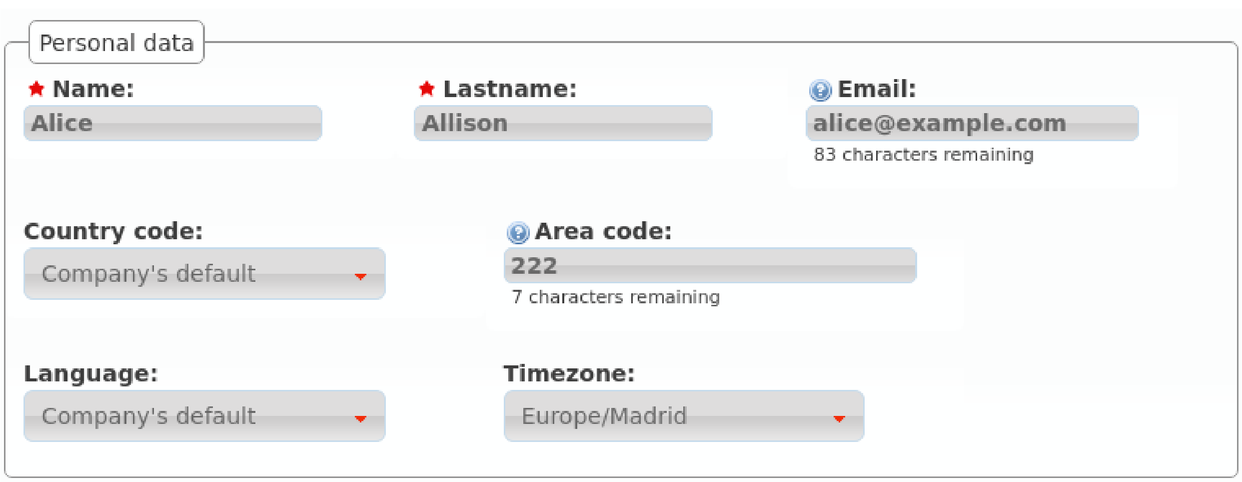
Company Configuration

10.1 Users configuration

The installation process creates *Alice* and *Bob* users, allowing us to test internals calls between them without too much effort.

We skipped most of the settings in **Users** configuration that we will described in this section.

10.1.1 Personal data



Name Used to identify this user in most of the screens. This is also the name that will be displayed in internal calls made from this user.

Lastname Most of the times this is used to complete the previous field.

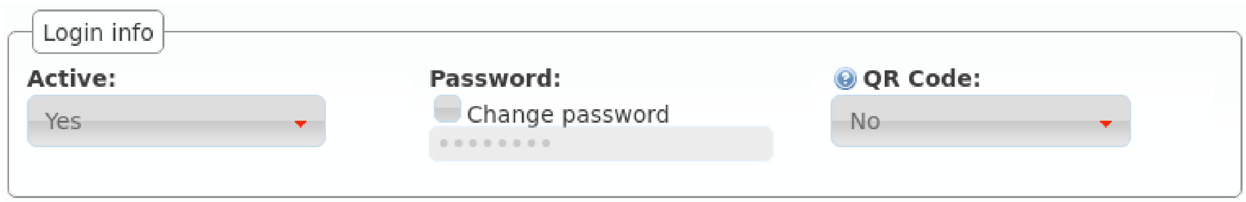
Email Email used to send the user's received voicemails. This is also used to identify the user in their portal.

Country code / Area code Defines the way the user calls and the way the numbers are presented to this user.

Language When a locution is played to this user, this language is used.

Timezone User portal call list times will use this timezone.

10.1.2 Login Info

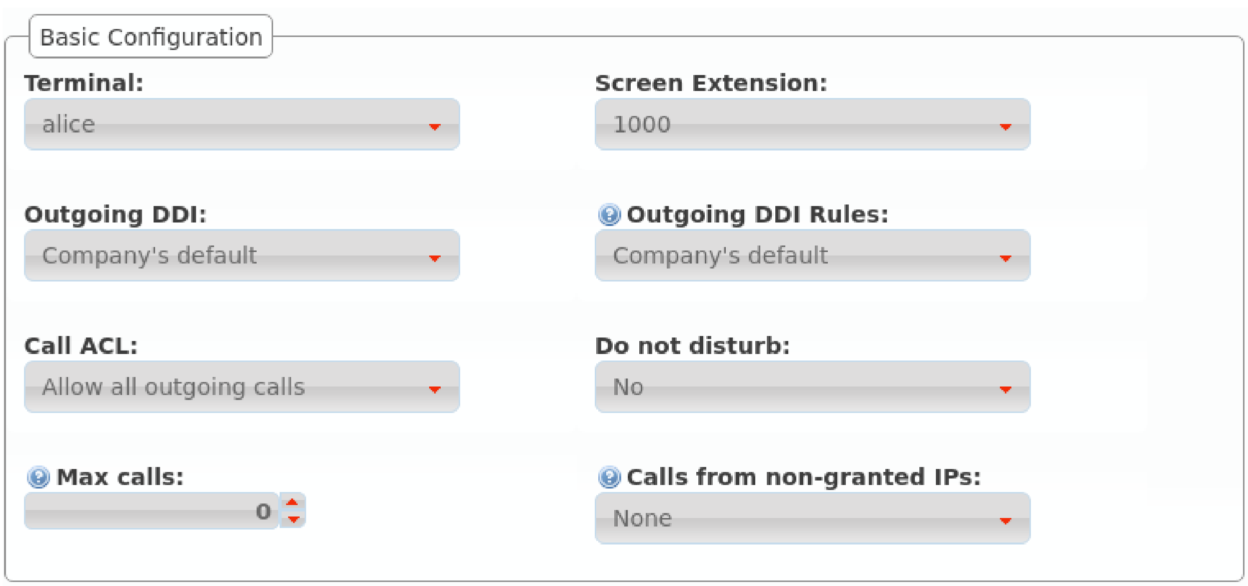


Active Allows administrators to grant or disable user's acces to the *user's portal*.

Password Password used to access the *user's portal*.

QR Code If enabled, a QR code for Grandstream Wave softphone configuration will be shown.

10.1.3 Basic Configuration



Terminal The available terminals created in *Terminals configuration* are listed here for assignment.

Screen Extension One of the available *Extensions* that this user will display when placing internal calls. While multiple extensions can be routed to the user, only one of them will be presented when the user calls.

Outgoing DDI As described in *Outgoing DDI configuration*, determines the number that will present when placing external outgoing calls.

Outgoing DDI Rules Manages exceptions to previous setting. Read *Outgoing DDI Rules* for further reference.

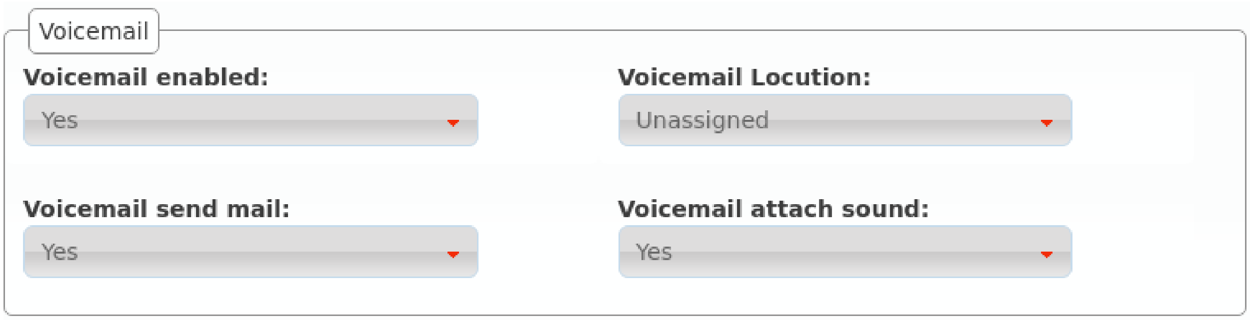
Call ACL One of the created *Call ACL* groups, described it the previous sections.

Do not disturb When this setting is enabled, the user won't receive any call but can still place calls.

Max Calls Limits the number of received calls if the user is handling simultaneously (inbound and outbound) more than the number set. Set 0 for unlimited calls.

Calls from non-granted IPs: Enable calling from non-granted IP addresses for this user. It limits the number of outgoing calls to avoid toll-fraud. 'None' value makes outgoing calls unlimited as long as company IP policy is fulfilled. Read *Roadwarrior users* for further reference.

10.1.4 Voicemail



VoiceMail enabled Enables or disables the **existance** of a users voicemail. This only makes the voicemail available to be routed as described in the section *forward to voicemail*.

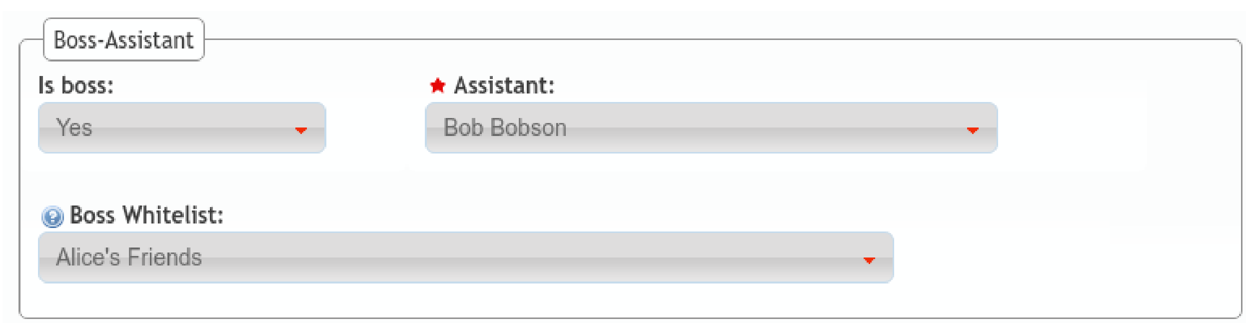
Voicemail Locution If set, this locution is played as voicemail welcome message when a voicemail for this user is going to be recorded. This only applies for call forwardings to voicemail described in the section *forward to voicemail*.

Email notification Send an email to the configured user address when a new voicemail is received.

Attach sounds: Attach the audio message to the sent email.

Note: If voicemail locution is not assigned, default locution will be used as long as the user has not recorded a custom message through the voicemail menu (calling to voicemail service code).

10.1.5 Boss-Assistant



This feature will turn the user into a boss that can only be directly call by:

- The selected assistant.
- Any origin that matches the white list.

The rest of the calls to *a boss* will be redirected to the assistant.

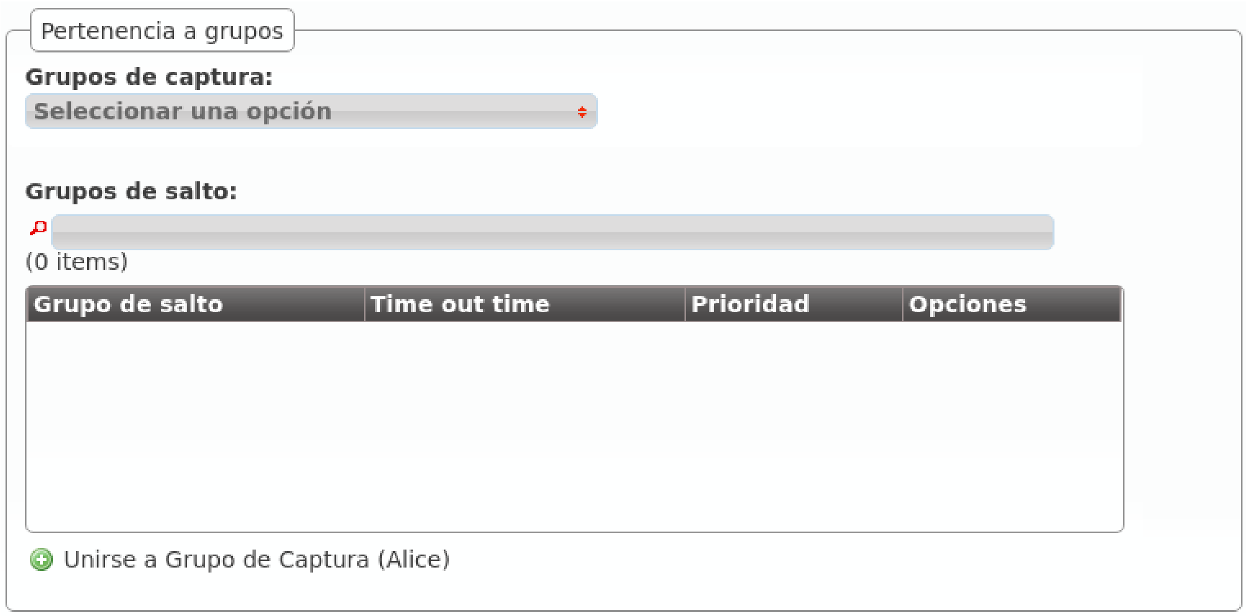
Is boss Determines if this user is a boss.

Assistant Who will receive the redirected calls of this boss.

Whitelist *Match Lists* with origins that are allowed to call directly to the boss.

With the setup in the image, every call to *Alice* will be redirected to *Bob*, except the ones placed by *Bob* itself and those coming from any origin that matches *Alice's friends* matchlist.

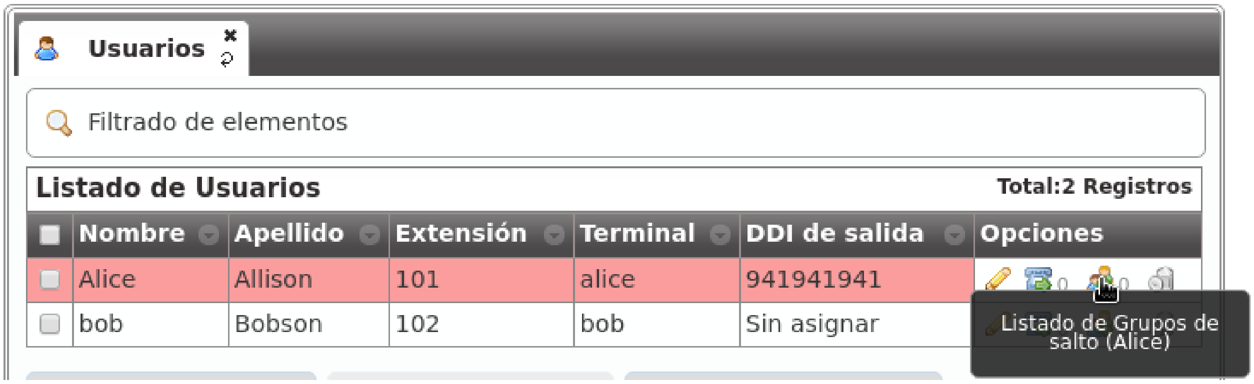
10.1.6 Group Configuration



As described in the sections *Hunt groups* and *Call pickup*, the user can be part of one or more huntgroups and pickup groups.

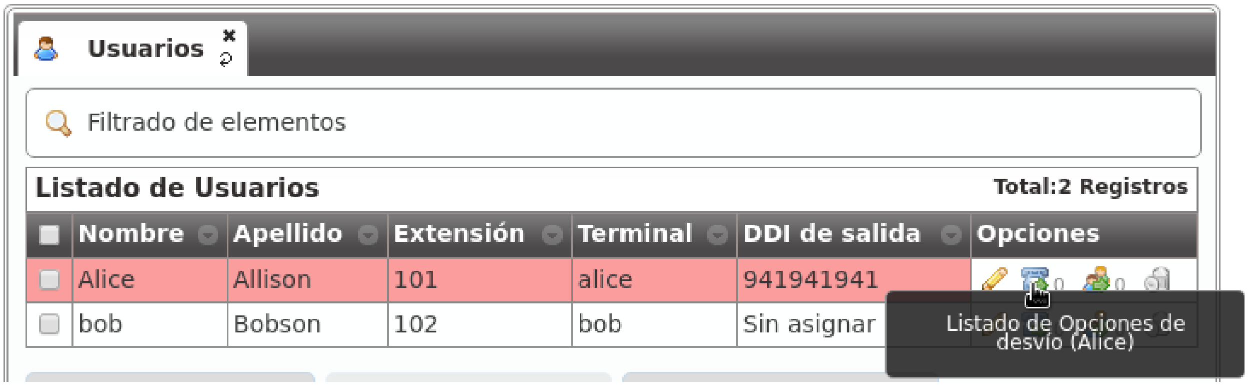
Those groups can be configured from the sections *Hunt groups* and *Call pickup* or the user's screen if the groups already exists.

You can also configure the user's **hunt groups** from the icon in each user line of the users list.



10.1.7 User Call Forward

The user's call forward can be configured in the following button:



For example, to forward all external calls that are not answered after 15 seconds, we could configure a call forward like this:

These are the fields and available values:

Call Type Determines if the forward must be applied to external, internal or any type of call.

Forward type

When this forward must be applied:

- Inconditional: always
- No answer: when the call is not answered in X seconds
- Busy: When the user is talking to someone (and call waiting is disabled), when *Do not disturb* is enabled or when the user rejects an incoming call.
- Not registered: when the user SIP terminal is not registered against IvozProvider.

Target type

What route will use the forwarded call.

- VoiceMail
- Number (external)
- Extension (internal)

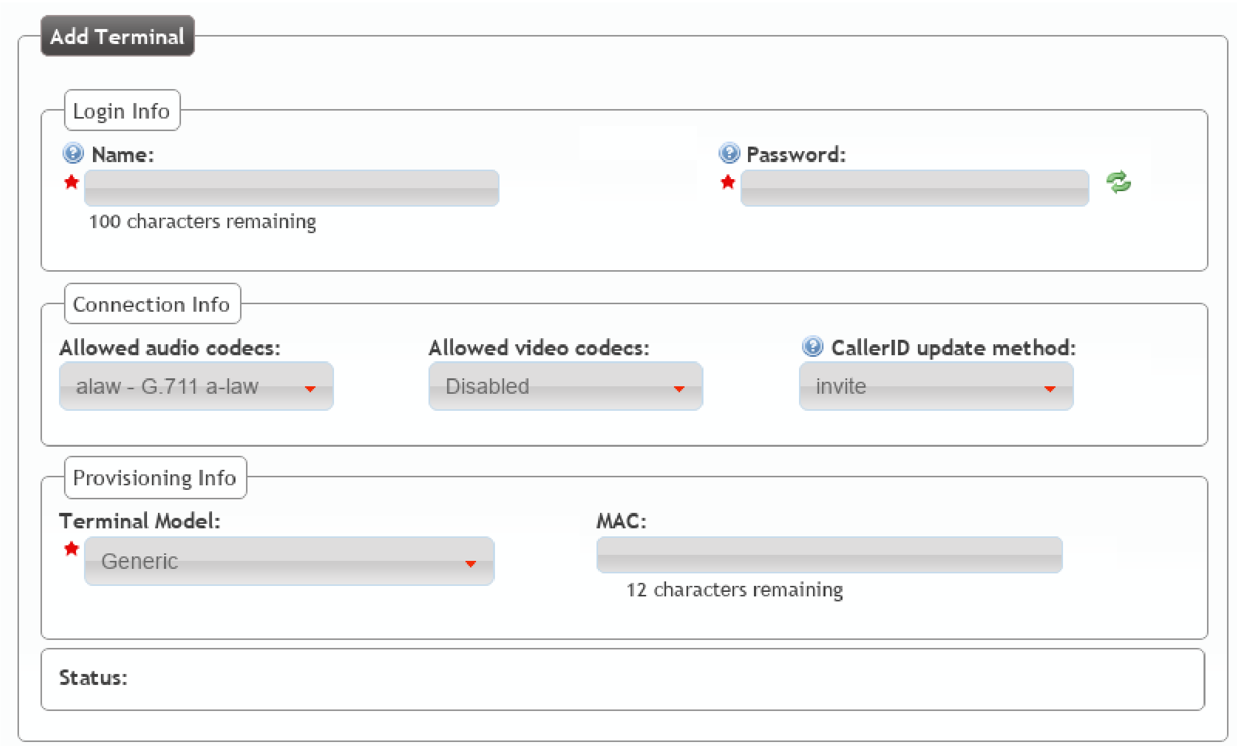
Hint: If we want to forward to other process, we can create an extension routed to that object and use the target type

Extension.

10.2 Terminals configuration

The section **Company configuration** > **Terminals** allows creating new SIP credentials that can be used by multiple SIP devices to place and receive calls from IvozProvider.

The best way to understand this section is creating a new item and see the fields that must be filled.



Name Username that will use the terminal during the SIP authentication phase with IvozProvider.

Password Password that will use the terminal to answer the SIP authentication challenge. You can use the automatic password generator to fullfill the secure password requirements.

Allowed/Disallowed codecs Determines what audio and video codecs will be used with the terminal.

CallerID update method Choose the SIP method the terminal prefers to received the session update information: INVITE or UPDATE. The help hint can be used as guide to configure different terminal manufacturers. Use *INVITE* in case of doubt.

Terminal model Determines the provisioning type that will receive this terminal. The section *terminal provisioning* will explain in depth the different models for automatic provision. If your device does not require provisioning, just select *Generic*.

MAC Optional field that is only required if you plan to use IvozProvider *terminal provisioning*. This is the phisical address of the network adapter of the SIP device.

Note: For **most of devices** that doesn't require provisioning just filling **username** and **password** will be enough.

Hint: Once the terminal has been created, most devices will only require the name, password and *Company SIP domain* in order to place calls.

10.3 Extensions

Note: **An extensions is**, by definition, **an internal number with an assigned logic**.

Create a new extension

Number The number that must be dialed by the internal user that will trigger the configured logic. It must have a minimum length of 2 and must be a number.

Route This select will allow us to choose the logic that will use this extension when is dialed from an internal user. Depending on the selected route, and additional select or input will be shown to select the hungroup, conference room, user, etc.

Warning: If an extension has a number that conflicts with an external number, this external number will be masked and, in practice, will be unavailable for the whole company.

10.4 Friends

Friends section in the **Company configuration** allows interconnection of IvozProvider with other SIP PBX systems through a SIP *trunk*. The most typical use case is when a company have multiple PBX systems that want to integrate in a single flow.

Warning: It's important to understand the difference between **Contract peering** defined by the **brand operator** to connect with the public network and **Friends**, defined by **company administrators** to connect the system with other PBXs.

10.4.1 What does this allow?

This sections allows not just communication between users at boths ends of the SIP *trunk*, but also:

- Users "from the other side" can call to the public network just like native Ivozprovider *Users*.
- Public network calls can be routed to the other SIP *trunk* end.

10.4.2 Types of friends

There are 2 main types of SIP PBX that can be integrate with IvozProvider:

- **Direct connection PBX**: IvozProvider must be able to talk SIP directly with this kind of friends by just redirecting the traffic to the proper port of the public IP address of the PBX.

- **PBX behind NAT**: Not directly accesible. This kind of PBX must register at IvozProvider (just like all the *Terminals* do).

10.4.3 What kind of calls can be routed through a *friend*?

IvozProvider must know what calls must be routed to the different defined *friends*. For that, **company administrator** will configure regular expressions that describe the numbers that *can be reached* through the **friend**.

Note: Internal *extensions* have priority over any expression defined in the *friends*.

To sum up, IvozProvider will route a call received by a *user* or a *friend* following this logic:

1. Destination matches an existing IvozProvider extension?
2. If not: Destination matches any *friend* regular expression?
3. If not: This is an external call.

10.4.4 Configuration

The **Friend** configuration is a merge between a **User** and a **Terminal**

Hint: **Friends** are so similar to **Users** that both talk SIP with the *Users SIP Proxy*.

This are the configurable settings of *friends*:

Name Name of the **friend**, like in **Terminals**. This will also be used in SIP messanges (sent **From User**).

Description Optional. Extra information for this **friend**.

Priority Used to solve conflicts while routing calls through **friends**. If a call destination **matches** more than one friend regular expresion the call will be routed through the friend with **less priority value**.

Password When the *friend* send requests, IvozProvider will authenticate it using this password. Like in terminals **using password IS A MUST**.

Direct connection If you choose 'Yes' here, you'll have to fill the protocol, address and port where this *friend* can be contacted.

Call ACL Similar to *internal users*, friends can place internal company calls without restriction (including Extension or other Friends). When calling to external numbers, this ACL will be checked if set.

Fallback Outgoing DDI External calls from this *friend* will be presented with this DDI, **unless the source presented by friend is a DDI that exists in DDIs section**.

Country and Area code Used for number transformation from and to this friend.

Allowed codecs Like a terminal, *friends* will talk the selected codec.

From domain Request from IvozProvider to this friend will include this domain in the From header.

Note: Calls to *friends* are considered internal. That means that ACLs won't be checked when calling a friend, no matter if the origin of the call is a user or another friend.

10.4.5 Asterisk as a friend

At the other end of a friend can be any kind of SIP entity. This section takes as example an Asterisk PBX system using SIP channel driver that wants to connect to IvozProvider.

register

If the system can not be directly access, Asterisk will have to register in the platform (like a terminal will do).

Configuration will be something like this:

```
register => friend-name:friend-password@ivozprovider-company.sip-domain.com
```

peer

```
[nombre-friend]
type=peer
host=ivozprovider-company.sip-domain.com
context=XXXXXX
disallow=all
allow=alaw
defaultuser=friend-name
secret=friend-password
fromdomain=ivozprovider-company.sip-domain.com
insecure=port,invite
```

Warning: *Friends*, like terminals, MUST NOT challenge IvozProvider. That's why the *insecure* setting is used here.

10.4.6 Summary

The key point is understanding that a *friend* has a direct relation with the extension-user-terminal trio:

- Can place calls to all internal extensions and other friends.
- Can place external calls that its ACL allows
- Display their configured outgoing DDI when calling to external entities
- Never challenge IvozProvider requests (don't request authentication on received requests)
- Answers IvozProvider authentication challenges (All request from them to IvozProvider must be autheticated for security reasons)
- Only connects with *Users SIP Proxy*, like terminals. In fact, SIP traffic from friends are identical to any other user terminal traffic in format.

10.5 DDIs

Country The country of the new created DDI. Used for E164 standarization.

DDI The number, without country code.

DDI Provider The *DDI Provider* that provides this number. This relation allow the platform to apply the required *Numeric transformations* in order to determine its standard form.

External Call Filter Allows configuration based on Calendars and Schedulers as shown in *External Call filters*. Leave empty if you don't need to apply any kind of filter.

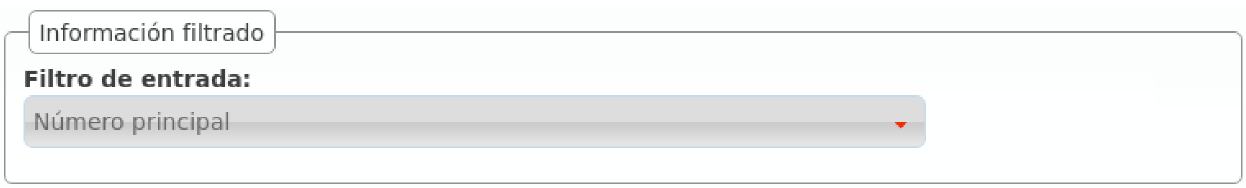
Route A DDI can have different *treatments*. For our current goul, set route to user and select *Alice*.

Record calls Can be used to record external calls (see *Call recording*).

Tarificate incoming calls This setting requires the external tarification module and allows tarification on special numbers. This module is not standard so don't hesitate in *contact us* if you are interested.

10.5.1 DDI external filters

We can assign a **external call filter** configured in *External Call filters*.



10.5.2 DDI routes

Once the call has passed all the checks in the filter (schedules and calendars) and after the welcome locution has been played (if there is any configured), we can route the call to the following processes:

- *Users configuration*
- *Hunt groups*
- *IVRs*
- *Conference rooms*
- *Conditional routes*
- *Queues*
- *Friends*

Hint: We can also route the DDI to a *Virtual Fax*, but this is something we will explain in the following block.

10.6 Outgoing DDI Rules

Most calling entities in IvozProvider require an outgoing DDI when placing calls to external numbers. This includes: Users, Friends, Faxes, Retail Accounts, and so on..

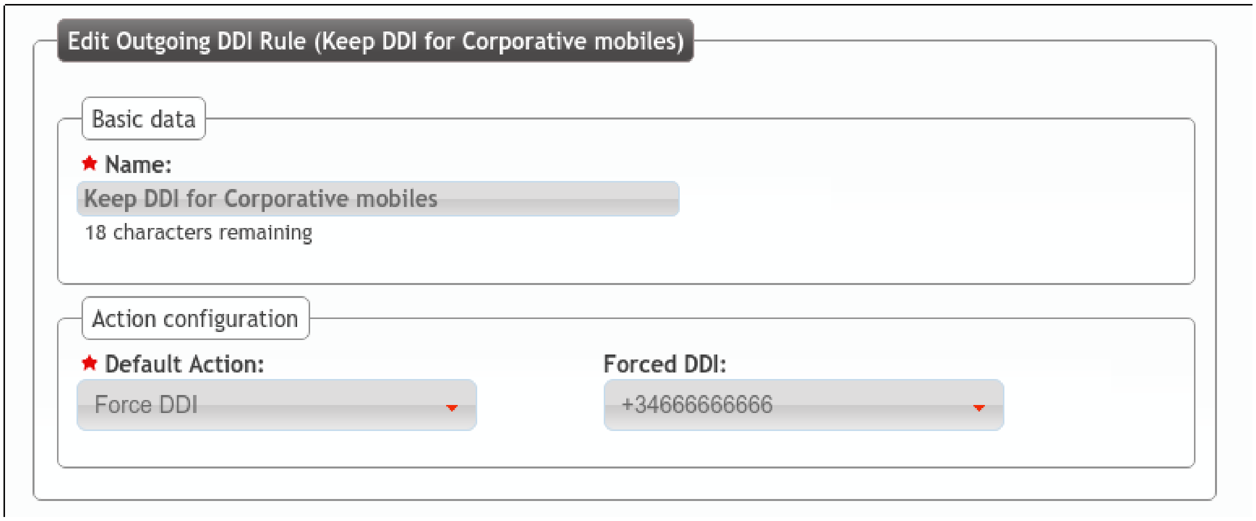
But there are some cases when a single outgoing DDI is not enough, and the presented DDI depends on the called number. To archive this dynamic outgoing DDI selection you can use Outgoing DDI rules.

Before creating a new rule, it would be required to frist group the destination numbers in *Match Lists*.

For this example, we will create a match list of corporative mobiles with all the mobile numbers of our company workers. When we call to those numbers, we will keep the original outgoing DDI assigned to the user, and for the rest of the cases we will force the DDI to the main company outgoing DDI.

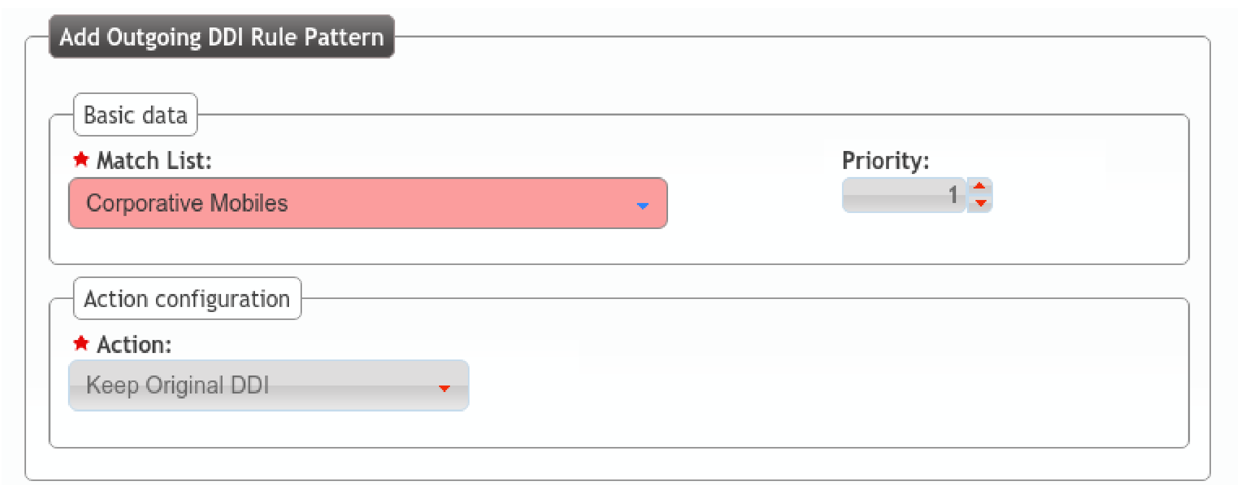
Create a new Outgoing DDI Rule

The main creation screen defines the action that will take place when no rule matches the dialed destination, so we define to force the main company DDI here.



Assign rule lists actions

Now we add a new rule that will match our mobiles to make the user's outgoing DDI be kept uptouched.



Assign rule to callers

At last, we have to configure who will use this rule to dynamically change it's presentation number. We can do this in the **Company's edit screen** or the **Users's edit screen**.

In this case, the User will present 777777777 DDI when calling Coporative mobiles and 666666666 when calling the rest of the external numbers.

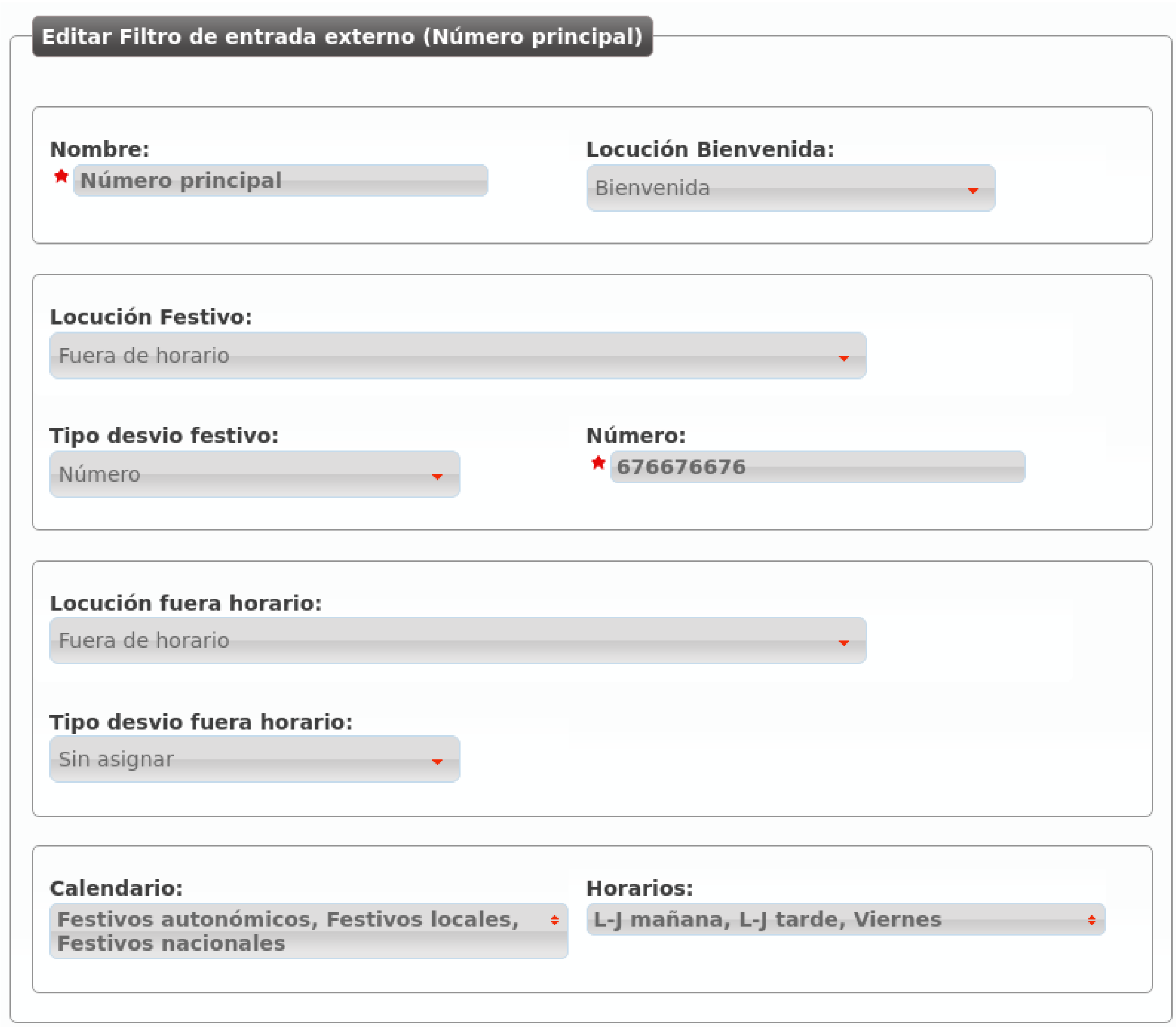
> **Attention:** Current implementation of Outgoing DDI rules won't work for diverted calls (out of schedule, holidays or user's call forward settings).

10.7 External Call filters

One of the most common task a company's administrator will do is to configure schedules and calendards to apply to existing *DDIs*.

Once we have our new created *Schedules* and *Calendars*, it's time to apply them in what we call **External call filter**.

The company admin can configure them in the following screen:



Name Descriptive name that will reference this filter in DDIs configuration.

Welcome locution This locution will be played if the call is not going to be forwarded by out of schedule or holiday filtering (in other words if the normal routing of the DDI is going to be applied).

Black list External origin will be checked against the associated *Match Lists*, if a coincidence is found, the call will be rejected immediately.

White list External origin will be checked against the associated *Match Lists*, if a coincidence is found, the call will be directly routed to the DDI destination, skipping the filter process. Take into account that black listed are checked before white lists.

Holiday locution The locution will be played when the day is maked as holiday in any of the calendars associated with the filter **if the calendar entry has no locution** for that day.

Holiday forward type After playing the above locution (if configured), call can be forwarded to a voicemail, external number or internal extension. For example, the filter of the image will redirect calls during holidays to the external number 676 676 676.

Out of schedule locution The locution will be played when, not being holiday, the current time is not in any of the time gaps defined in the schedules assigned to the filter.

Out of schedule forward type Like in the holidays forward, but for out of schedule. The image above won't apply any forward (and the call will be hanguped).

Calendars One or more calendars can be associated with the filter. The combination of all the callendars will be applied.

Schedules One or more schedules can be applied. The combination of all the time gaps defined in the schedules will be applied.

Attention: Holidays are processed **before** out of schedule events.

In the next section we will use this new created filter with *DDIs* so we can configure a welcome locution for normal days, and especial behaviours for hoildays and out of schedule events.

10.8 Conditional routes

Conditional routes allows changing a call logic depending on:

- Who is calling.
- What time is calling.
- What day is calling.

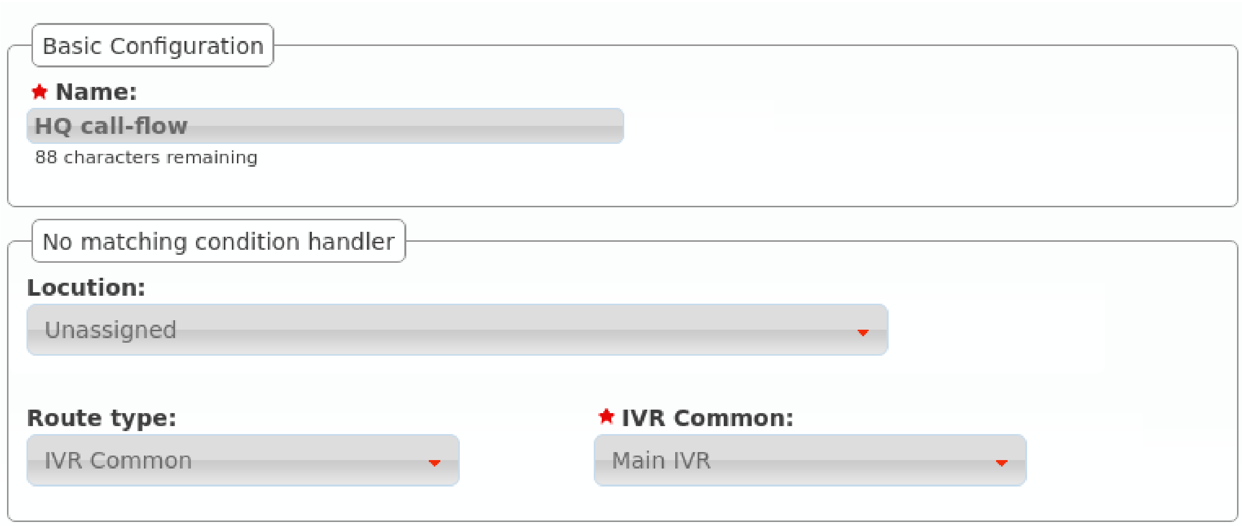
These routes are electable in three sections:

- DDIs
- Extensions
- IVR custom options

Tip: Remaining sections could use conditional routes creating an extension that point to a conditional route first, and routing to this extension.

10.8.1 Creating a conditional route

First of all we create a conditional route in **Conditional routes** section:

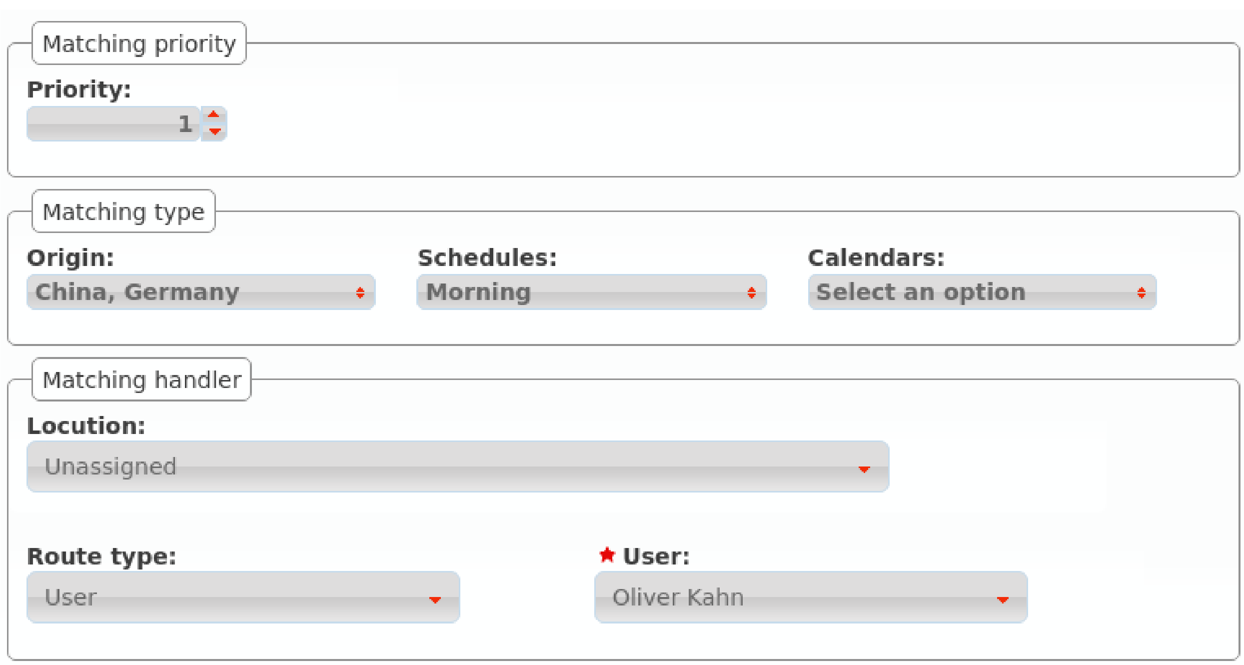


On creation we define what should be done with a call that does not satisfy any of the rules described below.

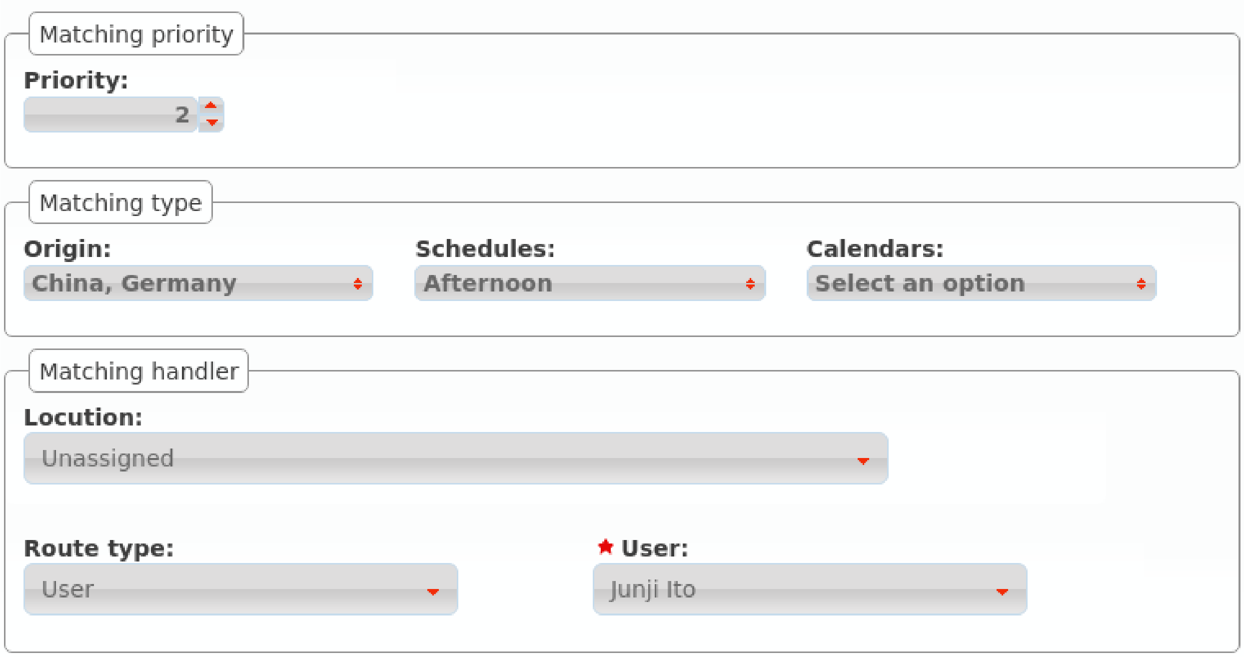
10.8.2 Adding rules

Once created, we need to add rules, for example:

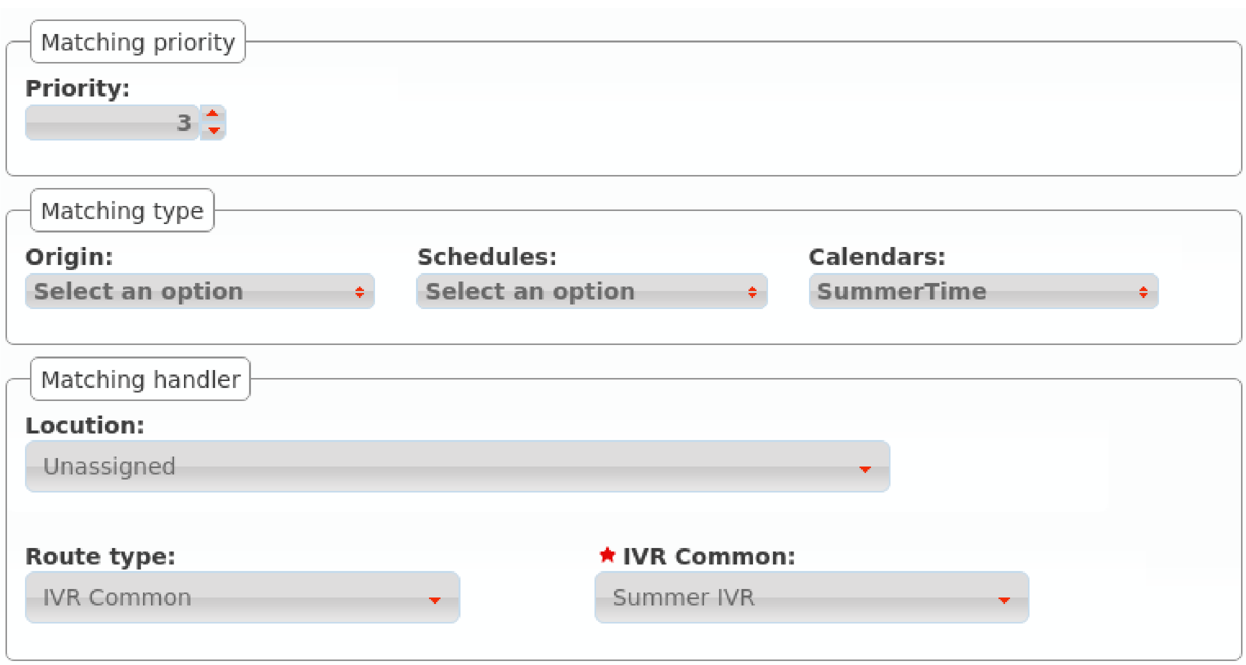
Calls from Japan and Germany received in the morning to an specific user



Calls from Japan and Germany received in the afternoon to another user



Override the reception IVR for summer days



With this example rules, our example conditional route will look like this:

List of Conditions (HQ call-flow) — Total: 3 Records

	Priority	Match	Locution	Route type	Target	Options
	1	China,Germany,Morning	Unassigned	User	Oliver Kahn	
	2	China,Germany,Afternoon	Unassigned	User	Junji Ito	
	3	SummerTime	Unassigned	IVR Common	Summer IVR	

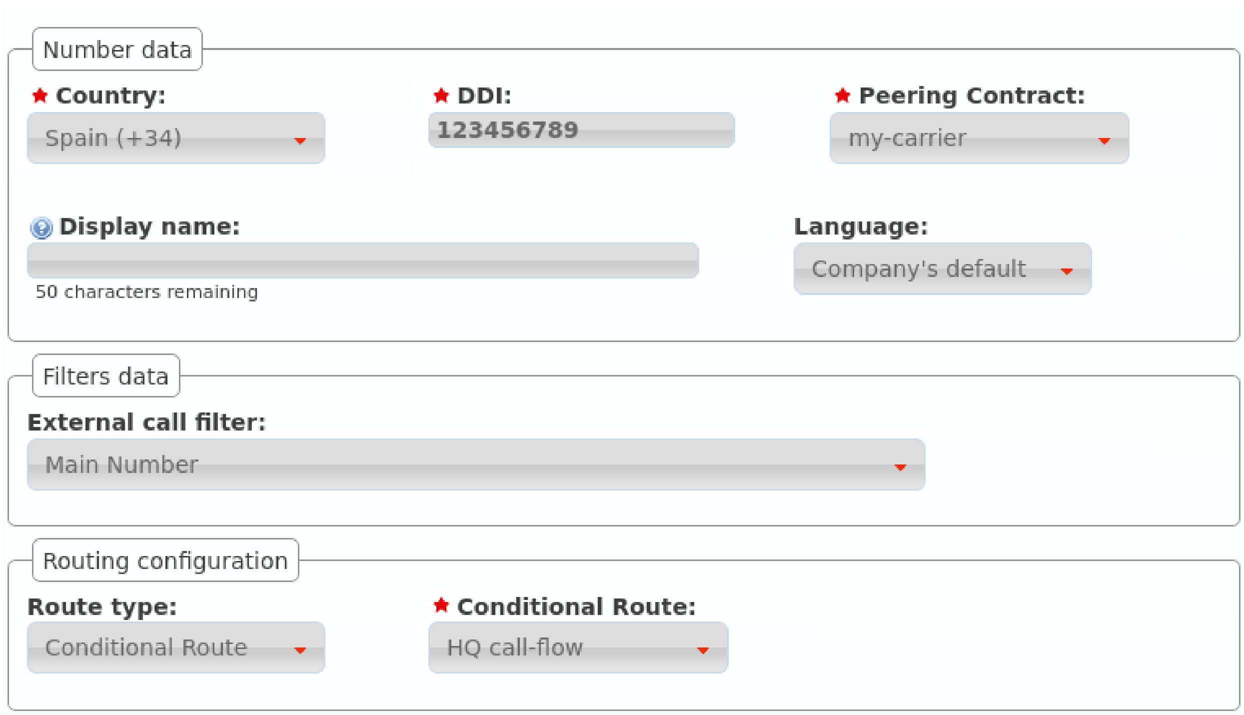
Some notes about this example:

- Rules are evaluated following the metric parameter. Once a rule matches, its logic is applied.
- Rules may have from 1 to 3 criteria:
  - None, one or more matchlist (pre-created, see *Match Lists*)
  - None, one or more schedules (pre-created, see *Schedules*)
  - None, one or more calendar (pre-created, see *Calendars*)
- These 3 criteria are combined (applying an AND logic).

10.8.3 Using a conditional route

The behaviour when an IVR option or an extension is routed to a conditional route is easy to understand, but using conditional routes with DDIs need an additional explanation.

Imagine this scenario:



DDI has an external call filter and is routed to the new conditional route.

When a call is received:

- External call filter is evaluated:
  - If current day is marked in any calendar, the holiday logic applies.

- If current time is not inside any time-gap, out-of-schedule logic applies.

- If external call filter logics have not applied, conditional route is evaluated.

Attention: Conditional route is not intented as an external call filter replacement. Filter is evaluated first, conditional route afterwards.

10.9 Match Lists

Mach Lists are designed to group well known numbers or patterns in order to use them in specific treatments.

Depending on the section used, this numbers can be matched with the origin or the destination of the call, so be sure to use distinctive names for your match lists.

For example, like mentioned in the previous section *External Call filters*, white and black lists contain one or more match lists. In this case, the **origin** of the call will be matched agains the list entries to determime if the treatment of **skipping** the filter or **rejecting** the call will be applied.

Note: Match lists themselfs have no behaviour associated, they only provide a common way for all process to determine if a number has a treatment.

Attention: Beware that numbers of a Match list are checked against origins or destinations depending on the configuration section that use them.

The section **Company configuration** > **Match Lists** allows to configure different items that will group the numbers and patterns.

The screen displayed to the company administrator looks like this:

List of Match Lists — Total: 2 Records

	Name	Options
	Corporative Mobiles	3
	Unwanted publicity agents	2

After creating a new Match list, you can include numbers and patterns.

List of Match List Patterns (Corporative Mobiles) — Total: 3 Records

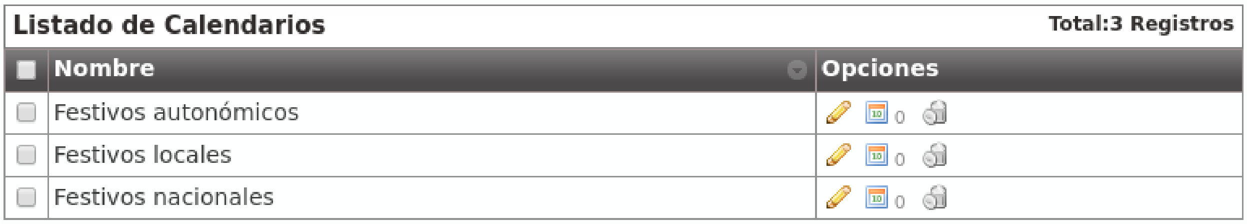
	Type	Match value	Description	Options
	Number	+42188888888881	Alice Alison Mobile	
	Number	+42188888888882	Bob Bobson	
	Regular Expression	^34[67]	Any Spanish mobile	

As shown, the match list can contain specific numbers or groups using Regular Expressions

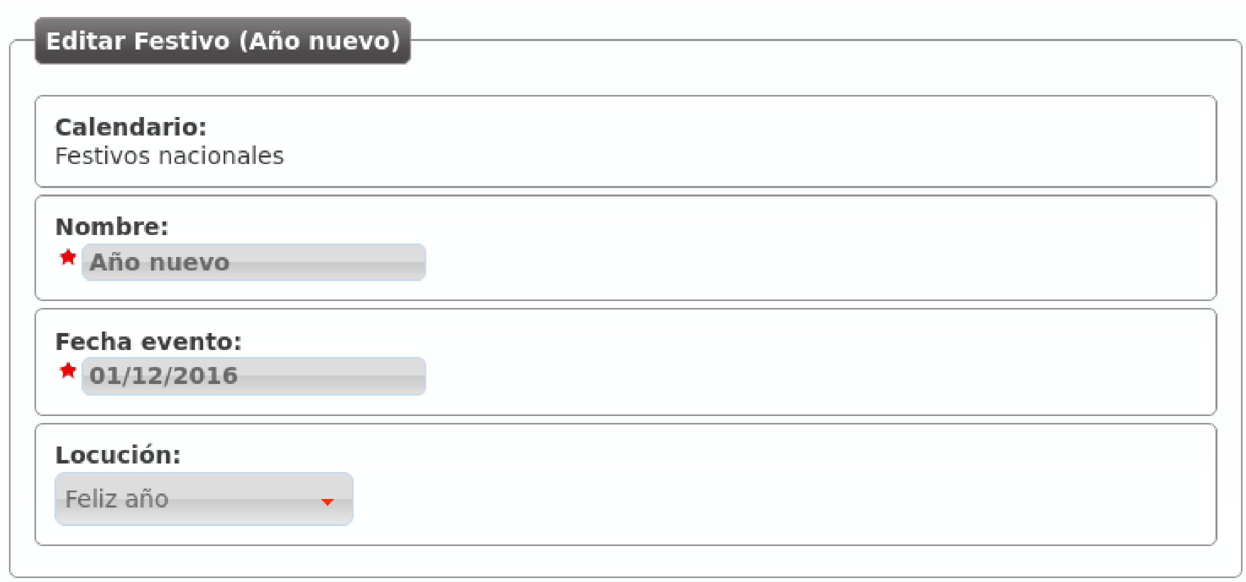
10.10 Calendars

Calenders are used to define what days are considered as holiday. Like schedules, multiples calendars can be combined.

Let's imagine three calendars with the following configuration:



Calendar creation process only requires a name. Once created, we can add what days will be holidays using the buttons in its row:



From this moment on, the calendar has the 1st of January of 2016 as holiday date with the locution "Happy New Year".

Warning: Calendars logic is opposite to Schedulers: If a day is not defined as holiday in any of the calendars, it will considered a normal day and no filtering will be applied.

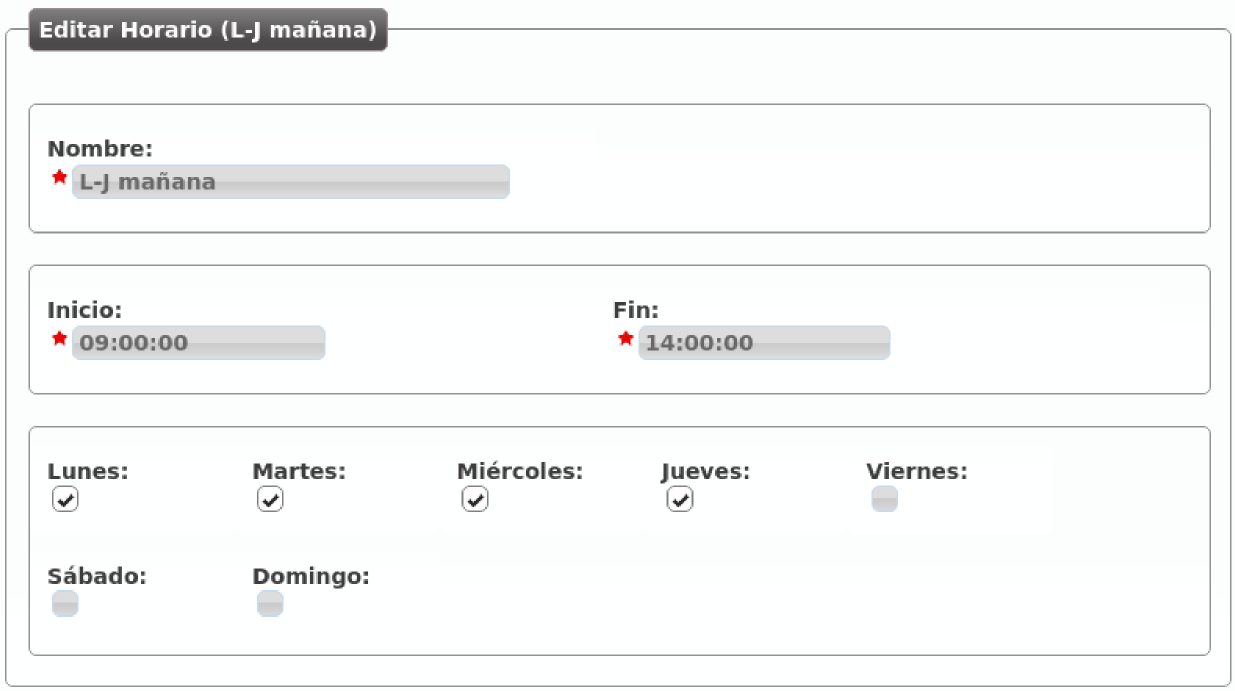
Hint: Holidays without special locutions will apply the external call filter holiday generic locution (see below).

Create a new External call filter

10.11 Schedules

The section **Company configuration** > **Schedule** allows to configure different time gaps when an *external DDI* will be available.

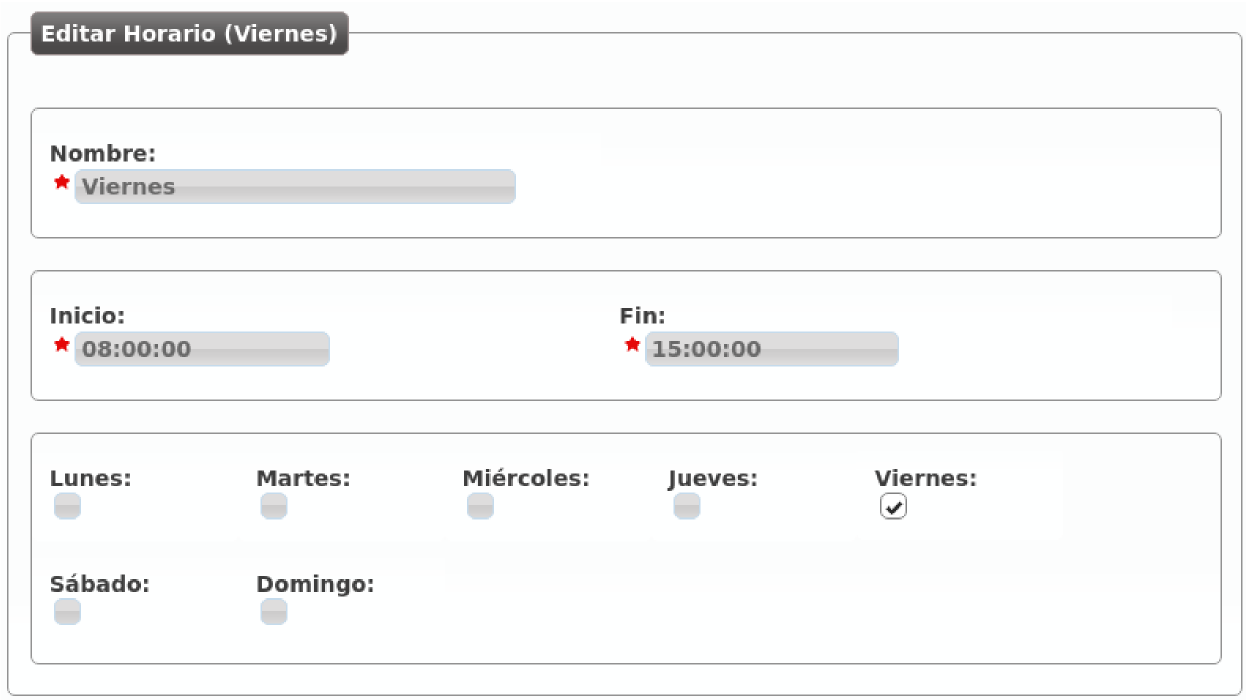
The screen displayed to the company administrator looks like this:



With the above configuration, we have defined a morning schedule that will be applied from Monday to Thursday.

We can also define an afternoon schedule for Monday to Thursday too:

And apply a different time gap for the Fridays:



We have the following time gaps that combined will determine our company office schedule.

Listado de Horarios — Total:3 Registros

	Nombre	Inicio	Fin	Opciones
	L-J mañana	09:00:00	14:00:00	
	L-J tarde	15:00:00	18:00:00	
	Viernes	08:00:00	15:00:00	

Warning: The schedule will be defined by combining the active time gaps: Any time outside this grouped gaps will be considered out-of-schedule.

10.12 Route locks

Route locks are a simple but powerful way to fork route logics when delivering calls. This fork is done depending on the state of the lock on a particular moment:

- **Opened**: green light, go ahead.
- **Closed**: red light, no trespassing allowed.

They are used as conditional route rule criteria (see how in *Conditional routes*).

10.12.1 Route lock creation

When you add a new route lock in **Route Locks** section, you are asked for the following fields:

Name This name will be used in conditional routes to identify the lock.

Description Just a description.

Status Set the initial status of the lock: opened or closed.

10.12.2 Route locks service codes

Although you can set the initial lock status on creation and change it using the admin portal too, the usual way to handle the status changes of a lock is to use the service codes listed in **Route locks** section.

These services codes have two parts:

- **Service code**: configured in **Services** section per brand/company.
- **Lock id**: immutable numeric id assigned to each lock.

Tip: There are 3 service codes available for most common operations on locks:

- Open Lock
- Close Lock
- Toggle Lock.

Read *Services* for further details.

10.13 Call ACL Control

The **Call ACLs** determines what users can call to external numbers.

Attention: The internal extensions are allowed to all users, the **Call ACLs only apply to external numbers**

The **Call ACL** setup has two different parts:

- Classify the call in different types based on **match lists**:
  - Brand level: **Brand Configuration** > **Match Lists**
  - Company level: **Company Configuration** > **Match Lists**
- Choose policies for groups of patterns: **Company Configuration** > **Call ACLs**

10.13.1 Call ACL Matchlits

The destination number is matched against the **ACL MatchLits** to determine the call permission.

Note: Brand matchlists can be used by any of its companies, so most common ACL Patterns (p.e. country prefixes) can be reused easily.

For more information of how MatchLists patterns are created, please refer to section *Match Lists*.

Attention: Regular expressions of Match List patterns must be in E.164 format.

10.13.2 Call ACL

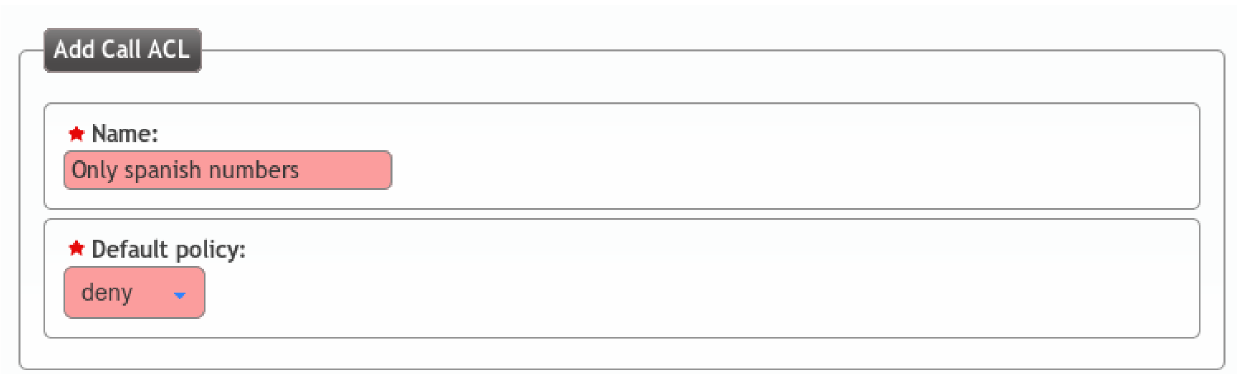
The **Call ACL** configuration is easier to explain with an example:

Imagine the following **match list** with this patterns:

List of Match List Patterns (Spanish Numbers) — Total: 2 Records

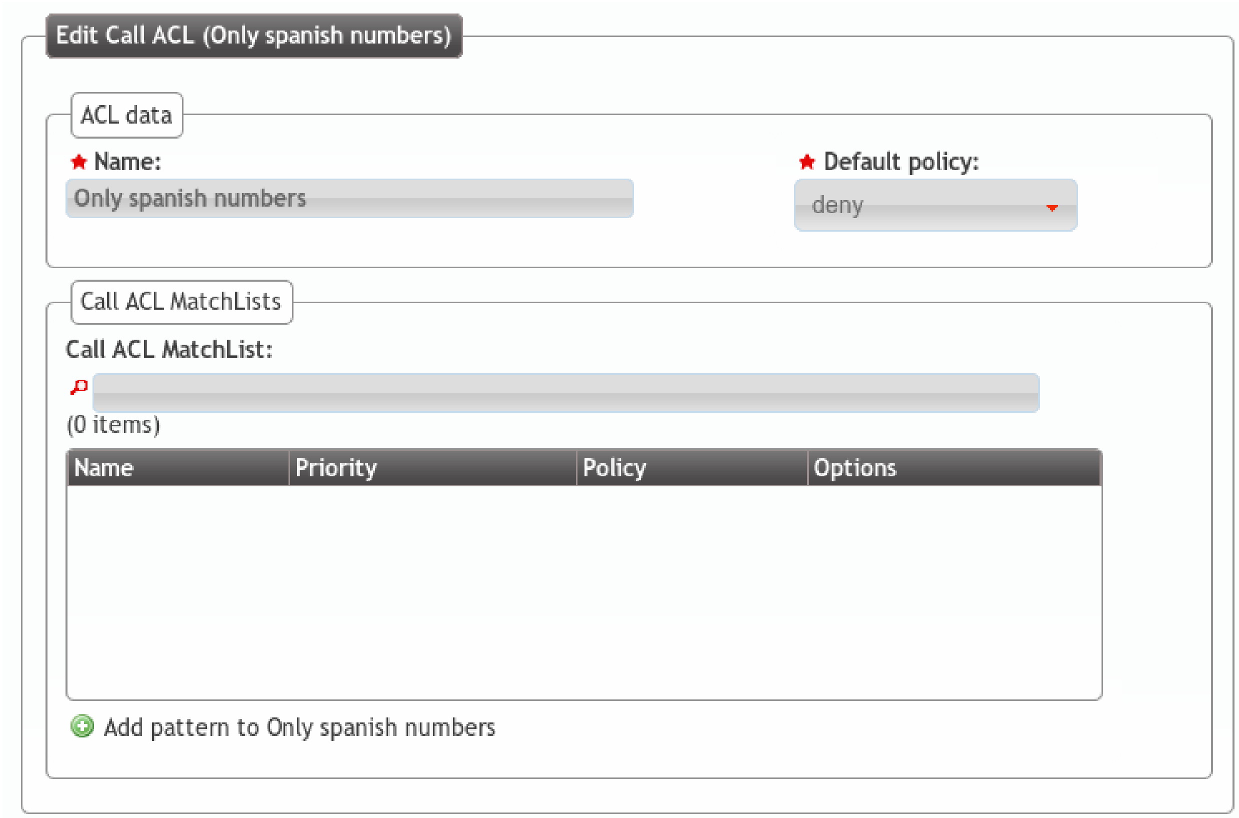
Type	Match value	Description	Options
Regular Expression	^+349[1-9]	Spanish Land Line	
Regular Expression	^+34[67]	Spanish Mobile	

We could create a **Call ACL** that only allow calling to this destinations:

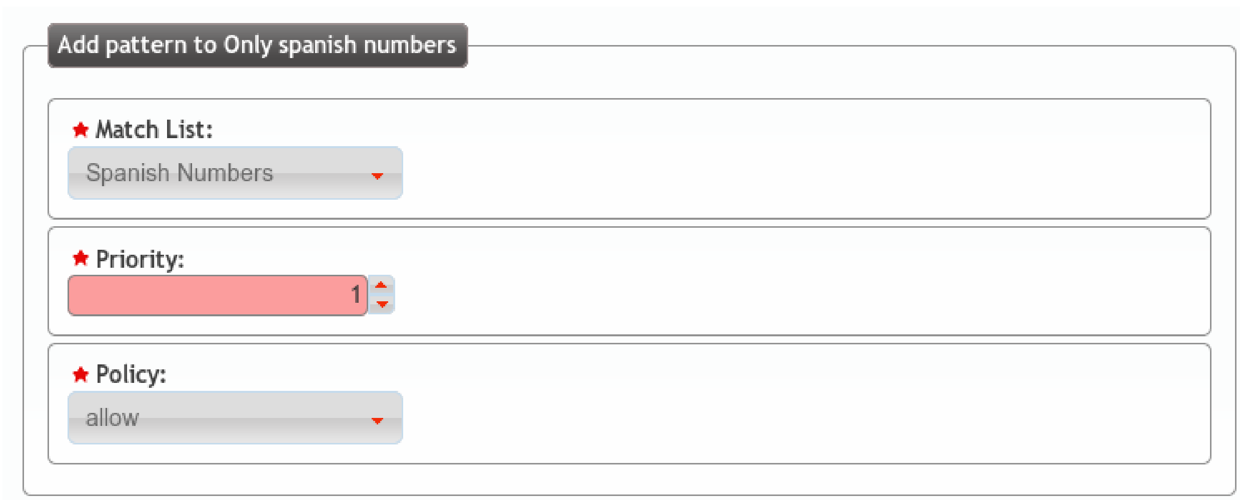


Note: The default policy determines what to do with the call when the destination number **does not match any ACL matchlists**.

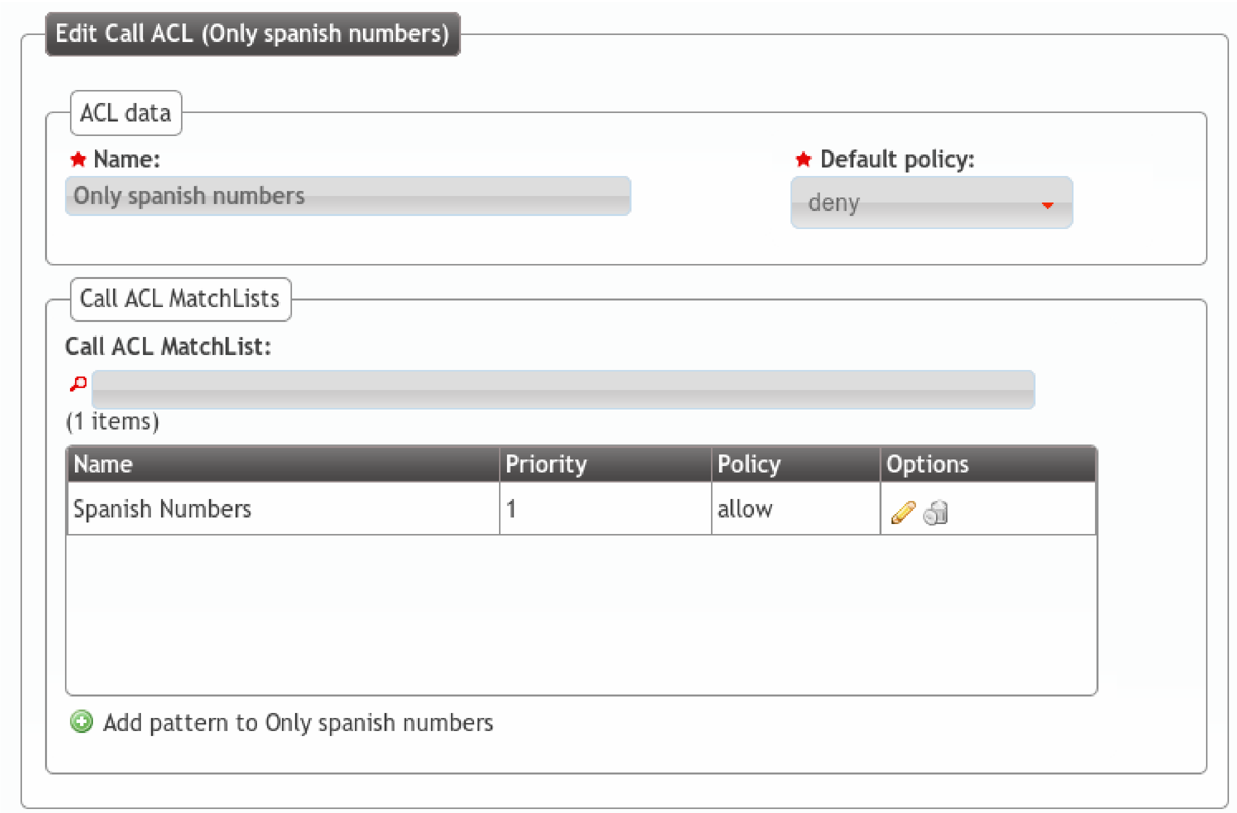
After creating the **Call ACL** we can edit it to add the required rules:



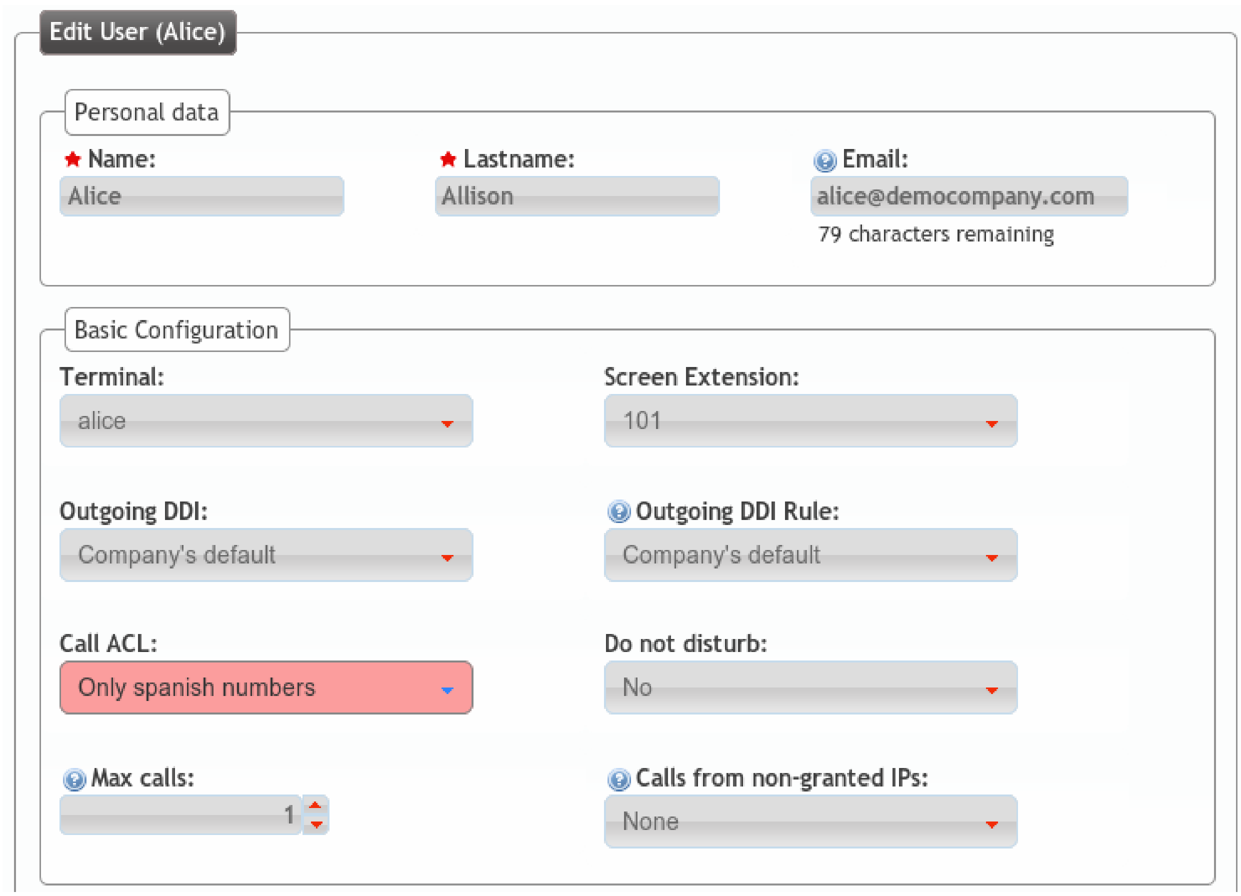
The **metric** determines the evaluation order of the rules and the action that will be applied if it *matches* the list (allow/deny).



Once we have added our spanish **Match List**, our **Call ACL** will look like this:



We only have to assign this ACL to the users in the section **Company configuration** > **Users**:



From this moment on, Alice will only be allowed to call internal extensions (they are always allowed) and spanish numbers.

10.14 Hunt groups

The hungroups allows configuring more complex *ringing* process that the traditional **call to a user**.

There are multiple types:

Ring all The call will make all the terminals of the group during a predefined time.

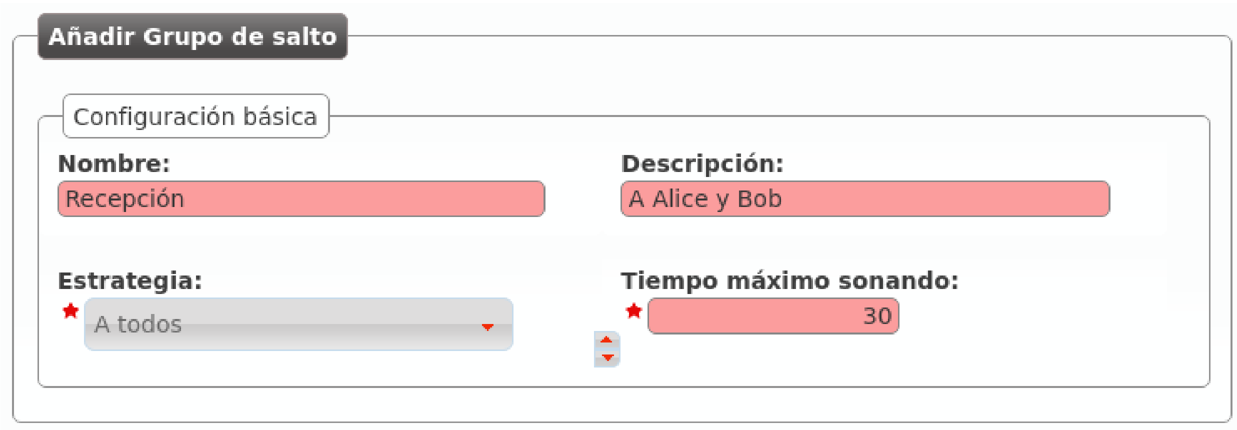
Sequential The call will *jump* from one user to another in a predefined order ringing during the configured time. If the call is not answered by any user of the group, it will be hanguped (or trigger the no answer logic).

Sequential (infinite) The call will *jump* from one user to another in a predefined order ringing during the configured time. If the call is not answered by any user of the group, the call will *jump* again to the first member of the group and keep looping.

Random The call will *jump* from one user to another in a random order, ringing during the configured time. If the call is not answered by any user of the group, it will be hanguped (or trigger the no answer logic).

Example 1: *Ringall* hunt group

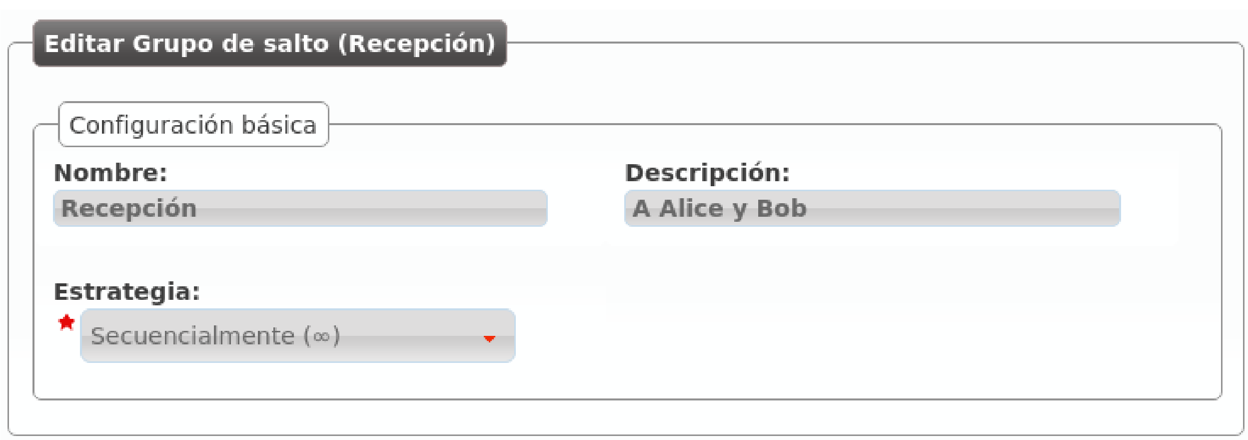
The following example will show how to create a hunt group that will call our 2 users at the same time during 30 seconds:



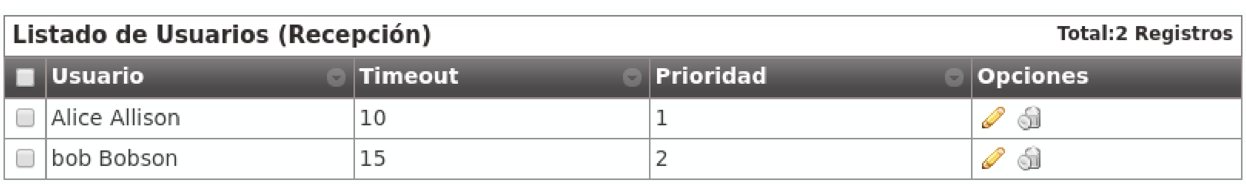
Pressing the proper icon, we can add Alice and Bob to the hunt group:

Example 2: *Sequential* hunt group

We will edit the hunt group to convert it into sequential: the call will ring Alice during 10 seconds, then it will ring Bob 15 seconds, repeting this process until one of them answers.

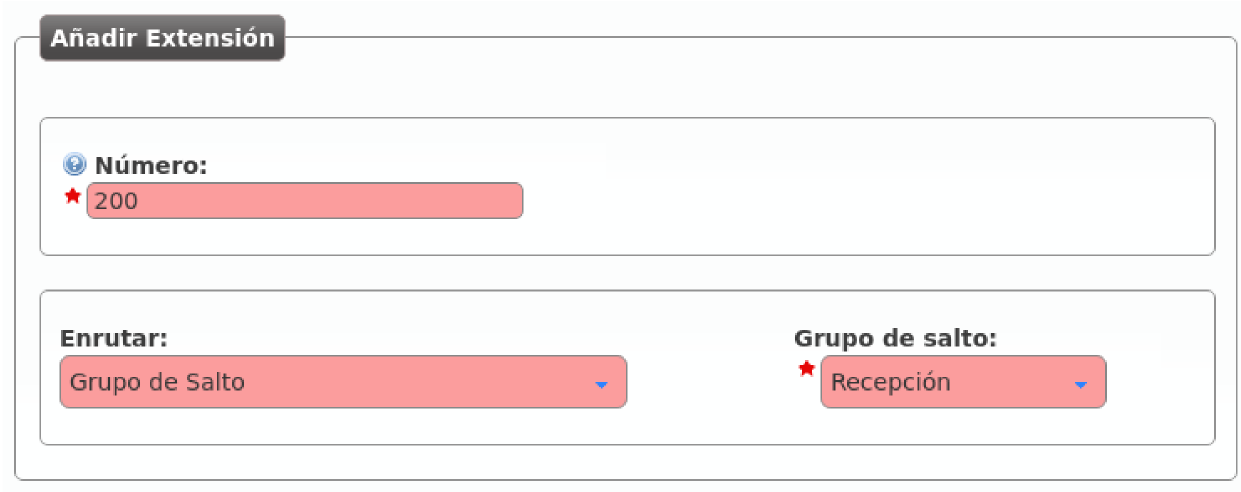


For this type of groups we have to configure priority (the call will *jump* from the users with lower number priority to the ones with higher number priority) and a *ringing* time for each user.



Hint: Hunt groups can be routed from any process of IvozProvider by simply adding an extension that route to them.

Let's create a new extension that routes to this hunt group:



10.15 Call pickup

Call pickup is the process where a user can answer a call that is being ringing in another terminal. No need to say that, somehow (sound, flashing lights, notification, etc) the users must know that the call is ringing elsewhere.

IvozProvider supports two kind of call pickups:

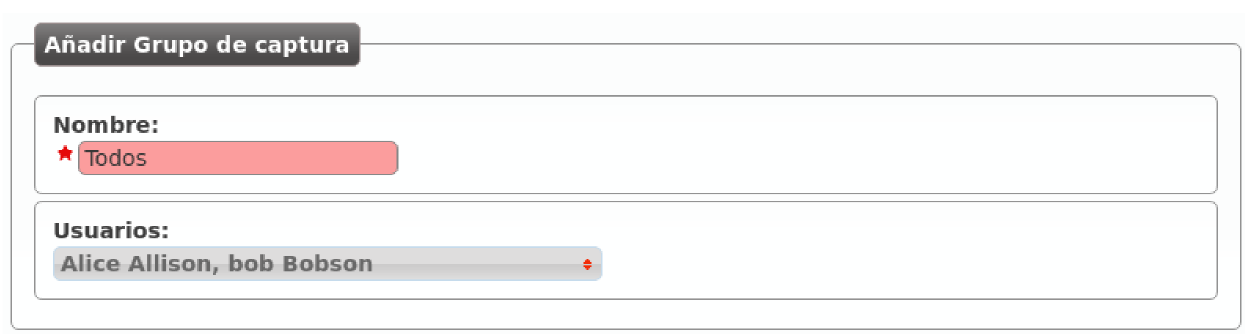
Direct pickup In this type of pickup, the user that is trying to capture the ringing call must include the extension of the target phone after the service code. For example, if the direct pickup code is *95, the user must dial *95101 to capture a call that is ringing in the extension 101.

Group pickup In this type of pickup, the user that is trying to capture the ringing call will just dial the service code. If anyone in any of the pickup groups of the user has a ringing call, it will be answered by the capturer.

10.15.1 Call pickup groups

In order to make **call group pickups**, the capturer user must be part of the same group that the target user that wants to capture.

The section **Pickup groups** allows the company administrator to configure what users will be in each group:



As shown in the section *Users configuration*, we can add or edit the groups of a user in the user's edit screen.

Note: A user can be part of multiple pickup groups. The system will take all of them into account when using the group pickup service.

10.15.2 Group pickup service code

IvozProvider supports 2 different configuration levels for defining the service codes for pickup:

- At brand level: **Brand configuration** > **Services**.
- At company level: **Company configuration** > **Services**.

The brand administrator can configure generic codes that all the companies will use. Companies can customize this codes if they are used to another ones.

The *following section* explains the services in depth, with all the additional services that can be accessed by dialing codes starting with *.

10.16 Interactive Voice Response (IVR)

IVRs are the most common way to make **audio menus** where the caller must choose the destination of the call by **pressing codes** based on the locutions instructions that will be played.

10.16.1 IVRs

IVRs support specifying actions for dialed digits, but also they can be also be used to route any existing company extension.

IVRs have the following fields:

Name Descriptive name of the IVR that will be used in other sections.

Timeout Time that caller has to enter the digits of the target extension.

Max digits Maximum number of digits allowed in this IVR.

Welcome locution This locution will be played as soon as the caller enters the IVR.

Success locution In case the dialed number matches one of the IVR entries or extension exists in the company (and allow extensions is enabled), this locution will be played (usually something like 'Connecting, please wait...').

Allow dialing extensions When this setting is enabled, the caller can directly press the extension that must previously know (or the welcome locution suggests) and the system will automatically connect with that extension.

Excluded Extensions When Allow extensions is enabled, you can exclude some extensions to be directly dialed adding them to the exclusion list.

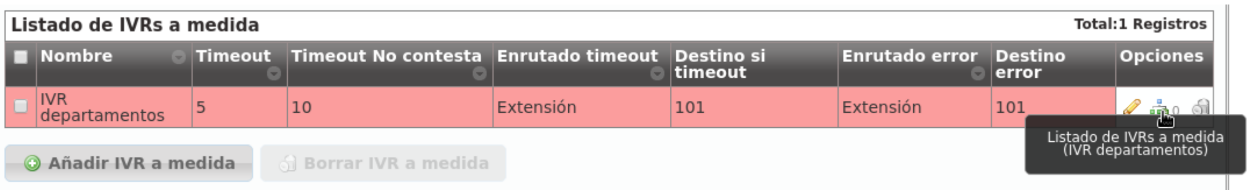
No input process If the caller does not input any digit in the timeout value, the no input process will trigger, playing the configured locution and redirecting the call to another number, extension or voicemail.

Error process If the dialed extension does not match any IVR entry, any company extensions (when allow extensions is enabled), or it matches one of the extensions in the excluded Extensions list, the error process will trigger, playing the configured locution and redirecting the call to another number, extension or voicemail.

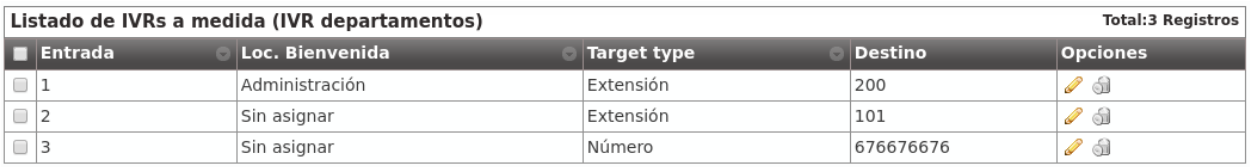
10.16.2 IVR Entries

Hint: The most common usage for IVR is combining them with a welcome locution that says something like 'Press 1 to contact XXX, Press 2 to contact YYY, ...”

The process of each entry of the IVR can be defined in the following button:



In this example, the caller can dial 1, 2 or 3 (the rest will be considered as an error and will trigger the **Error process**):



- 1: Call to the internal extension 200, created in *previous section* that routes to hunt group *Reception*.
- 2: Call to the internal extension 101.
- 3: Route this call to the external number 676 676 676.

Note: Each of the IVR entries supports a locution that, if set, will be played instead of the IVR **success locution**. This way, you can configure a generic locution (like 'Connecting....') or a custom one for a given entry (like 'Connecting reception department, please wait...').

Entries are regular expressions

You can specify IVR entries as Regular Expressions. If entry is just a numeric value, it will be handled as a sequence of digits, otherwise it will be handled a regular expression. This can be handy if you have the same behaviour for a group of dialed numbers.

10.17 Queues

Easy queue behaviour was included in IvozProvider in 1.3 version. It is a simple approach with **the unique goal to provide the capability to handle more calls than users attending them**.

Warning: Queues and callcenter are close terms but different. **IvozProvider is not a suitable product for callcenters**, as it does not provide advanced features that are crucial to them (reports, RT visualization, queue related stat, etc.).

In distributed installations using Queues is only compatible with an static assignment or 'hash based' distribution (see **Distribute method** *here*).

Hint: Brand operators can choose which Companies have queues (see **Features** in *Brand Configuration* and *Company Configuration*).

10.17.1 Queue configuration

This are the settings related to a queue:

Name Use to reference this queue

Weight Priorizes calls to an agent that attends calls in two (or more) calls. The higher, the more priorized.

Strategy How will the queue deliver the calls? Calling to all agents, calling to a random one?

Member call seconds Defines how long will a call to an agent last.

Member rest seconds Seconds between calls for an agent.

Announce Select a locution and its frequency. Caller waiting in the call will listen to this locution.

Timeout configuration Limits the time that a call can wait in a queue and the following behaviour.

Full Queue configuration Limits the amount of people waiting in a call and the behaviour when this limit it reached.

Apart from creating a queue, you have to assign users to it. This users will have a **penalty: a user will not be selected to deliver a call if any user with lower penalty is available**.

Hint: A call can be sent to a queue selecting it in the "Route type" selectors available in multiple sections of IvozProvider (extension to queue, DDI to queue, etc.)

10.17.2 Queue strategy

The queue strategy **always applies to current penalty members** starting with the smallest penalty value and only going to the next penalty if all members of current one are busy or unavailable.

Ring all The call will make all the members of the current priority during a predefined time.

Least recent The call will *jump* from one member to another in a predefined order based on the last time the member attended a call. Members whose latest call is older will be called first.

Fewer calls The call will *jump* from one member to another in a predefined order based on the number of atteded calls. Members that have attedended less calls will be called first.

Random The call will *jump* from one member to another in a random order, ringing during the configured time.

Round Robin memory The call will *jump* from one member to another in a predefined order starting past the last member that attended a call.

Linear The call will *jump* from one member to another in a predefined order based on the creation time of the member.

10.18 Conference rooms

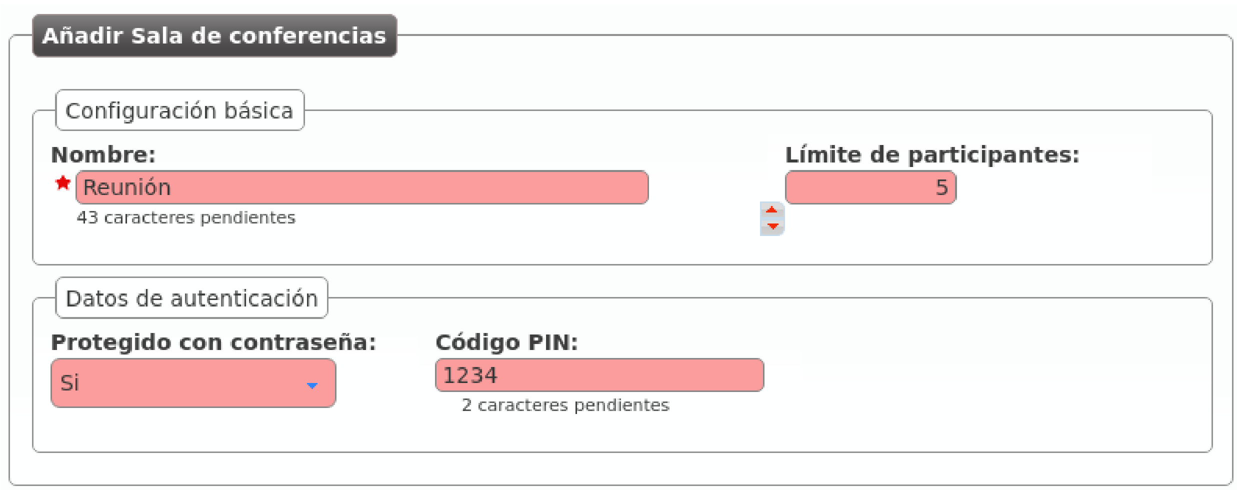
IvozProvider supports Conference rooms that can be configured in the section **Company configuration > Conference rooms**.

In distributed installations using Conferences is only compatible with an static assignment or 'hash based' distribution (see **Distribute method** *here*).

Hint: Brand operators can choose which Companies have conferences (see **Features** in *Brand Configuration* and *Company Configuration*).

Create a new audio conference

The following image shows the process of creating a new confrence room:



Name Name that will used to identify this conference room in other sections

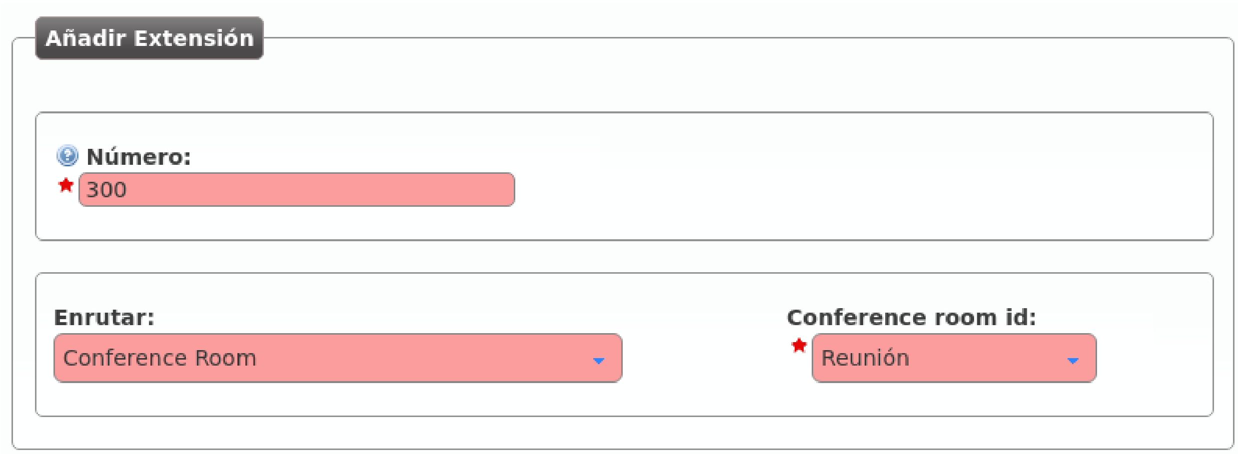
Max members Maximum number of participants in the conference. When this limit is reached, join requests will be rejected.

Pin protected Conference rooms can be pin protected. The pin will be requested before entering and must be numeric.

Note: Member limit can be disabled by setting it to 0.

Route an extension or DDI to the conference

In order to enter a conference there must be a number that is route to them:



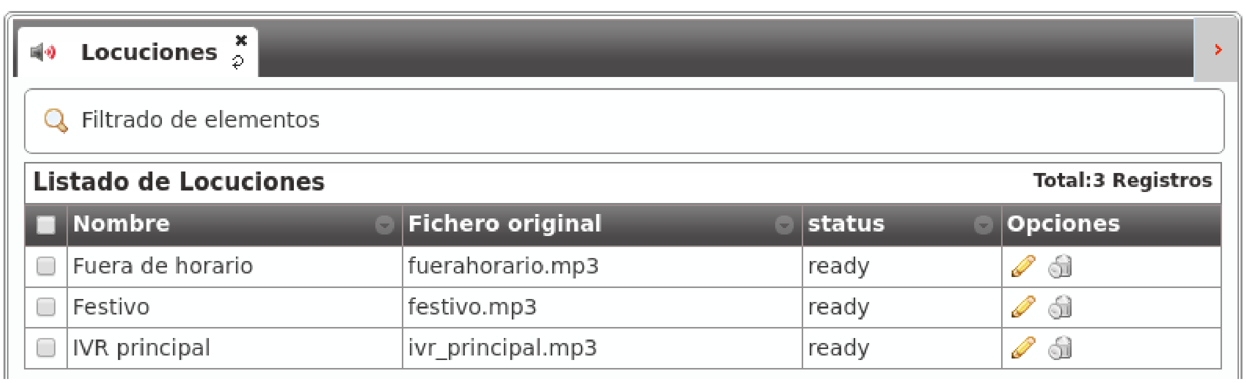
In the following section we will see how to configure a *external DDI* to a conference room so it can be used by external callers.

Hint: There are other ways to make external callers join a conference room without using a DDI: it can be assigned to an Extension. This way, any user can transfer the call to the conference extension, or can be routed, for exmaple using an IVR entry.

10.19 Locutions

The locutions of the platform are created and uploaded just like the files of *Music on Hold*.

The section **Company configuration** > **Locutions** allows the company admin to choose the sounds that will be played in many configuration places (IVR, etc) accross the platform.



Attention: Locutions can be recorded from any terminal by dialing the Recording extension displayed in their edit screen.

Hint: The main difference between a **locution** and **music on hold** is that the administrator chooses when the first one will be played (out of schedule, IVRs, and so on) and the second one will be played when a call is holded by an user.

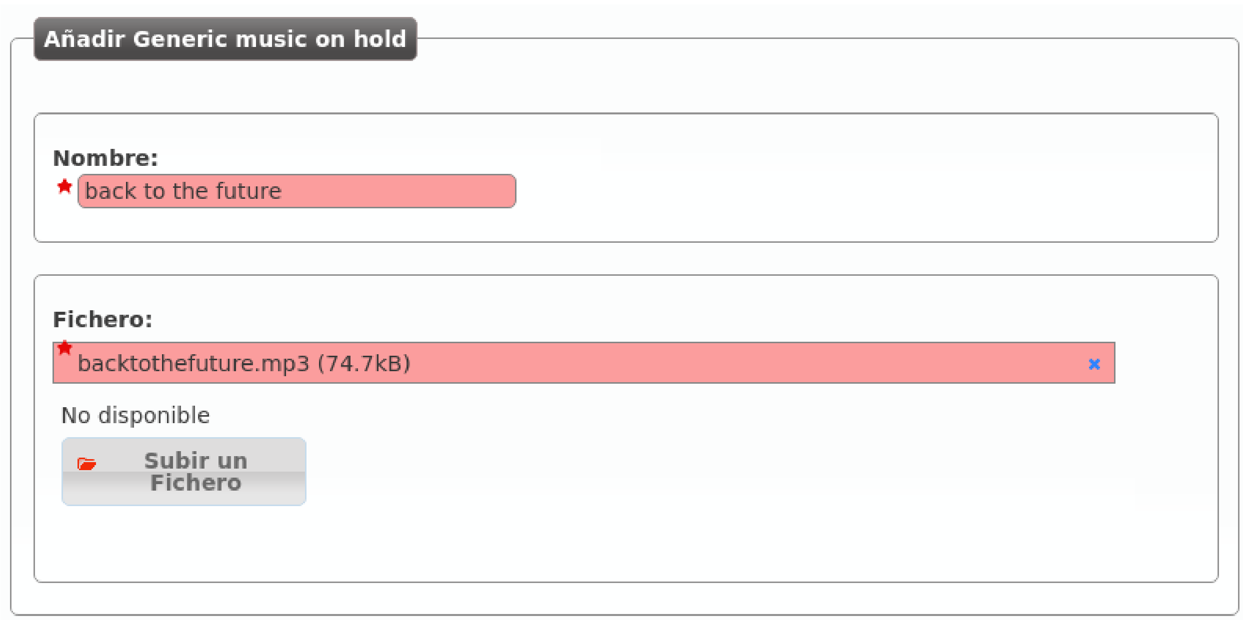
10.20 Music on Hold

The music on hold will be played when the user holds the call and the other member waits until the call is resumed.

If a company has defined a music on hold, it will be played. Otherwise, the one defined by the brand administrator. If none of this is configured, a global music will be played.

Note: Multiple files can be added to be played as Music on Hold. The system will choose them randomly for each call.

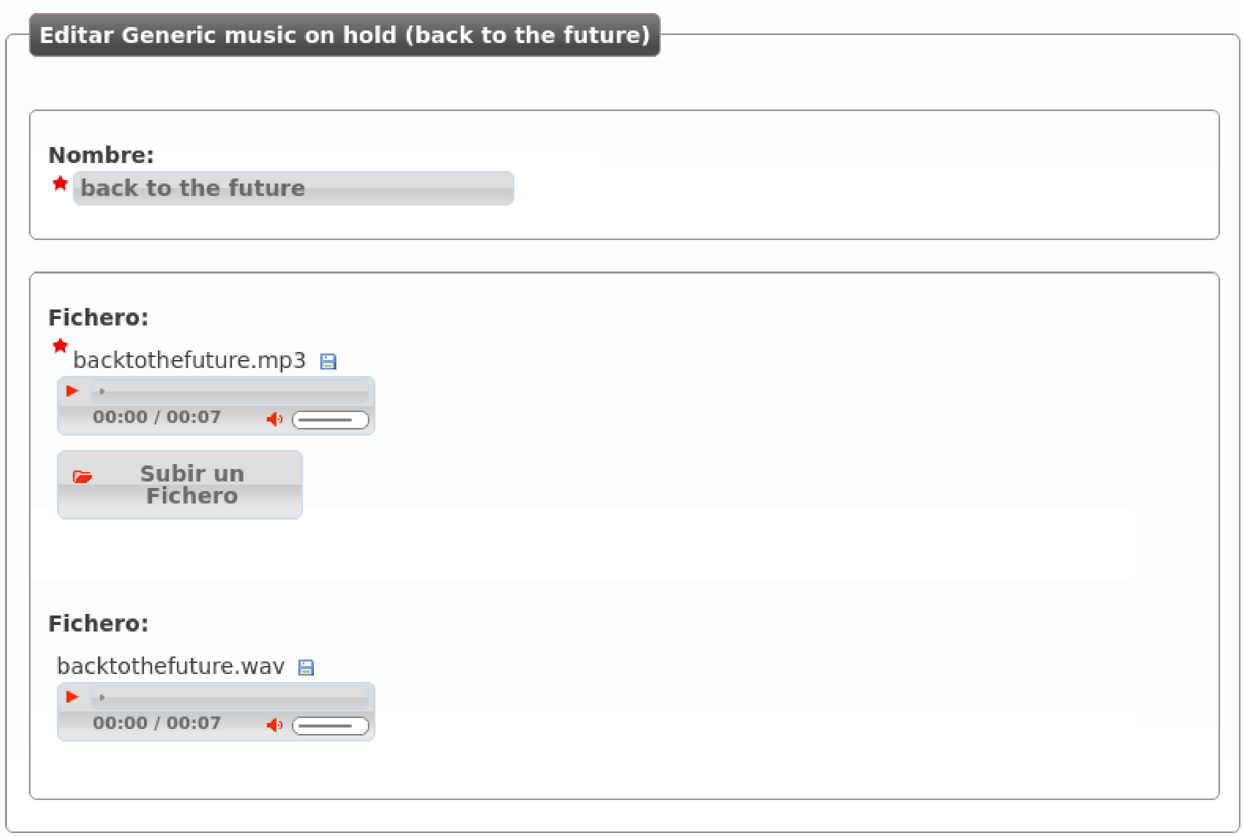
Add a new music on hold



Once the music has been *encoded* the **Status** fill will display *ready* and the music will be used for the next calls.

Tip: IvozProvider supports most of the common audio formats and *encodes* them to the optimal format for the platform.

After the *encoding*, we can download both the original and the converted version in the edit screen.



10.21 Call recording

> **Attention:** Beware that local legislation may enforce to announce that the call is being recorded (sometimes to both parties). You should include a recording disclaimer in your welcome locutions for DDIs with automatic recording enabled.

IvozProvider supports two different ways of recording calls:

- **Automatic recordings** for the incoming/outgoing calls that use a *External DDI*.
- **On demand recordings** requested by a user during a call.

10.21.1 Automatic DDI recordings

In this type of recording, **the whole conversation will be recorded**: from the start until it finishes.

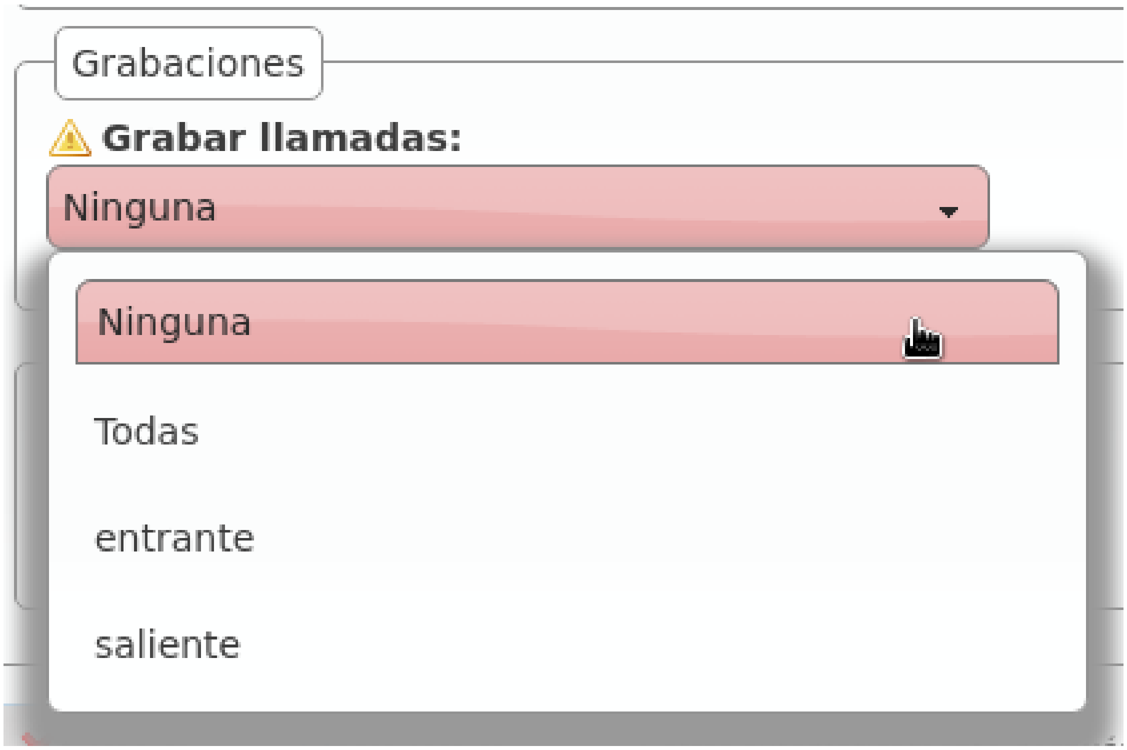
Two different scenarios:

- **Incoming calls to a DDI**: The call will continue until the external dialer hangups (no matter whom is talking to).
- **Outgoing calls using a DDI** as *Outgoing DDI*: the recording will continue as long as the external destination keeps in the conversation.

Attention: Take into account that the call will be recorded while the external entity is present, even it the call is being transfered between multiple users of the platform.

Record all the calls of a DDI

To enable this feature, edit the DDI and configure the field under the section recording data:



There are 4 available options:

- Disable recordings
- Enable incoming recordings
- Enable outgoing recordings
- Enable all call recordings

10.21.2 On demand recordings

The *on-demand* recordings must be enabled by the *brand administator* for the companies that request it. This can be done in the companey edit screen:

Warning: Contrary to the *Services* mentioned in the previous section, the on demand record are actived within a conversation.

Contrary to automatic ones, on demand recording can be stoped using the same process that started them.

Activated using the *Record* key

Some terminals (for example, *Yealink*) support sending a SIP INFO message during the conversation with a special *Record* header (see reference). This is not a standard for the protocol, but being Yealink one of the supported manufacturers of the solution, we include this kind of on-demand recording.

Important: For this recording requests, the configured code doesn't matter but the company still must have on demand records enabled.

To start or stop this kind of recordings, just press the Record key in the terminal and the system will handle the sent message.

Activated using *DTMF* codes

The more traditional approach for this feature is to press a combination of keys during the call. Some notification will be played and the recording will start or stop. This combination is sent to the system using DTMF tones using the same audio stream that the conversation (as mentioned in RFC 4733).

IvozProvider supports this kind of on demand record activation but with an important downside. In order to capture this codes, the pbx must process each audio packet to detect the code, avoiding the direct flow of media between the final endpoints.

Important: Enabling this record mode highly affects the performance of the platform. Use at your own risk.

Activated using a frustated blind transfer

There is a tricky way to access this feature for terminals that does not support the INFO message and don't want its audio to be parsed:

Danger: This method is a workaround for those terminals that does not support the native *Record* key activation (recommended). Take into account that not all terminals will behave the same way while performing the transfer described in this section.

The keys for this methods are:

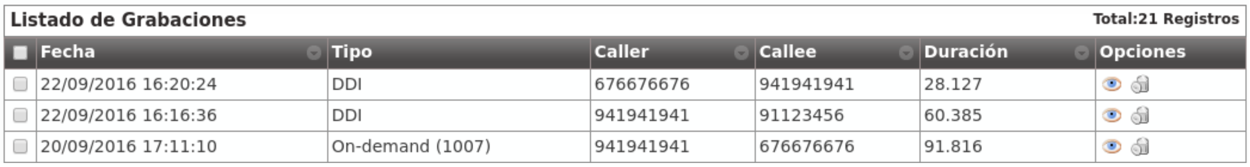
- It's not activated using a code during the conversation.
- It's activated making a **blind transfer** to the on demand record code.
- The system will understand this as a request to record and will reject the transfer.
- The user will continue with the existing call and keep talking.

Why this method does even exist?

The reason behind this tricky method is based, as explained in the previous block, on the design of the *Platform general architecture* and the *RTP audio flow*.

10.21.3 Recordings list

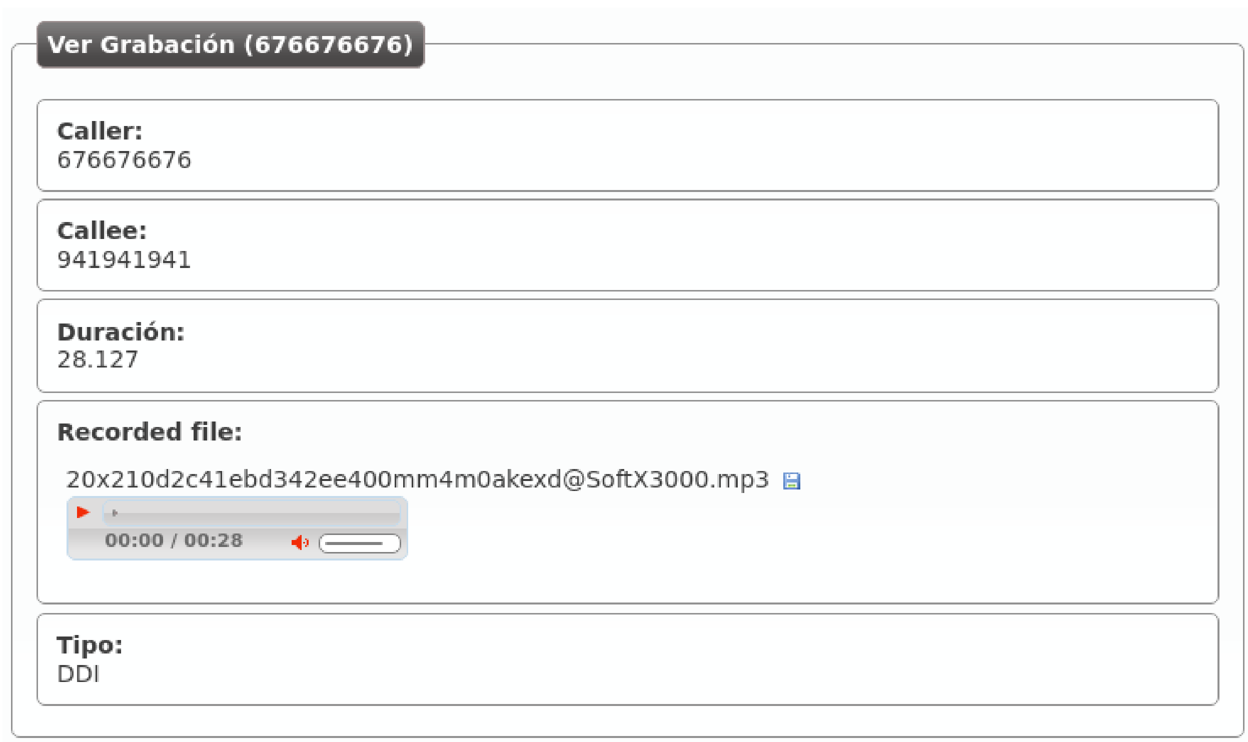
The *company administrator* can access to all the recordings in the section **Company configuration** > **Recordings**:



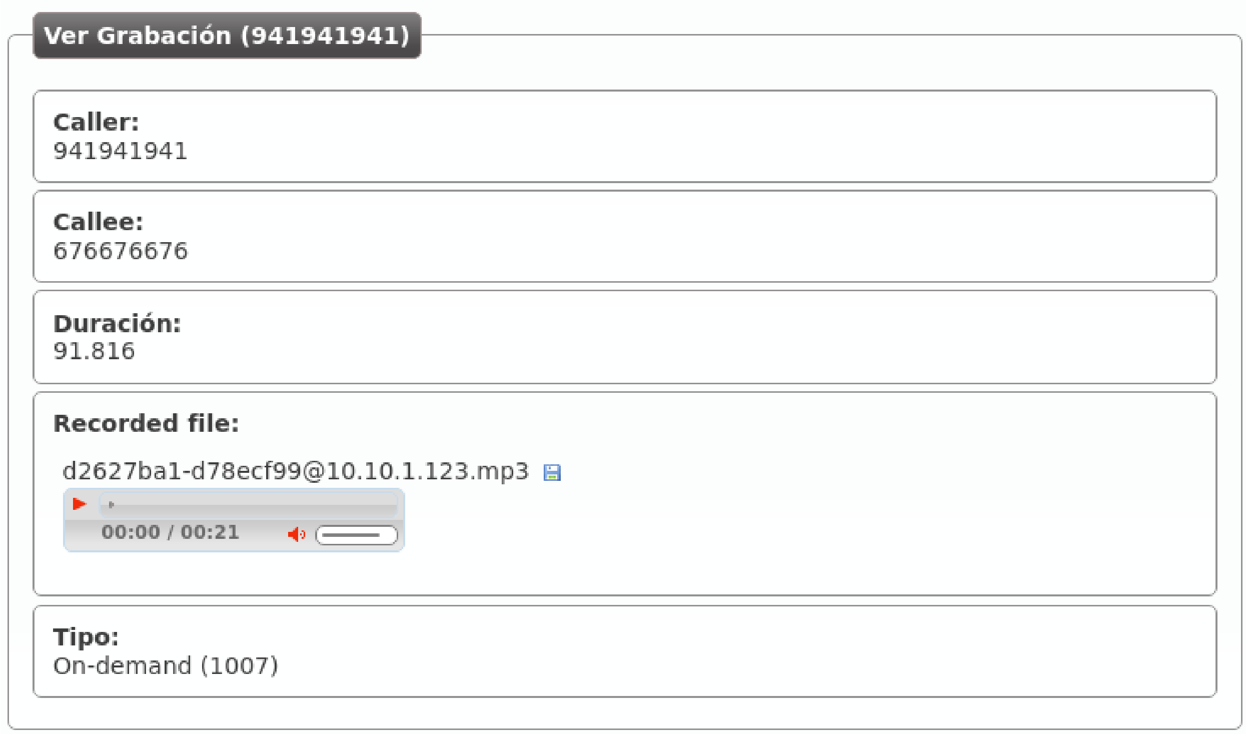
Listado de Grabaciones — Total:21 Registros

	Fecha	Tipo	Caller	Callee	Duración	Opciones
	22/09/2016 16:20:24	DDI	676676676	941941941	28.127	
	22/09/2016 16:16:36	DDI	941941941	91123456	60.385	
	20/09/2016 17:11:10	On-demand (1007)	941941941	676676676	91.816	

Recordings can be heard from the *web* or downloaded in MP3 format:



If the recording has been started on demand, it will also include the user that requested it:

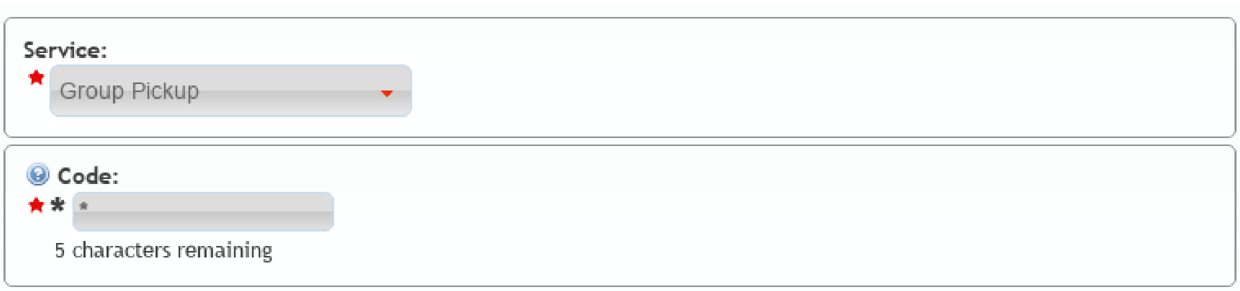


10.22 Company Services

> **Danger:** Services defined in this section **are not accessible during a conversation**. They are activated by **calling the codes**, not using DTMF codes while talking.

Each company can *customize* the default values assigned by the *brand operator* using the section **Company configuration > Services** and changing the codes listed there.

Company that wants to capture using ** instead of the default *95:



Hint: Services deleted by the *company admin* will not available to users.

10.23 Virtual Fax System

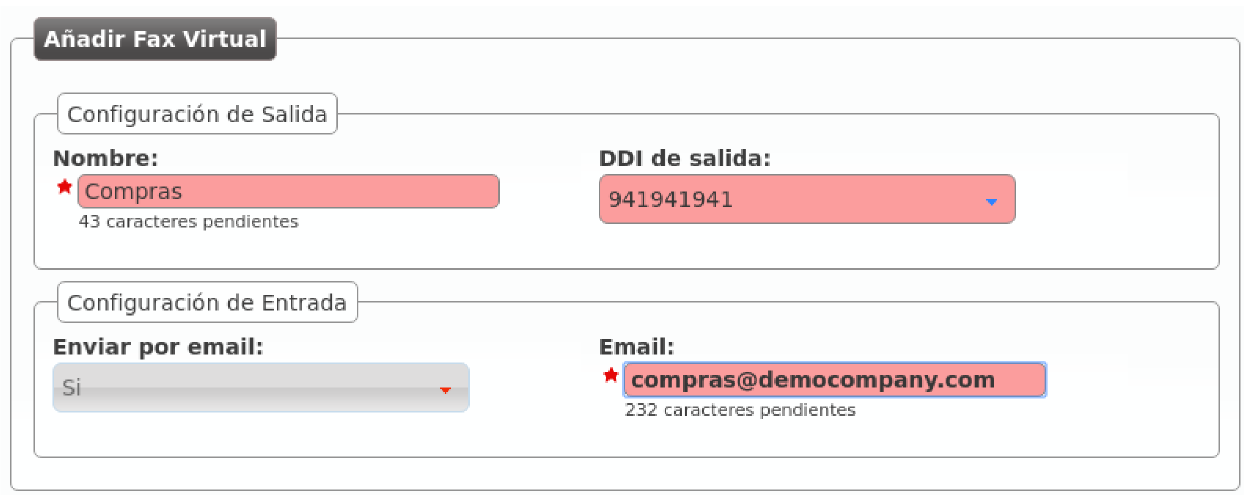
IvozProvider includes a simple but efficient *faxing* solution that allows:

- Sending PDF files via Fax.
- Receiving faxes through email or check them through the web portal.

Error: IvozProvider uses T.38 for both sending and receiving faxes. Brand Operator must use *peering contracts* that have support for it.

10.23.1 Creating a virtual fax

This is the interface that turns up when we create a new fax in section **Company configuration** > **Virtual Faxes**:



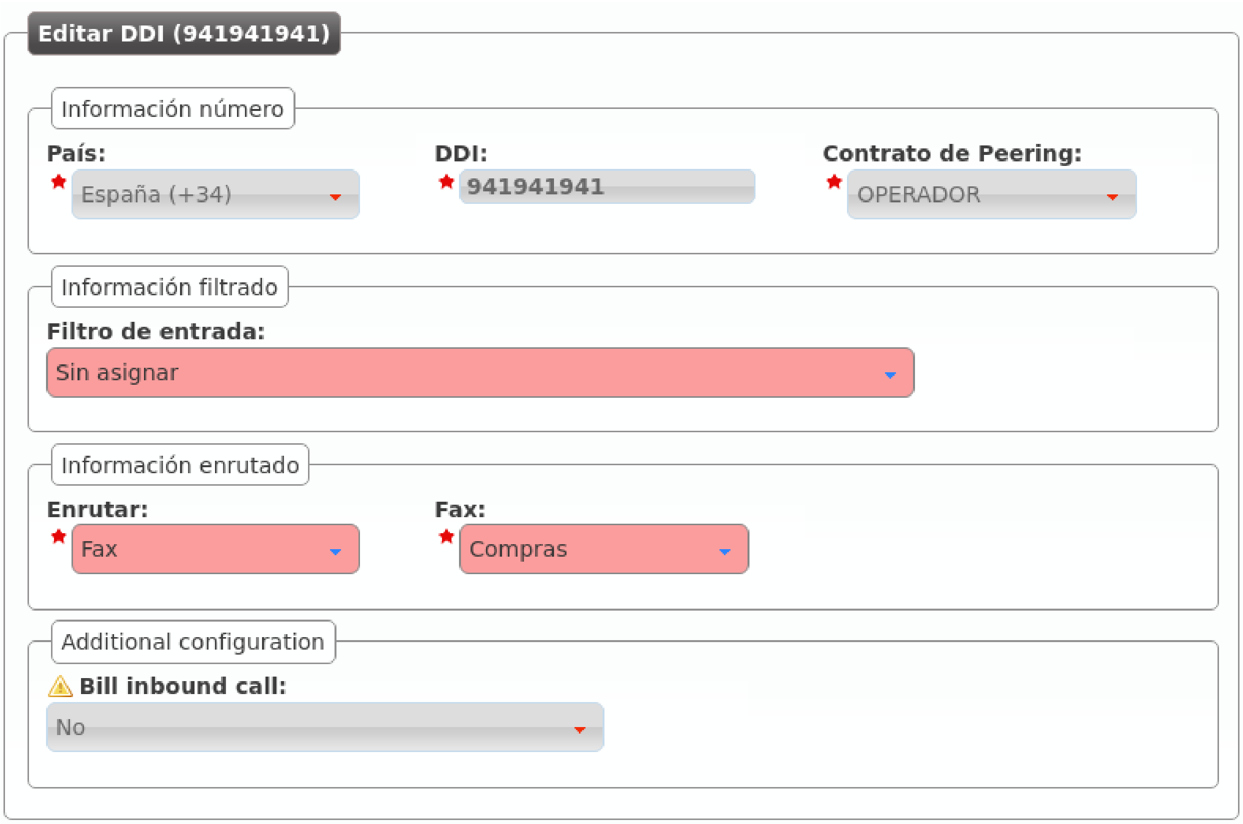
Fields are nearly self-explanatory:

Name Used by remaining section to reference a fax

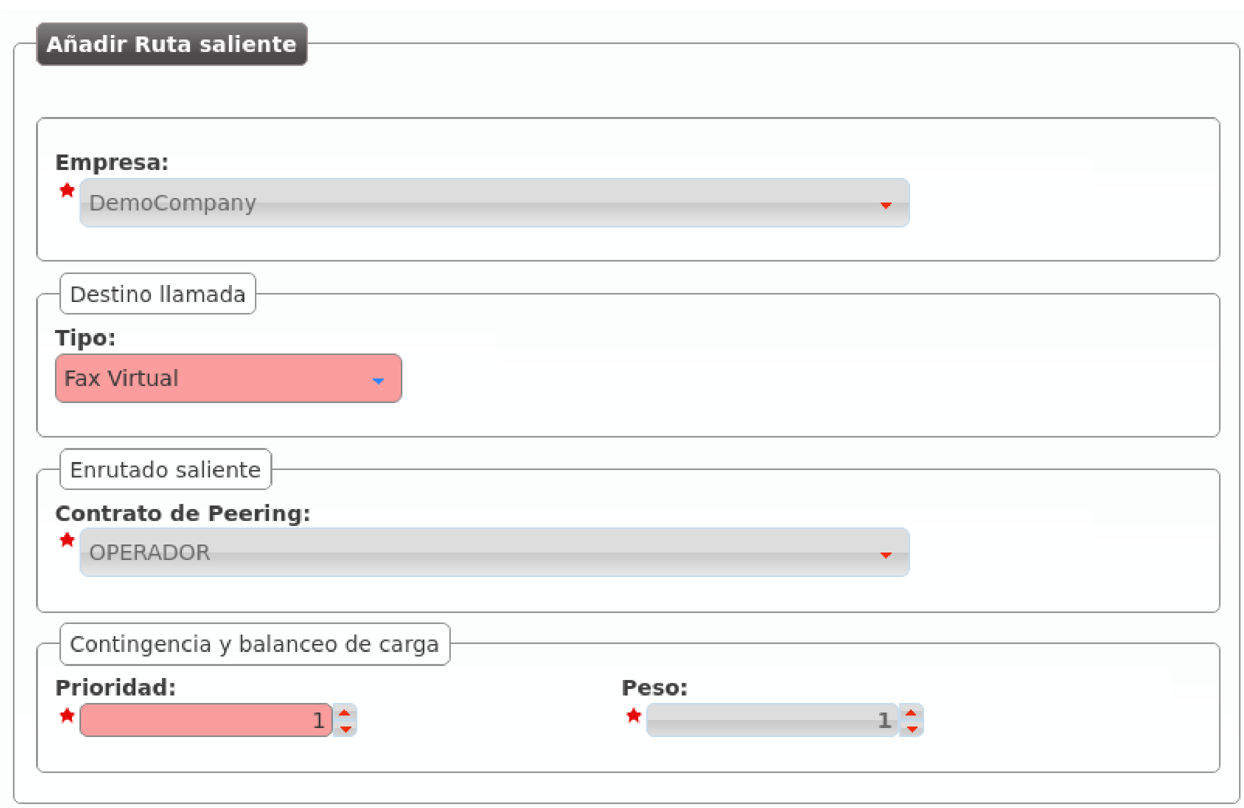
Email Email address when we want to receive incoming faxes (if we check 'Send by email')

Outbound DDI DDI used as source number for outgoing faxes

To receive faxes in this DDI, we need to point it to our new fax in the section **DDIs**:



Brand Operator can choose one or more *Outgoing Routes* for sending faxes:



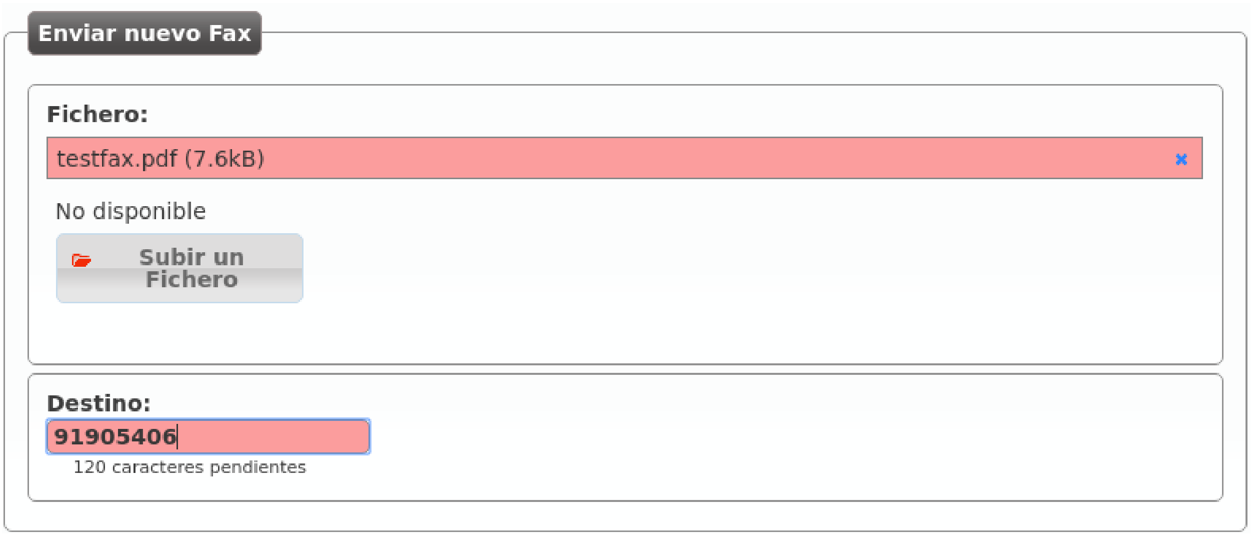
This route applies to all faxes sent by selected company (or for all companies).

Note: *load-balancing* y *failover* logics described in *previous sections* apply to faxes too.

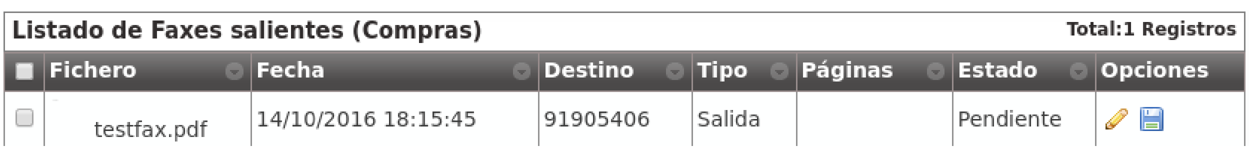
Important: If no fax-specific route is defined, faxes will be routed using standard call routes.

10.23.2 Sending a fax

Sending a fax is an easy task. First, we upload de PDF file and set the destination:



The list shows the fax and its status:

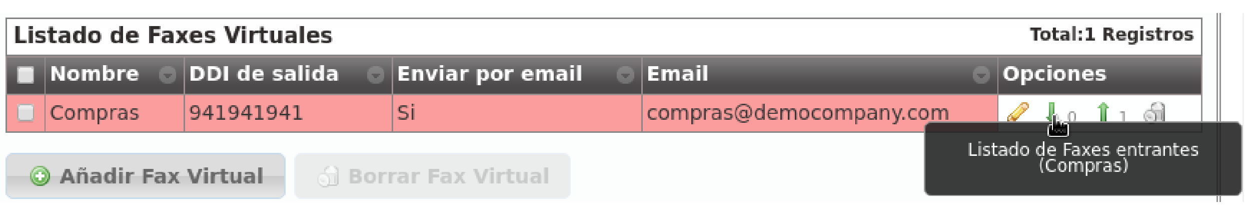


Listado de Faxes salientes (Compras) — Total:1 Registros

Fichero	Fecha	Destino	Tipo	Páginas	Estado	Opciones
testfax.pdf	14/10/2016 18:15:45	91905406	Salida		Pendiente	

10.23.3 Incoming faxes display

Apart from being received by mail, faxes can be watched and downloaded within the web portal too:



Listado de Faxes Virtuales — Total:1 Registros

Nombre	DDI de salida	Enviar por email	Email	Opciones
Compras	941941941	Si	compras@democompany.com	

10.24 Call Registry

Lists all the calls of the client:

Listado de Registros de llamadas — Total:297 Registros | Registros por página: 50

Fecha	Tipo	Subtype	Caller	Callee	Duración	Opciones
23/09/2016 12:31:56	interna		1007	3219	8	
23/09/2016 12:31:23	interna		1007	1020	37	
23/09/2016 12:30:28	interna		1007	1020	6	
23/09/2016 12:27:41	interna		1007	1020	20	
23/09/2016 12:07:46	interna		3219	1007	43	
23/09/2016 12:07:00	interna		1020	3219	88	
23/09/2016 12:06:39	interna		1007	1020	78	
23/09/2016 12:05:21	interna		1007	1020	15	

Note: CSV export makes possible to download the list for its later analysis.

CHAPTER 11

Residential Configuration

Residential clients are a special type of company that only provide a connectivity service with carriers through residential devices.

> **Attention:** Contrary to the Virtual PBX companies, all Residential clients use the brand domain to unequivocally identify their accounts. You'll need to configure Brand's domain to use this feature.

Hint: Residential clients can be enabled per Brand basis via Features.

The goal of this section will be describe each of the configuration settings associated with Residential clients included in IvozProvider.

11.1 Residential Devices

Residential Devices are the main routable option in Retail clients. More or less like *Friends* are to Virtual PBX Companies, accounts contain the required configurable options to provide a SIP connectivity service with IvozProvider and an external SIP entity.

> **Warning:** Although both **Contract peering** and **Retail accounts** are defined by the **brand operator**, the first ones are designed to connect with the public network while the second ones connect the system with other SIP agents.

11.1.1 Types of residential devices

There are 2 main types of SIP PBX that can use residential with IvozProvider:

- **Direct connection PBX**: IvozProvider must be able to talk SIP directly with this kind of accounts by just redirecting the traffic to the proper port of the public IP address of the PBX.
- **PBX behind NAT**: Not directly accesible. This kind of PBX must register at IvozProvider (just like all the *Terminals* do).

11.1.2 What kind of calls can be routed through a *Residential Devices*?

Contrary to Friends, **Residential Devices** have some simplifications and limitations.

- Residential Devices only route their assigned DDIs
- Residential Devices only place externals calls to Contract Peerings
- Residential Devices only receive external calls from Contract Peerings

11.1.3 Residential Devices Configuration

These are the configurable settings of *Retail accounts*:

Name Name of the **residential device**. This name must be unique in the whole brand so it's recommended to use some kind of secuential identifier. This will also be used in SIP messages (sent **From User**).

Description Optional. Extra information for this *residential device*.

Password When the *residential device* send requests, IvozProvider will authenticate it using this password. Like in other SIP agents in IvozProvider **using password IS A MUST**.

Direct connection If you choose 'Yes' here, you'll have to fill the protocol, address and port where this *residential device* can be contacted.

Fallback Outgoing DDI External calls from this *residential device* will be presented with this DDI, **unless the source presented matches a DDI belonging to the account**.

Country and Area code Used for number transformation from and to this residential device.

Allowed codecs Like a other SIP entities, *residential devices* will talk the selected codec.

From domain Request from IvozProvider to this account will include this domain in the From header.

11.2 Residential DDI filters

Residential External Filters can be assigned to DDIs to temporary forward calls to an external number.

11.2.1 Filters Configuration

This are the configurable settings of *Residential external filters*:

Name Name of the filter.

Number External Destination for this filter.

> **Attention:** Calls forwarded by a filter will keep the original caller identification, adding the fordwarding info in a SIP *Diversion* header.

11.3 Residential DDIs

DDIs are the external entry point from Contract Peerings to Residential Clients that can be routed through Residential Accounts.

We can assign an **external call filter** configured in *previous section*. Contrary to Virtual PBX External Call fiters, Residential DDIs filters only allow static redirection to another external number.

11.3.1 Residential DDI routes

Residential DDIs can only be routed to a *Residential Devices* or *Virtual Fax*.

Hint: Routing a DDI through a Residential device will allow to place external calls from that device presenting that DDI as origin.

11.3.2 Residential Recordings

If Residential Client has *Recordings* feature enabled, DDIs can also record incoming and/or outgoing calls.

CHAPTER 12

Retail Configuration

Retail clients are a special type of company that only provide a connectivity service with contract peerings through retail accounts.

Attention: Contrary to the Virtual PBX companies, all Retail clients use the brand domain to unequivocally identify their accounts. You'll need to configure Brand's domain to use this feature.

Hint: Retail clients can be enabled per Brand basis via Features.

The goal of this section will be describe each of the configuration settings associated with Retail clients included in IvozProvider.

12.1 Retail Accounts

Retail Accounts are the main routable option in Retail clients. More or less like *Friends* are to Virtual PBX Companies, accounts contain the required configurable options to provide a SIP connectivity service with IvozProvider and an external SIP entity.

Warning: Although both **Contract peering** and **Retail accounts** are defined by the **brand operator**, the first ones are designed to connect with the public network while the second ones connect the system with other SIP agents.

12.1.1 Types of retail accounts

There are 2 main types of SIP PBX that can use retail with IvozProvider:

- **Direct connection PBX**: IvozProvider must be able to talk SIP directly with this kind of accounts by just redirecting the traffic to the proper port of the public IP address of the PBX.
- **PBX behind NAT**: Not directly accesible. This kind of PBX must register at IvozProvider (just like all the *Terminals* do).

12.1.2 What kind of calls can be routed through a *Retail Account*?

Contrary to Friends, **Retail Accounts** have some simplifications and limitations.

- Retail Accounts only route their assigned DDIs
- Retail Accounts only place externals calls to Contract Peerings
- Retail Accounts only receive external calls from Contract Peerings

12.1.3 Retail Accounts Configuration

These are the configurable settings of *Retail accounts*:

Name Name of the **retail account**. This name must be unique in the whole brand so it's recommended to use some kind of secuential identifier. This will also be used in SIP messages (sent **From User**).

Description Optional. Extra information for this *retail account*.

Password When the *retail account* send requests, IvozProvider will authenticate it using this password. Like in other SIP agents in IvozProvider **using password IS A MUST**.

Direct connection If you choose 'Yes' here, you'll have to fill the protocol, address and port where this *retail account* can be contacted.

Fallback Outgoing DDI External calls from this *retail account* will be presented with this DDI, **unless the source presented matches a DDI belonging to the account**.

Country and Area code Used for number transformation from and to this retail account.

Allowed codecs Like a other SIP entities, *retail accounts* will talk the selected codec.

From domain Request from IvozProvider to this account will include this domain in the From header.

12.1.4 Asterisk as an account client

At the other end of a account can be any kind of SIP entity. This section takes as example an Asterisk PBX system using SIP channel driver that wants to connect to IvozProvider.

Account register

If the system can not be directly access, Asterisk will have to register in the platform (like a terminal will do).

Configuration will be something like this:

```
register => account-name:account-password@ivozprovider-brand.sip-domain.com
```

Account peer

```
[name-peer]
type=peer
host=ivozprovider-brand.sip-domain.com
context=XXXXXX
disallow=all
allow=alaw
defaultuser=account-name
secret=account-password
```

```
fromdomain=ivozprovider-brand.sip-domain.com
insecure=port,invite
```

Warning: *Account clients* MUST NOT challenge IvozProvider. That's why the *insecure* setting is used here.

12.2 Retail DDIs

DDIs are the external entry point from DDI Providers to Retail Clients that can be routed through Retail Accounts.

12.2.1 Retail DDI routes

Retail DDIs can only be routed to a *Retail Accounts*

Hint: Routing a DDI through a Retail account will allow to place external calls from that account presenting that DDI as origin.

CHAPTER 13

User Portal

IvozProvider provides a web portal where final users can do the following actions:

- See all calls he or she has been involved.
- Configure call forwards:
  - To voicemail
  - To an internal extension
  - To an external number
- Enable functionalities:
  - Call waiting
  - Do Not Disturb
- See the state of his or her SIP device registration

13.1 Access URLs

Prior to accessing to user portal, the URL addresses must be configured (domains in these URLs must point to any of the public IP addresses of the platform).

2 roles can perform this task:

13.1.1 God operator

In the section **Platform configuration > Brands** you can configure as many user URLs as you wish, using the button **Portal list** of each brand.

Note: URLs are linked to brands and god operator may choose where to create one shared user portal URL for all the companies of a brand or creating one per company.

Warning: URLs MUST be HTTPS.

This section also allows setting a logo per URL, a theme and a phrase to use as the title of user portal.

Hint: This allows creating corporative user portals.

13.1.2 Brand Operator

Brand Operator can also perform this same task in order to configure the user portal URLs of his companies.

This way, he can choose whether to configure one URL per Company (with custom domains, logos, theme and title) or sharing a global URL for all of them.

The section to do this is **Brand configuration > Portal URLs**.

13.2 Access credentials

Access credentials to user portal is configured in **Company configuration > Users** section.

Specifically:

- **Login information** block, the access of each user is enabled or disabled.
- You can set the **Password** too.
- To log in the user portal, the user must use his/her email address.

Warning: The **email** of each user MUST be **globally unique**.

CHAPTER 14

Security elements

14.1 Firewall

IvozProvider does not currently include a firewall but...

Danger: We **strongly encourage any production installation to implement a firewall** to protect the platform from the wild Internet.

The protection method could be:

- Local firewall based on iptables
- External firewall
- Both

14.1.1 Exposed ports/services

These are the **ports IvozProvider needs to expose** to work properly:

SIP signalling:

- Port 5060 (TCP/UDP)
- Port 5061 (TCP)
- Port 7060 (TCP/UDP) y 7061 TCP (just in case both ProxyUsers and ProxyTrunks share IP)

RTP audioflow:

- Port range 13000-19000 UDP

Web portal and provisioning:

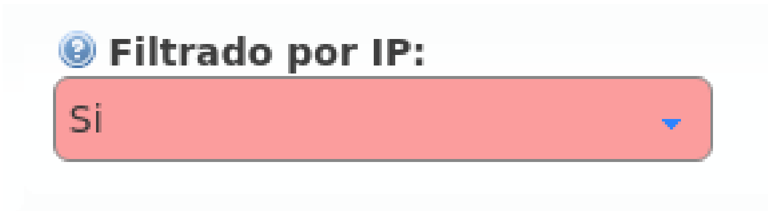
- Ports TCP 443, 1443 y 2443

Hint: We recommend using **iptables geoIP module** to drop connections from countries where we don't have any users.

14.2 Authorized company IP ranges

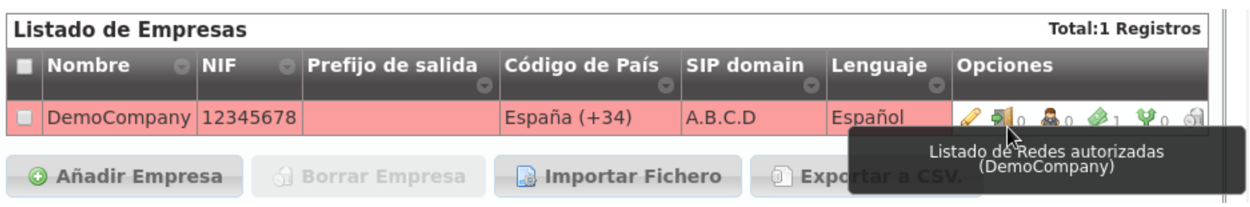
During the Company creating process, we skipped the security mechanism that **limits the IP addresses or ranges that the company terminals can use in their terminals**.

This can be activated in the section **Brand configuration** > **Virtual PBXs**:

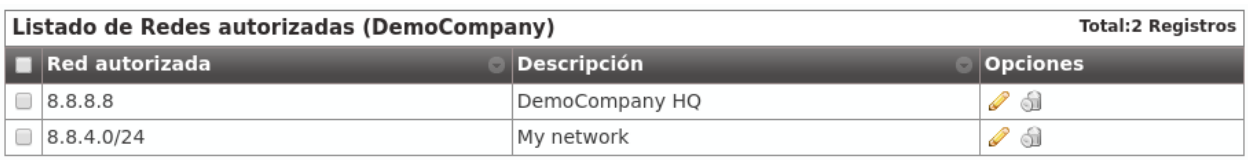


Rest of the users won't be allowed to connect from another network, even if the credentials are valid.

> **Warning:** Once the filter has been activated you **MUST** add networks or valid IP addresses, otherwise, all the calls will be rejected.



Both IP addresses or ranges can be used, in CIDR format (IP/mask):



Important: This mechanism limits the origin of the users of a company, it doesn't filter origin from **Contract Peerings**.

14.2.1 Roadwarrior users

Some companies have roadwarrior users that travel often and connect from external networks, forcing Companies to disable the IP filter security mechanism.

To solve this issue, there is a user option called **Calls for non-granted IPs** that enables these users to call from non-granted IPs while their companies are still protected with IP filter mechanism.

When users like theese call from non-granted IPs, their amount of concurrent outgoing calls are limited to 1, 2 or 3 to avoid being a security breach.

> **Warning:** Only calls generated by this kind of user (both internals and externals) are counted and limited, received calls are not affected by this setting (they are controlled with **MaxCalls** setting).

To sum up, with this feature:

- There are users that are allowed to make a fixed amount of calls from non-granted IPs.
- This calls from non-granted IPs are counted and limited.

Example 1 - Company without IP check

It doesn't matter if the user is allowed to make calls from non-granted IPs, as there are no non-granted IPs.

Example 2 - Company with IP check

- If the user is calling from one of the allowed IPs, it doesn't matter if the user is allowed to make calls from non-granted IPs: this calls are not counted nor limited.
- If the user is NOT calling from one of the allowed IPs, it is verified the amount of calls that this user is allowed to make. If the user is allowed to make calls from non-granted IPs and has not exceeded his limit, the call is granted and counted.

Note: If **Calls for non-granted IPs** is set to *None* the user must fulfill the IP policy of his company. See *Antiflood trusted IPs*.

14.3 Concurrent call limit

Another security mechanism can avoid that compromised credentials are used to establish hundreds of calls in little time. This mechanism **limits the number of external calls** of each company.

Note: This mechanism only takes into account the external channels, both incoming or outgoing external calls.

This can be configured in the company edit screen:

Tip: To disable this mechanism, set its value to 0.

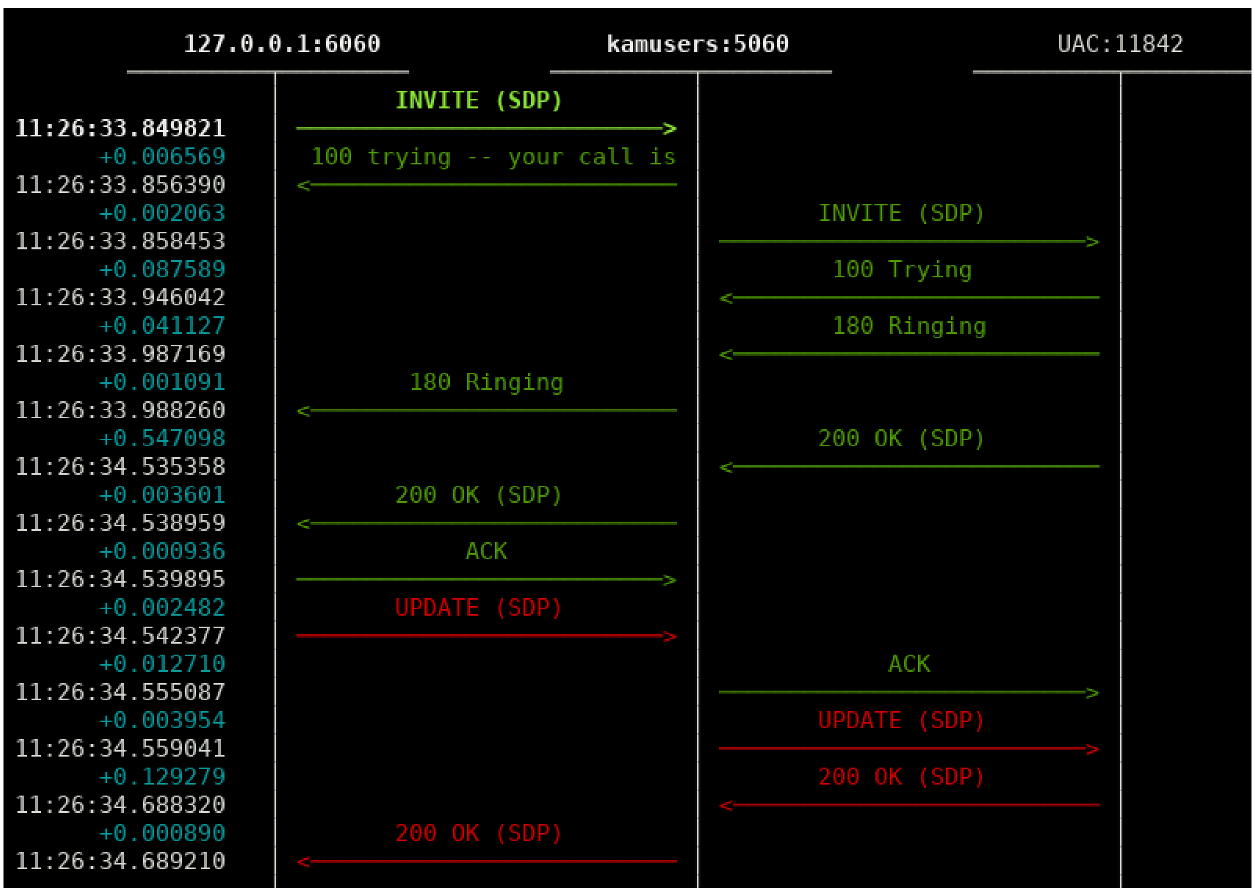
Maintenance and troubleshooting

This section described the tools included in IvozProvider to troubleshot any problems you may have:

15.1 Analyzing SIP traffic

Although all production IvozProvider installations mantained by Irontec include a Homer SIP Capture Server, it is not installed in the standlone version of IvozProvider. The reason behind this is that we prefer awesome SIPCAPTURE stack running on an additional machine.

sngrep Ncurses SIP Messages flow viewer developed by Irontec is currently the preferred tool to inspect SIP traffic included in IvozProvider.



15.1.1 sngrep

See live SIP traffic (all):

```
sngrep
```

See live SIP traffic related to calls:

```
sngrep -c
```

See live SIP traffic and capture RTP too:

```
sngrep -c -r
```

For more reference, visit sngrep official site.

15.1.2 Other capturing tools

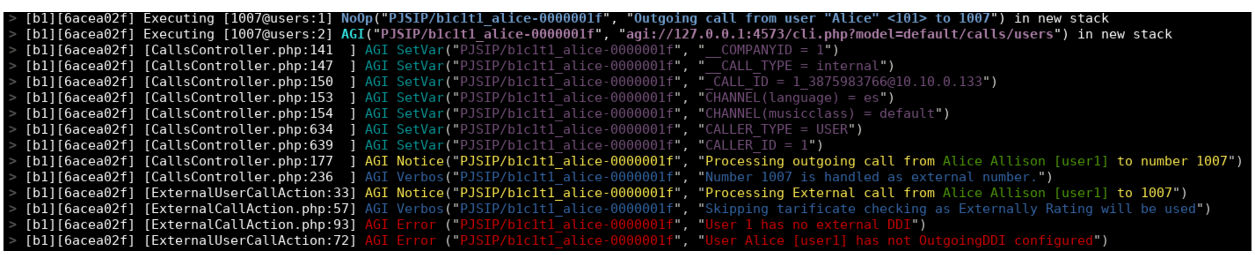
Although sngrep is our preferred capturing tool, IvozProvider ships other tools to capture SIP/RTP traffic, such as tcpdump and ngrep.

15.2 Log viewer

Although all production IvozProvider installations mantained by Irontec include a Graylog server, journalctl is currently the unique tool to inspect logs generated by different elements of the solution in the past.

15.2.1 Asterisk CLI

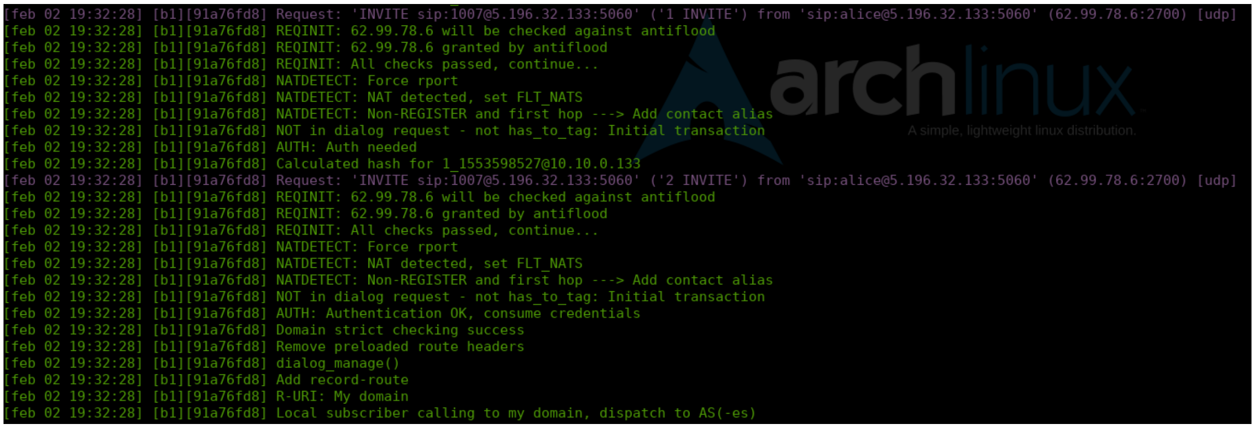
Asterisk CLI gives tons of realtime information too and are formatted beautifully to detect possible configuration errors:



You can access Asterisk CLI typing *ast* in the shell.

15.2.2 Kamailio realtime log viewing

You can see Kamailio logs in realtime too typing *kamtail-proxyusers* and *kamtail-proxytrunks* in the shell:



```
[feb 02 19:32:28] [b1][91a76fd8] Request: 'INVITE sip:1007@5.196.32.133:5060' ('1 INVITE') from 'sip:alice@5.196.32.133:5060' (62.99.78.6:2700) [udp]
[feb 02 19:32:28] [b1][91a76fd8] REQINIT: 62.99.78.6 will be checked against antiflood
[feb 02 19:32:28] [b1][91a76fd8] REQINIT: 62.99.78.6 granted by antiflood
[feb 02 19:32:28] [b1][91a76fd8] REQINIT: All checks passed, continue...
[feb 02 19:32:28] [b1][91a76fd8] NATDETECT: Force rport
[feb 02 19:32:28] [b1][91a76fd8] NATDETECT: NAT detected, set FLT_NATS
[feb 02 19:32:28] [b1][91a76fd8] NATDETECT: Non-REGISTER and first hop ---> Add contact alias
[feb 02 19:32:28] [b1][91a76fd8] NOT in dialog request - not has_to_tag: Initial transaction
[feb 02 19:32:28] [b1][91a76fd8] AUTH: Auth needed
[feb 02 19:32:28] [b1][91a76fd8] Calculated hash for 1_1553598527@10.10.0.133
[feb 02 19:32:28] [b1][91a76fd8] Request: 'INVITE sip:1007@5.196.32.133:5060' ('2 INVITE') from 'sip:alice@5.196.32.133:5060' (62.99.78.6:2700) [udp]
[feb 02 19:32:28] [b1][91a76fd8] REQINIT: 62.99.78.6 will be checked against antiflood
[feb 02 19:32:28] [b1][91a76fd8] REQINIT: 62.99.78.6 granted by antiflood
[feb 02 19:32:28] [b1][91a76fd8] REQINIT: All checks passed, continue...
[feb 02 19:32:28] [b1][91a76fd8] NATDETECT: Force rport
[feb 02 19:32:28] [b1][91a76fd8] NATDETECT: NAT detected, set FLT_NATS
[feb 02 19:32:28] [b1][91a76fd8] NATDETECT: Non-REGISTER and first hop ---> Add contact alias
[feb 02 19:32:28] [b1][91a76fd8] NOT in dialog request - not has_to_tag: Initial transaction
[feb 02 19:32:28] [b1][91a76fd8] AUTH: Authentication OK, consume credentials
[feb 02 19:32:28] [b1][91a76fd8] Domain strict checking success
[feb 02 19:32:28] [b1][91a76fd8] Remove preloaded route headers
[feb 02 19:32:28] [b1][91a76fd8] dialog_manage()
[feb 02 19:32:28] [b1][91a76fd8] Add record-route
[feb 02 19:32:28] [b1][91a76fd8] R-URI: My domain
[feb 02 19:32:28] [b1][91a76fd8] Local subscriber calling to my domain, dispatch to AS(-es)
```