The sip:provider CE Handbook mr5.5.10

Sipwise GmbH
<support@sipwise.com>

Contents

1 Introduction

1.1 About this Handbook

This handbook describes the architecture and the operational steps to install, operate and modify the Sipwise sip:provider CE.

In various chapters, it describes the system architecture, the installation and upgrade procedures and the initial configuration steps to get your first users online. It then dives into advanced preference configurations such as rewrite rules, call blockings, call forwards, etc.

There is a description of the customer self-care interface, how to configure the billing system and how to provision the system via the provided APIs.

Finally, it describes the internal configuration framework, the network configuration and gives hints about tweaking the system for security and performance.

1.2 What is the sip:provider CE?

The sip:provider CE is a SIP based Open Source Class5 VoIP soft-switch platform providing rich telephony services. It offers a wide range of features to end users (call forwards, voicemail, conferencing, call blocking, click-to-dial, call-lists showing near-realtime accounting information, etc.), which can be configured by them using the customer-self-care web interface. For operators, it offers a fully web-based administrative panel, allowing them to configure users, peerings, billing profiles, etc., as well as viewing real-time statistics of the system. For tight integration into existing infrastructures, it provides a powerful REST API.

The sip:provider CE can be installed in a few steps within a couple of minutes and requires no knowledge about configuration files of specific software components.

1.3 The Advantages of the sip:provider CE

Opposed to other free VoIP software, the sip:provider CE is not a single application, but a whole software platform, the Sipwise NGCP (Sipwise Next Generation Communication Platform), which is based on Debian GNU/Linux.

Using a highly modular design approach, the NGCP leverages popular open-source software like MySQL, NGINX, Kamailio, SEMS, Asterisk, etc. as its core building blocks. These blocks are glued together using optimized and proven configurations and workflows and are complemented by functionality developed by Sipwise to provide fully-featured and easy to operate VoIP services.

After downloading and starting the installer, it will fetch and install all the required Debian packages from the relevant Debian repositories. The installed applications are managed by the NGCP Configuration Framework, which makes it possible to change system parameters in a single place, so administrators don't need to have any knowledge of the dozens of different configuration files of the different packages. This provides a very easy and bullet-proof way of operating, changing and tweaking the otherwise quite complex system.

Once configured, integrated web interfaces are provided for both end users and administrators to use the sip:provider CE. By using the provided provisioning and billing APIs, it can be integrated tightly into existing OSS/BSS infrastructures to optimize workflows.

1.4 Who is the sip:provider CE for?

The sip:provider CE is specifically tailored to companies and engineers trying to start or experiment with a fully-featured SIP-based VoIP service without having to go through the steep learning curve of SIP signalling, integrating the different building blocks to make them work together in a reasonable way and implementing the missing components to build a business on top of that.

In the past, creating a business-ready VoIP service included installation and configuration of SIP software like Asterisk, OpenSER, Kamailio, etc., which can get quite difficult when it comes to implementing advanced features. It required implementing different web interfaces, billing engines and connectors to existing OSS/BSS infrastructure. These things are now obsolete due to the CE, which covers all these requirements.

1.5 Getting Help

1.5.1 Community Support

We have set up the *spce-user* mailing list, where questions are answered on a best-effort basis and discussions can be started with other community users.

1.5.2 Commercial Support

If you need professional help setting up and maintaining the sip:provider CE, send an email to support@sipwise.com.

Sipwise also provides training and commercial support for the platform. Additionally, we offer a migration path to the sip:provider PRO appliance, which is the commercial, carrier-grade version of the sip:provider CE. If the user base grows on the CE, this will allow operators to migrate seamlessly to a highly available and scalable platform with defined service level agreements, phone support and on-call duty. Please visit www.sipwise.com for more information on commercial offerings.

2 Architecture

The sip:provider CE platform is one single node running all necessary components of the system. The components are outlined in the following figure:

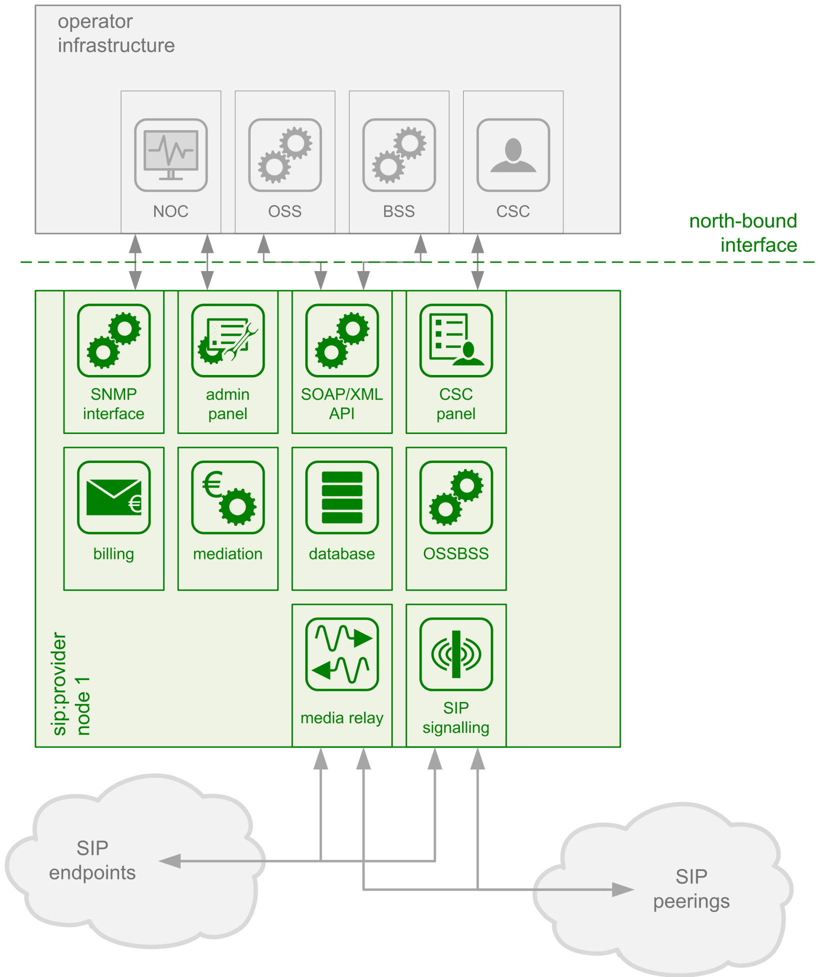

Figure 1: Architecture Overview

The main building blocks of the sip:provider CE are:

- SIP Signaling and Media Relay
- Provisioning
- Mediation and Billing

2.1 SIP Signaling and Media Relay

In SIP-based communication networks, it is important to understand that the signaling path (e.g. for call setup and tear-down) is completely independent of the media path. On the signaling path, the involved endpoints negotiate the call routing (which user calls which endpoint, and via which path - e.g. using SIP peerings or going through the PSTN - the call is established) as well as the media attributes (via which IPs/ports are media streams sent and which capabilities do these streams have - e.g. video using H.261 or Fax using T.38 or plain voice using G.711). Once the negotiation on signaling level is done, the endpoints start to send their media streams via the negotiated paths.

The components involved in SIP and Media on the sip:provider CE are shown in the following figure:

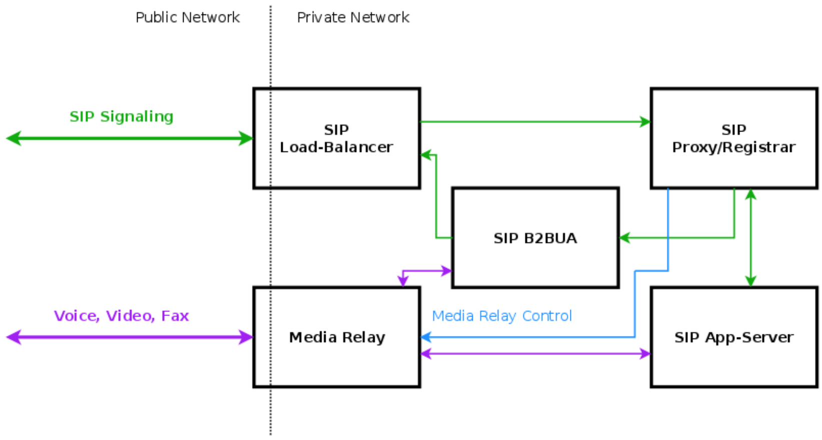

Figure 2: SIP and Media Relay Components

2.1.1 SIP Load-Balancer

The SIP load-balancer is a Kamailio instance acting as ingress and egress point for all SIP traffic to and from the system. It's a high-performance SIP proxy instance based on Kamailio and is responsible for sanity checks of inbound SIP traffic. It filters broken SIP messages, rejects loops and relay attempts and detects denial-of-service and brute-force attacks and gracefully handles them to protect the underlying SIP elements. It also performs the conversion of TLS to internal UDP and vice versa for secure signaling between endpoints and the sip:provider CE, and does far-end NAT traversal in order to enable signaling through NAT devices.

The load-balancer is the only SIP element in the system which exposes a SIP interface to the public network. Its second leg binds in the switch-internal network to pass traffic from the public internet to the corresponding internal components.

The name load-balancer comes from the fact that in the commercial version, when scaling out the system beyond just one pair of servers, the load-balancer instance becomes its own physical node and then handles multiple pairs of proxies behind it.

On the public interface, the load-balancer listens on port 5060 for UDP and TCP, as well as on 5061 for TLS connections. On the internal interface, it speaks SIP via UDP on port 5060 to the other system components, and listens for XMLRPC connections on TCP port 5060, which is used by the OSSBSS system to control the daemon.

Its config files reside in `/etc/ngcp-config/templates/etc/kamailio/lb/`, and changes to these files are applied by executing `ngcpcfg apply` *`my commit message`*.

Tip

The SIP load-balancer can be managed via the commands `/etc/init.d/kamailio-lb start`, `/etc/init.d/kamailio-lb stop` and `/etc/init.d/kamailio-lb restart`. Its status can be queried by executing `/etc/init.d/kamailio-lb status`. Also `ngcp-kamctl lb` and `ngcp-sercmd lb` are provided for querying kamailio functions, for example: `ngcp-sercmd lb htable.dump ipban`.

2.1.2 SIP Proxy/Registrar

The SIP proxy/registrar (or short *proxy*) is the work-horse of the sip:provider CE. It's also a separate Kamailio instance running in the switch-internal network and is connected to the provisioning database via MySQL, authenticates the endpoints, handles their registrations on the system and does the call routing based on the provisioning data. For each call, the proxy looks up the provisioned features of both the calling and the called party (either subscriber or domain features if it's a local caller and/or callee, or peering features if it's from/to an external endpoint) and acts accordingly, e.g. by checking if the call is blocked, by placing call-forwards if applicable and by normalizing numbers into the appropriate format, depending on the source and destination of a call.

It also writes start- and stop-records for each call, which are then transformed into call detail records (CDR) by the mediation system.

If the endpoints indicate negotiation of one or more media streams, the proxy also interacts with the *Media Relay* to open, change and close port pairs for relaying media streams over the sip:provider CE, which is especially important to traverse NAT.

The proxy listens on UDP port 5062 in the system-internal network. It cannot be reached directly from the outside, but only via the SIP load-balancer.

Its config files reside in `/etc/ngcp-config/templates/etc/kamailio/proxy/`, and changes to these files are applied by executing `ngcpcfg apply` *`my commit message`*.

Tip

The SIP proxy can be controlled via the commands `/etc/init.d/kamailio-proxy start`, `/etc/init.d/kamailio-proxy stop` and `/etc/init.d/kamailio-proxy restart`. Its status can be queried by executing `/etc/init.d/kamailio-proxy status`. Also `ngcp-kamctl proxy` and `ngcp-sercmd proxy` are provided for querying kamailio functions, for example: `ngcp-kamctl proxy ul show`.

2.1.3 SIP Back-to-Back User-Agent (B2BUA)

The SIP B2BUA (also called SBC within the system) decouples the first call-leg (calling party to sip:provider CE) from the second call-leg (sip:provider CE to the called party).

The software part used for this element is SEMS.

This element is typically optional in SIP systems, but it is always used for SIP calls (INVITE) that don't have the sip:provider CE as endpoint. It acts as application server for various scenarios (e.g. for feature provisioning via Vertical Service Codes and as Conferencing Server) and performs the B2BUA decoupling, topology hiding, caller information hiding, SIP header and Media feature filtering, outbound registration, outbound authentication and call length limitation as well as Session Keep-Alive handler.

Due to the fact that typical SIP proxies (like the load-balancer and proxy in the sip:provider CE) do only interfere with the content of SIP messages where it's necessary for the SIP routing, but otherwise leave the message intact as received from the endpoints, whereas the B2BUA creates a new call leg with a new SIP message from scratch towards the called party, SIP message sizes are reduced significantly by the B2BUA. This helps to bring the message size under 1500 bytes (which is a typical default value for the MTU size) when it leaves the sip:provider CE. That way, chances of packet fragmentation are quite low, which reduces the risk of running into issues with low-cost SOHO routers at customer sides, which typically have problems with UDP packet fragmentation.

The SIP B2BUA only binds to the system-internal network and listens on UDP port 5080 for SIP messages from the load-balancer or the proxy, on UDP port 5040 for control messages from the cli tool and on TCP port 8090 for XMLRPC connections from the OSSBSS to control the daemon.

Its configuration files reside in `/etc/ngcp-config/templates/etc/ngcp-sems`, and changes to these files are applied by executing `ngcpcfg apply` *`my commit message`*.

Tip

The SIP B2BUA can be controlled via the commands `/etc/init.d/ngcp-sems start`, `/etc/init.d/ngcp-sems stop` and `/etc/init.d/ngcp-sems restart`. Its status can be queried by executing `/etc/init.d/ngcp-sems status`

2.1.4 SIP App-Server

The SIP App-Server is an Asterisk instance used for voice applications like Voicemail and Reminder Calls. Asterisk uses the MySQL database as a message spool for voicemail, so it doesn't directly access the file system for user data. The voicemail plugin is a slightly patched version based on Asterisk 1.4 to make Asterisk aware of the sip:provider CE internal UUIDs for each subscriber. That way a SIP subscriber can have multiple E164 phone numbers, but all of them terminate in the same voicebox.

The App-Server listens on the internal interface on UDP port 5070 for SIP messages and by default uses media ports in the range from UDP port 10000 to 20000.

The configuration files reside in `/etc/ngcp-config/templates/etc/asterisk`, and changes to these files are applied by executing `ngcpcfg apply` *`my commit message`*.

Tip

The SIP App-Server can be controlled via the commands `/etc/init.d/asterisk start`, `/etc/init.d/asterisk stop` and `/etc/init.d/asterisk restart`. Its status can be queried by executing `/etc/init.d/asterisk status`

2.1.5 Media Relay

The Media Relay (also called *rtpengine*) is a Kernel-based packet relay, which is controlled by the SIP proxy. For each media stream (e.g. a voice and/or video stream), it maintains a pair of ports in the range of port number 30000 to 40000. When the media streams are negotiated, rtpengine opens the ports in user-space and starts relaying the packets to the addresses announced by the endpoints. If packets arrive from different source addresses than announced in the SDP body of the SIP message (e.g. in case of NAT), the source address is implicitly changed to the address the packets are received from. Once the call is established and the rtpengine has received media packets from both endpoints for this call, the media stream is pushed into the kernel and is then handled by a custom Sipwise iptables module to increase the throughput of the system and to reduce the latency of media packets.

The rtpengine internally listens on UDP port 12222 for control messages from the SIP proxy. For each media stream, it opens two pairs of UDP ports on the public interface in the range of 30000 and 40000 per default, one pair on even port numbers for the media data, and one pair on the next odd port numbers for metadata, e.g. RTCP in case of RTP streams. Each endpoint communicates with one dedicated port per media stream (opposed to some implementations which use one pair for both endpoints) to avoid issues in determining where to send a packet to. The rtpengine also sets the QoS/ToS/DSCP field of each IP packet it sends to a configured value, 184 (0xB8, *expedited forwarding*) by default.

The kernel-internal part of the rtpengine is facilitated through an *iptables* module having the target name `RTPENGINE`. If any additional firewall or packet filtering rules are installed, it is imperative that this rule remains untouched and stays in place. Otherwise, if the rule is removed from iptables, the kernel will not be able to forward the media packets and forwarding will fall back to the user-space daemon. The packets will still be forwarded normally, but performance will be much worse under those circumstances, which will be especially noticeable when a lot of media streams are active concurrently. See the section on *Firewalling* for more information.

The rtpengine configuration file is `/etc/ngcp-config/templates/etc/default/ngcp-rtpengine-daemon`, and changes to this file are applied by executing `ngcpcfg apply` *`my commit message`*. The UDP port range can be configured via the `config.yml` file under the section `rtpproxy`. The QoS/ToS value can be changed via the key `qos.tos_rtp`.

Tip

The Media Relay can be controlled via the commands `/etc/init.d/ngcp-rtpengine-daemon start`, `/etc/init.d/ngcp-rtpengine-daemon stop` and `/etc/init.d/ngcp-rtpengine-daemon restart`. Its status can be queried by executing `/etc/init.d/ngcp-rtpengine-daemon status`

2.2 Redis Database

The redis database is used as a high-perfomance key/value storage for global system data. This includes calls information and concurrent calls counters for customers and subscribers, etc..

3 Initial Installation

3.1 Prerequisites

For an initial installation of the sip:provider CE, it is mandatory that your production environment meets the following criteria:

HARDWARE REQUIREMENTS

- Recommended: Dual-core, x86_64 compatible, 3GHz, 4GB RAM, 128GB HDD
- Minimum: Single-core, x86_64 compatible, 1GHz, 2GB RAM, 16GB HDD

SUPPORTED OPERATING SYSTEMS

- Debian 9 (stretch) 64-bit

INTERNET CONNECTION

- Hardware needs connection to the Internet

Important
Only **Debian 9 (stretch) 64-bit** is currently supported as a host system for the sip:provider CE.

Important
It is **HIGHLY** recommended that you use a **dedicated server** (either a physical or a virtual one) for sip:provider CE, because the installation process will wipe out existing MySQL databases and modify several system configurations.

3.2 Using the NGCP install CD (recommended)

The custom Sipwise NGCP install CD provides the ability to easily install sip:provider CE, including automatic partitioning and installation of the underlying Debian system. You can install the latest available or Long Term Support (LTS) sip:provider CE release using sip:provider CE install CD image (checksums: sha1, md5).

Important
The NGCP install CD automatically takes care of partitioning, any present data will be overwritten! While the installer prompts for the disk that should be used for installation before its actual execution, it's strongly recommended to boot the ISO in an environment with empty disks or disks that you don't plan to use for anything else than the newly installed NGCP system.

To configure network options please choose the according boot menu entries with either `DHCP` or `static NW` (static network configuration) in its name. Press the `<tab>` key on the menu entry you want to boot, then adjust the `ip=...` and `dns=...` boot options as needed.

Tip

When DHCP is available in your infrastructure then you shouldn't have to configure anything, just choose `DHCP` and press enter. If network configuration still doesn't work as needed a console based network configuration system will assist you in setting up your network configuration.

Also, you can use sip:provider CE install CD to boot the Grml (Debian based live system) rescue system, check RAM using a memory testing tool or install plain Debian squeeze/wheezy/jessie/stretch system for manual installation using NGCP installer.

3.3 Using the NGCP installer

3.3.1 Installing the Operating System

You need to install Debian 9 (stretch) 64-bit on the server. A **basic** installation without any additional task selection (like *Desktop System*, *Web Server* etc.) is sufficient.

Tip

Sipwise recommends using the latest Netinstall ISO as installation medium.

Important

If you use other kinds of installation media (e.g. provided by your hosting provider), prepare for some issues that might come up during installation. For example, you might be forced to manually resolve package dependencies in order to install the sip:provider CE. Therefore, it is HIGHLY RECOMMENDED to use a clean Debian installation to simplify the installation process.

Note

If you installed your system using the Debian CDs/DVDs (so neither using the NGCP install CD nor the Debian Netinstall ISO) apt-get might prompt to insert disk to proceed during NGCP installation. The prompt won't be visible for you and installation hangs. Please disable the cdrom entries in /etc/apt/sources.list and enable a Debian mirror (e.g. http://http.debian.net/debian) instead.

3.3.1.1 Using special Debian setups

If you plan to install the sip:provider CE on Virtual Hosting Providers like *Dreamhost* with their provided Debian installer, you might need to manually prepare the system for the NGCP installation, otherwise the installer will fail installing certain package versions required to function properly.

Using Dreamhost Virtual Private Server

A Dreamhost virtual server uses apt-pinning and installs specific versions of MySQL and apache, so you need to clean this up beforehand.

Note

Apache is not used by default since mr3.6.1, still better to remove pinned Apache version.

```
apt-get remove --purge mysql-common ndn-apache22
mv /etc/apt/preferences /etc/apt/preferences.bak
apt-get update
apt-get dist-upgrade
```

Warning

Be aware that this step will break your web-based system administration provided by Dreamhost. Only do it if you are certain that you won't need it.

3.3.2 Installing the sip:provider CE

The sip:provider CE is based on the *Sipwise NGCP*, so download and install the latest *Sipwise NGCP* installer package:

```
PKG=ngcp-installer-mr5.5.10.deb
wget http://deb.sipwise.com/spce/${PKG}
dpkg -i ${PKG}
```

Run the installer as root user:

```
ngcp-installer
```

Note

You can find the previous versions of *Sipwise NGCP* installer package here.

The installer will ask you to confirm that you want to start the installation. Read the given information **carefully**, and if you agree, proceed with *y*.

The installation process will take several minutes, depending on your network connection and server performance. If everything goes well, the installer will (depending on the language you use), show something like this:

```
Installation finished. Thanks for choosing NGCP sip:provider Community Edition.
```

During the installation, you can watch the background processing by executing the following command on a separate console:

```
tail -f /tmp/ngcp-installer.log
```

3.4 Using a pre-installed virtual machine

For quick test deployments, pre-installed virtualization images are provided. These images are intended to be used for quick test, not recommended for production use.

3.4.1 Vagrant box for VirtualBox

Vagrant is an open-source software for creating and configuring virtual development environments. Sipwise provides a so called Vagrant base box for your service, to easily get direct access to your own sip:provider CE Virtual Machine without any hassles.

Note

The following software must be installed to use Vagrant boxes:

- VirtualBox (v.5.1.26+ is recommended)
- Vagrant v.1.9.7+

Get your copy of sip:provider CE by running:

```
vagrant init spce-mr5.5.10 http://deb.sipwise.com/spce/images/sip_provider_CE_mr5.5.10 ↩
    _vagrant.box
vagrant up
```

As soon as the machine is up and ready you should have your local copy of sip:provider CE with the following benefits:

- all the software and database are automatically updated to the latest available version
- the system is configured to use your LAN IP address (received over DHCP)
- basic SIP credentials to make SIP-2-SIP calls out of the box are available

Use the following command to access the terminal:

```
vagrant ssh
```

or login to Administrator web-interface at https://127.0.0.1:1443/login/admin (with default user *administrator* and password *administrator*).

There are two ways to access VM resources, through NAT or Bridge interface:

Note

a.b.c.d is IP address of VM machine received from DHCP; x.y.z.p is IP address of your host machine

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Table 1: Vagrant based VirtualBox VM interfaces:

Description	Host-only address	LAN address	Notes
SSH	ssh://127.0.0.1:2222	ssh://a.b.c.d:22 or ssh://x.y.z.p:2222	Also available via "*vagrant ssh*"
Administrator interface	https://127.0.0.1:1443/-login/admin	https://a.b.c.d:1443/login/-admin or https://x.y.z.p:1443/login/-admin	
New Customer self care interface	https://127.0.0.1:1443	https://a.b.c.d:1443 or https://x.y.z.p:1443	new self-care interface based on powerful ngcp-panel framework
Old Customer self care interface	https://127.0.0.1:22443	https://a.b.c.d:443 or https://x.y.z.p:22443	will be removed in upcoming releases
Provisioning interfaces	https://127.0.0.1:2443	https://a.b.c.d:2443 or https://x.y.z.p:2443	
SIP interface	not available	sip://a.b.c.d:5060	Both TCP and UDP are available.

Note

VM ports smaller then 1024 mapped to ports 22<vm_port> through NAT, otherwise root on host machine requires to map them. It means SSH port 22 mapped to port 2222, WEB port 443 → 22443.

VM IP address (a.b.c.d), as well as SIP credentials will be printed to terminal during "vagrant up" stage, e.g.:

```
[20_add_sip_account] Adding SIP credentials...
[20_add_sip_account]   - removing domain 192.168.1.103 with subscribers
[20_add_sip_account]   - adding domain 192.168.1.103
[20_add_sip_account]   - adding subscriber 43991002@192.168.1.103 (pass: 43991002)
[20_add_sip_account]   - adding subscriber 43991003@192.168.1.103 (pass: 43991003)
[20_add_sip_account]   - adding subscriber 43991004@192.168.1.103 (pass: 43991004)
[20_add_sip_account]   - adding subscriber 43991005@192.168.1.103 (pass: 43991005)
[20_add_sip_account]   - adding subscriber 43991006@192.168.1.103 (pass: 43991006)
[20_add_sip_account]   - adding subscriber 43991007@192.168.1.103 (pass: 43991007)
[20_add_sip_account]   - adding subscriber 43991008@192.168.1.103 (pass: 43991008)
[20_add_sip_account]   - adding subscriber 43991009@192.168.1.103 (pass: 43991009)
[20_add_sip_account] You can USE your VM right NOW:  https://192.168.1.103:1443/login/admin
```

To turn off your sip:provider CE virtual machine, just type:

```
vagrant halt
```

To completely remove sip:provider CE virtual machine, use:

```
vagrant destroy
vagrant box remove spce-mr5.5.10
```

Further documentation for Vagrant is available at the official Vagrant website.

Vagrant usage tips:

- Default SSH login is *root* and password is *sipwise*. SSH connection details can be displayed via:

```
vagrant ssh-config
```

- You can download a Vagrant box for VirtualBox from here manually (checksums: sha1, md5).

3.4.2 VirtualBox image

You can download a VirtualBox image from here (checksums: sha1, md5). Once you have downloaded the file you can import it to VirtualBox via its import utility.

The format of the image is *ova*. If you have VirtualBox 3.x running, which is not compatible with *ova* format, you need to extract the file with any *tar* compatible software and import the *ovf* file which is inside the archive.

On Linux, you can do it like this:

```
tar xvf sip_provider_CE_mr5.5.10_virtualbox.ova
```

On Windows, right-click on the ova file, choose *Open with* and select *WinZIP* or *WinRAR* or any other application able to extract *tar* archives. Extract the files to any place and import the resulting *ovf* file in VirtualBox.

Considerations when using this virtual machine:

- You will need a 64bit guest capable VirtualBox setup.
- The root password is *sipwise*
- You should use *bridge mode* networking (adjust your bridging interface in the virtual machine configuration) to avoid having the sip:provider CE behind NAT.
- You'll need to adjust your timezone and keyboard layout.
- The network configuration is set to DHCP. You'll need to change it to the appropriate static configuration.
- As the virtual image is a static file, it won't contain the most updated versions of our software. Please upgrade the system via apt as soon as you boot it for the first time.

3.4.3 VMware image

You can download a VMware image from here (checksums: sha1, md5). Once you have downloaded the file just extract the *zip* file and copy its content to your virtual machines folder.

Considerations when using this virtual machine:

- You will need a 64bit guest capable vmware setup.
- The root password is *sipwise*
- You'll need to adjust your timezone and keyboard layout.
- The network configuration is set to DHCP. You'll need to change it to the appropriate static configuration.
- As the virtual image is a static file, it won't contain the most updated versions of our software. Please upgrade the system via apt as soon as you boot it for the first time.

3.4.4 Amazon EC2 image

Sipwise provides AMI (Amazon Machine Images) images in all Amazon EC2 regions for the latest and LTS sip:provider CE releases. Please find the appropriate AMI ID for your region in release announcement.

Note
The following documentation will use Amazon region *eu-west-1* with AMI ID *ami-8bef6cfc* as an example. Please find the appropriate AMI ID for your region in the latest release announcement.

As a next step please visit https://console.aws.amazon.com/ec2/v2/home?region=eu-west-1 with your EC2 account.

Choose "Launch Instance":

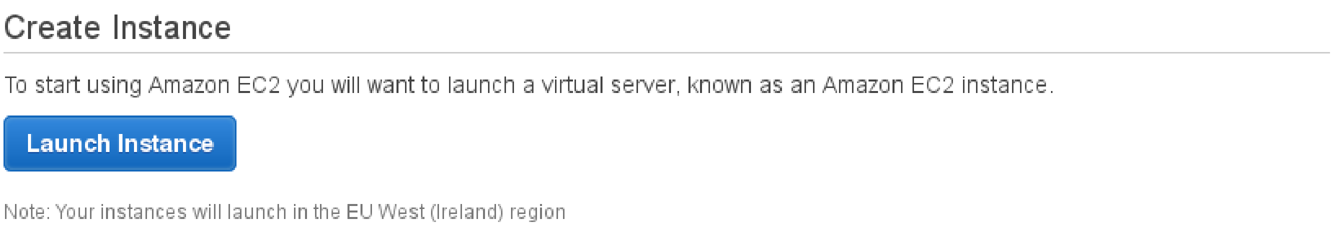


Figure 3: Launch Amazon EC2 Instance for your region

Select "Community AMIs" option, enter "ami-8bef6cfc" inside the search field and press "Select" button:

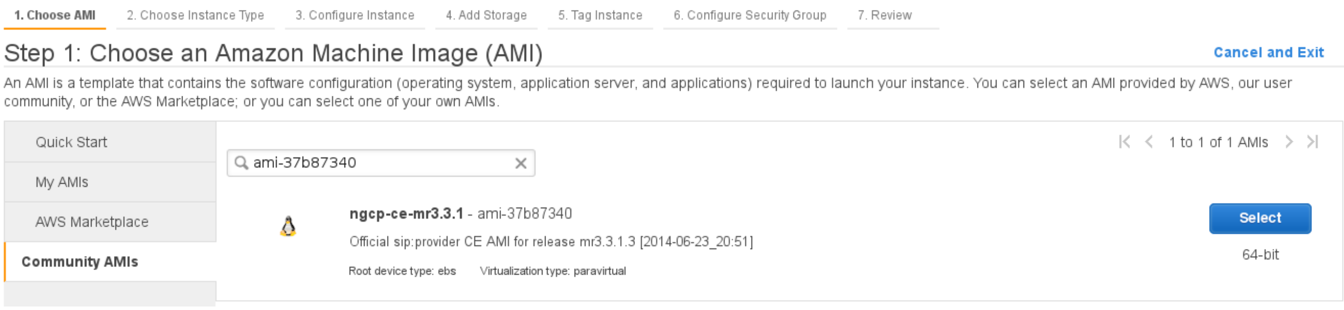


Figure 4: Choose sip:provider CE image (different for each region)

Select the Instance Type you want to use for running sip:provider CE (recommended: >=2GB RAM):

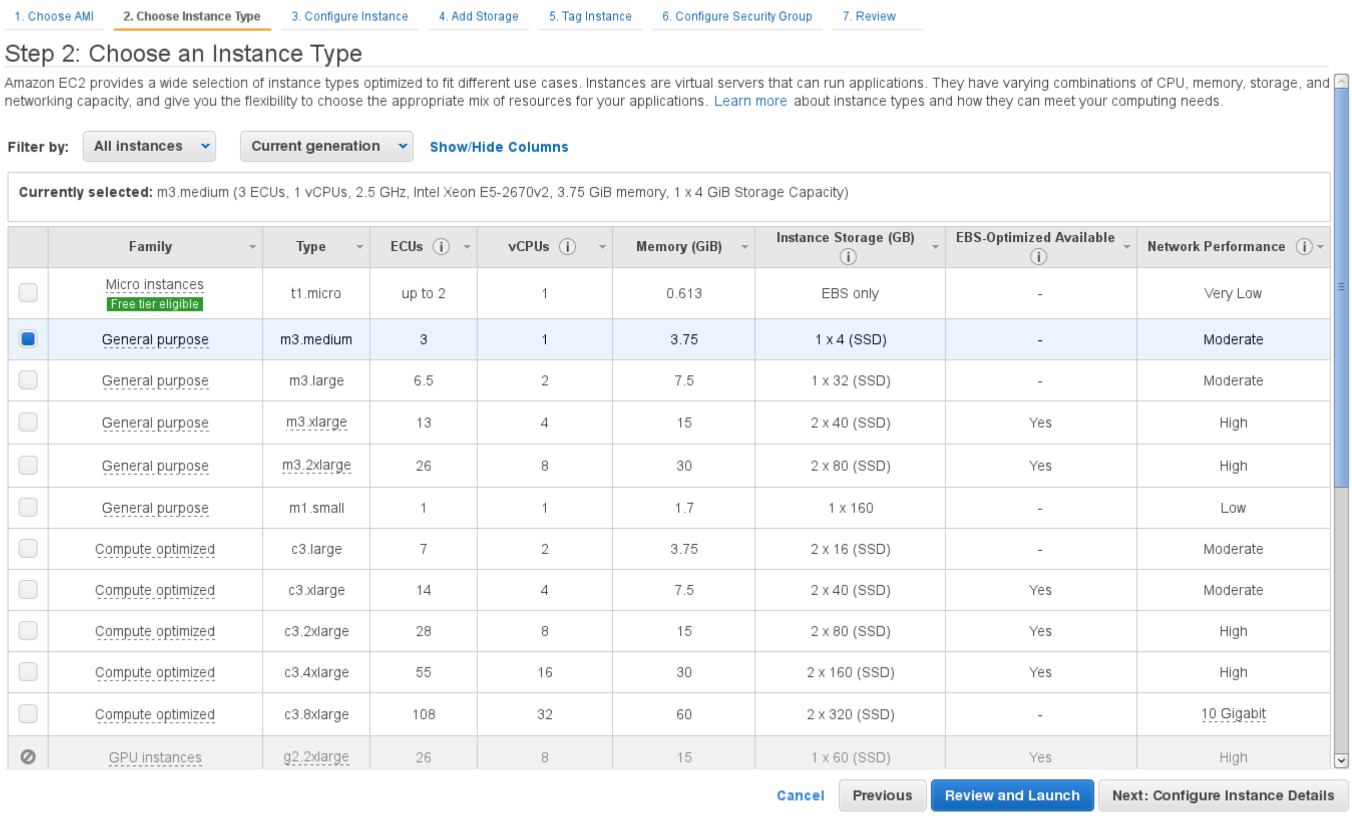


Figure 5: Choose Amazon EC2 instance

Tip

Do not forget to tune necessary sip:provider CE performance parameters depending on Amazon EC2 instance type and performance you are looking for. Sipwise image is tunned for minimum performance to fit Micro instances. Feel free to read more about sip:provider CE performance tuning in Section 16.7.

Run through next configuration options

- Configure Instance: optional (no special configuration required from sip:provider CE)
- Add Storage: choose >=8GB disk size (no further special configuration required from sip:provider CE)
- Tag Instance: optional (no special configuration required from sip:provider CE)
- Configure Security Group: create a new security group (SSH on port 22, HTTPS on port 443, TCP on ports 1443, 2443, 1080 and 5060 as well as UDP on port 5060 are suggested)

Note

Please feel free to restrict the *Source* options in your Security Group to your own (range of) IP addresses.

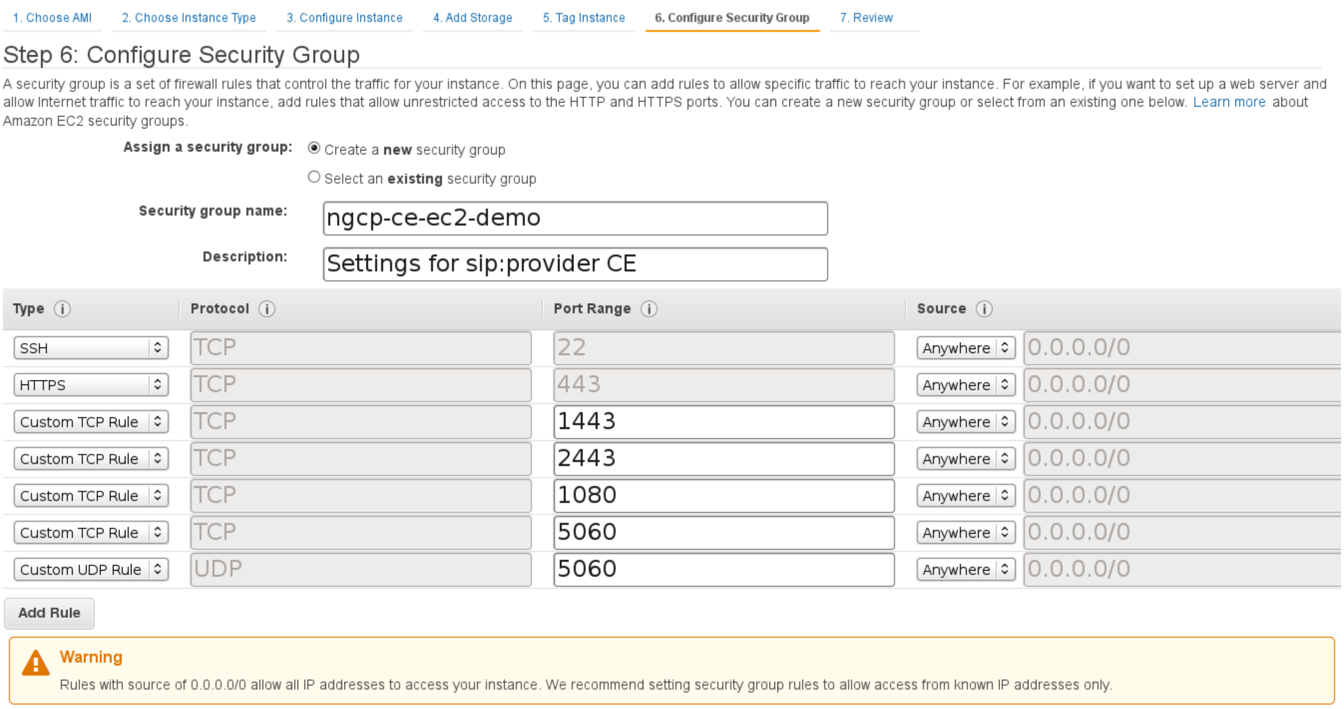


Figure 6: Configure Security Group

Finally Review instance launch and press "Launch" button:

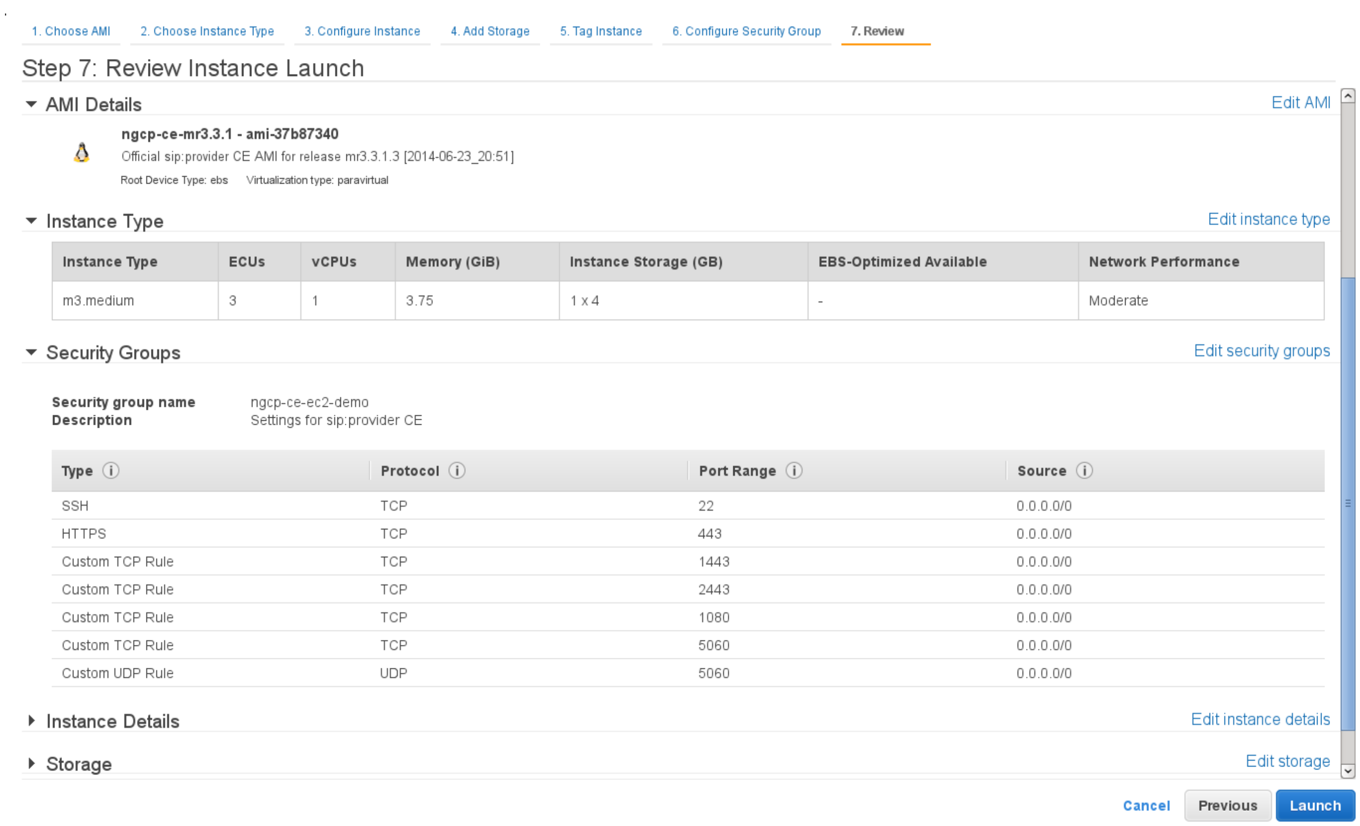


Figure 7: Launch Amazon EC2 instance with sip:provider CE

Choose an existing key pair which you want to use for logging in, or create a new one if you don't have one.

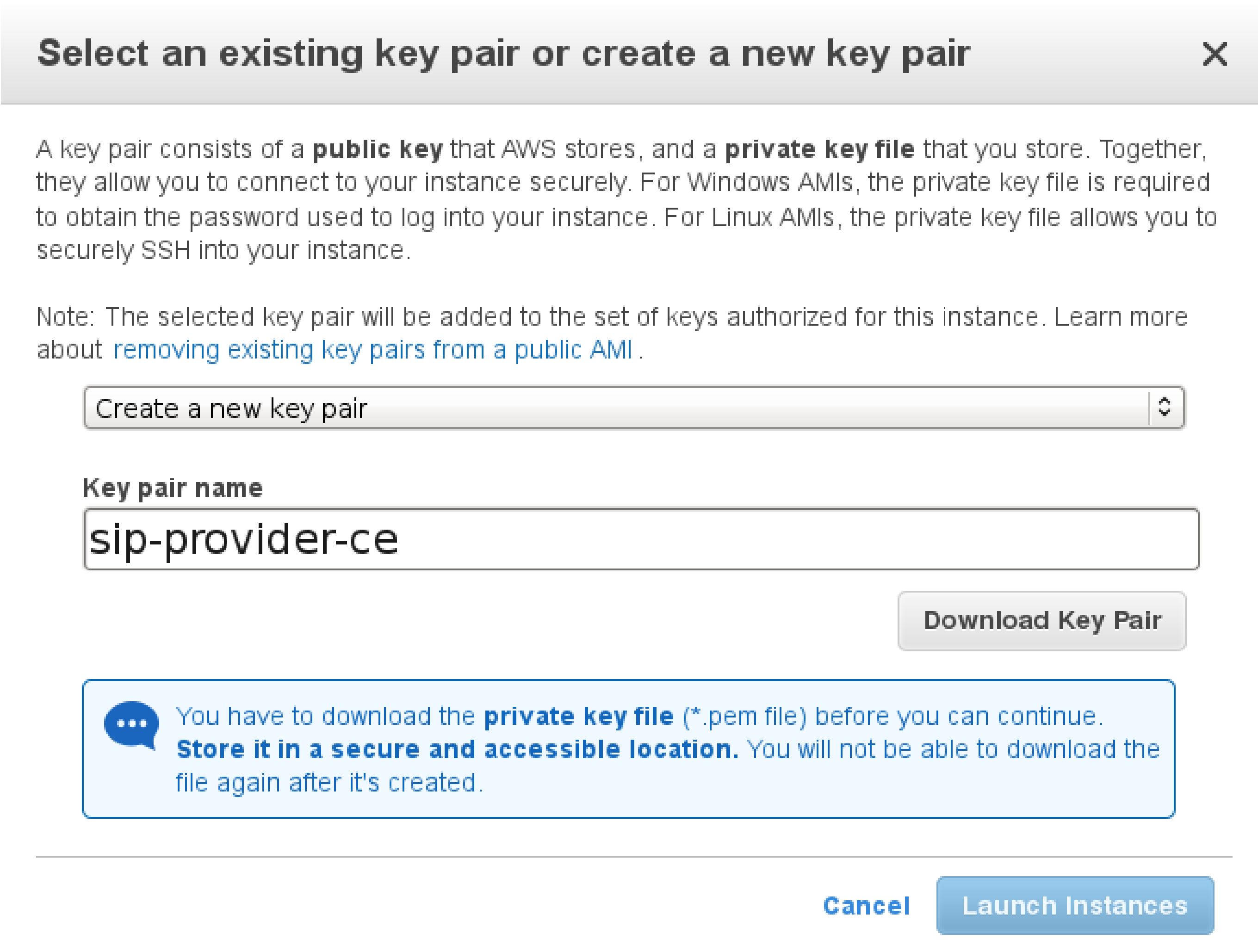


Figure 8: Choose key pair to access sip:provider CE

You should have a running instance after a few seconds/minutes now (check DNS name/IP address).

Figure 9: Running Amazon EC2 sip:provider CE instance

First step should be logging in to the Admin panel (username *administrator*, password *administrator*) and changing the default password: https://$DNS:1443/login/admin and then follow Section 16 to secure your installation.

Logging in via SSH should work now, using the key pair name (being sip-provider-ce.pem as $keypair in our example) and the

DNS name/IP address the system got assigned.

```
ssh -i $keypair.pem admin@$DNS
```

Now you can increase your privileges to user *root* for further system configuration:

```
sudo -s
```

Don’t forget to add the Advertised IP for kamailio lb instance, since it’s required by the Amazon EC2 network infrastructure:

```
ngcp-network --set-interface=eth0 --advertised-ip=<your_public_amazon_ip>
```

and apply your changes:

```
ngcpcfg apply 'add advertised-ip on interface eth0'
```

Now feel free to use your newly started Amazon EC2 sip:provider CE instance!

Warning

Do not forget to stop unnecessary instance(s) to avoid unexpected costs (see http://aws.amazon.com/ec2/pricing/).

4 Initial System Configuration

After the installation went through successfully, you are ready to adapt the system parameters to your needs to make the system work properly.

4.1 Network Configuration

If you have only one network card inside your system, its device name is *eth0*, it’s configured and only IPV4 is important to you then there should be nothing to do for you at this stage. If multiple network cards are present, your network card does *not* use *eth0* for its device name or you need IPv6 then the only parameter you need to change at this moment is the listening address for your SIP services.

To do this, you have to specify the interface where your listening address is configured, which you can do with the following command (assuming your public interface is *eth0*):

```
ngcp-network --set-interface=eth0 --ip=auto --netmask=auto --hwaddr=auto
ngcp-network --move-from=lo --move-to=eth0 --type=web_ext --type=sip_ext --type=rtp_ext -- ←
    type=ssh_ext --type=web_int
```

If you want to enable IPv6 as well, you have to set the address on the proper interface as well, like this (assuming you have an IPv6 address *fdda:5cc1:23:4:0:0:0:1f* on interface *eth0*):

```
ngcp-network --set-interface=eth0 --ipv6='FDDA:5CC1:23:4:0:0:0:1F'
```

Tip

Always use a full IPv6 address with 8 octets. Leaving out zero octets (e.g. `FDDA:5CC1:23:4::1F`) is **not allowed**.

Important

You should use the IPv6 address in **upper-case** because LB (kamailio) handles the IPv6 addresses internally in upper-case format.

If you haven't fully configured your network interfaces, do this by adapting also the file */etc/network/interfaces*:

```
vim /etc/network/interfaces
```

Add or adapt your interface configuration accordingly. For example, if you just want to use the system in your internal network 192.168.0.0/24, it could look something like this:

```
auto lo
iface lo inet loopback

auto eth0
iface eth0 inet static
        address 192.168.0.10
        netmask 255.255.255.0
        gateway 192.168.0.1
        dns-nameservers 8.8.8.8
        dns-search yourdomain.com
```

```
/etc/init.d/networking restart
```

4.2 Apply Configuration Changes

In order to apply the changes you made to */etc/ngcp-config/config.yml*, you need to execute the following command to re-generate your configuration files and to automatically restart the services:

```
ngcpcfg apply 'added network interface'
```

Tip

At this point, your system is ready to serve.

4.3 Start Securing Your Server

During installation, the system user *cdrexport* is created. This jailed system account is supposed to be used to export CDR files via sftp/scp. Set a password for this user by executing the following command:

```
passwd cdrexport
```

The installer has set up a MySQL database on your server. You need to set a password for the MySQL root user to protect it from unauthorized access by executing this command:

```
mysqladmin password <your mysql root password>
```

For the *Administrative Web Panel* located at *https://<your-server-ip>:1443/login/admin*, a default user *administrator* with password *administrator* has been created. Connect to the panel (accept the SSL certificate for now) using those credentials and change the password of this user by going to *Settings*→*Administrators* and click the *Edit* when hovering over the row.

4.4 Configuring the Email Server

The NGCP installer will install *mailx* (which has *Exim4* as MTA as a default dependency) on the system, however the MTA is not configured by the installer. If you want to use the *Voicemail-to-Email* feature of the Voicebox, you need to configure your MTA properly. If you are fine to use the default MTA *Exim4*, execute the following command:

```
dpkg-reconfigure exim4-config
```

Depending on your mail setup in your environment (whether to use a smarthost or not), configure Exim accordingly. In the most simple setup, apply the following options when prompted for it:

- **General type of mail configuration:** `internet site; mail is sent and received directly using SMTP`
- **System mail name:** the FQDN of your server, e.g. `ce.yourdomain.com`
- **IP-addresses to listen on for incoming SMTP connections:** `127.0.0.1`
- **Other destinations for which mail is accepted:** the FQDN of your server, e.g. `ce.yourdomain.com`
- **Domains to relay mail for:** leave empty
- **Machines to relay mail for:** leave empty
- **Keep number of DNS-queries minimal (Dial-on-Demand)?** `No`
- **Delivery method for local mail:** `mbox format in /var/mail/`
- **Split configuration into small files?** `No`

Important

You are free to install and configure any other MTA (e.g. postfix) on the system, if you are more comfortable with that.

4.5 Advanced Network Configuration

You have a typical test deployment now and you are good to go, however you may need to do extra configuration depending on the devices you are using and functionality you want to achieve.

4.5.1 Audiocodes devices workaround

As reported by many users, Audiocodes devices suffer from a problem where they replace `127.0.0.1` address in Record-Route headers (added by the sip:provider CE's internal components) with its own IP address. The problem has been reported to Audiocodes but as of end 2012 the fixed firmware is not available yet so supposedly the whole range of Audiocodes devices, including but not limited to the MP202, MP252 CPEs as well as Audiocodes media gateways, is malfunctioning. As a workaround, you may change the internal IP address from `127.0.0.1` to some dummy network interface. Please execute the following command (in this example `192.168.2.2` is a new internal IP address):

Enable module dummy at */etc/ngcp-config/config.yml*:

```
modules:
  - enable: yes
    name: dummy
    options: numdummies=2
```

```
ifconfig dummy0 192.168.2.2 netmask 255.255.255.0
```

Adapt your */etc/network/interfaces* file accordingly:

```
auto dummy0
iface dummy0 inet static
address 192.168.2.2
netmask 255.255.255.0
```

Update the network configuration in the sip:provider CE:

```
ngcp-network --set-interface=dummy0 --ip=auto --netmask=auto
ngcp-network --move-from=lo --move-to=dummy0 --type=sip_int --type=web_int
```

Refer to the Network Configuration chapter for more details about the `ngcp-network` tool.

Apply configuration:

```
ngcpcfg apply 'audiocodes devs workaround'
```

4.6 What's next?

To test and use your installation, you need to follow these steps now:

1. Create a SIP domain

2. Create some SIP subscribers
3. Register SIP endpoints to the system
4. Make local calls and test subscriber features
5. Establish a SIP peering to make PSTN calls

Please read the next chapter for instructions on how to do this.

5 VoIP Service Administration Concepts

5.1 Contacts

A contact contains information such as the name, the postal and email addresses, and others. A contact's main purpose is to identify entities (resellers, customers, peers and subscribers) it is associated with.

A person or an organization may represent a few entities and it is handy to create a corresponding organization's contact beforehand and use it repeatedly when creating new entities. In this case we suggest populating the **External #** field to distinguish between customers associated with the same contact.

Reseller	Contact	Customer	External #
Default	Rylic Longstaff	DTS	0007
		Morning Times	0008
TelephOne	Clare Fenn	Lantern Co	—
	Ike Leonard	City Bank	—

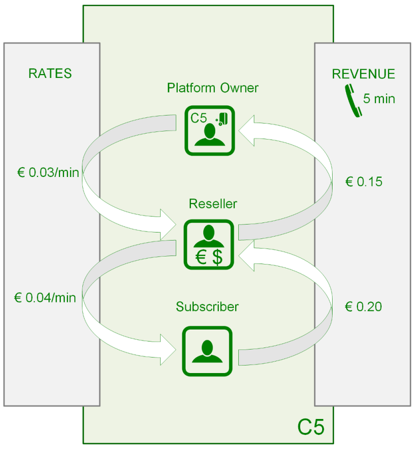
Note that the only required contact field is **email**. For contacts associated with customers, it will be used for sending invoices and notifications such as password reset, new subscriber creation and others. A contact for a subscriber is created automatically but only if you specify an email address for this subscriber. It is mainly used to send notification messages, e.g. in case of a password reset.

5.2 Resellers

The reseller model allows you to expand your presence in the market by including virtual operators in the sales chain. A virtual operator can be a company without its own VoIP platform and even without a technical background, but with sales presence in a market. You define such a company as a reseller in the platform: grant limited access to the administrative web interface (the reseller administrator will only see his own customers, domains and billing profiles) and define wholesale rates for this reseller. Then, the reseller is free to operate under its own brand, make up its retail rates, establish the customer base and resell your services to its customers. The reseller's profit is a margin between the wholesale and retail rates.

Let us consider an example:

- You operate in Munich and provide residential and business services.
- A company Cheap Call that has a strong presence in Frankfurt offers to resell your services under its own brand in this city.
- You define wholesale rates for Cheap Call, such as calls to Argentina at €0,03.
- Cheap Call defines its retail price and offers calls to Argentina at €0,04.
- When one of Cheap Call's subscribers makes a 5-minute call to Argentina, this subscriber will be charged €0,20.
- You will get €0,15 revenue and Cheap Call's profit will be €0,20 - €0,15 = €0,05.



A reseller usually uses dedicated IP addresses or SIP domain names to provide services. Also, a reseller can rebrand the self-care web interface for its customers and select languages per SIP domain that allows the reseller to operate even in multiple countries.

5.3 SIP Domain

A SIP domain represents an external Internet address where your subscribers register their SIP phones to make calls or send messages. The SIP domain also contains particular default configuration for all the subscribers registered with this SIP domain. A SIP domain can be a regular FQDN (e.g. sip.yourdomain.com) or a NAPTR/SRV record. Using IP addresses for SIP domains in production is **strongly discouraged**.

5.3.1 Additional SIP Domains

You can create as many SIP domains as required to satisfy your networking or marketing requirements, e.g.:

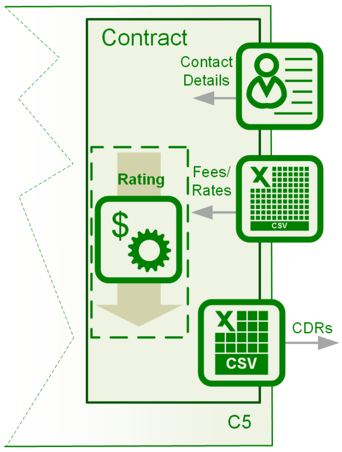
- A dedicated SIP domain is *suggested* per CloudPBX customer.
- A separate SIP domain may be dedicated to every whitelabel reseller.
- Multiple SIP domains may be used to provide services in different countries or regions.
- Multiple SIP domains may be used to brand your own services.

Domain	Purpose
sip.yourdomain.com	Your own domain for retail customers
sip.enterprise.com	Your big customer with Cloud PBX
sip.reseller.com	Your white-label reseller
sip.yourdomain.de	Your domain for providing a new service in another country

5.4 Contracts

A contract is a combination of a *contact* and a *billing profile*, hence it represents a business contract for your resellers and peering partners.

Contracts can be created in advance on the *Reseller and Peering Contracts* page, or immediately during creation of a peer or a reseller.



Note that the *customer* entity (described below) is a special type of the contract. A customer entity has an email and an invoice templates in addition to a contact and a billing profile.

5.5 Customers

A customer is a physical or legal entity whom you provide the VoIP service with and send invoices to. Here are the main features of a customer:

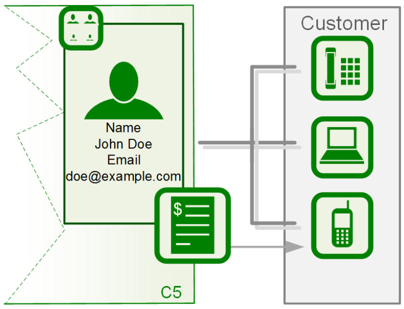
- Contains the contact and legal information. For example, an address or an email address for invoicing.
- Associated with a billing profile (to define fees per destination) and tracks the balance (used mostly for post-paid customers).
- Contains a certain number of subscribers who actually use the service and whose calls appear in the customer's list of CDRs.
- Provides some default parameters for all its subscribers. For example, voice prompts and call restriction.

Here are two common examples of the customer model:

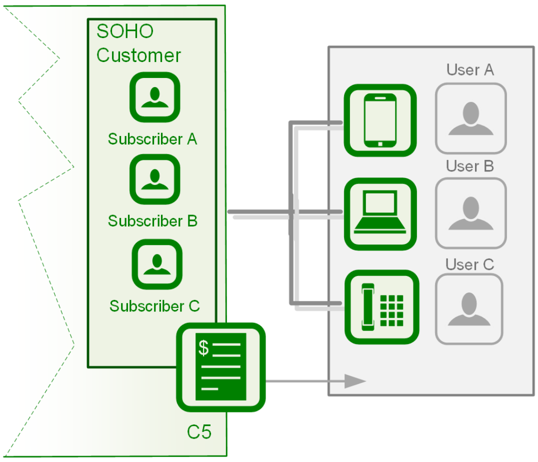
5.5.1 Residential and SOHO customers

With this service you provide your residential and SOHO customers with one or multiple numbers and offer the service on a post-paid basis.

For a residential customer you usually create one *customer* entity with one *subscriber* under it. A residential customer can register multiple devices with the same number thus having a convenient Viber or Skype-like service: any device can be used to make a call and all of them will ring simultaneously when there is an incoming call. At the end of the billing period, you send an invoice to the customer.

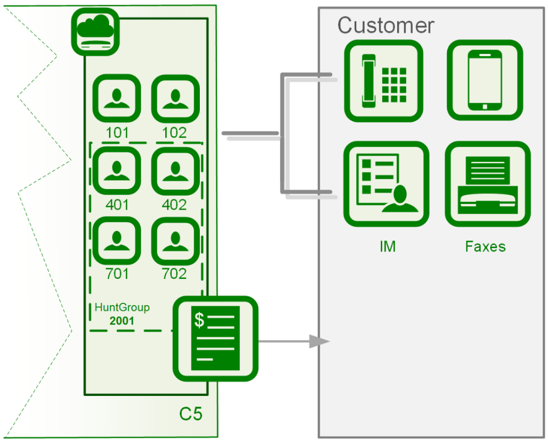


For SOHO customers you usually create multiple subscribers under the same customer and assign every subscriber a dedicated number to allow users make and receive calls. A common invoice will contain calls of all the subscribers.



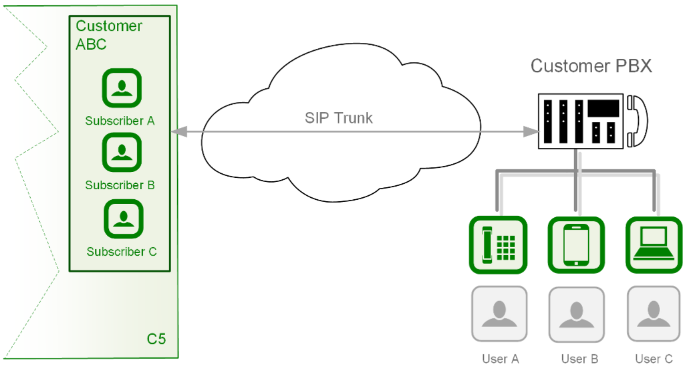
5.5.2 Business customers with the Cloud PBX service

In this case you create a Customer and all the required entities under it to reflect the company's structure: subscribers, extensions, hunt groups, auto-attendant menus, etc.



5.5.3 SIP Trunking

If a customer PBX can register itself with C5, you create a regular subscriber for it and configure a standard username/password authentication. Multiple PBX users can then send and receive calls.



Legacy PBX devices that are not capable of passing the *challenge*-based authentication can be authenticated by the IP address. Optionally, every user of such a PBX can be authenticated separately by the FROM header and the IP address. For more details, refer to the Trusted Sources section.

5.5.4 Mobile subscribers

The pre-paid model works perfectly for mobile application users. In this case you generally create a single subscriber under a customer.

5.5.5 Pre-paid subscribers who use your calling cards

In this case you will most likely create a single subscriber under a customer, although multiple subscribers would work as well. In the latter case, they will share and top-up the common balance. Notice that the *customer* entity itself does not contain any technical configuration for the VoIP service authentication and instead contains other entities called *subscribers*, which do.

5.6 Subscribers

Every subscriber represents a SIP line or a SIP trunk. For example, in the residential services a subscriber entity is dedicated to every user. In the SIP trunking scenario, a subscriber can be used to authenticate all VoIP traffic from the remote PBX device.

In the following table logical subscriber types and their purpose are described.

Service	Subscriber Type	Purpose	Features
Residential	Regular subscriber	A regular VoIP service	Requires a DID number to receive calls from outside of your network
Enterprise (CloudPBX)	Pilot subscriber	A base number for the enterprise customer; Lists all extra numbers (aliases)	Configures the rest of customer subscribers in its self-care web interface

Service	Subscriber Type	Purpose	Features
	Extension	Extra numbers (DIDs, "implicit" extensions) for the enterprise customer	Can be dialed in different ways; The number configuration builds on top of the Pilot subscriber
	PBX Group	Forwards incoming calls to multiple extensions	Ringing policy defines in which order the extensions will ring
SIP Trunk	Digest authentication	Dynamically registers a remote IP PBX device	Handles multiple users behind the IP PBX device
	IP authentication	IP authentication of legacy IP PBX devices incapable of registering with the platform	Might require Trusted Subscriber and Trusted Source configuration
Prepaid	Regular subscriber with prepaid billing profile	Authorization of services based on customer balance; Disconnection of calls on "zero balance"	Voucher and cache top-up; Billing Profile Packages

Tip

Subscriber **Aliases** can provide Extra DIDs or extension numbers to a subscriber.

5.7 SIP Peerings

A SIP peering is your interconnection with the external VoIP or PSTN network. Usually, a VoIP service provider has at least a few termination partners to offer its subscribers calls to virtually any landline and mobile destination.

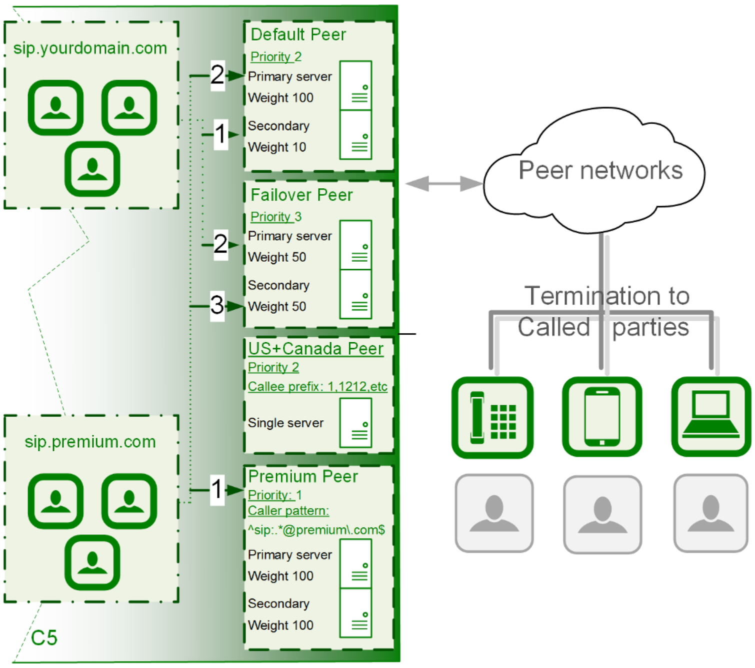
SIP peerings also enable incoming calls to your platform. For example, if you rent a pool of DID numbers from a SIP peer and offer them to your residential and business customers.

An interconnection with your termination partners and DID number providers can include multiple servers and enable both outbound and inbound calls, hence such a configuration is called a *SIP peering group*. You configure at least one SIP peering group for every partner and the main principle here is that all servers in a group terminate calls to the same set of listed destinations.

Any SIP peering group is associated with a *contract* for reconciliation and billing purposes and includes two main technical configurations:

- Peering Servers Represent connections to/from your SIP peering's network. The parameters include an IP address and/or a hostname of the remote part. For outbound calls, this is the destination address where to send calls to and for inbound calls it is an IP authorization of the remote server.
- Outbound/Inbound Peering Rules Outbound rules define through which SIP peering group a call from a specific subscriber will be sent for termination to a specific destination.

The example below shows four SIP peering groups with different priorities, callee prefixes (actual destinations offered by this SIP peering) and callee / called patterns (fine-tuning which callee request URIs and caller URIs are allowed through this SIP peering group).



The figure shows how calls from premium subscribers can in the first place be routed through a dedicated SIP peering group unavailable to regular subscribers.

See the Routing Order Selection section for details about call routing.

Inbound rules allow filtering out incoming INVITE requests arriving from the corresponding SIP peering servers.

6 VoIP Service Configuration Scenario

A basic VoIP service configuration is fast, easy and straight-forward. Provided that your network and required DNS records have been preconfigured, the configuration of a VoIP service can be done purely via the administrative web interface. The configuration mainly includes the following steps:

- Reseller creation (optional)
- SIP domain configuration
- Customer creation
- Subscribers provisioning

Let us assume you are using the *1.2.3.4* IP address with an associated *sip.yourdomain.com* domain to provision VoIP services. This allows you to provide an easy-to-remember domain name instead of the IP address as the proxy server. Also, your subscribers' URIs will look like *1234567@sip.yourdomain.com*.

Tip

Using an IP address instead of an associated FQDN (domain name) for a SIP domain is not suggested as it could add extra administrative work if you decide to relocate your servers to another datacenter or just change IP addresses.

Go to the *Administrative Web Panel* (*Admin Panel*) running on *https://<ip>:1443/login/admin* and follow the steps below. The default web panel user and password are *administrator*, if you have not already changed it in Changing Administrator Password Section 4.3.

6.1 Creating a SIP Domain

A SIP domain is a connection point for your subscribers. The SIP domain also contains specific default configuration for all its subscribers.

Tip

Thoroughly plan your domain names policy in advance and take into account that: 1) the name of a SIP domain cannot be changed after creating it in the administrative web panel; 2) subscribers cannot be moved from one domain to another and must be recreated.

To create a SIP domain, follow these steps:

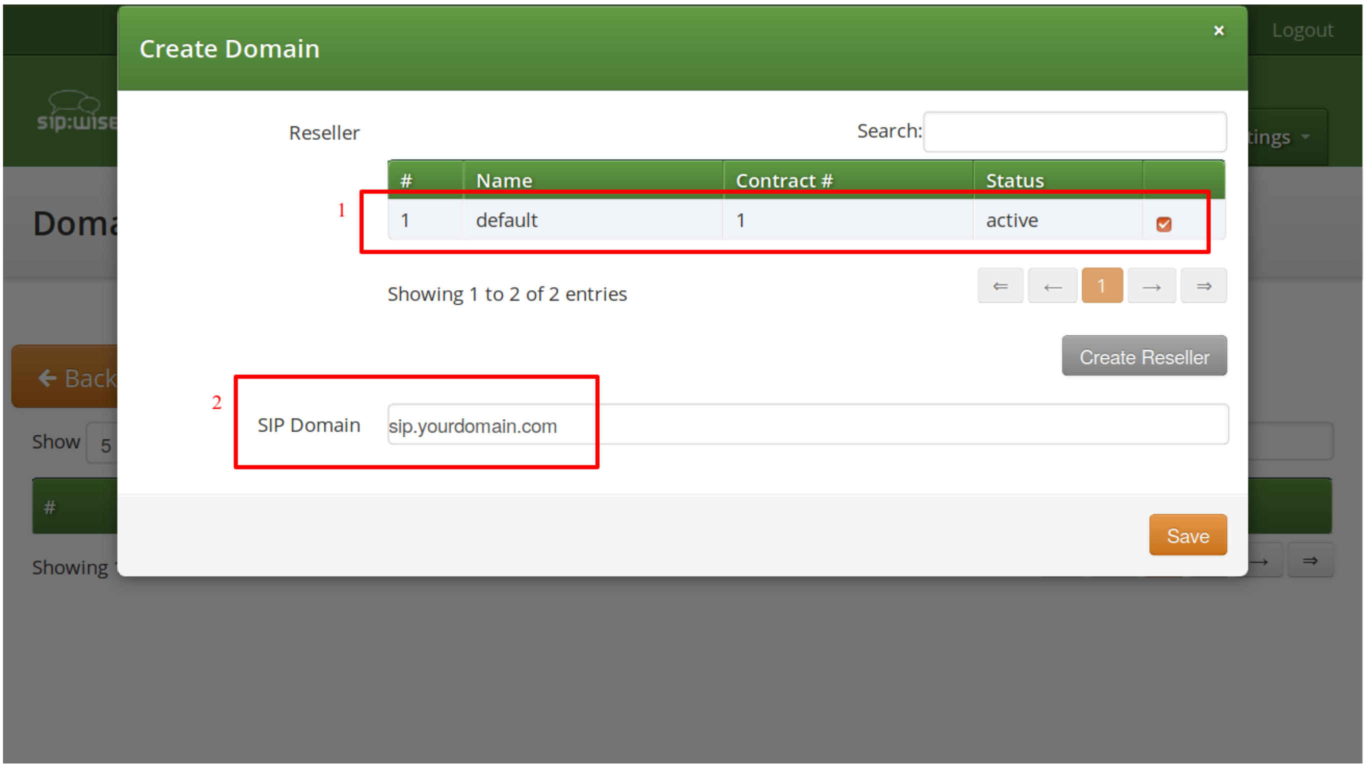
1. Firstly, configure an FQDN on your DNS server for it.

   The domain name must point to the physical IP address you are going to use for providing the VoIP service. A good approach is to create an SRV record:

```
SIP via UDP on port 5060
SIP via TCP on port 5060
SIP via TCP/TLS on port 5061
```

2. Create a new SIP domain in the administrative web panel.

   Go to the *Domains* page and create a new SIP Domain using the FQDN created above.



   Select a *Reseller* who will own the subscribers in this SIP domain. Use the *default* virtual reseller if you provide services directly. Enter your SIP domain name and press *Save*.

3. Adjust the new SIP domain's preferences if necessary.

You can create multiple SIP domains reusing the existing IP address or adding a new one. Extra SIP domains are required e.g., if you would like to host a virtual operator on your platform, create separate domains for providing services in different countries or just offer a new service.

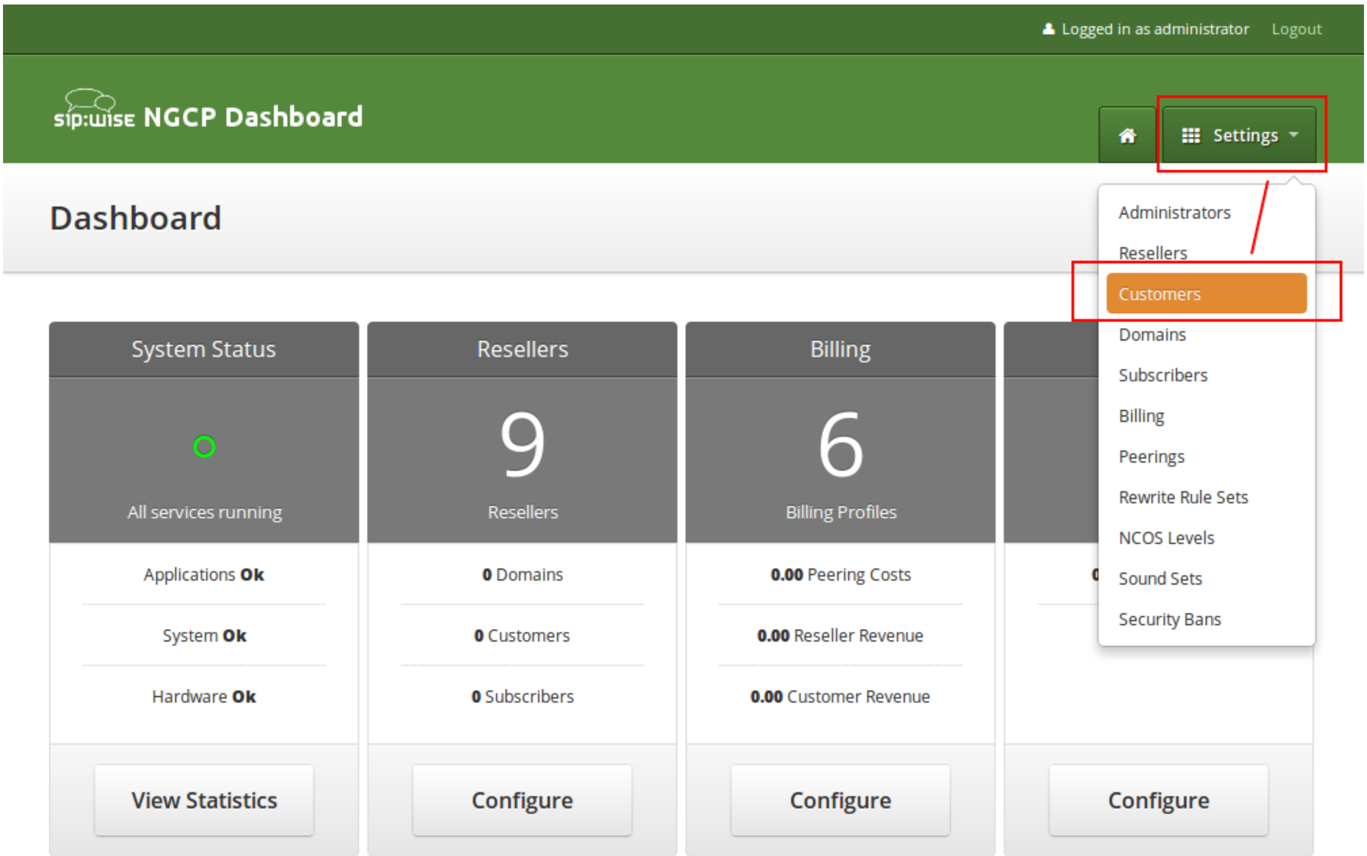
6.2 Creating a Customer

A Customer is a special type of contract acting as legal and billing information container for SIP subscribers. A customer can have one or more SIP subscriber entities that represent SIP lines.

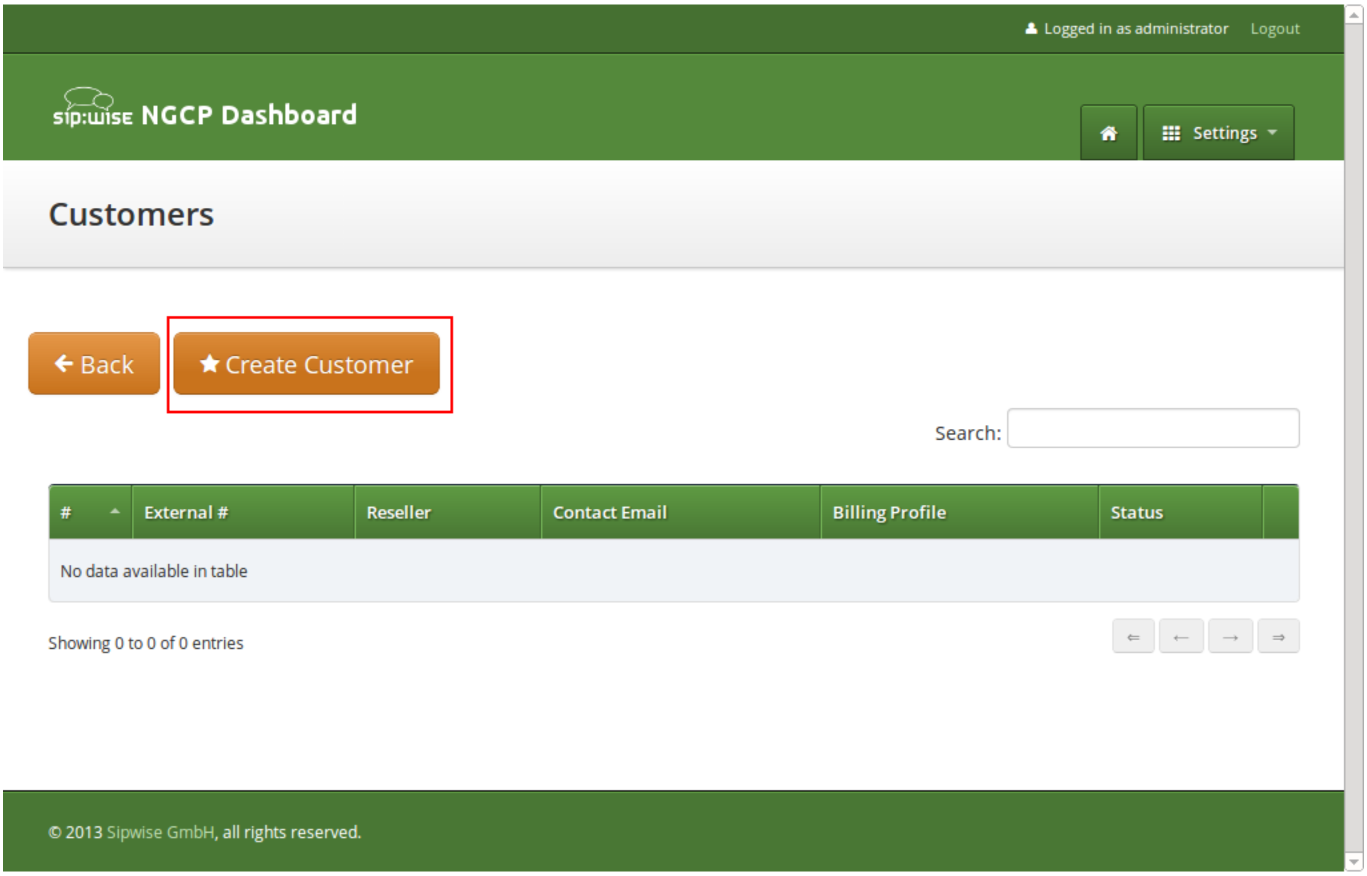
Tip

For correct billing, notification and invoicing, create a customer with a single SIP subscriber for the residential service (as it normally has only one telephone line) and a customer with multiple SIP subscribers to provide a service to a company with many telephone lines.

To create a Customer, go to *Settings*→*Customers*.



Click on *Create Customer*.

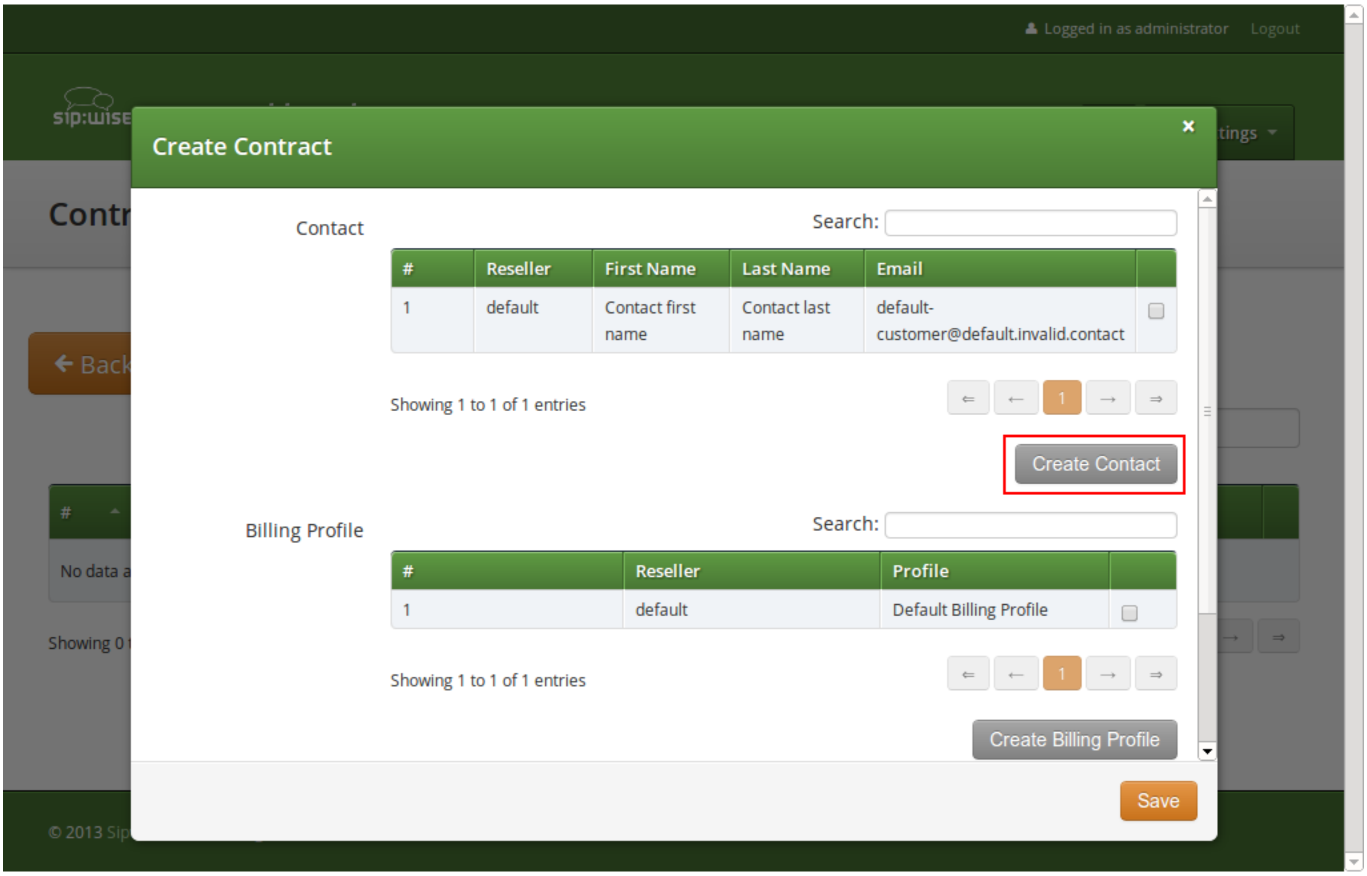


Each *Customer* has a *Contact* — a container for the personal and legal information that identifies a private or corporate customer.

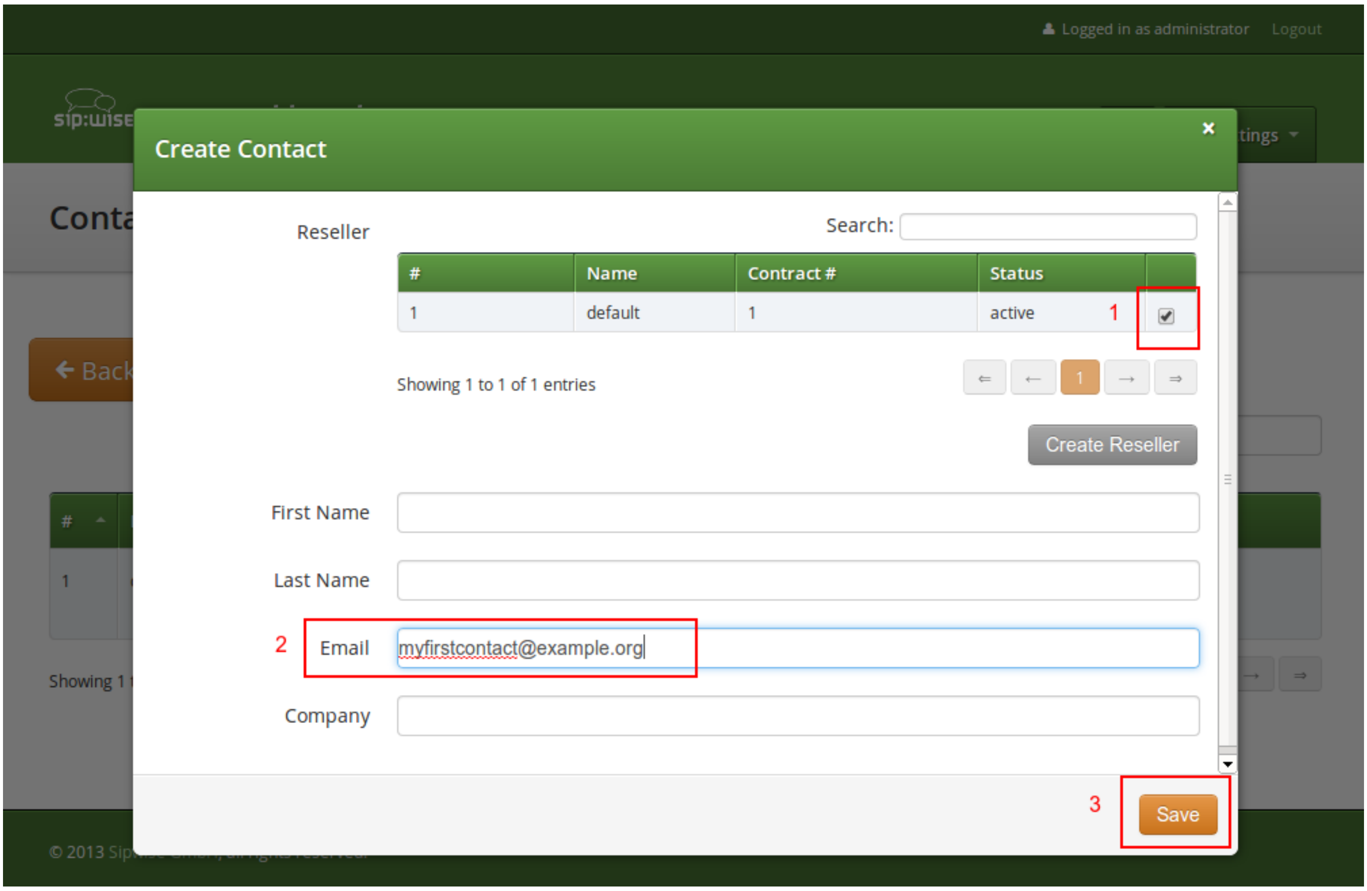
Tip

Create a dedicated *Contact* for every *Customer* as it contains specific data e.g., name, address and IBAN that identifies this customer.

Click on *Create Contact* to create a new *Contact*.

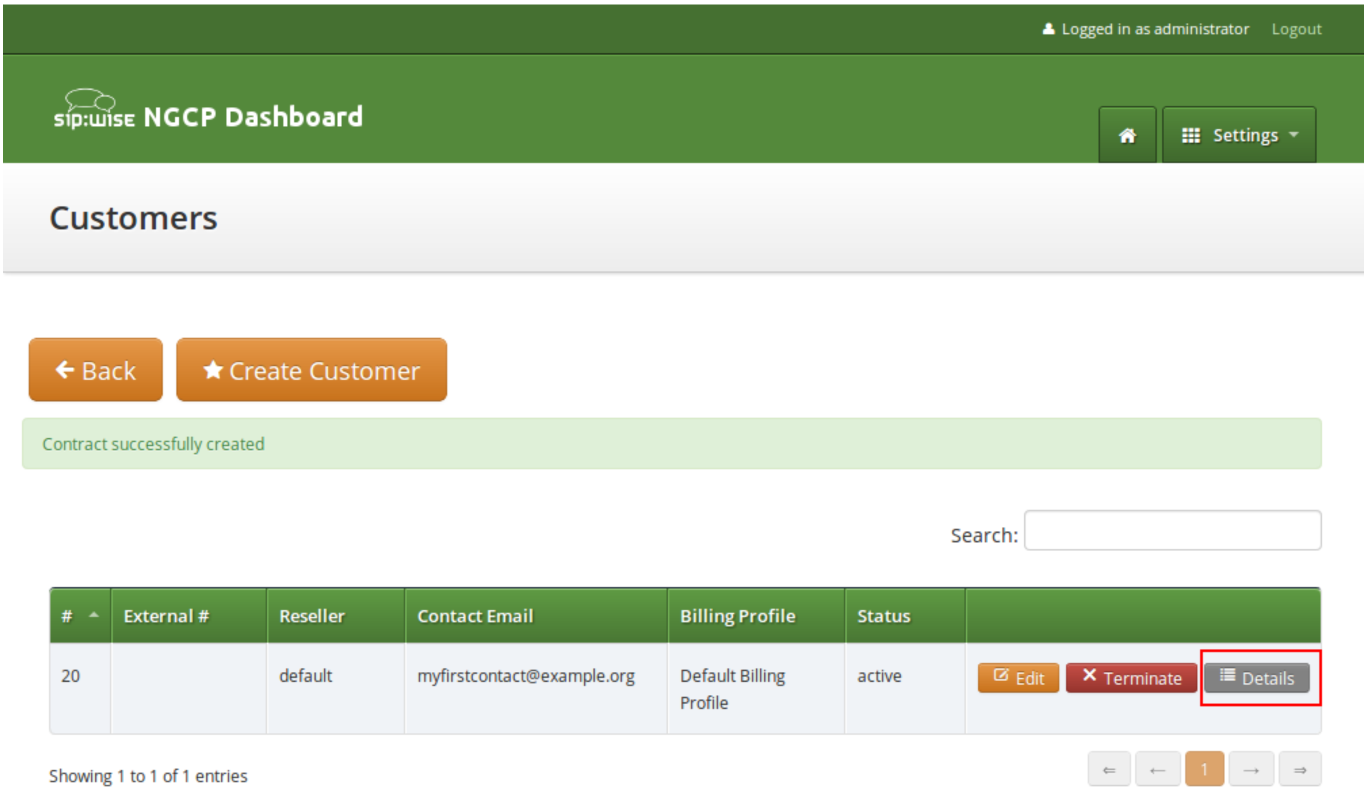


Select the required *Reseller* and enter the contact details (at least an *Email* is required), then press *Save*.



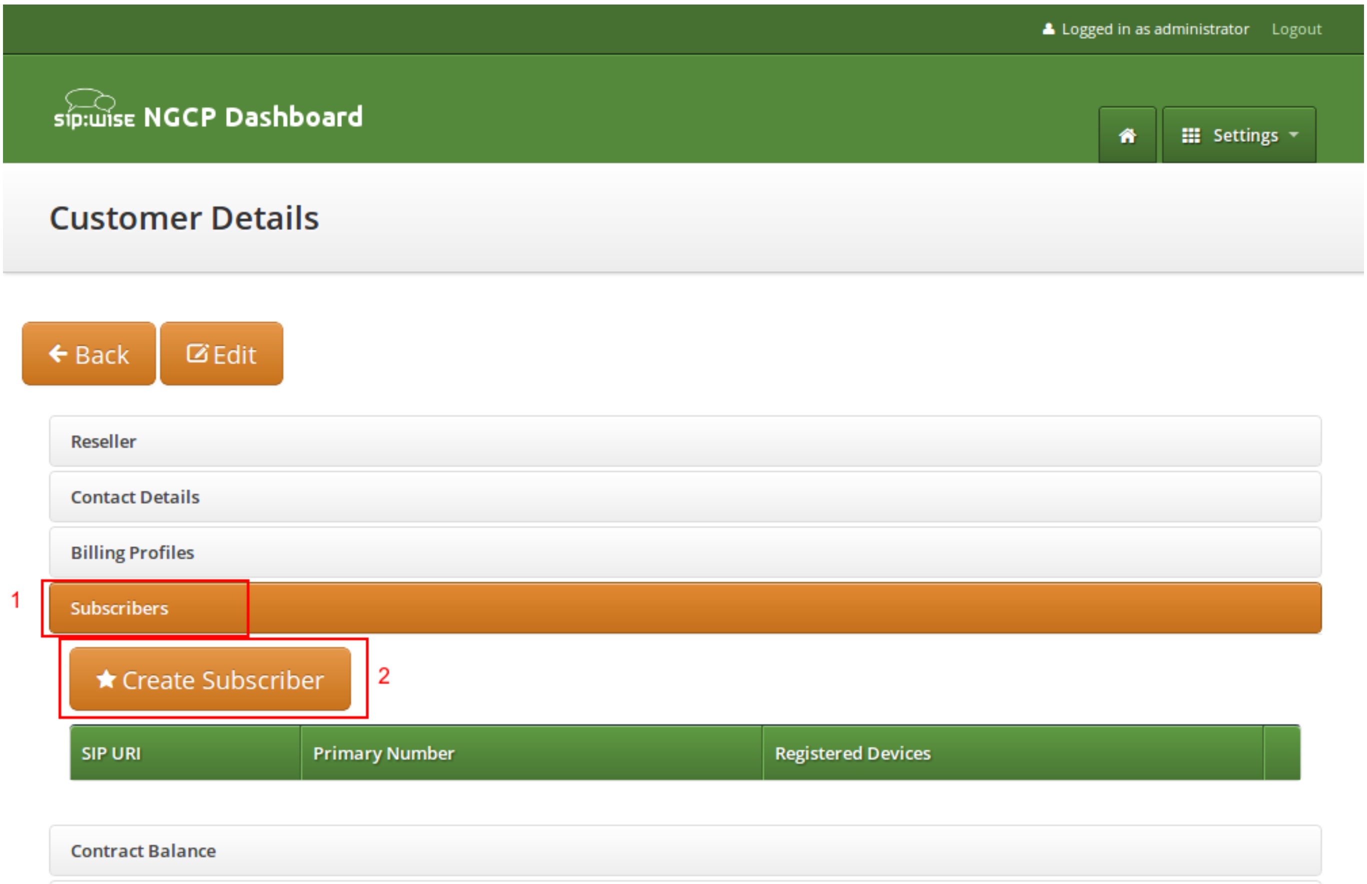
You will be redirected back to the *Customer* form. The newly created *Contact* is selected by default now, so only select a *Billing Profile* and press *Save*.

You will now see your first *Customer* in the list. Hover over the customer and click *Details* to make extra configuration if necessary.



6.3 Creating a Subscriber

In your *Customer* details view, click on the *Subscribers* row, then click *Create Subscriber*.



Select a *SIP Domain* created earlier and specify required and optional parameters:

- **Domain**: The domain part of the SIP URI for your subscriber.
- **E164 Number**: This is the telephone number mapped to the subscriber, separated into *Country Code (CC)*, *Area Code (AC)* and *Subscriber Number (SN)*. For the first tests, you can set an imaginary number here and change it later when you get number blocks assigned by your PSTN interconnect partner. So in our example, we'll use *43* as CC, *99* as AC and *1001* as SN to form the imaginary number *+43 99 1001*.

Tip

This number can actually be used to place calls between local subscribers, even if you don't have any PSTN interconnection. This comes in handy if you use phones instead of soft-clients for your tests. The format in which this number can be dialled, so the subscriber is reached is defined in Section 6.7.

Important

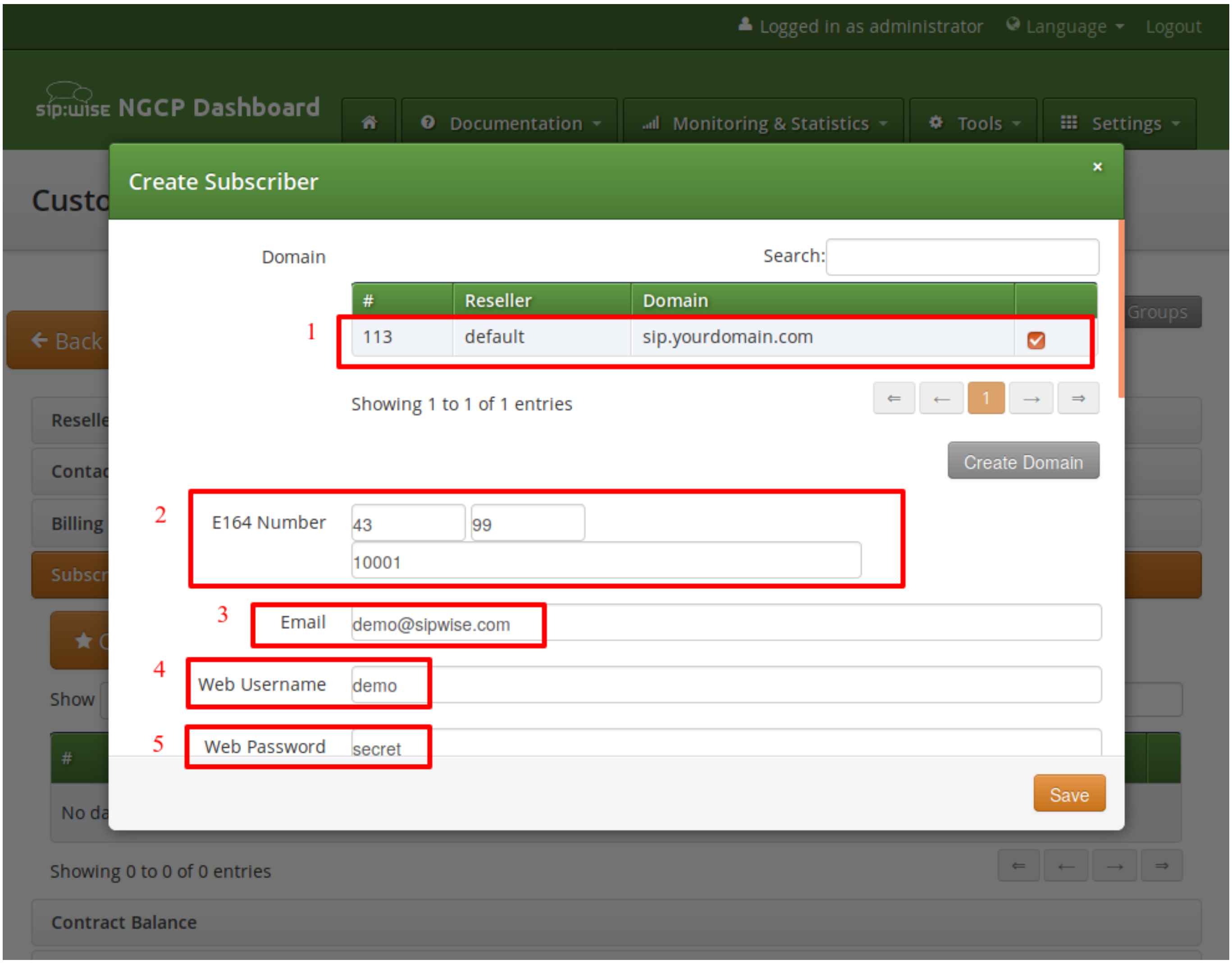
NGCP allows a single subscriber to have multiple E.164 numbers to be used as aliases for receiving incoming calls. Also, NGCP supports so-called "implicit" extensions. If a subscriber has phone number 012345, but somebody calls 012345100, then NGCP first tries to send the call to number 012345100 (even though the user is registered as 012345). If NGCP then receives the 404 - Not Found response, it falls back to 012345 (the user-part with which the callee is registered).

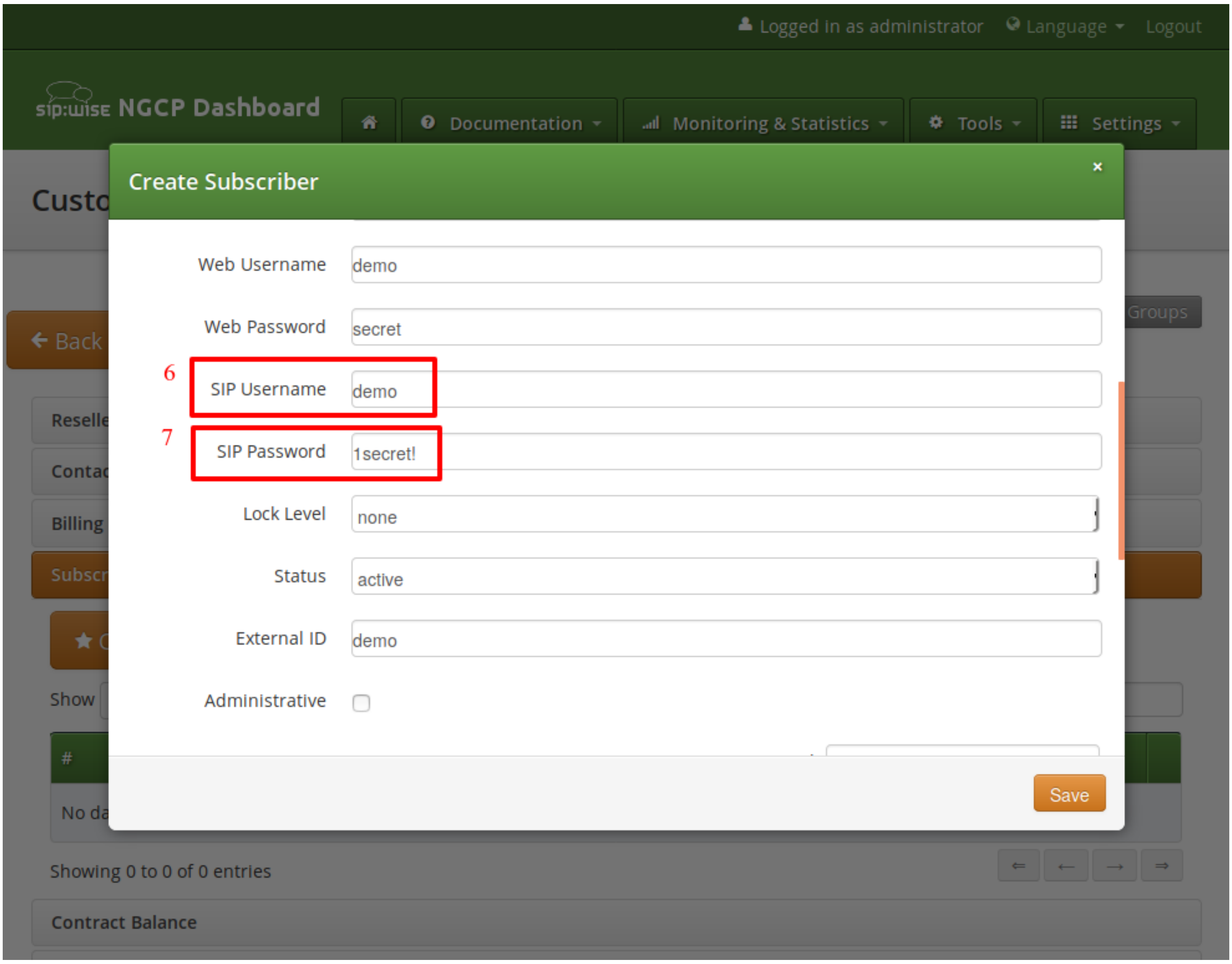
- **Email**: An email address for sending service-related notifications to.
- **Web Username**: This is the user part of the username the subscriber may use to log into her *Customer Self Care Interface*. The user part will be automatically suffixed by the SIP domain you choose for the **SIP URI**. Usually, the web username is identical to the **SIP URI**, but you may choose a different naming schema.

Caution

The web username needs to be unique. The system will return a fault if you try to use the same web username twice.

- **Web Password**: This is the password for the subscriber to log into her *Customer Self Care Interface*. It must be at least 6 characters long.
- **SIP Username**: The user part of the SIP URI for your subscriber.
- **SIP Password**: The password of your subscriber to authenticate on the SIP proxy. It must be at least 6 characters long.
- **Status**: You can lock a subscriber here, but for creating one, you will most certainly want to use the *active* status.
- **External ID**: You can provision an arbitrary string here (e.g. an ID of a 3rd party provisioning/billing system).
- **Administrative**: If you have multiple subscribers in one account and set this option for one of them, this subscriber can administrate other subscribers via the *Customer Self Care Interface*.


Logged in as administrator
Language
Logout
sip:wise NGCP Dashboard
Documentation
Monitoring & Statistics
Tools
Settings
Create Subscriber
Domain
Search:
Reseller Domain
1
113 default sip.yourdomain.com
Showing 1 to 1 of 1 entries
Create Domain
2
E164 Number
43
99
10001
3
Email
demo@sipwise.com
4
Web Username
demo
5
Web Password
secret
Save
Showing 0 to 0 of 0 entries
Contract Balance



Repeat the creation of *Customers* and *Subscribers* for all your test accounts. You should have at least 3 subscribers to test the functionality of the NGCP.

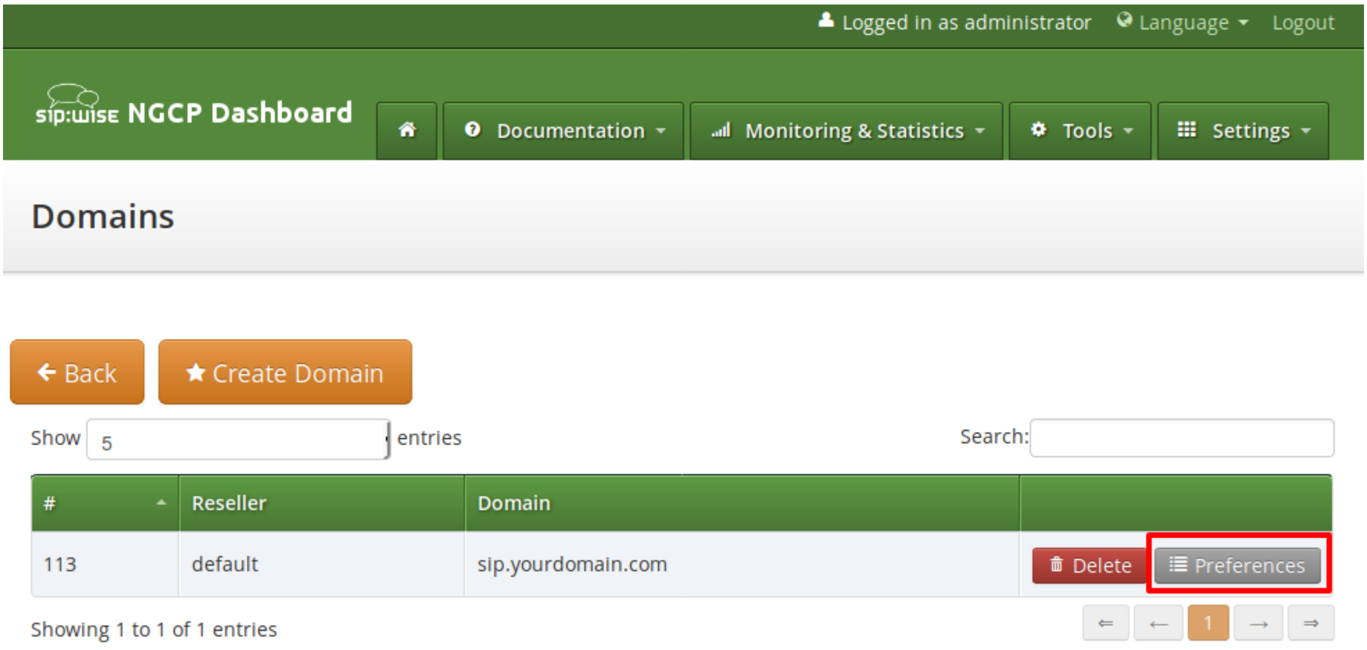
Tip

At this point, you're able to register your subscribers to the NGCP and place calls between these subscribers.

You should now revise the *Domain* and *Subscriber* Preferences.

6.4 Domain Preferences

The *Domain Preferences* are the default settings for *Subscriber Preferences*, so you should set proper values there if you don't want to configure each subscriber separately. You can later override these settings in the *Subscriber Preferences* if particular subscribers need special settings. To configure your *Domain Preferences*, go to *Settings*→*Domains* and click on the *Preferences* button of the domain you want to configure.

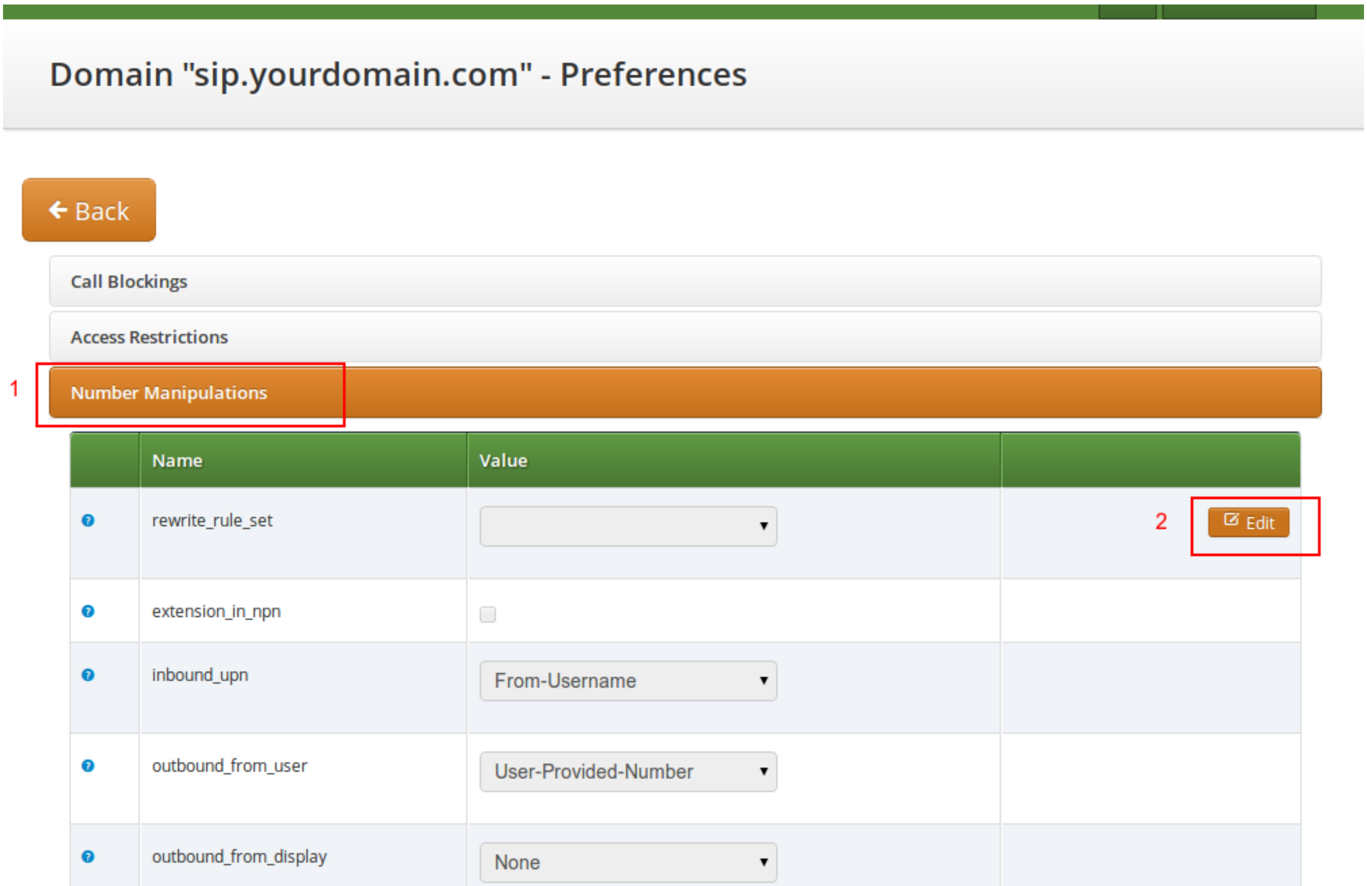


The most important settings are in the *Number Manipulations* group.

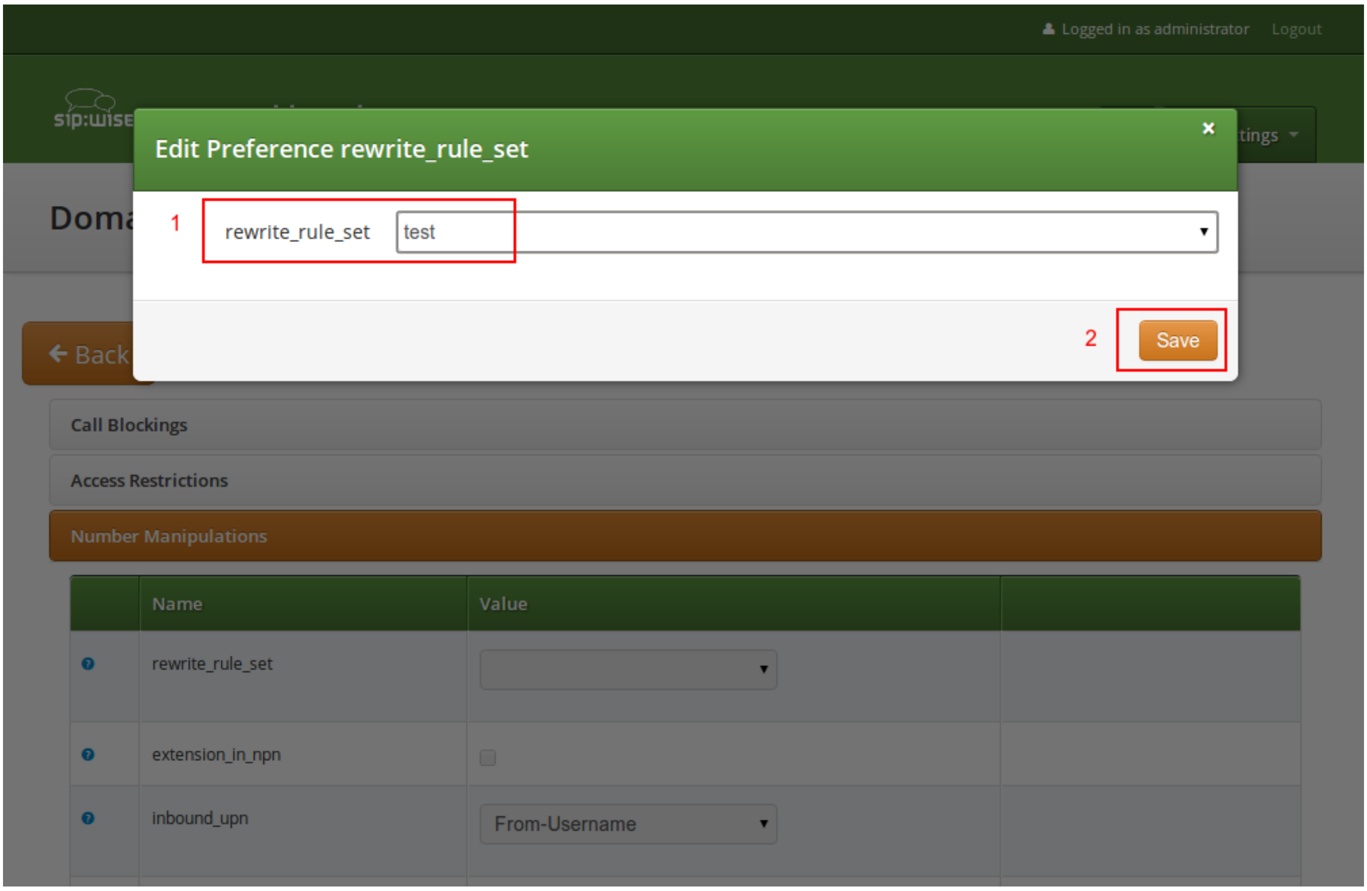
Here you can configure the following:

- for incoming calls - which SIP message headers to take numbers from
- for outgoing calls - where in the SIP messages to put certain numbers to
- for both - how these numbers are normalized to E164 format and vice versa

To assign a *Rewrite Rule Set* to a *Domain*, create a set first as described in Section 6.7, then assign it to the domain by editing the *rewrite_rule_set* preference.



Select the *Rewrite Rule Set* and press *Save*.



Then, select the field you want the *User Provided Number* to be taken from for inbound INVITE messages. Usually the *From-Username* should be fine, but you can also take it from the *Display-Name* of the From-Header, and other options are available as well.

6.5 Subscriber Preferences

You can override the *Domain Preferences* on a subscriber basis as well. Also, there are *Subscriber Preferences* which don't have a default value in the *Domain Preferences*.

To configure your *Subscriber*, go to *Settings*→*Subscribers* and click *Details* on the row of your subscriber. There, click on the *Preferences* button on top.

You want to look into the *Number Manipulations* and *Access Restrictions* options in particular, which control what is used as user-provided and network-provided calling numbers.

- For outgoing calls, you may define multiple numbers or patterns to control what a subscriber is allowed to send as user-provided calling numbers using the *allowed_clis* preference.
- If *allowed_clis* does not match the number sent by the subscriber, then the number configured in *cli* (the network-provided number) preference will be used as user-provided calling number instead.
- You can override any user-provided number coming from the subscriber using the *user_cli* preference.

Note

Subscribers preference *allowed_clis* will be synchronized with subscribers primary number and aliases if *ossbss→provisioning→auto_allow_cli* is set to **1** in */etc/ngcp-config/config.yml*.

Note

Subscribers preference *cli* will be synchronized with subscribers primary number if *ossbss→provisioning→auto_sync_cli* is set to **yes** in */etc/ngcp-config/config.yml*.

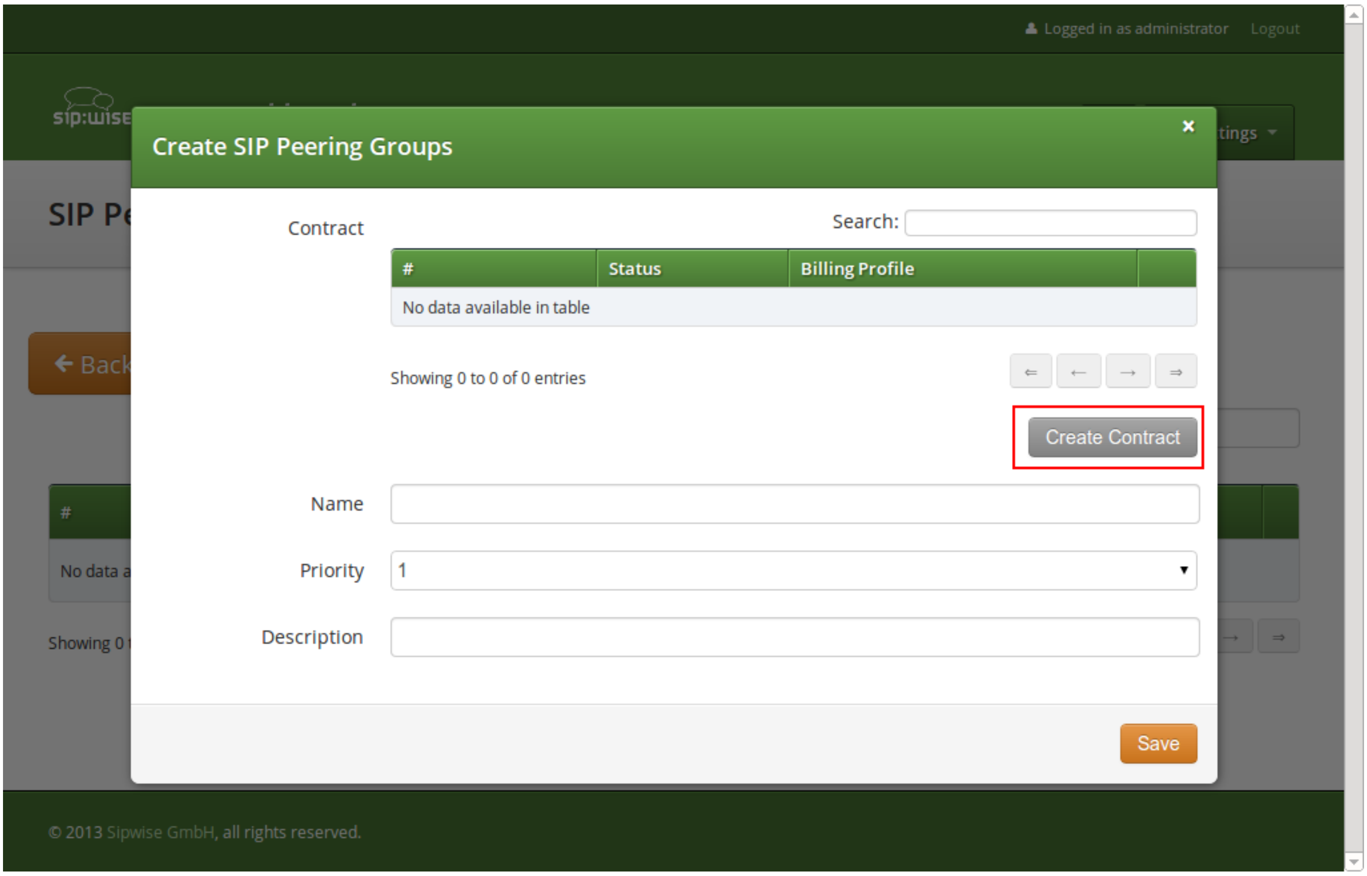
6.6 Creating Peerings

If you want to terminate calls at or allow calls from 3rd party systems (e.g. PSTN gateways, SIP trunks), you need to create SIP peerings for that. To do so, go to *Settings→Peerings*. There you can add peering groups, and for each peering group add peering servers and rules controlling which calls are routed over these groups. Every peering group needs a peering contract for correct interconnection billing.

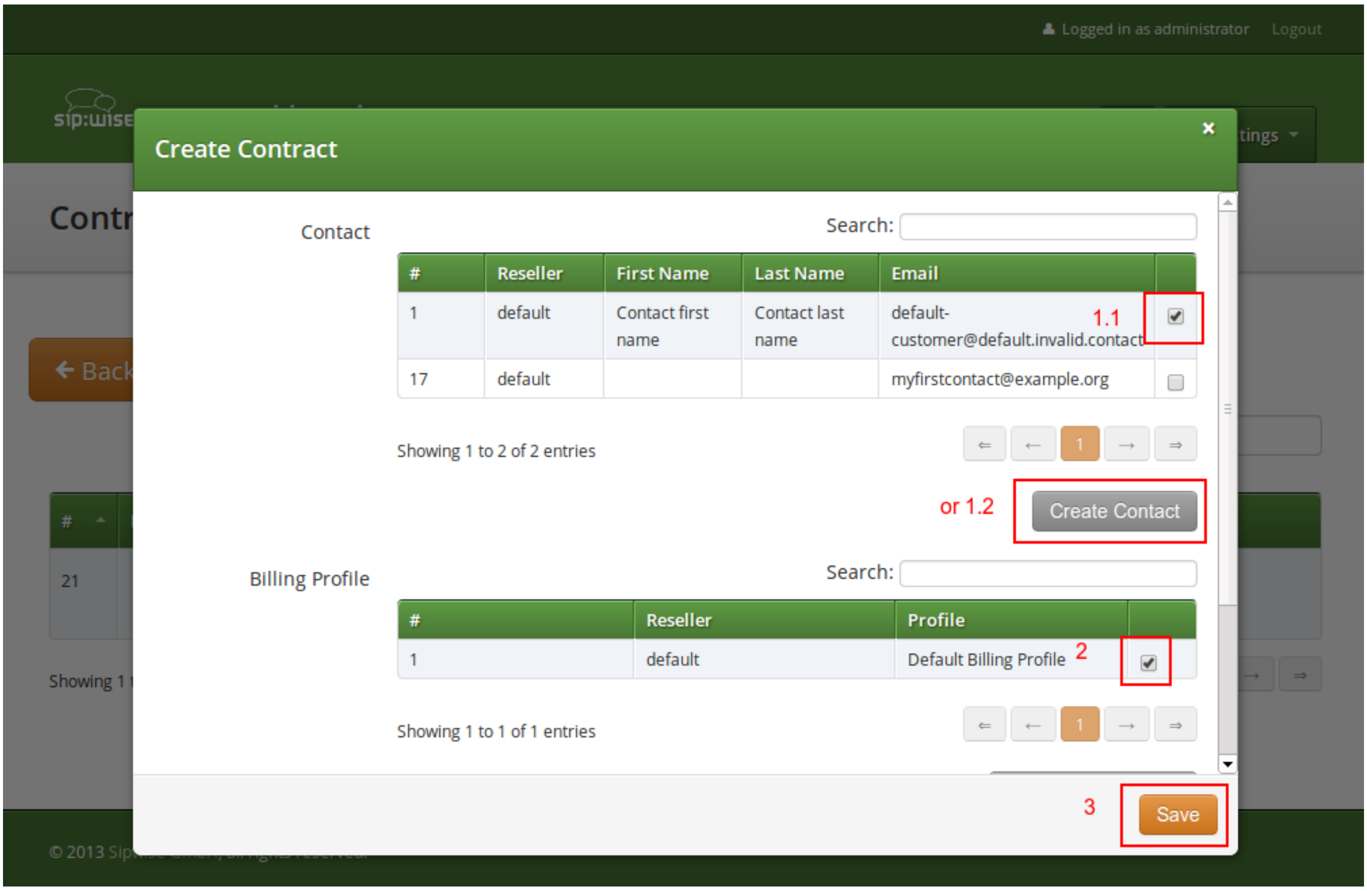
6.6.1 Creating Peering Groups

Click on *Create Peering Group* to create a new group.

In order to create a group, you must select a peering contract. You will most likely want to create one contract per peering group.

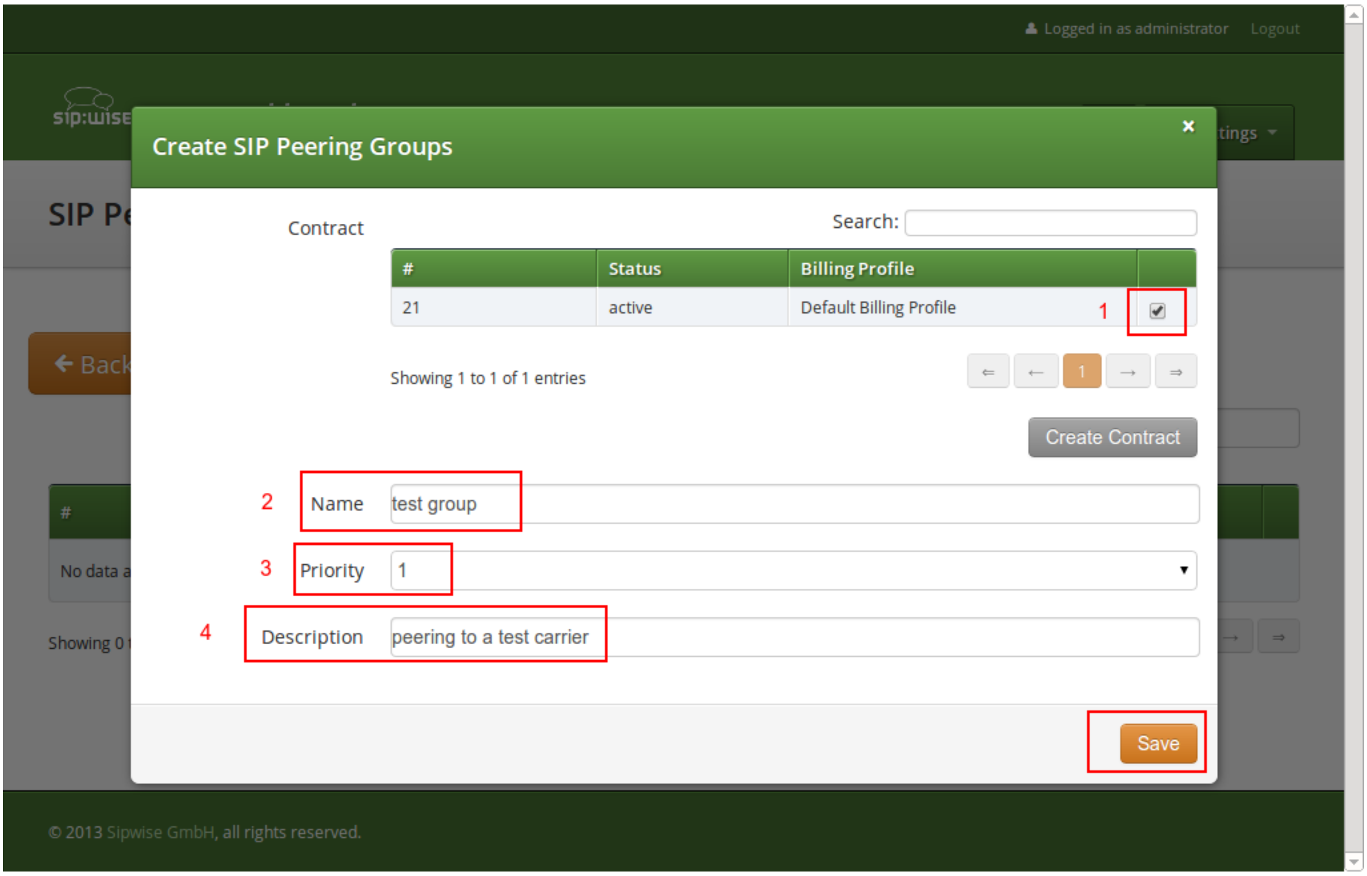


Click on *Create Contract* create a *Contact*, then select a *Billing Profile*.



Click *Save* on the *Contacts* form, and you will get redirected back to the form for creating the actual *Peering Group*. Put a name, priority and description there, for example:

- **Peering Contract:** select the id of the contract created before
- **Name:** `test group`
- **Priority:** `1`
- **Description:** `peering to a test carrier`



The *Priority* option defines which *Peering Group* to favor (Priority **1** gives the highest precedence) if two peering groups have peering rules matching an outbound call. *Peering Rules* are described below.

Then click *Save* to create the group.

6.6.2 Creating Peering Servers

In the group created before, you need to add peering servers to route calls to and receive calls from. To do so, click on *Details* on the row of your new group in your peering group list.

To add your first *Peering Server*, click on the *Create Peering Server* button.

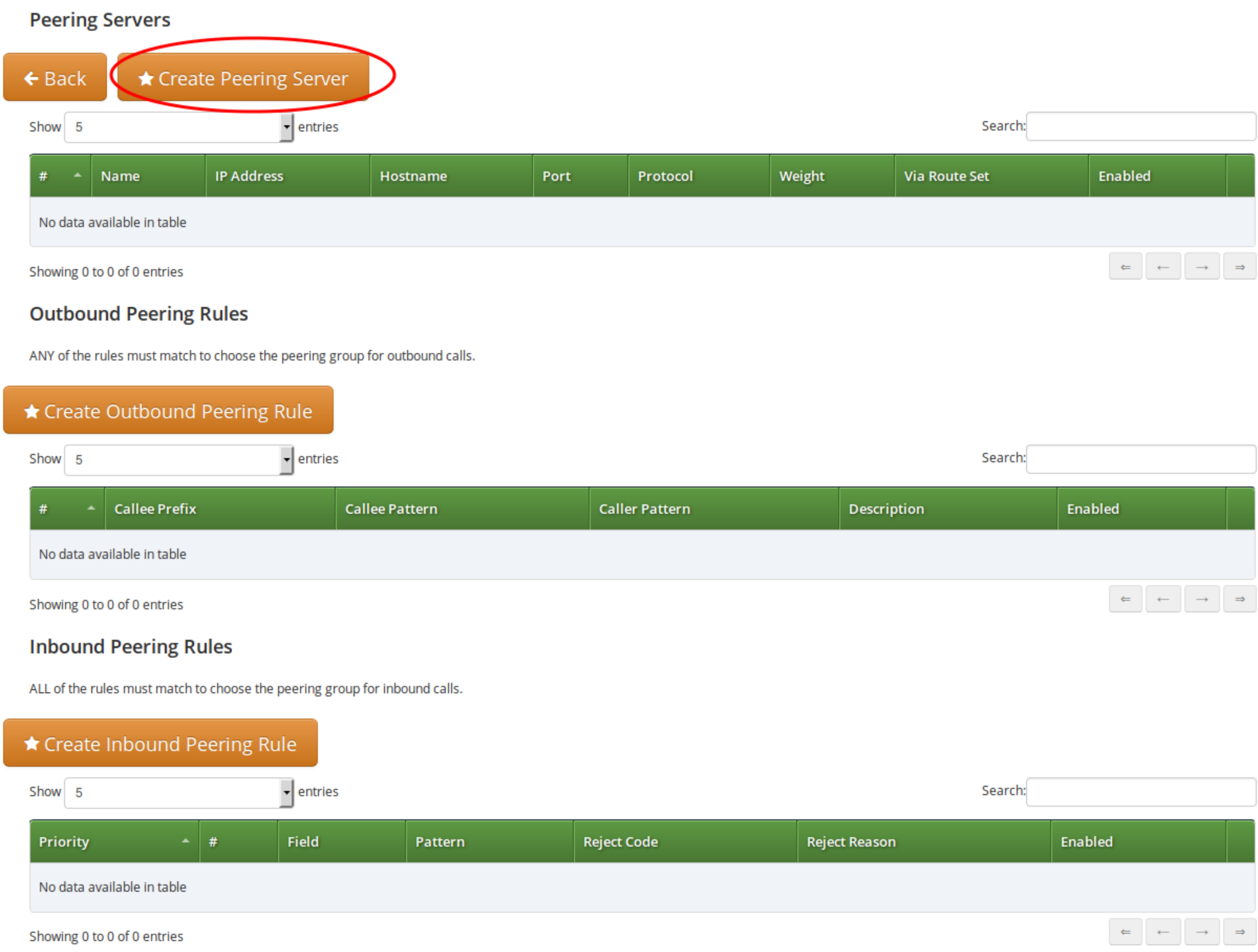


Figure 10: Create Peering Server

In this example, we will create a peering server with IP *2.3.4.5* and port *5060*:

- **Name**: `test-gw-1`
- **IP Address:** `2.3.4.5`
- **Hostname:** leave empty
- **Port:** `5060`
- **Protocol:** `UDP`
- **Weight:** `1`
- **Via Route:** `None`

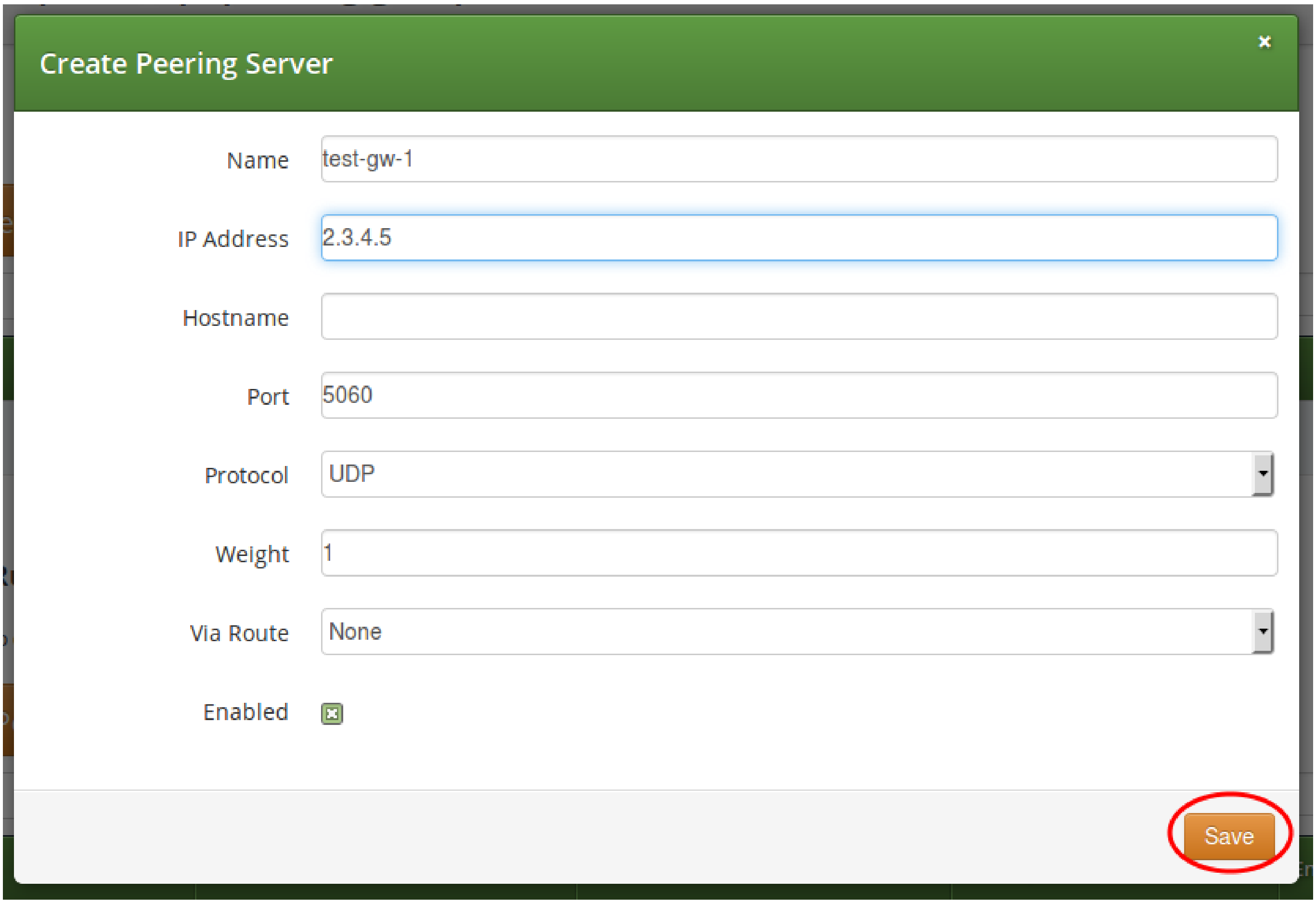


Figure 11: Peering Server Properties

Click *Save* to create the peering server.

Tip

The *hostname* field for a peering server is optional. Usually, the IP address of the peer is used as the **domain** part of the Request URI. Fill in this field if a peer requires a particular hostname instead of the IP address. The IP address must always be given though as the request will always be sent to the specified IP address, no matter what you put into the *hostname* field.

Tip

If you want to add a peering server with an IPv6 address, enter the address without surrounding square brackets into the *IP Address* column, e.g. *::1*.

You can force an additional hop (e.g. via an external SBC) towards the peering server by using the *Via Route* option. The available options you can select there are defined in `/etc/ngcp-config/config.yml`, where you can add an array of SIP URIs in `kamailio→lb→external_sbc` like this:

```
kamailio:
  lb:
```

```
external_sbc:
  - sip:192.168.0.1:5060
  - sip:192.168.0.2:5060
```

Execute `ngcpcfg apply` *`added external sbc gateways`*, then edit your peering server and select the hop from the *Via Route* selection.

Once a peering server has been created, this server can already send calls to the system.

6.6.2.1 Outbound Peering Rules

> **Important**
>
> To be able to send outbound calls towards the servers in the *Peering Group*, you also need to define *Outbound Peering Rules*. They specify which source and destination numbers are going to be terminated over this group. To create a rule, click the *Create Outbound Peering Rule* button.

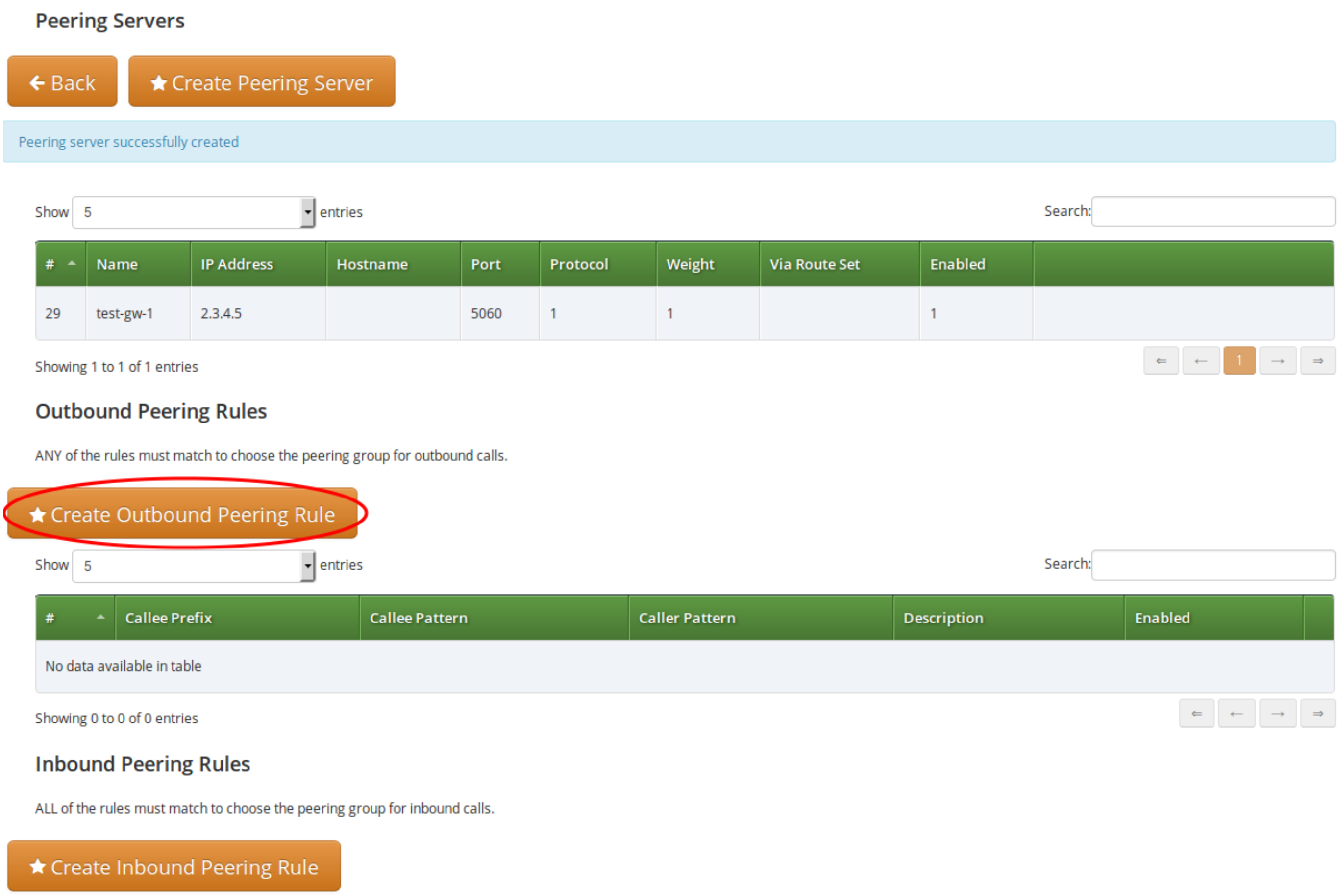


Figure 12: Create Outbound Peering Rule

Since the previously created peering group will be the only one in our example, we have to add a default rule to route *all* calls via this group. To do so, create a new peering rule with the following values:

- **Callee Prefix:** leave empty
- **Callee Pattern:** leave empty
- **Caller Pattern:** leave empty
- **Description:** `Default Rule`

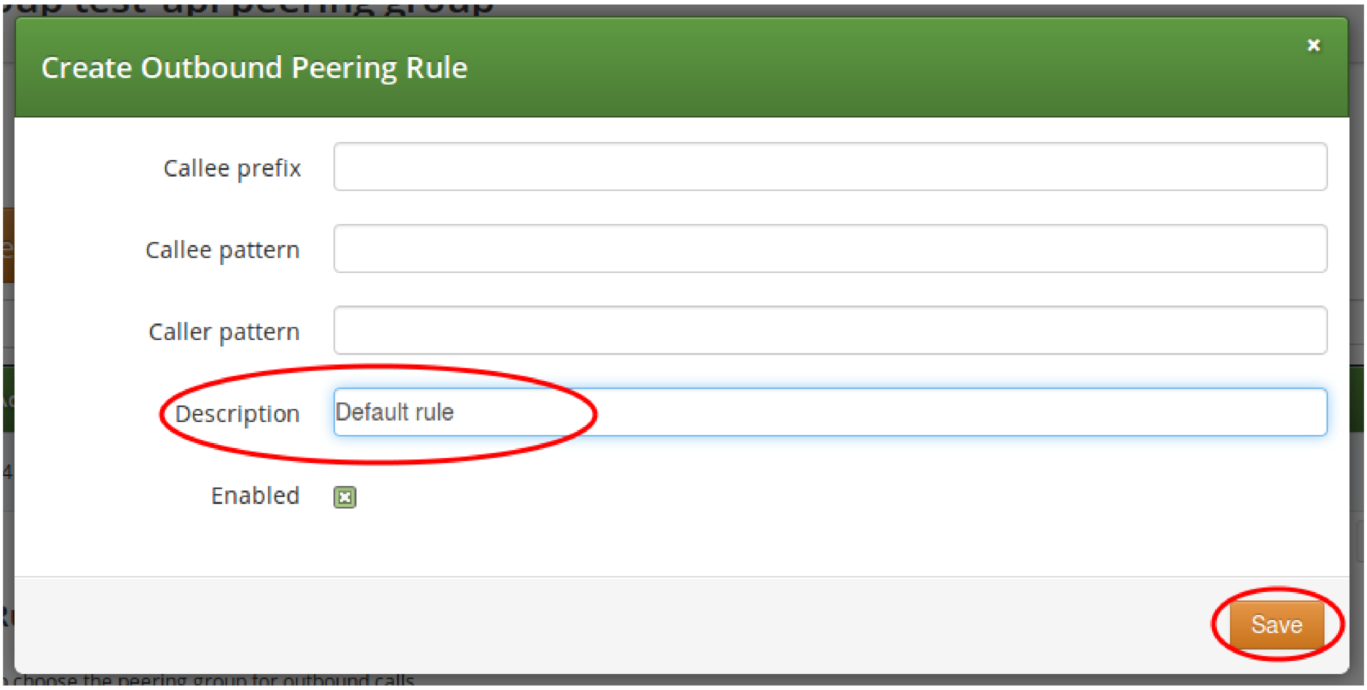


Figure 13: Outbound Peering Rule Properties

Then click *Save* to add the rule to your group.

Tip

In contrast to the callee/caller pattern, the callee prefix has a regular alphanumeric string and can not contain any regular expression.

Tip

If you set the caller or callee rules to refine what is routed via this peer, enter all phone numbers in full E.164 format, that is *<cc><ac><sn>*.

Tip

The *Caller Pattern* field covers the whole URI including the subscriber domain, so you can only allow certain domains over this peer by putting for example `@example\.com` into this field.

6.6.2.2 Inbound Peering Rules

Starting from *mr5.0* release, Sipwise NGCP supports filtering SIP INVITE requests sent by SIP peers. The system administrator may define one or more matching rules for SIP URIs that are present in the headers of SIP INVITE requests, and select which SIP header (or part of the header) must match the pattern declared in the rule.

If the incoming SIP INVITE message has the proper headers, NGCP will accept and further process the request. If the message does not match the rule it will be rejected.

Caution

An incoming SIP INVITE message must match **all the inbound peering rules** so that NGCP does not reject the request.

In order to **create an inbound peering rule** you have to select a peering group, press *Details* and then press *Create Inbound Peering Rule* button.

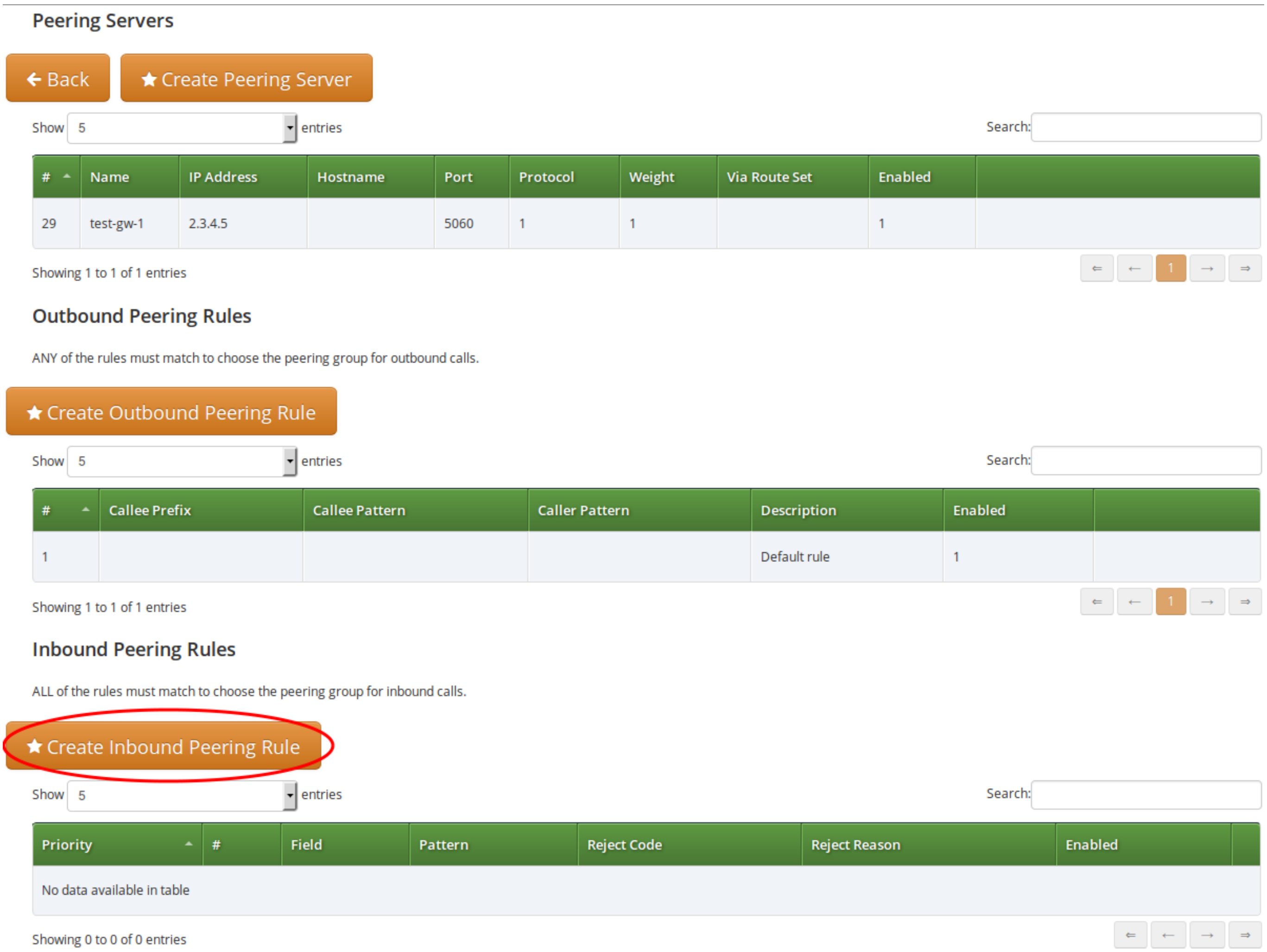


Figure 14: Create Inbound Peering Rule

An inbound peering rule has the following **properties**:

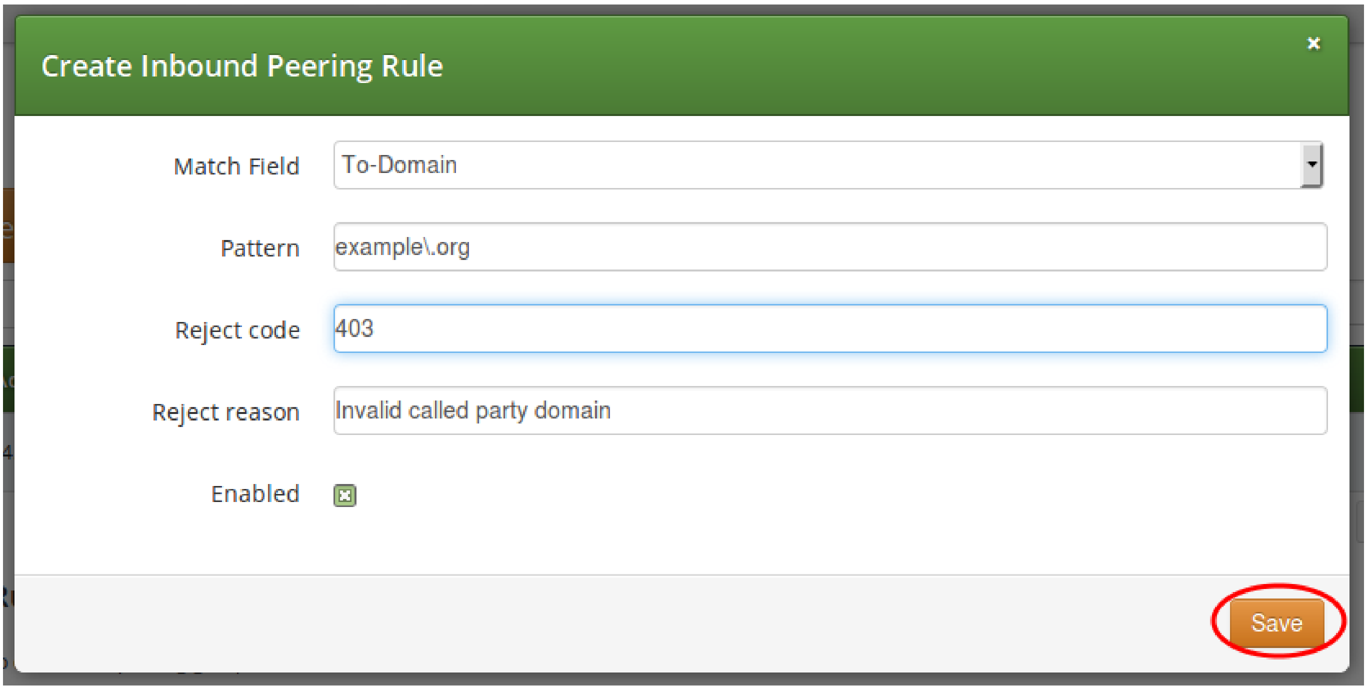


Figure 15: Inbound Peering Rule Properties

- `Match Field`: select which header and which part of that header in a SIP INVITE message will be checked for matching the pattern
- `Pattern`: a POSIX regular expression that defines the accepted value of a header; example: `^sip:.+@example\.org$` — this will match a SIP URI that contains "example.org" in the domain part
- `Reject code`: optional; a SIP status code that will be sent as a response to an INVITE request that does not match the pattern; example: `403`
- `Reject reason`: optional; an arbitrary text that will be included in the SIP response sent with the *reject code*
- `Enabled`: a flag to enable / disable the particular inbound peering rule

Note
Both of the properties `Reject code` and `Reject reason` must be left empty if a peering server (i.e. a specific IP address) is part of more peering groups. Such a configuration is useful when an incoming SIP INVITE request needs to be treated differently in the affected peering groups, based on its content, and that's why if the INVITE message only partly matches an inbound peering rule it should not simply be rejected.

When all settings for a peering group are done the details of the group look like:

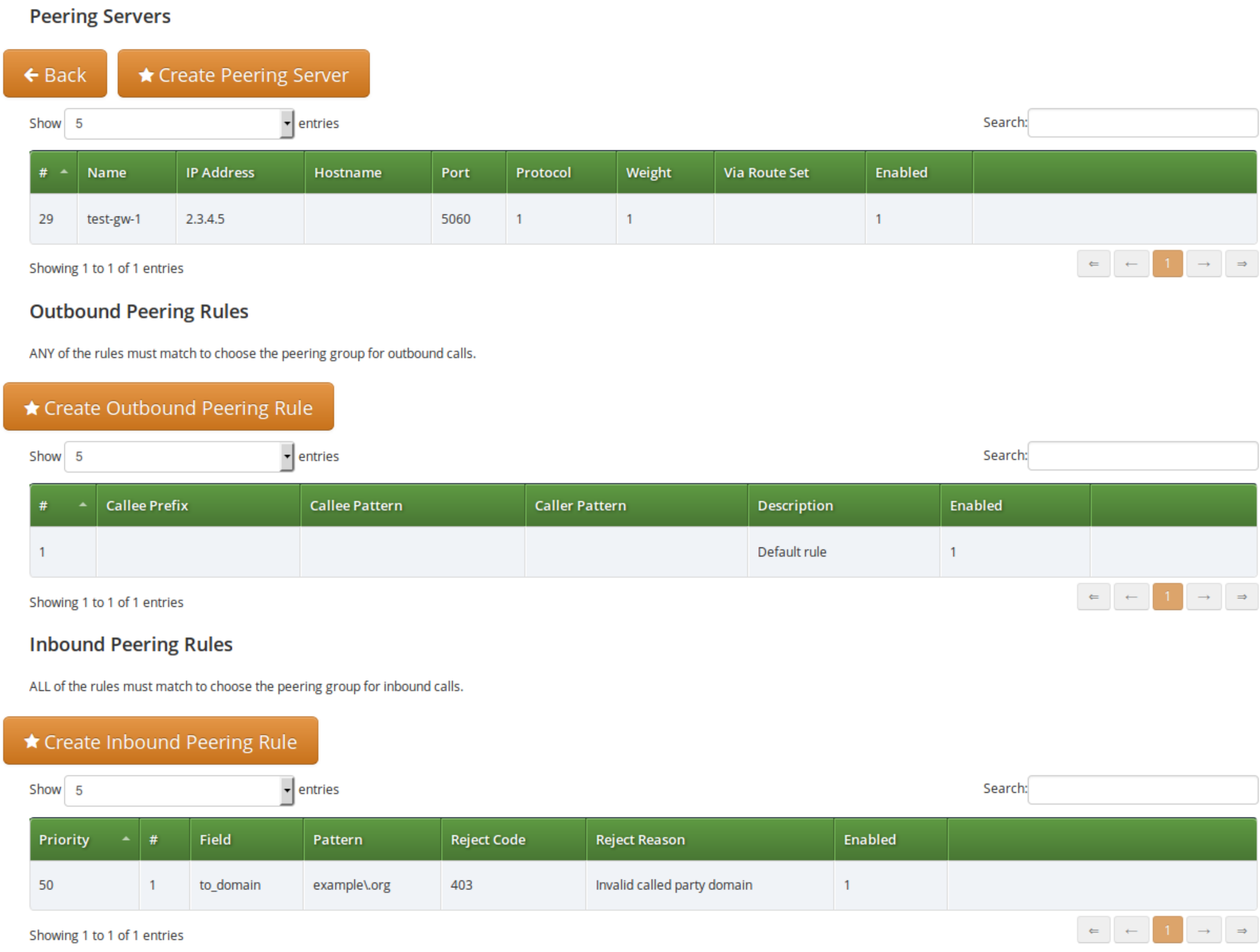


Figure 16: Peering Servers Overview

6.6.2.3 Routing Order Selection

The selection of peering groups and peering servers for outgoing calls is done in the following way:

1. All peering groups that meet the following criteria configured in the outbound peering rule are added to the list of routes for a particular call:
   - Callee's username matches *callee prefix*
   - Callee's URI matches *callee pattern*
   - Caller's URI matches *caller pattern*
2. When all matching peering groups are selected, they are ordered by *callee prefix* according to the **longest match basis** (sometimes referred to as the **longest pattern match** or **maximum pattern length match**). One or more peering group with longest *callee prefix* match will be given first positions on the list of routes.

3. Peering groups with the same *callee prefix* length are further ordered by ***Priority***. Peering group(s) with the higher priorities will occupy higher positions.

> **Important**
> Priority **1** gives the *highest* precedence to the corresponding peering group. Hence, a lower priority value will put the peering group higher in the list of routes (compared to other peering groups with the same *callee prefix* length).

Priority can be selected from **1** (highest) to **9** (lowest).

4. All peering servers in the peering group with the highest priority (e.g. priority **1**) are tried one-by-one starting from the highest server weight. Peering groups with lower priorities or with shorter *callee prefix* will be used only for fail-over.

   The ***weight*** of the peering servers in the selected peering group will influence the order in which the servers within the group will be tried for routing the outbound call. The weight of a server can be set in the range from **1** to **127**.

> **Important**
> Opposite to the peering group priority, a peering server with a higher weight value has a *higher* precedence, but the server weight rather sets a probability than a strict order. E.g. although a peering server with weight **127** has the highest chance to be the first in the list of routes, another server with a lower weight (e.g. **100**) sometimes will be selected first.

In order to find out this probability knowing the weights of peering servers, use the following script:

```
#!/usr/bin/php
<?php

   // This script can be used to find out actual probabilities
   // that correspond to a list of peering weights.

if ($argc < 2) {
  echo "Usage: lcr_weight_test.php <list of weights (integers 1-254)>\n";
  exit;
 }

$iters = 10000;

$rands = array();
for ($i = 1; $i <= $iters; $i++) {
  $elem = array();
  for ($j = 1; $j < $argc; $j++) {
    $elem["$j"] = $argv[$j] * (rand() >> 8);
  }
  $rands[] = $elem;
 }

$sorted = array();
```

```
foreach ($rands as $rand) {
  asort($rand);
  $sorted[] = $rand;
 }

$counts = array();
for ($j = 1; $j < $argc; $j++) {
  $counts["$j"] = 0;
 }

foreach ($sorted as $rand) {
  end($rand);
  $counts[key($rand)]++;
 }

for ($j = 1; $j < $argc; $j++) {
  echo "Peer with weight " . $argv[$j] . " has probability " . $counts["$j"]/$iters . "\n";
 }
?>
```

Let us say you have 2 peering servers, one with weight 1 and another with weight 2. At the end — running the script as below — you will have the following traffic distribution:

```
# lcr_weight_test.php 1 2

 Peer with weight 1 has probability 0.2522
 Peer with weight 2 has probability 0.7478
```

If a peering server replies with SIP codes `408`, `500` or `503`, or if a peering server doesn't respond at all, the next peering server in the current peering group is tried as a fallback. All the servers within the group are tried one after another until the call succeeds. If no more servers are left in the current peering group, the next group which matches the outbound peering rules is used.

Note

The Sipwise NGCP may use a slightly different approach in selecting the appropriate peering server if the *peer probing* feature is enabled. See the details in Section 7.10 of the handbook.

6.6.2.4 Least Cost Routing (LCR) Configuration

The default call routing uses statically configured peering group priorities to decide where to send the calls. This solution is useful when you have an external SBC that makes all the routing decisions and is described in the Routing Order Selection section. Sipwise C5 also allows you routing calls to the cheapest SIP peers saving your termination cost.

To enable LCR routing, do the following:

- Upload the billing fees provided by your peers to the corresponding peering billing profiles

- Enable the LCR module in config.yml (kamailio.proxy.perform_peer_lcr: yes)

When the LCR routing is enabled, the selection of peering groups would be the following:

1. All peering groups that meet the following criteria configured in the outbound peering rule are added to the list of routes for a particular call (for pure LCR you might want to omit these filters leaving them blank):
   - Callee's username matches *callee prefix*
   - Callee's URI matches *callee pattern*
   - Caller's URI matches *caller pattern*
2. When all matching peering groups are selected, the longest matching *callee prefix* is selected from each of them. And the peering groups are *temporary* ordered according to the longest matching prefix and priority.
3. Then, the LCR module re-orders the peering groups starting from the lowest termination cost to the highest (ignoring the prefix length and peering group priorities).
4. The platform will first route the call to the servers of the first peering group in this list. If no peering server can terminate the call, the call would fail-over to the second peering group from the list and so on.

Note

The peering servers in every peering group are sorted and tried according to their weight as described in the previous section.

Let us consider a short example. There are two peering groups (PG1 and PG2) that can deliver calls to New York (e.g. 12121234567) and they have the following rates:

Peering Group	Prefix	Cost	Description
PG1	1	0.02	USA & Canada
PG2	1	0.05	USA & Canada
	1212	0.03	New York, USA

PG1 has only one rate that matches the dialed number, so that it will be taken into account, PG2 has two rates and the longest will be selected. The call will be routed to PG1 servers first as it has a cheaper price and can fail-over to PG2 servers.

The Sipwise C5 LCR feature together with the codec filtering, media transcoding, header manipulations, SIP, and RTP encryption and other SBC features make an external SBC unnecessary. This simplifies your VoIP network and cuts deployment and operation costs.

6.6.3 Authenticating and Registering against Peering Servers

6.6.3.1 Proxy-Authentication for outbound calls

If a peering server requires the sip:provider CE to authenticate for outbound calls (by sending a 407 as response to an INVITE), then you have to configure the authentication details in the *Preferences* view of your peer host.

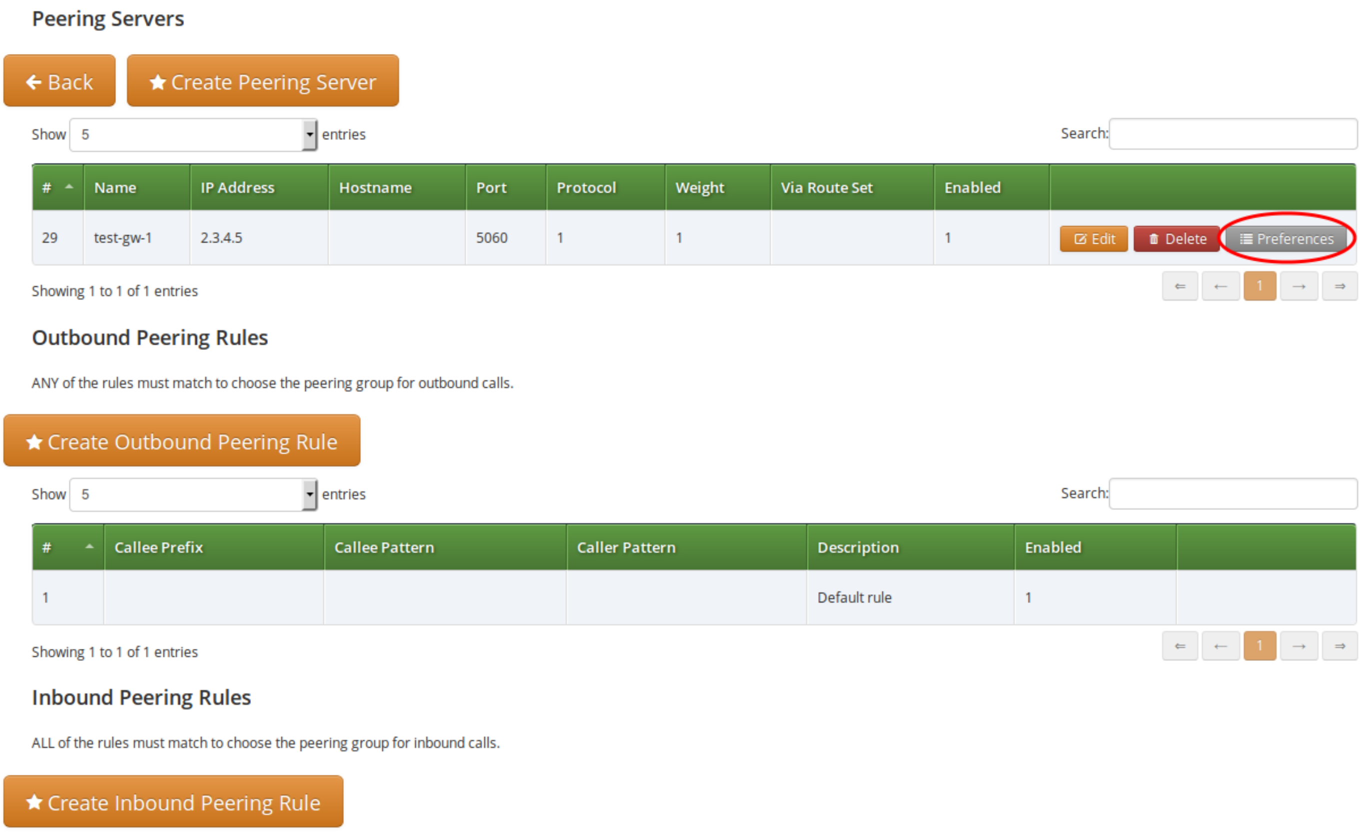
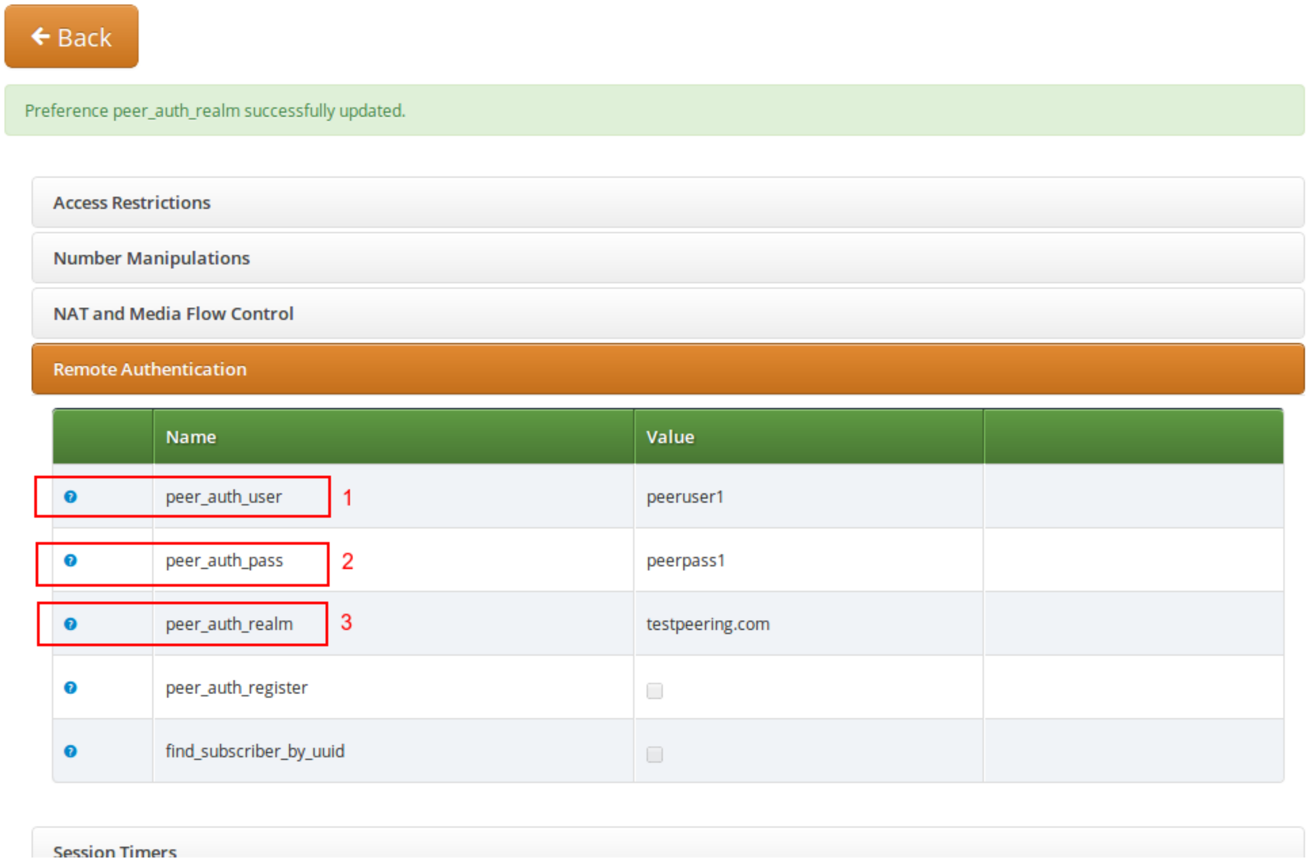


Figure 17: Select Peering Server Preferences

To configure this setting, open the *Remote Authentication* tab and edit the following three preferences:

- **peer_auth_user:** `<username for peer auth>`
- **peer_auth_pass:** `<password for peer auth>`
- **peer_auth_realm:** `<domain for peer auth>`



Important

If you do NOT authenticate against a peer host, then the caller CLI is put into the `From` and `P-Asserted-Identity` headers, e.g. `"+4312345" <sip:+4312345@your-domain.com>`. If you DO authenticate, then the `From` header is `"+4312345" <sip:your_peer_auth_user@your_peer_auth_realm>` (the CLI is in the Display field, the peer_auth_user in the From username and the peer_auth_realm in the From domain), and the `P-Asserted-Identity` header is as usual like `<sip:+4312345@your-domain.com>`. So for presenting the correct CLI in *CLIP no screening* scenarios, your peering provider needs to extract the correct user either from the `From Display-Name` or from the `P-Asserted-Identity URI-User`.

Tip

You will notice that these three preferences are also shown in the *Subscriber Preferences* for each subscriber. There you can override the authentication details for all peer host if needed, e.g. if every user authenticates with his own separate credentials at your peering provider.

Tip

If **peer_auth_realm** is set, the system may overwrite the Request-URI with the peer_auth_realm value of the peer when sending the call to that peer or peer_auth_realm value of the subscriber when sending a call to the subscriber. Since this is rarely a desired behavior, it is disabled by default starting with NGCP release 3.2. If you need the replacement, you should set *set_ruri_to_peer_auth_realm: 'yes'* in */etc/ngcp-config/config.yml*.

6.6.3.2 Registering at a Peering Server

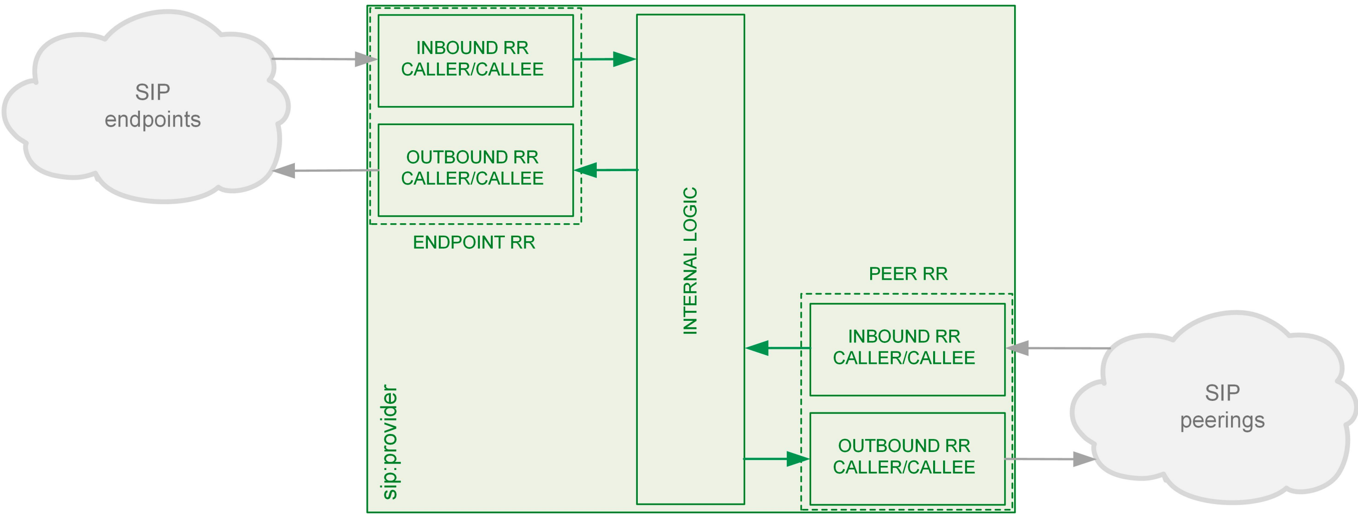
Unfortunately, the credentials configured above are not yet automatically used to register the sip:provider CE at your peer hosts. There is however an easy manual way to do so, until this is addressed.

Configure your peering servers with the corresponding credentials in */etc/ngcp-config/templates/etc/ngcp-sems/etc/reg_agent.conf.tt2*, then execute *ngcpcfg apply 'added upstream credentials'*.

> **Important**
> Be aware that this will force SEMS to restart, which will drop all calls.

6.7 Configuring Rewrite Rule Sets

On the NGCP, every phone number is treated in E.164 format *<country code><area code><subscriber number>*. Rewrite Rule Sets is a flexible tool to translate the caller and callee numbers to the proper format before the routing lookup and after the routing lookup separately. The created Rewrite Rule Sets can be assigned to the domains, subscribers and peers as a preference. Here below you can see how the Rewrite Rules are used by the system:



As from the image above, following the arrows, you will have an idea about which type of Rewrite Rules are applied during a call. In general:

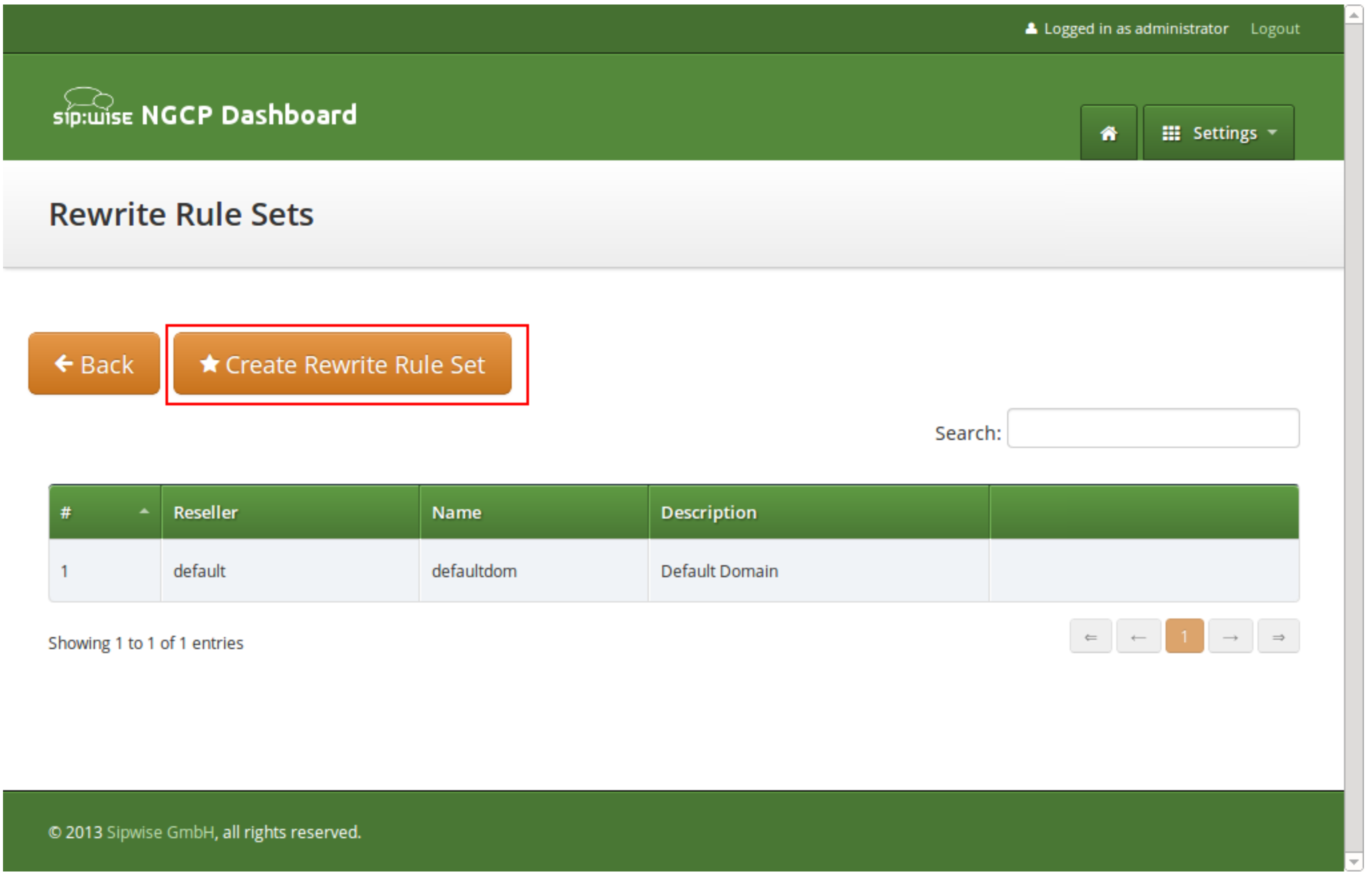
- Call from local subscriber A to local subscriber B: Inbound RR from local Domain/Subscriber A and Outbound Rewrite Rules from local Domain/Subscriber B.
- Call from local subscriber A to the peer: Inbound RR from local Domain/Subscriber A and Outbound Rewrite Rules from the peer.
- Call from peer to local subscriber B: Inbound RR from the Peer and Outbound Rewrite Rules from local Domain/Subscriber B.

You would normally begin with creating a Rewrite Rule Set for your SIP domains. This is used to control what an end user can dial for outbound calls, and what is displayed as the calling party on inbound calls. The subscribers within a domain inherit Rewrite Rule Sets of that domain, unless this is overridden by a subscriber Rewrite Rule Set preference.

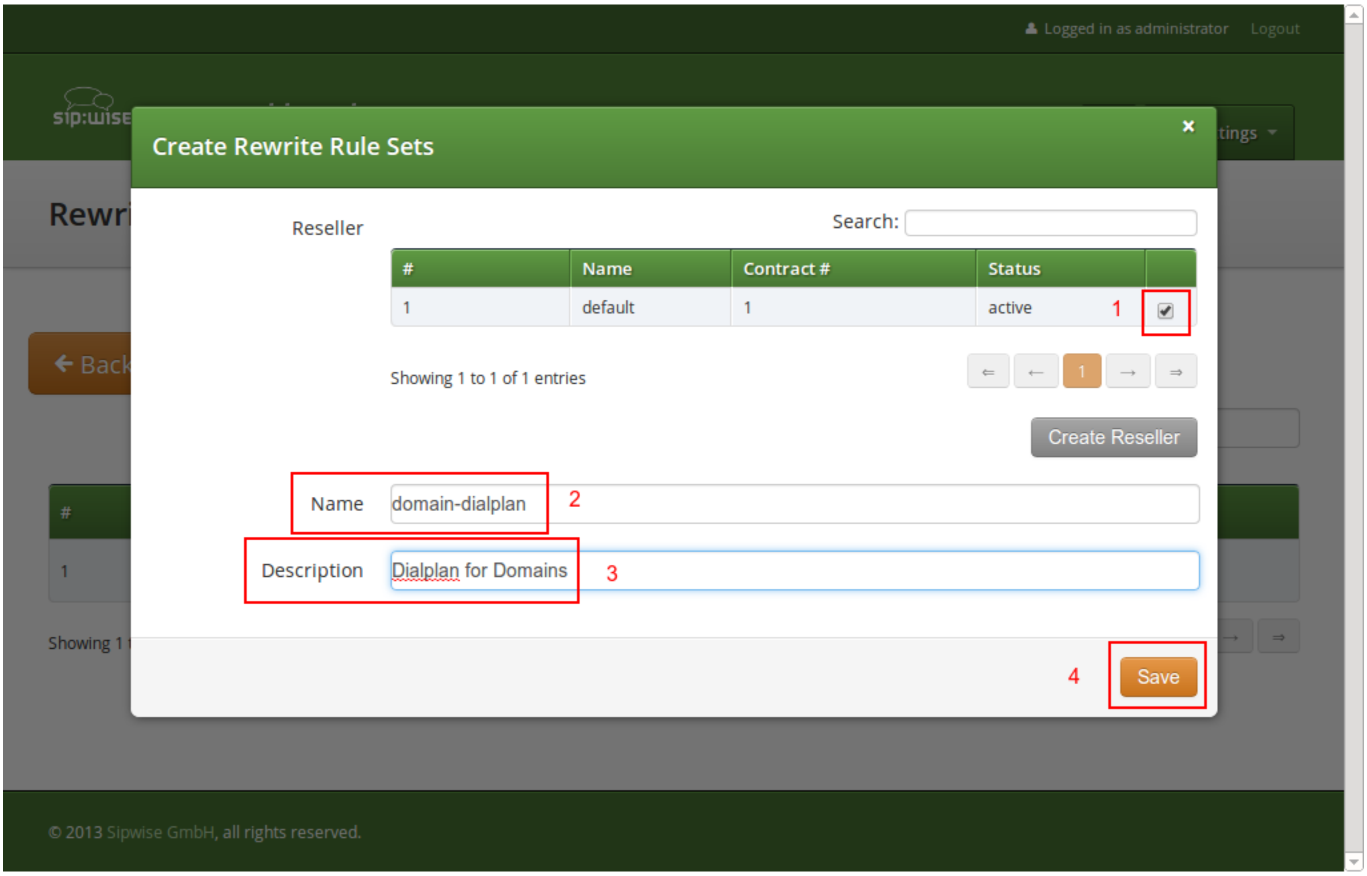
You can use several special variables in the Rewrite Rules, below you can find a list of them. Some examples of how to use them are also provided in the following sections:

- `${caller_cc}` : This is the value taken from the subscriber's preference CC value under Number Manipulation
- `${caller_ac}` : This is the value taken from the subscriber's preference AC value under Number Manipulation
- `${caller_emergency_cli}` : This is the value taken from the subscriber's preference emergency_cli value under Number Manipulation
- `${caller_emergency_prefix}` : This is the value taken from the subscriber's preference emergency_prefix value under Number Manipulation
- `${caller_emergency_suffix}` : This is the value taken from the subscriber's preference emergency_suffix value under Number Manipulation

To create a new Rewrite Rule Set, go to *Settings→Rewrite Rule Sets*. There you can create a Set identified by a name. This name is later shown in your peer-, domain- and user-preferences where you can select the rule set you want to use.

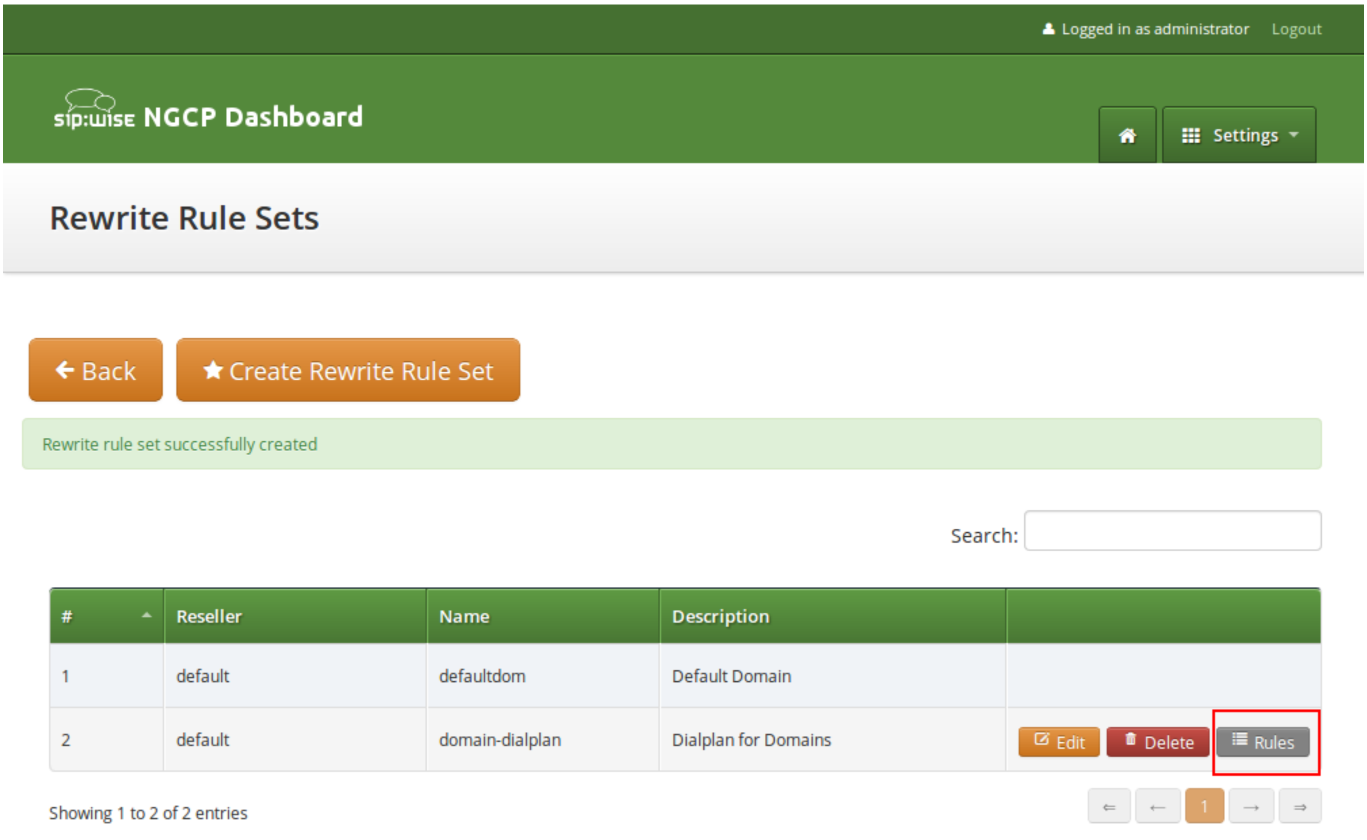


Click *Create Rewrite Rule Set* and fill in the form accordingly.



Press the *Save* button to create the set.

To view the *Rewrite Rules* within a set, hover over the row and click the *Rules* button.



The rules are ordered by *Caller* and *Callee* as well as direction *Inbound* and *Outbound*.

Tip

In Europe, the following formats are widely accepted: *+<cc><ac><sn>*, *00<cc><ac><sn>* and *0<ac><sn>*. Also, some countries allow the areacode-internal calls where only subscriber number is dialed to reach another number in the same area. Within this section, we will use these formats to show how to use rewrite rules to normalize and denormalize number formats.

6.7.1 Inbound Rewrite Rules for Caller

These rules are used to normalize user-provided numbers (e.g. passed in *From Display Name* or *P-Preferred-Identity* headers) into E.164 format. In our example, we'll normalize the three different formats mentioned above into E.164 format.

To create the following rules, click on the *Create Rewrite Rule* for each of them and fill them with the values provided below.

STRIP LEADING 00 OR +

- Match Pattern: `^(00|\+)([1-9][0-9]+)$`
- Replacement Pattern: `\2`
- Description: `International to E.164`
- Direction: `Inbound`

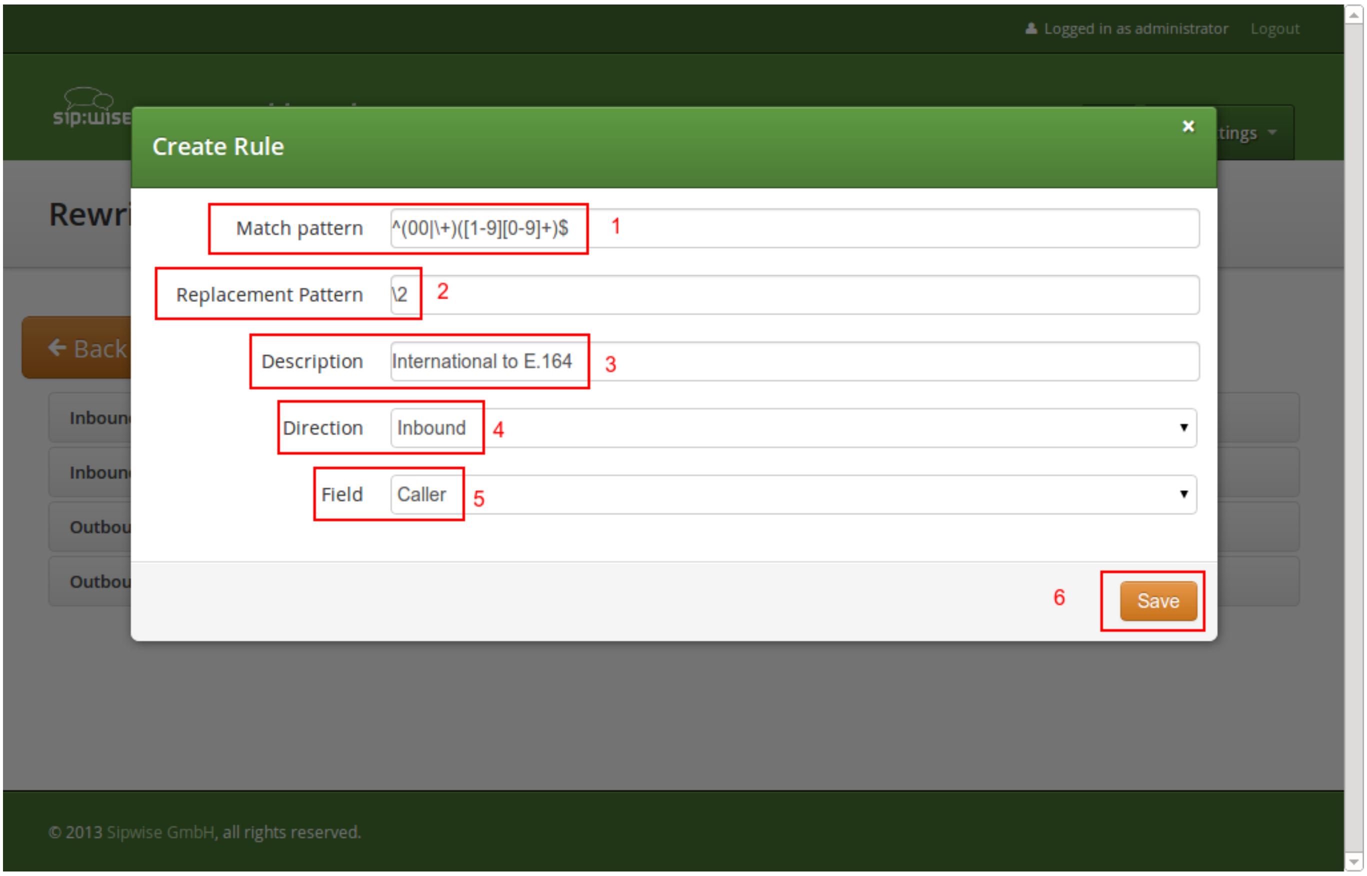
- Field: `Caller`

REPLACE 0 BY CALLER'S COUNTRY CODE:

- Match Pattern: `^0([1-9][0-9]+)$`
- Replacement Pattern: `${caller_cc}\1`
- Description: `National to E.164`
- Direction: `Inbound`
- Field: `Caller`

NORMALIZE LOCAL CALLS:

- Match Pattern: `^([1-9][0-9]+)$`
- Replacement Pattern: `${caller_cc}${caller_ac}\1`
- Description: `Local to E.164`
- Direction: `Inbound`
- Field: `Caller`



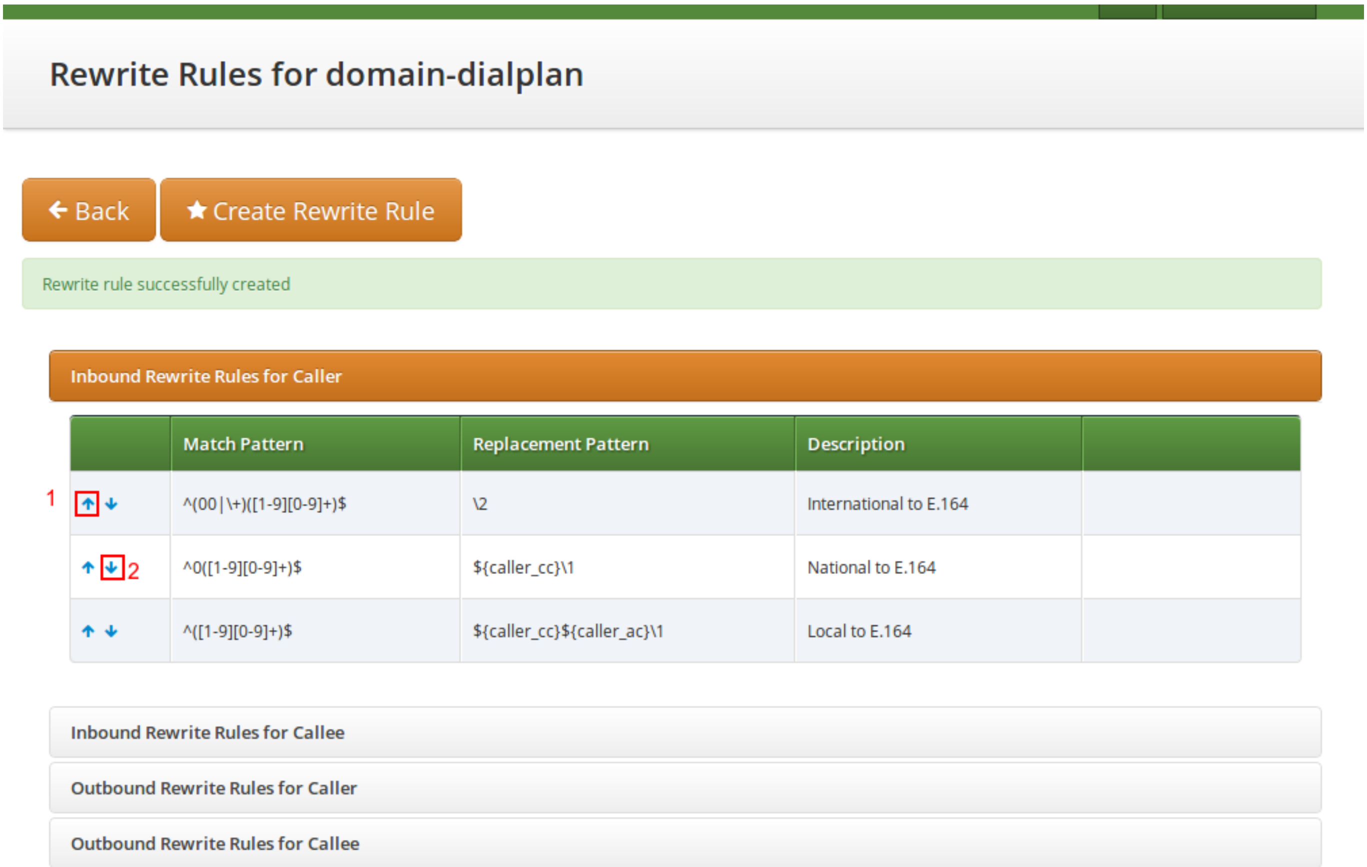
Normalization for national and local calls is possible with special variables `${caller_cc}` and `${caller_ac}` that can be used in Replacement Pattern and are substituted by the country and area code accordingly during the call routing.

Important

These variables are only being filled in when a call originates from a subscriber (because only then the cc/ac information is known by the system), so you can not use them when a calls comes from a SIP peer (the variables will be just empty in this case).

Tip

When routing a call, the rewrite processing is stopped after the first match of a rule, starting from top to bottom. If you have two rules (e.g. a generic one and a more specific one), where both of them would match some numbers, reorder them with the up/down arrows into the appropriate position.



6.7.2 Inbound Rewrite Rules for Callee

These rules are used to rewrite the number the end user dials to place a call to a standard format for routing lookup. In our example, we again allow the three different formats mentioned above and again normalize them to E.164, so we put in the same rules as for the caller.

STRIP LEADING `00` OR +

- Match Pattern: `^(00|\+)([1-9][0-9]+)$`
- Replacement Pattern: `\2`

- Description: `International to E.164`
- Direction: `Inbound`
- Field: `Callee`

Replace `0` by caller's country code:

- Match Pattern: `^0([1-9][0-9]+)$`
- Replacement Pattern: `${caller_cc}\1`
- Description: `National to E.164`
- Direction: `Inbound`
- Field: `Callee`

Normalize areacode-internal calls:

- Match Pattern: `^([1-9][0-9]+)$`
- Replacement Pattern: `${caller_cc}${caller_ac}\1`
- Description: `Local to E.164`
- Direction: `Inbound`
- Field: `Callee`

Tip

Our provided rules will only match if the caller dials a numeric number. If he dials an alphanumeric SIP URI, none of our rules will match and no rewriting will be done. You can however define rules for that as well. For example, you could allow your end users to dial `support` and rewrite that to your support hotline using the match pattern `^support$` and the replace pattern `43800999000` or whatever your support hotline number is.

6.7.3 Outbound Rewrite Rules for Caller

These rules are used to rewrite the calling party number for a call to an end user. For example, if you want the device of your end user to show *0<ac><sn>* if a national number calls this user, and *00<cc><ac><sn>* if an international number calls, put the following rules there.

Replace Austrian country code `43` by `0`

- Match Pattern: `^43([1-9][0-9]+)$`
- Replacement Pattern: `0\1`
- Description: `E.164 to Austria National`

- Direction: `Outbound`
- Field: `Caller`

PREFIX `00` FOR INTERNATIONAL CALLER

- Match Pattern: `^([1-9][0-9]+)$`
- Replacement Pattern: `00\1`
- Description: `E.164 to International`
- Direction: `Outbound`
- Field: `Caller`

Tip

Note that both of the rules would match a number starting with `43`, so reorder the national rule to be above the international one (if it's not already the case).

6.7.4 Outbound Rewrite Rules for Callee

These rules are used to rewrite the called party number immediately before sending out the call on the network. This gives you an extra flexibility by controlling the way request appears on a wire, when your SBC or other device expects the called party number to have a particular tech-prefix. It can be used on calls to end users too if you want to do some processing in intermediate SIP device, e.g. apply legal intercept selectively to some subscribers.

PREFIX `SIPSP#` FOR ALL CALLS

- Match Pattern: `^([0-9]+)$`
- Replacement Pattern: `sipsp#\1`
- Description: `Intercept this call`
- Direction: `Outbound`
- Field: `Callee`

6.7.5 Emergency Number Handling

There are 2 ways to handle calls from local subscribers to emergency numbers in NGCP:

- *Simple* emergency number handling: inbound rewrite rules append an emergency tag to the called number, this will be recognised by NGCP's call routing logic and the call is routed directly to a peer. Please read the next section for details of simple emergency number handling.
- An emergency *number mapping* is applied: a dedicated emergency number mapping database is consulted in order to obtain the most appropriate routing number of emergency services. This logic ensures that the caller will contact the geographically closest emergency service. Please visit the Emergency Mapping Section 7.5 section of the handbook for more details.

6.7.5.1 Simple Emergency Number Handling Overview

The overview of emergency call processing is as follows:

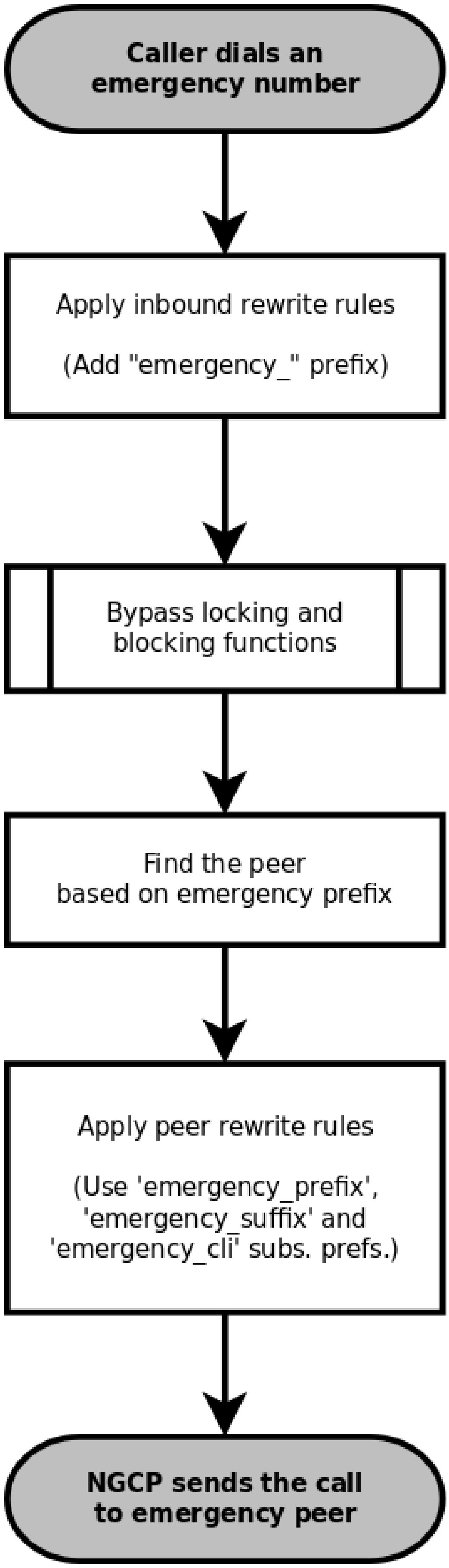


Figure 18: Simple Emergency Call Handling

Configuring Emergency Numbers is also done via Rewrite Rules.

6.7.5.2 Tagging Inbound Emergency Calls

For Emergency Calls from a subscriber to the platform, you need to define an *Inbound Rewrite Rule For Callee*, which adds a prefix `emergency_` to the number (and can rewrite the number completely as well at the same time). If the proxy detects a call to a SIP URI starting with `emergency_`, it will enter a special routing logic bypassing various checks which might make a normal call fail (e.g. due to locked or blocked numbers, insufficient credits or exceeding the max. amount of parallel calls).

TAG AN EMERGENCY CALL

- Match Pattern: `^(911|112)$`
- Replacement Pattern: `emergency_\1`
- Description: `Tag Emergency Numbers`
- Direction: `Inbound`
- Field: `Callee`

To route an Emergency Call to a Peer, you can select a specific peering group by adding a peering rule with a *callee prefix* set to `emergency_` to a peering group.

6.7.5.3 Normalize Emergency Calls for Peers

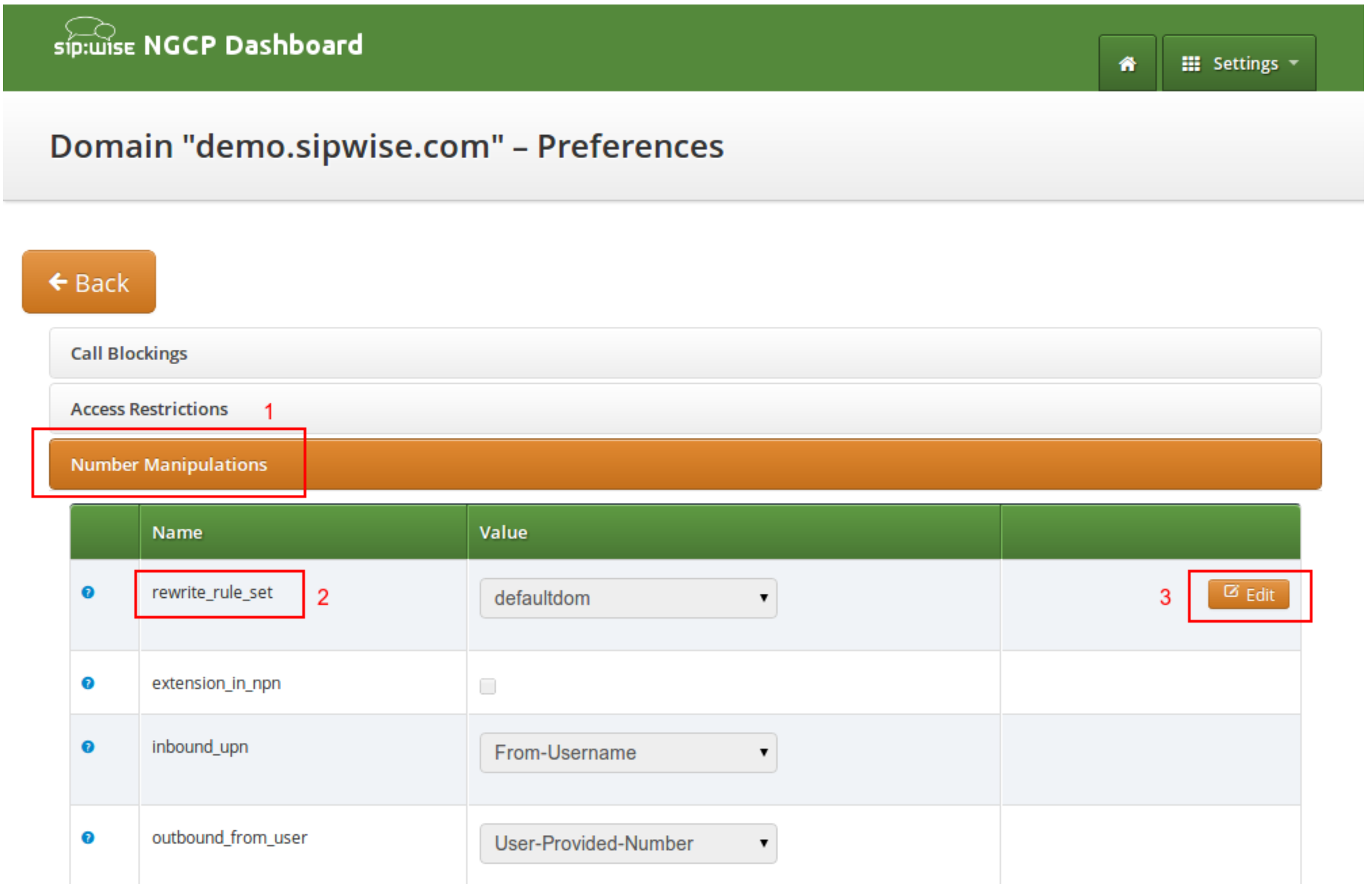
In order to normalize the emergency number to a valid format accepted by the peer, you need to assign an *Outbound Rewrite Rule For Callee*, which strips off the `emergency_` prefix. You can also use the variables `${caller_emergency_cli}`, `${caller_emergency_prefix}` and `${caller_emergency_suffix}` as well as `${caller_ac}` and `${caller_cc}`, which are all configurable per subscriber to rewrite the number into a valid format.

NORMALIZE EMERGENCY CALL FOR PEER

- Match Pattern: `^emergency_(.+)$`
- Replacement Pattern: `${caller_emergency_prefix}${caller_ac}\1`
- Description: `Normalize Emergency Numbers`
- Direction: `Outbound`
- Field: `Callee`

6.7.6 Assigning Rewrite Rule Sets to Domains and Subscribers

Once you have finished to define your Rewrite Rule Sets, you need to assign them. For sets to be used for subscribers, you can assign them to their corresponding domain, which then acts as default set for all subscribers. To do so, go to *Settings→Domains* and click *Preferences* on the domain you want the set to assign to. Click on *Edit* and select the Rewrite Rule Set created before.



You can do the same in the *Preferences* of your subscribers to override the rule on a subscriber basis. That way, you can finely control down to an individual user the dial-plan to be used. Go to *Settings*→*Subscribers*, click the *Details* button on the subscriber you want to edit, the click the *Preferences* button.

6.7.7 Creating Dialplans for Peering Servers

For each peering server, you can use one of the Rewrite Rule Sets that was created previously as explained in Section 6.7 (keep in mind that special variables `${caller_ac}` and `${caller_cc}` can not be used when the call comes from a peer). To do so, click on the name of the peering server, look for the preference called *Rewrite Rule Sets*.

If your peering servers don't send numbers in E.164 format *<cc><ac><sn>*, you need to create *Inbound Rewrite Rules* for each peering server to normalize the numbers for caller and callee to this format, e.g. by stripping leading + or put them from national into E.164 format.

Likewise, if your peering servers don't accept this format, you need to create *Outbound Rewrite Rules* for each of them, for example to append a + to the numbers.

6.7.8 Call Routing Verification

The sip:provider CE provides a utility that helps with the verification of call routing among local subscribers and peers. It is called *Call Routing Verification* and employs rewrite rules and peer selection rules, in order to process calling and called numbers or SIP users and find the appropriate peer for the destination.

The *Call Routing Verification* utility performs only basic number processing and does not invoke the full number manipulation logic applied on real calls. The goal is to enable testing of rewrite rules, rather than validate the complete number processing.

- What is considered during the test:
  - subscriber preferences: `cli` and `allowed_clis`
  - domain / subscriber / peer rewrite rules
- What is not taken into account during the test:
  - other subscriber or peer preferences
  - LNP (Local Number Portability) lookup on called numbers; LNP rewrite rules

You can access the utility following the path on Admin web interface: *Tools* → *Call Routing Verification*.

Expected input data

- `Caller number/uri`: 2 formats are accepted in this field:
  - A simple **phone number** in international (`00431..`, `+431..`) or E.164 (`431..`) format.
  - A SIP **URI** in `username@domain` format (without adding "sip:" at the beginning).
- `Callee number/uri`: The same applies as for `Caller number/uri`.
- `Caller Type`: Select `Subscriber` or `Peer`, depending on the source of the call.
- `Caller Subscriber` or `Caller Peer`: Optionally, you can select the subscriber or peer explicitly. Without the explicit selection, however, the *Call Routing Verification* tool is able to find the caller in the database, based on the provided number / URI.
- `Caller RWR Override, Callee RWR Override, Callee Peer Override`: The caller / callee rewrite rules and peer selection rules defined in domain, subscriber and peer preferences are used for call processing by default. But you can also override them by explicitly selecting another rewrite or peer selection rule.

Examples

1. Using only phone numbers and explicit subscriber selection
   - Input Data:

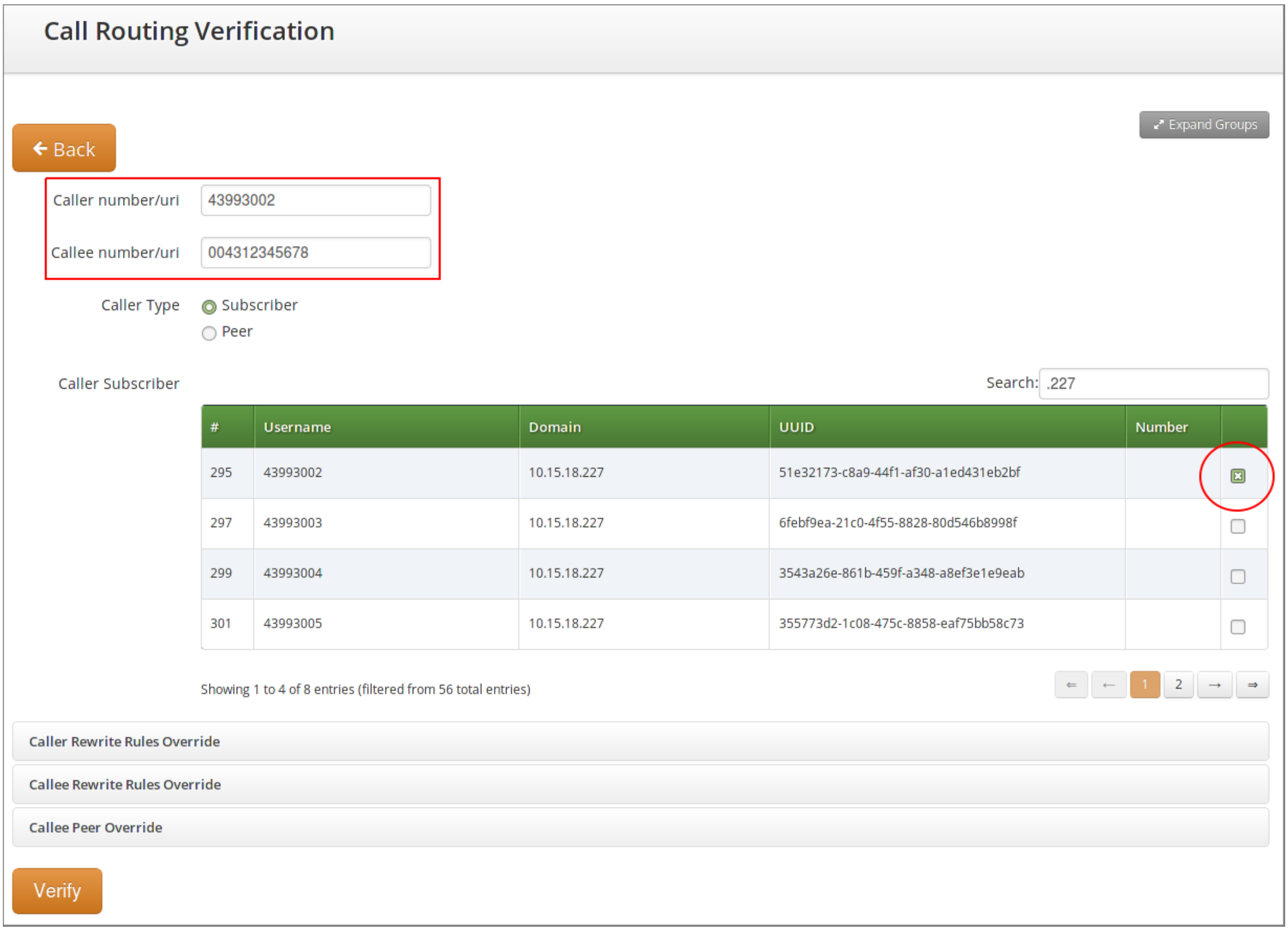


Figure 19: Call Routing Verif. - Only Numbers - Input

- Result:

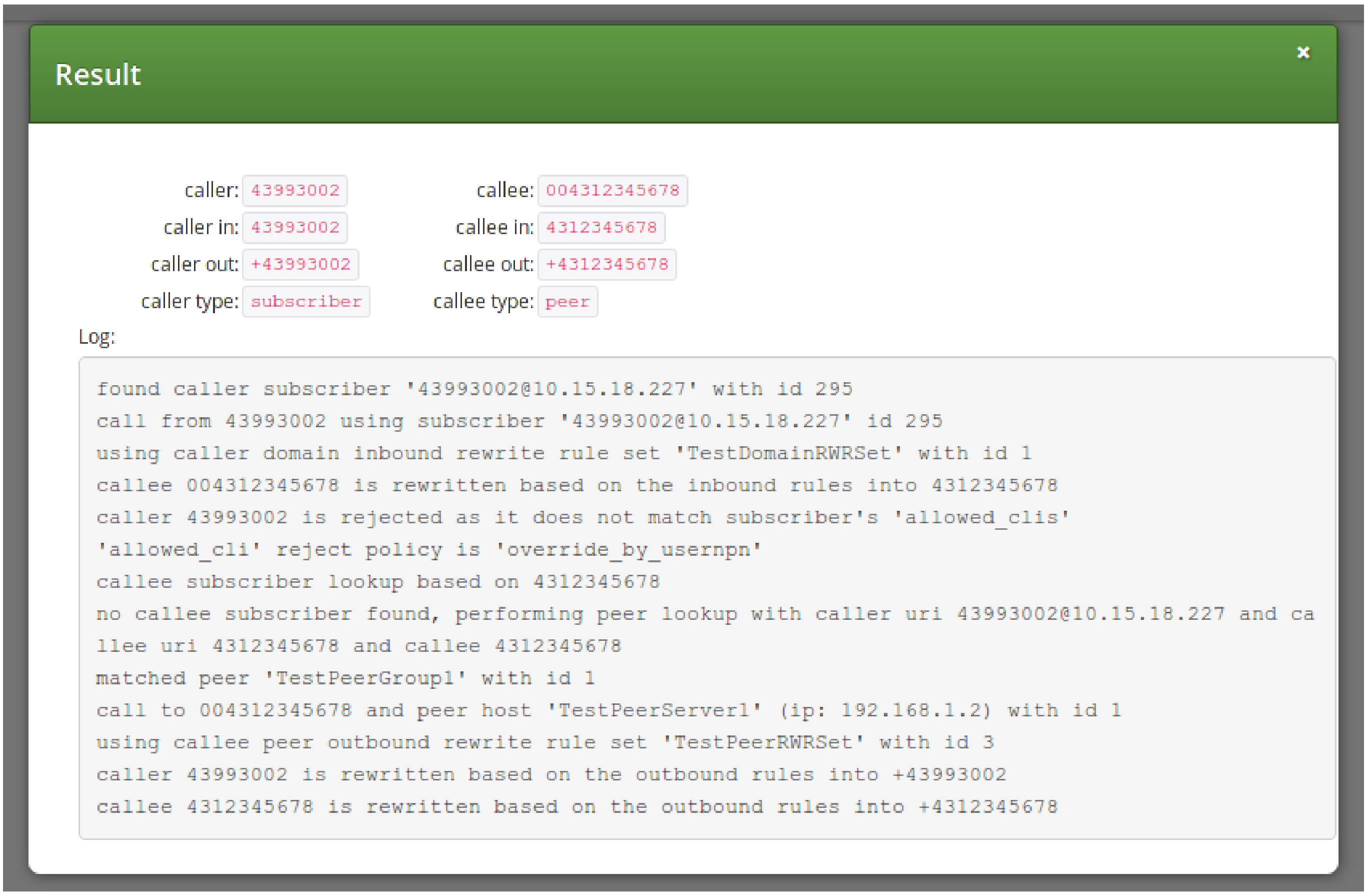


Figure 20: Call Routing Verif. - Only Numbers - Result

2. Using phone number and URI, without explicit subscriber selection

    • Input Data:

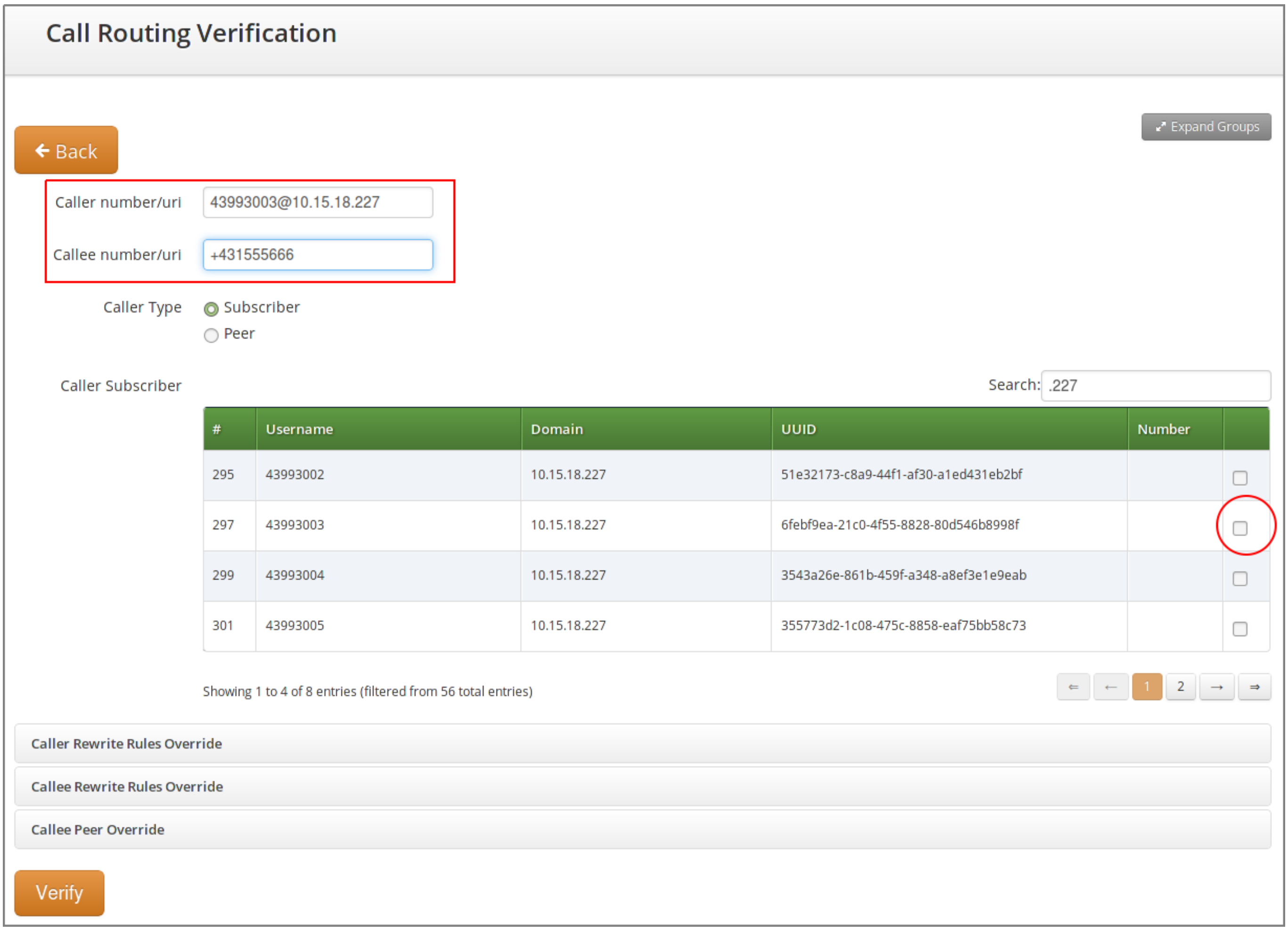


Figure 21: Call Routing Verif. - Number and URI - Input

- Result:

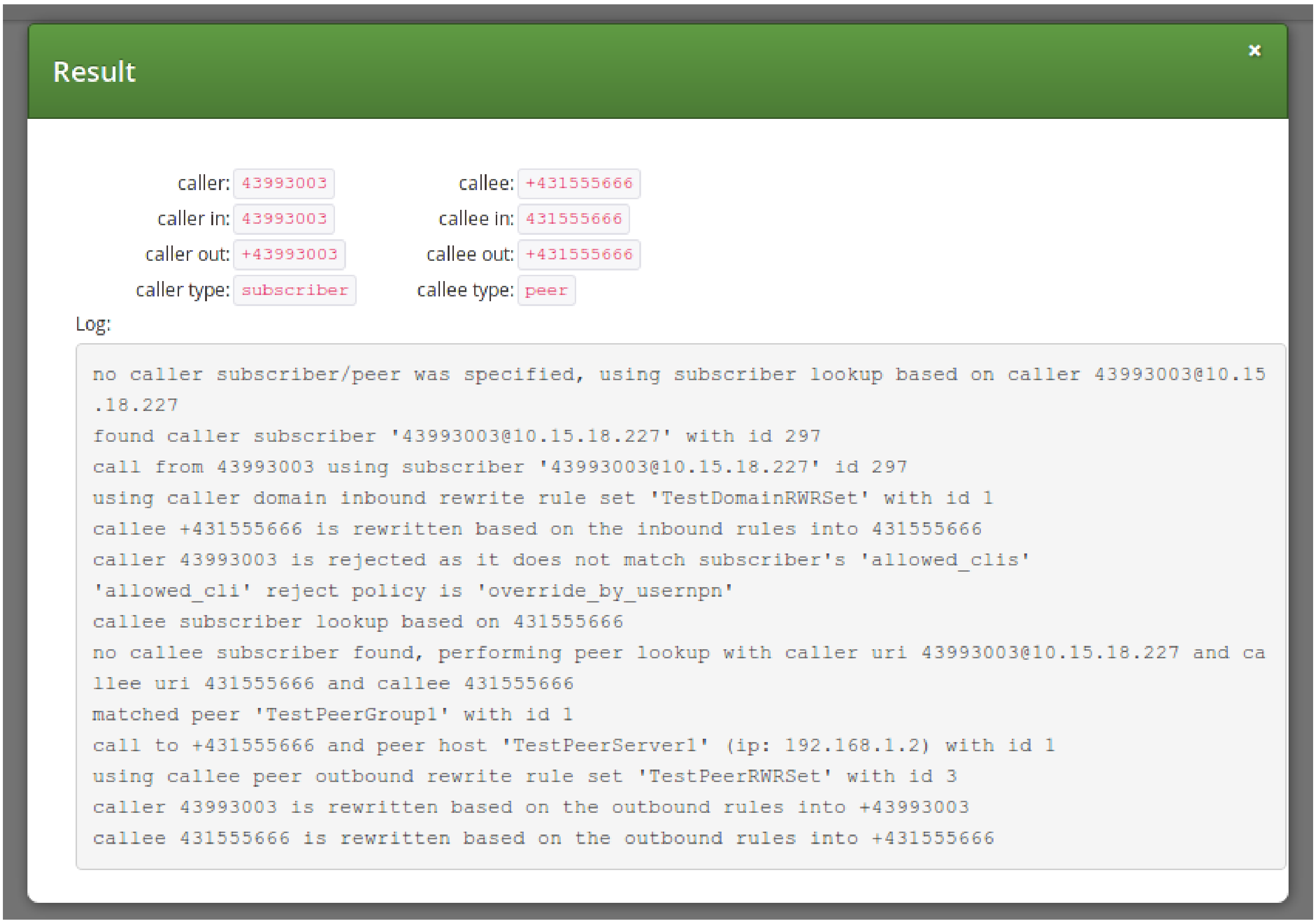


Figure 22: Call Routing Verif. - Number and URI - Result

7 Features

The sip:provider CE provides plenty of subscriber features to offer compelling VoIP services to end customers, and also to cover as many deployment scenarios as possible. In this chapter, we provide the features overview and describe their function and use cases.

7.1 Managing System Administrators

The sip:provider CE offers the platform operator with an easy to use interface to manage users with administrative privileges. Such users are representatives of resellers, and are entitled to manage configuration of services for *Customers, Subscribers, Domains, Billing Profiles* and other entities on Sipwise NGCP.

Administrators, as user accounts, are also used for client authentication on the REST API of NGCP.

There is a single administrator, whose account is enabled by default and who belongs to the *default reseller*. This user is the *superuser* of the NGCP administrative web interface (the so-called "admin panel"), and he has the right to modify administrators of other *Resellers* as well.

7.1.1 Configuring Administrators

Configuration of access rights of system administrators is possible through the admin panel of NGCP. In order to do that, please navigate to *Settings* → *Administrators*.

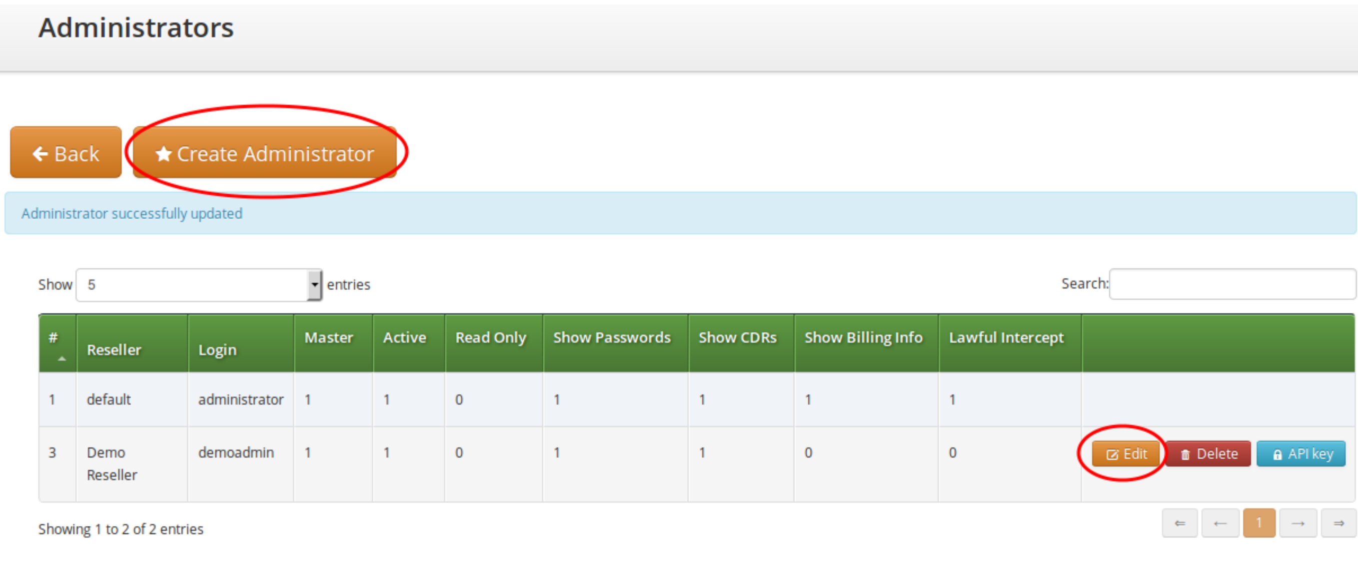


Figure 23: List of System Administrators

You have 2 options:

- If you'd like to **create** a new administrator user press *Create Administrator* button.

- If you'd like to **update** an existing administrator user press *Edit* button in its row.

There are some generic attributes that have to be set for each administrator:

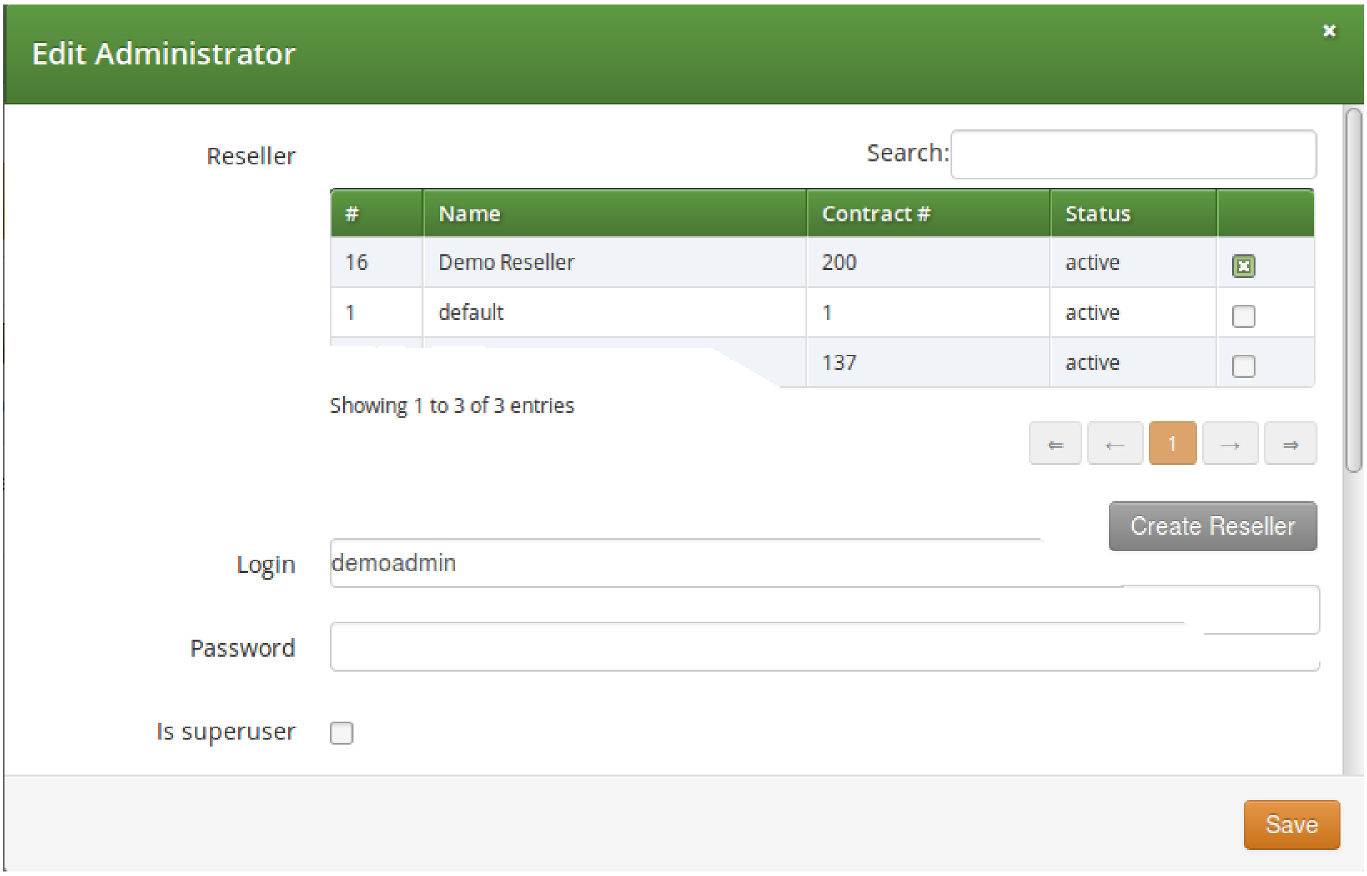


Figure 24: Generic System Administrator Attributes

- *Reseller*: each administrator user must belong to a *Reseller*. There is always a default reseller (ID: `1`, Name: `default`), but the administrator has to be assigned to his real reseller, if such an entity (other than `default`) exists.
- *Login*: the login name of the administrator user
- *Password*: the password of the administrator user for logging in the admin panel, or for authentication on REST API

The second set of attributes is a list of access rights that are discussed in subsequent section of the handbook.

7.1.2 Access Rights of Administrators

The various access rights of administrators are shown in the figure and summarized in the table below.

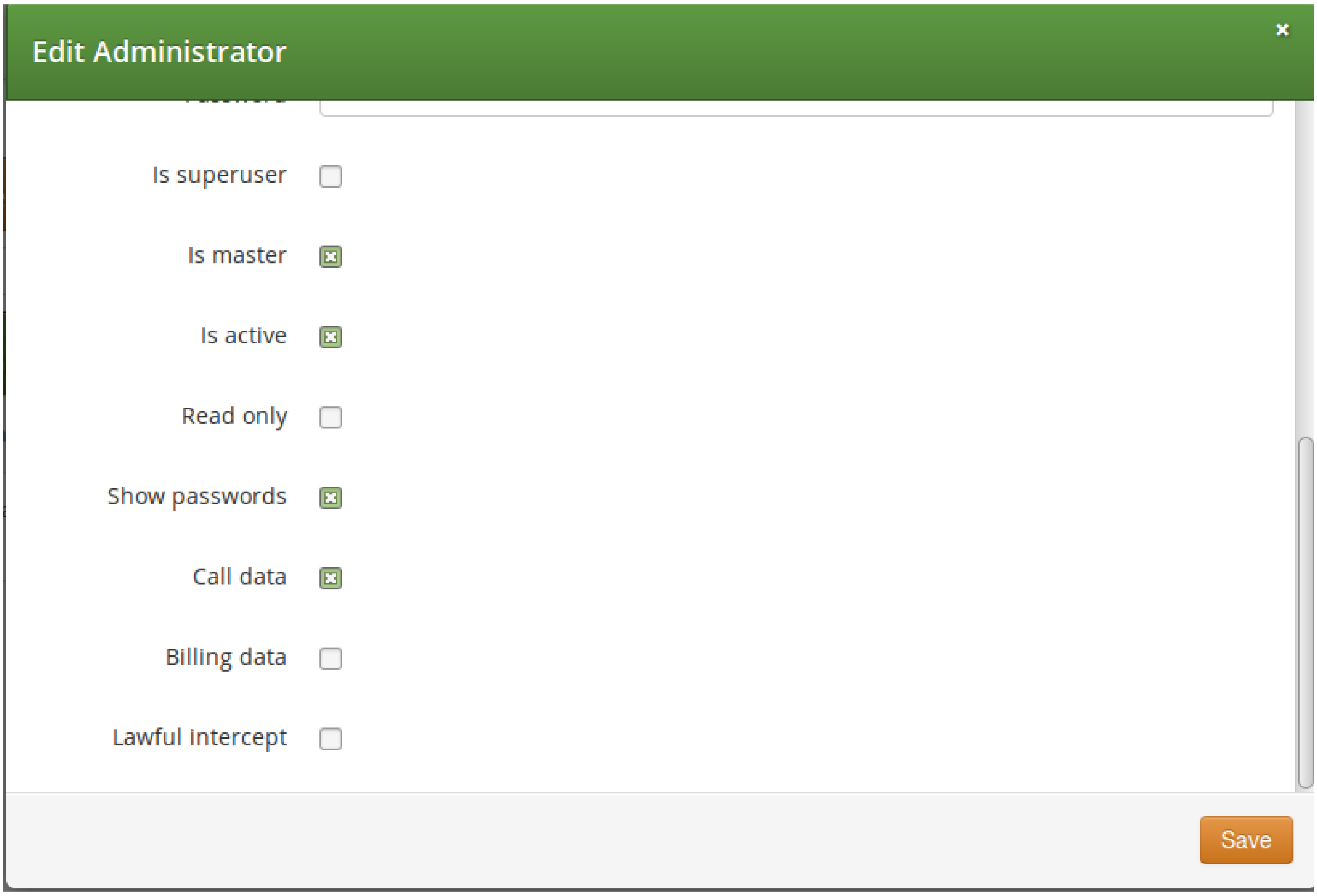


Figure 25: Access Rights of System Administrators

Table 2: Access Rights of System Administrators

Label in admin list	Access Right	Description
not shown	`Is superuser`	The user is allowed to modify data on Reseller level and — among others — is able to modify administrators of other resellers. There should be only 1 user on Sipwise NGCP with this privilege.
Master	`Is master`	The user is allowed to create, delete or modify other Admins who belong to the same Reseller.
Active	`Is active`	The user account is active, i.e. the admin user can login on the web panel or authenticate himself on REST API; otherwise user authentication will fail.

Table 2: (continued)

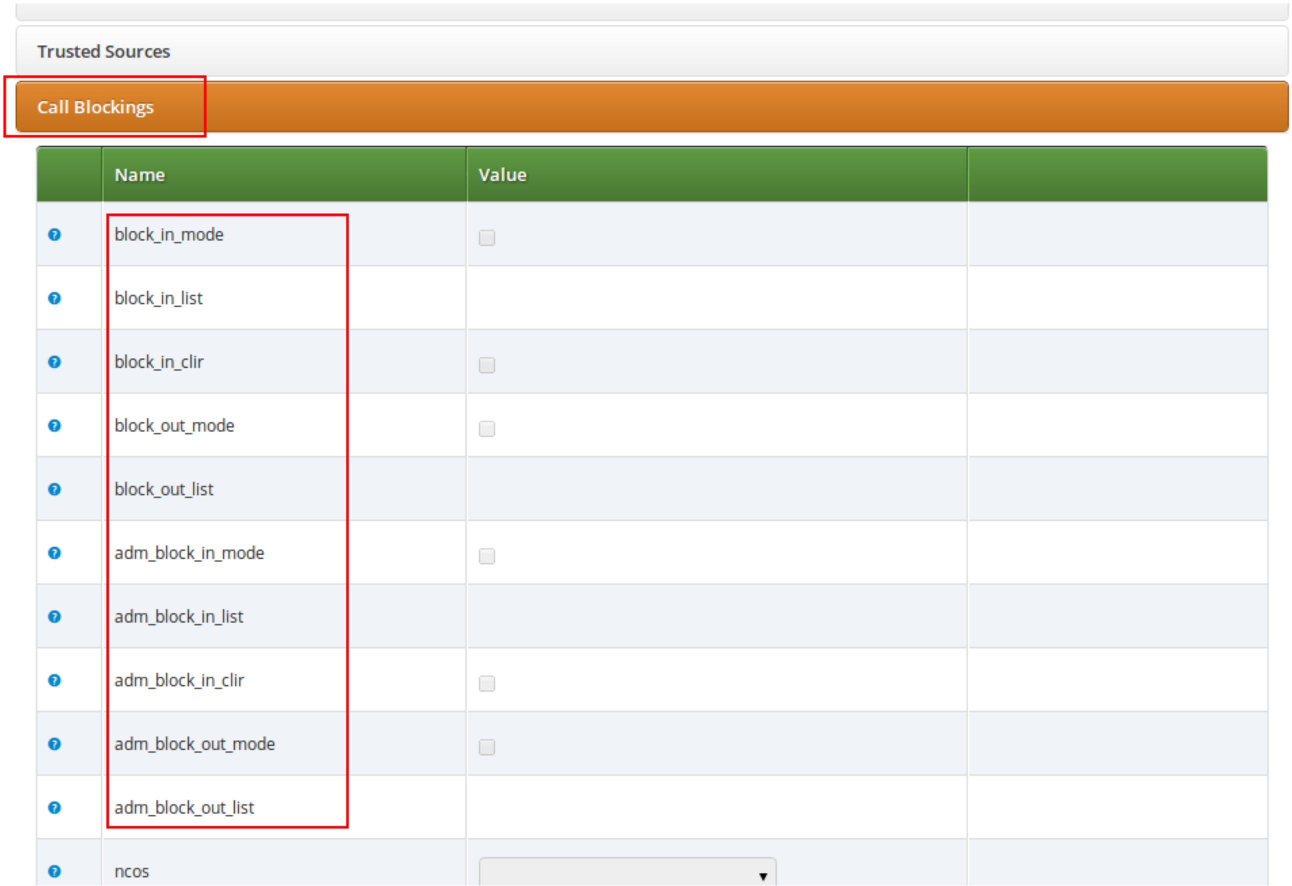
Label in admin list	Access Right	Description
Read Only	`Read only`	The user will only be able to list various data but is not allowed to modify anything. • For the **web interface** this means that *Create…* and *Edit* buttons will be hidden or disabled. • For the **REST API** this means that only `GET, HEAD, OPTIONS` HTTP request methods are accepted, and NGCP will reject those targeting data modification: `PUT, PATCH, POST, DELETE`.
Show Passwords	`Show passwords`	The user sees subscriber passwords (in plain text) on the web interface. **Note** Admin panel user passwords are stored in an unreadable way (cryptographic hash digest) in the database, while subscriber passwords are basically always stored in plain text. The latter happens on purpose, e.g. to make subscriber data migration possible.
Show CDRs	`Call data`	This privilege has effect on 2 items that will be displayed on admin panel of NGCP, when *Subscriber* → *Details* is selected: 1. PBX *Groups* list 2. *Captured Dialogs* list
Show Billing Info	`Billing data`	Some REST API resources that are related to billing are disabled: HTTP requests on `/api/vouchers, /api/topupcash` and `/api/topupvoucher` resources are rejected.
Lawful Intercept	`Lawful intercept`	If the privilege is selected then the REST API for interceptions (that is: `/api/interceptions`) is enabled; if the privilege is not selected then the interceptions API is disabled. **Note** This means that besides enabling LI in `config.yml` configuration file one also needs to enable the API via the LI privilege of an administrator user, so that NGCP can really provide LI service.

7.2 Access Control for SIP Calls

There are two different methods to provide fine-grained call admission control to both subscribers and admins. One is *Block Lists*, where you can define which numbers or patterns can be called from a subscriber to the outbound direction and which numbers or patterns are allowed to call a subscriber in the inbound direction. The other is *NCOS Levels*, where the admin predefines rules for outbound calls, which are grouped in certain levels. The subscriber can then just choose the level, or the admin can restrict a subscriber to a certain level. Also sip:provider CE offers some options to restrict the IP addresses that subscriber is allowed to use the service from. The following sections describe these features in detail.

7.2.1 Block Lists

Block Lists provide a way to control which users/numbers can call or be called, based on a subscriber level, and can be found in the *Call Blockings* section of the subscriber preferences.



Block Lists are separated into *Administrative Block Lists* (*adm_block_**) and *Subscriber Block Lists* (*block_**). They both have the same behaviour, but Administrative Block Lists take higher precedence. Administrative Block Lists are only accessible by the system administrator and can thus be used to override any Subscriber Block Lists, e.g. to block certain destinations. The following break-down of the various block features apply to both types of lists.

7.2.1.1 Block Modes

Block lists can either be *whitelists* or *blacklists* and are controlled by the User Preferences *block_in_mode*, *block_out_mode* and their administrative counterparts.

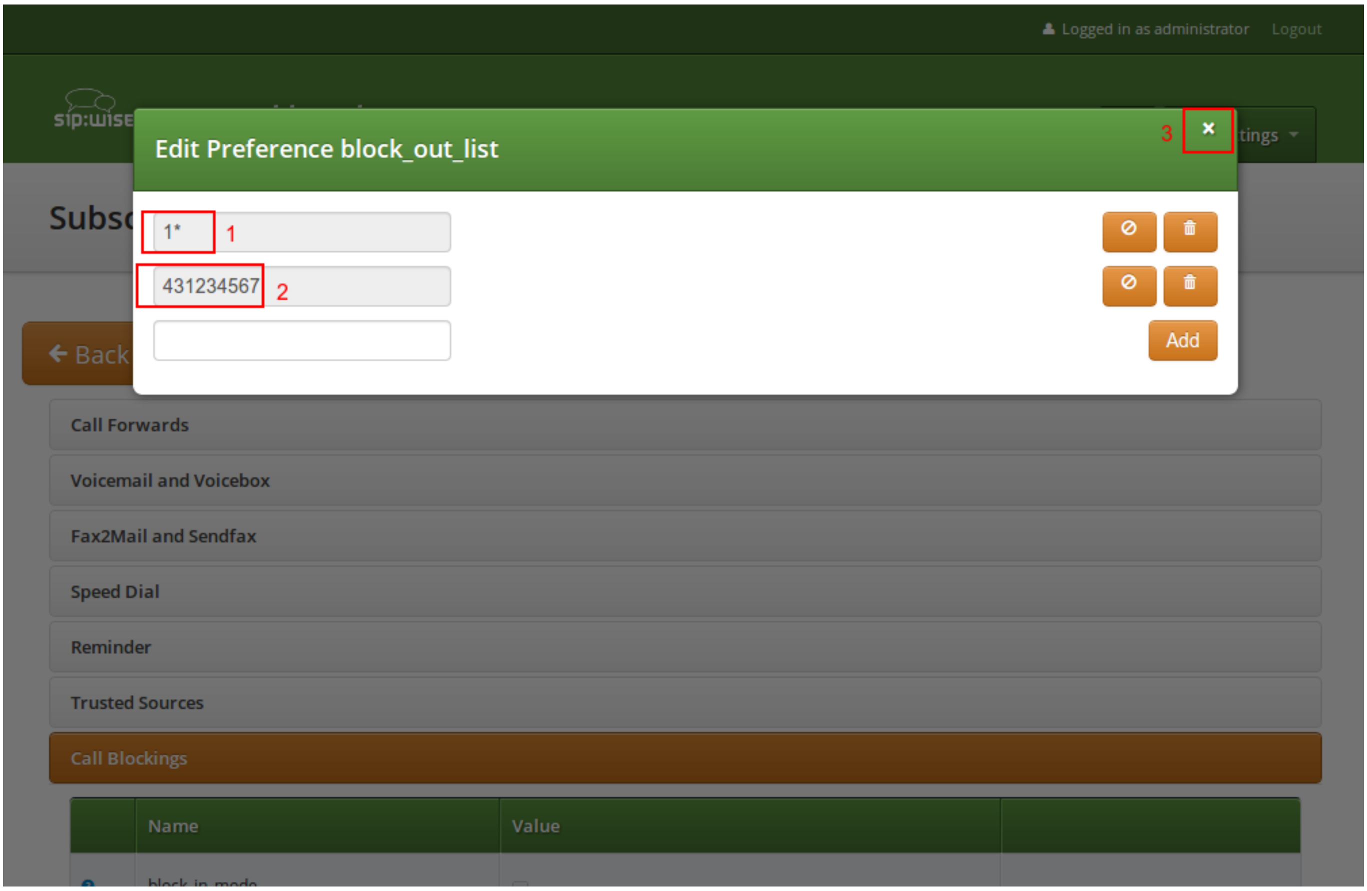
- The *blacklist* mode (option is not checked tells the system to **allow anything except the entries in the list**. Use this mode if you just want to block certain numbers and allow all the rest.
- The *whitelist* mode indicates to **reject anything except the entries in the list**. Use this mode if you want to enforce a strict policy and allow only selected destinations or sources.

You can change a list mode from one to the other at any time.

7.2.1.2 Block Lists

The list contents are controlled by the User Preferences *block_in_list*, *block_out_list* and their administrative counterparts. Click on the *Edit* button in the *Preferences* view to define the list entries.

In block list entries, you can provide shell patterns like `*` and `[]`. The behavior of the list is controlled by the *block_xxx_mode* feature (so they are either allowed or rejected). In our example above we have *block_out_mode* set to *blacklist*, so all calls to US numbers and to the Austrian number +431234567 are going to be rejected.



Click the *Close* icon once you're done editing your list.

7.2.1.3 Block Anonymous Numbers

For incoming call, the User Preference *block_in_clir* and *adm_block_in_clir* controls whether or not to reject incoming calls with number supression (either "[Aa]nonymous" in the display- or user-part of the From-URI or a header *Privacy: id* is set). This flag is independent from the Block Mode.

7.2.2 NCOS Levels

NCOS Levels provide predefined lists of allowed or denied destinations for outbound calls of local subscribers. Compared to *Block Lists*, they are much easier to manage, because they are defined on a global scope, and the individual levels can then be assigned to each subscriber. Again there is the distinction for user- and administrative-levels.

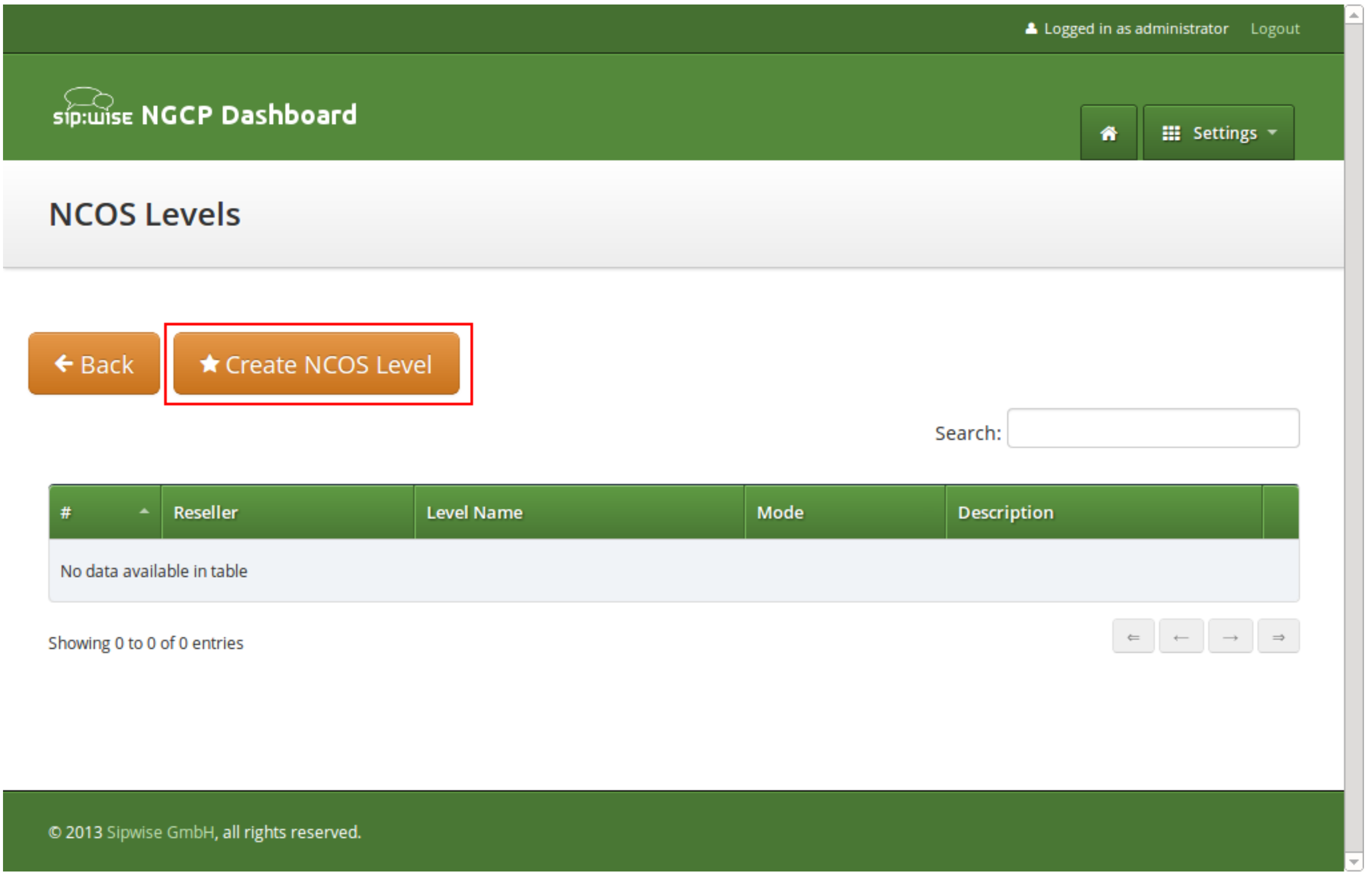
If case of a conflict, when the Block Lists feature allows a number and NCOS Levels rejects the same number or vice versa, the number will be rejected.

NCOS levels can either be *whitelists* or *blacklists*.

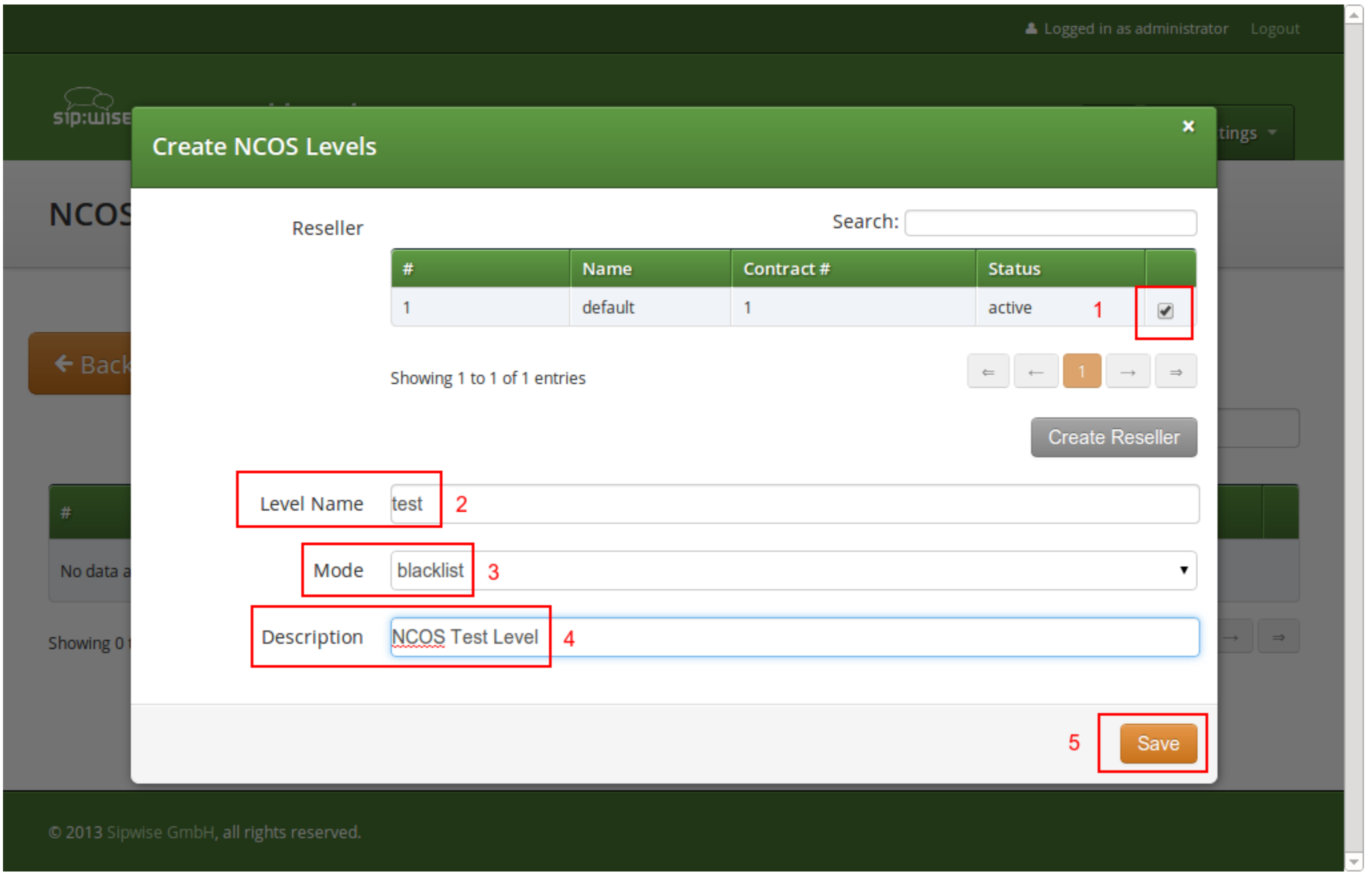
- The *blacklist* mode indicates to **allow everything except the entries in this level**. This mode is used if you want to just block certain destinations and allow all the rest.
- The *whitelist* mode indicates to **reject anything except the entries in this level**. This is used if you want to enforce a strict policy and allow only selected destinations.

7.2.2.1 Creating NCOS Levels

To create an NCOS Level, go to *Settings→NCOS Levels* and press the *Create NCOS Level* button.

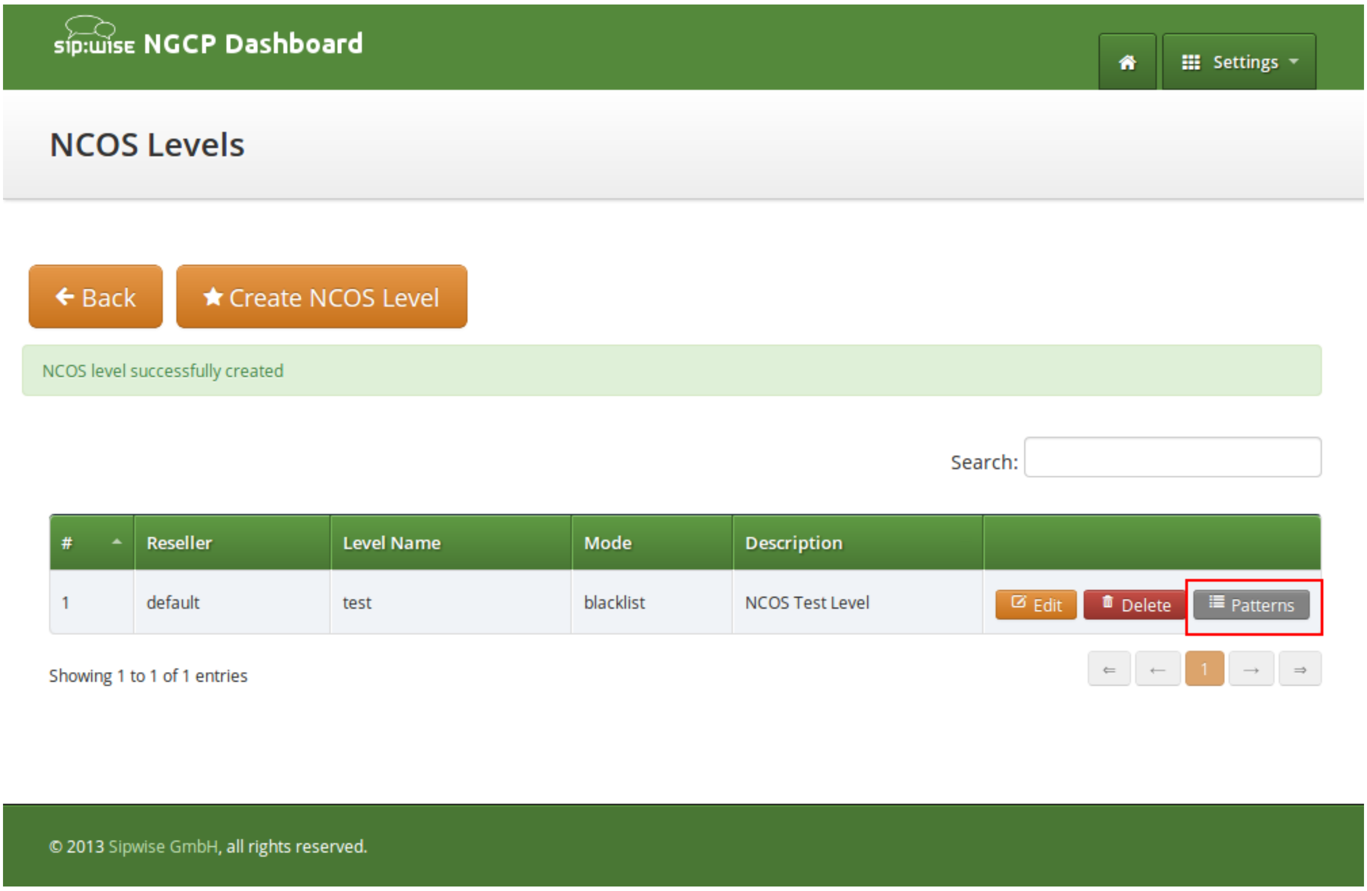


Select a reseller, enter a name, select the mode and add a description, then click the *Save* button.



7.2.2.2 Creating Rules per NCOS Level

To define the rules within the newly created NCOS Level, click on the *Patterns* button of the level.



There are 2 groups of patterns where you can define matching rules for the selected NCOS Level:

- NCOS Number Patterns: here you can define number patterns that will be matched against the called number and allowed or blocked, depending on whitelist / blacklist mode. The patterns are regular expressions.
- NCOS LNP Carriers: here you can select predefined *LNP Carriers* that will be allowed (whitelist mode) or prohibited (blacklist mode) to route calls to them. (See Section 7.4.1 in the handbook for the description of LNP functionality)

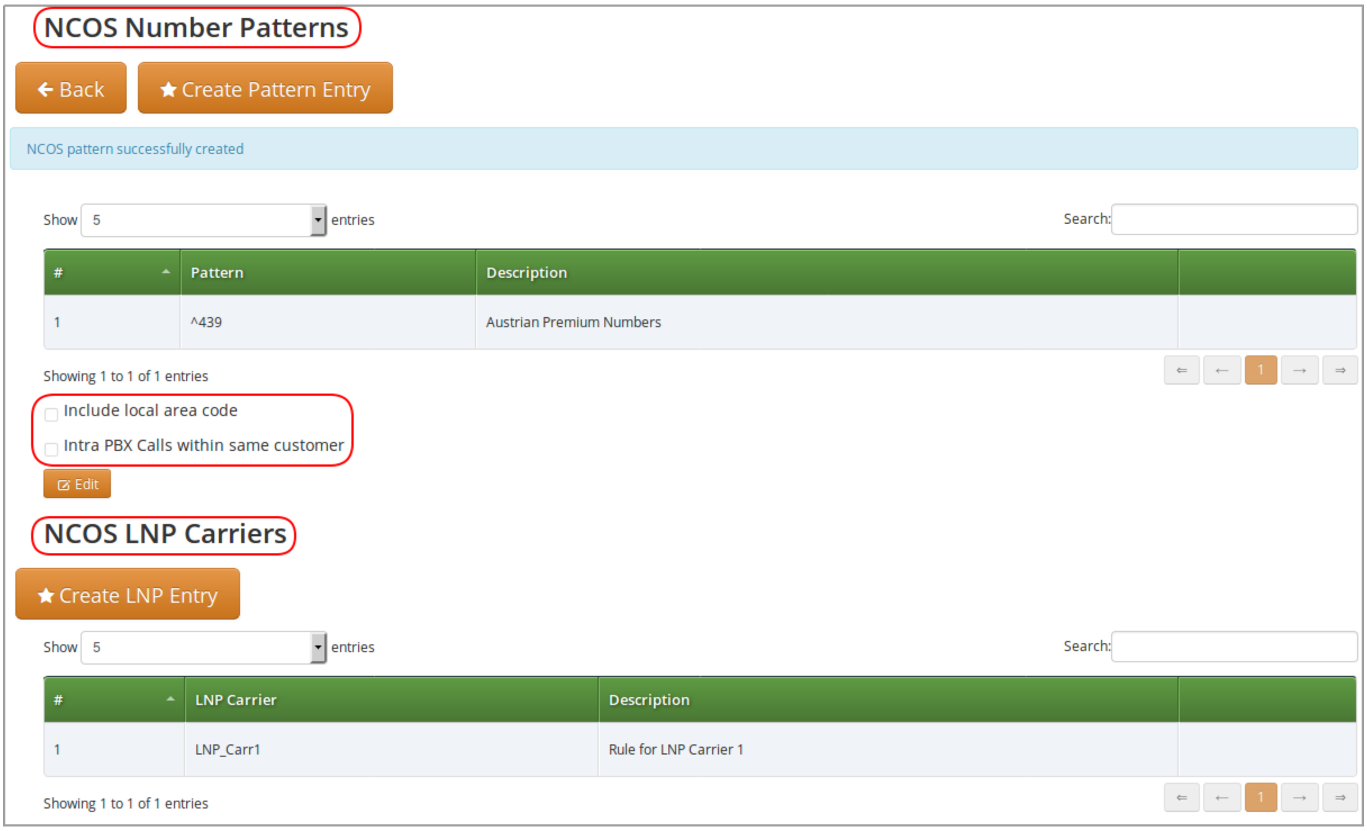


Figure 26: NCOS Patterns List

In the ***NCOS Number Patterns*** view you can create multiple patterns to define your level, one after the other. Click on the *Create Pattern Entry* Button on top and fill out the form.

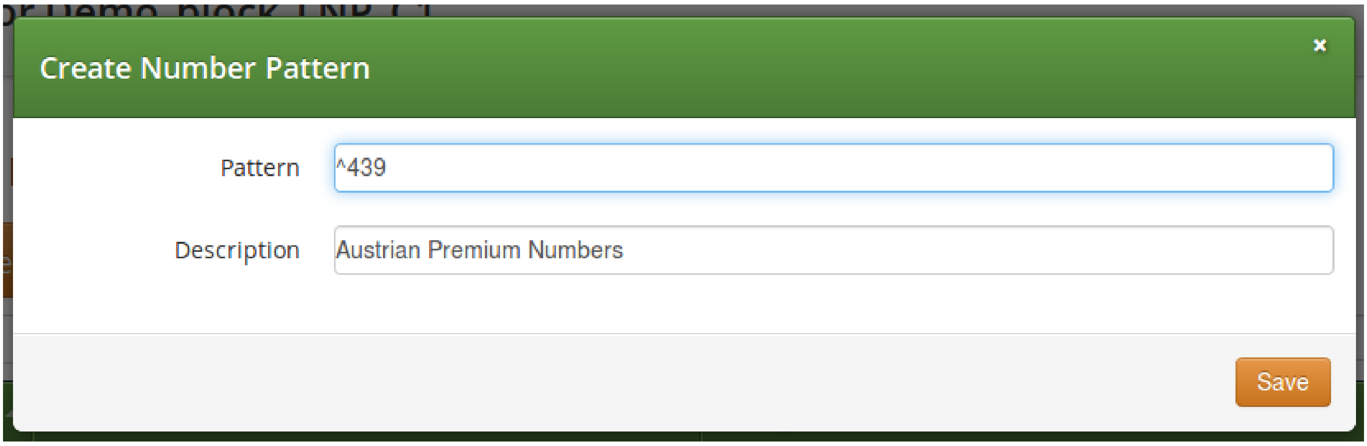


Figure 27: Create NCOS Number Pattern

In this example, we block (since the mode of the level is *blacklist*) all numbers starting with `439`. Click the *Save* button to save the entry in the level.

There are ***2 options*** that help you to easily define specific number ranges that will be allowed or blocked, depending on whitelist / blacklist mode:

- *Include local area code*: all subscribers within the caller's local area, e.g. if a subscriber has country-code 43 and area-code 1, then selecting this checkbox would result in the implicit number pattern: `^431`.
- *Intra PBX calls within same customer*: all subscribers that belong to the same PBX customer as the caller himself.

In the ***NCOS LNP Carriers*** view you can select specific LNP Carriers — i.e. carriers that host the called ported numbers — that will be allowed or blocked for routing calls to them (whitelist / blacklist mode, respectively).

Sipwise NGCP performs number matching always with the dialed number and not with the number generated after LNP lookup that is: either the original dialed number prefixed with an LNP carrier code, or the routing number.

An example of *NCOS LNP Carrier* pattern definition:

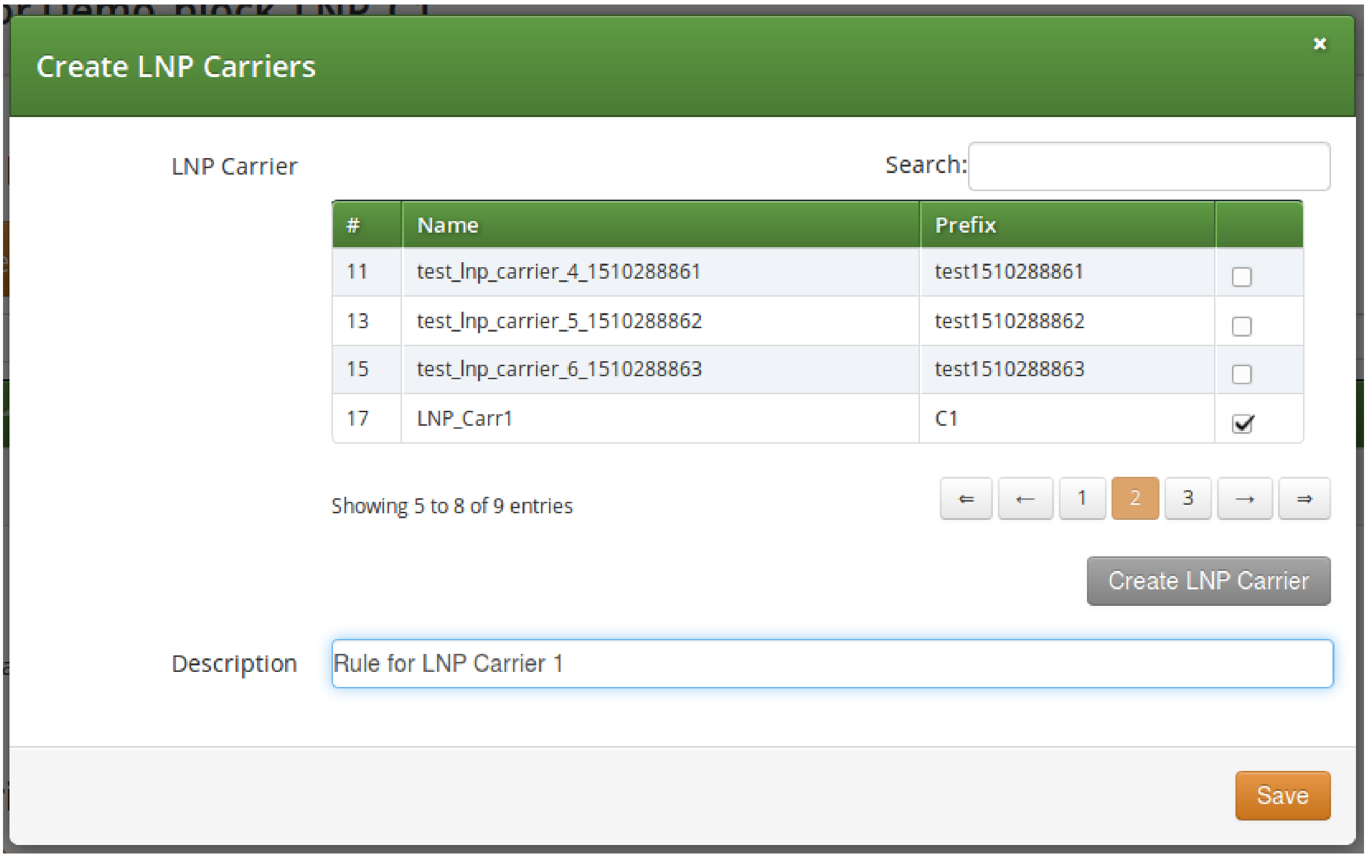


Figure 28: Create NCOS LNP Carrier

In the above example we created a rule that blocks calls to "LNP_Carr1" carrier, supposing we use blacklist mode of the NCOS Level.

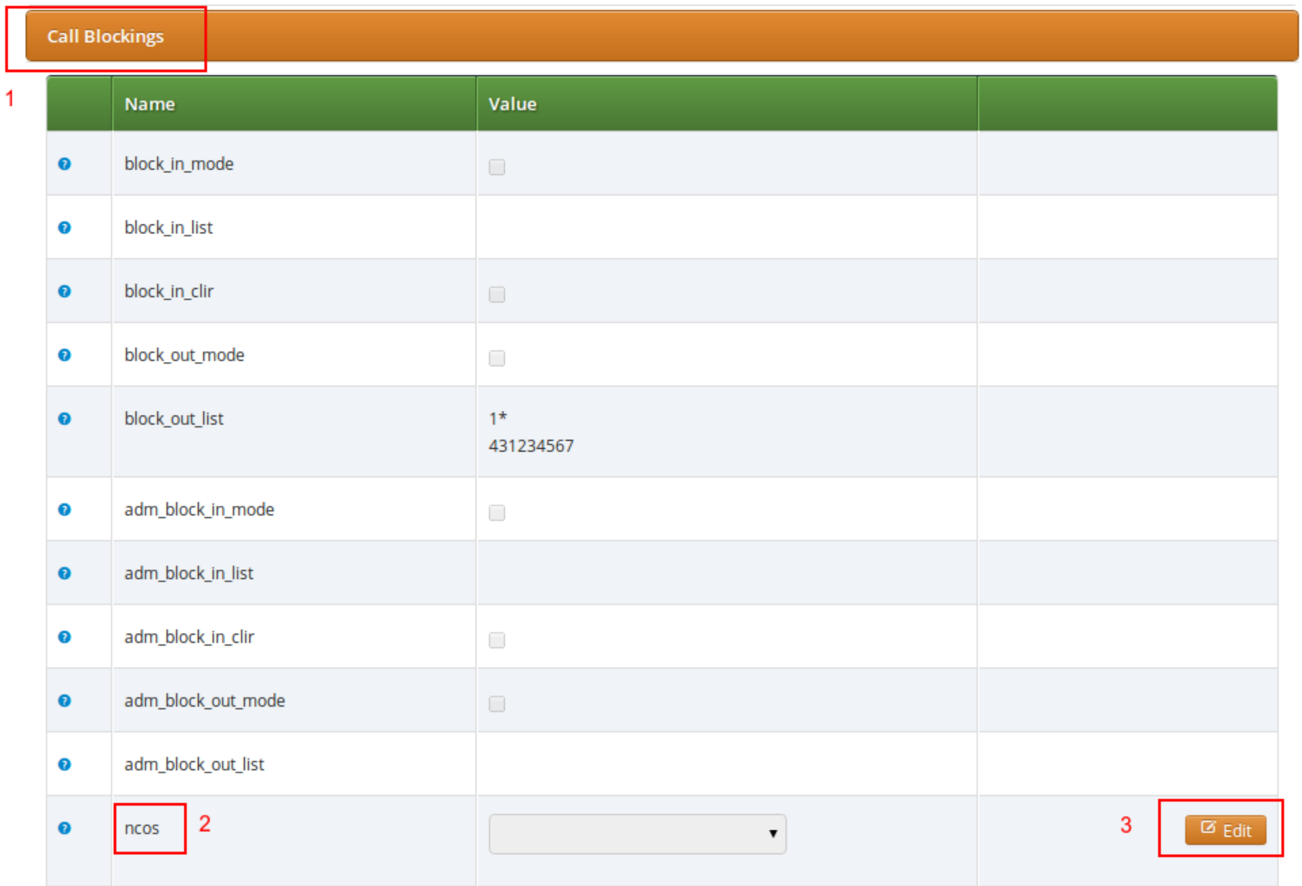
> **Note**
>
> Currently NGCP does not support filtering of individual phone numbers in addition to LNP Carrier matching. In other words: combining phone number and LNP Carrier patterns is not possible.

Tip

There might be situations when phone number patterns may not be strictly aligned with telephony providers, for instance in case of full number portability in a country. In such cases using *NCOS LNP Carriers* patterns still allows for defining NCOS levels that allow / block calls to mobile numbers, for example. In order to achieve this goal you have to list all LNP carriers in the NCOS patterns that are known to host mobile numbers.

7.2.2.3 Assigning NCOS Levels to Subscribers/Domains

Once you've defined your NCOS Levels, you can assign them to local subscribers. To do so, navigate to *Settings→Subscribers*, search for the subscriber you want to edit, press the *Details* button and go to the *Preferences* View. There, press the *Edit* button on either the *ncos* or *adm_ncos* setting in the *Call Blockings* section.



You can assign the NCOS level to all subscribers within a particular domain. To do so, navigate to *Settings→Domains*, select the domain you want to edit and click *Preferences*. There, press the *Edit* button on either *ncos* or *admin_ncos* in the *Call Blockings* section.

Note: if both domain and subscriber have same NCOS preference set (either *ncos* or *adm_ncos*, or both) the subscriber's preference is used. This is done so that you can override the domain-global setting on the subscriber level.

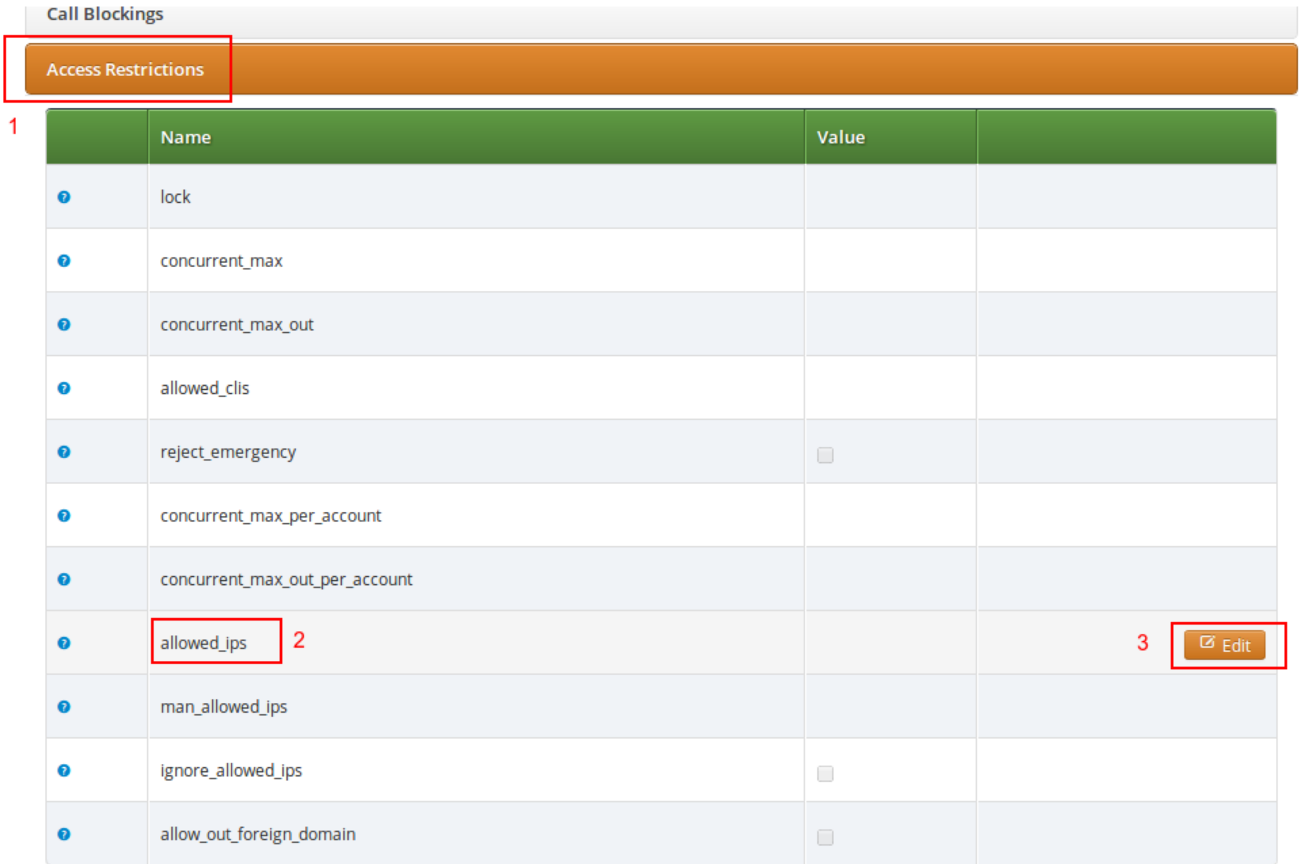
7.2.2.4 Assigning NCOS Level for Forwarded Calls to Subscribers/Domains

In some countries there are regulatory requirements that prohibit subscribers from forwarding their numbers to special numbers like emergency, police etc. While the sip:provider CE does not deny provisioning Call Forward to these numbers, the administrator can prevent the incoming calls from being actually forwarded to numbers defined in the NCOS list: just select the appropriate NCOS level in the domain's or subscriber's preference *adm_cf_ncos*. This NCOS will apply only to the Call Forward from the subscribers and not to the normal outgoing calls from them.

7.2.3 IP Address Restriction

The sip:provider CE provides subscriber preference *allowed_ips* to restrict the IP addresses that subscriber is allowed to use the service from. If the REGISTER or INVITE request comes from an IP address that is not in the allowed list, the sip:provider CE will reject it with a 403 message. Also a voice message can be played when the call attempt is rejected (if configured).

By default, *allowed_ips* is an empty list which means that subscriber is not restricted. If you want to configure a restriction, navigate to *Settings*→*Subscribers*, search for the subscriber you want to edit, press *Details* and then *Preferences* and press *Edit* for the *allowed_ips* preference in the *Access Restrictions* section.



Press the Edit button to the right of empty drop-down list.

You can enter multiple allowed IP addresses or IP address ranges one after another. Click the *Add* button to save each entry in the list. Click the *Delete* button if you want to remove some entry.

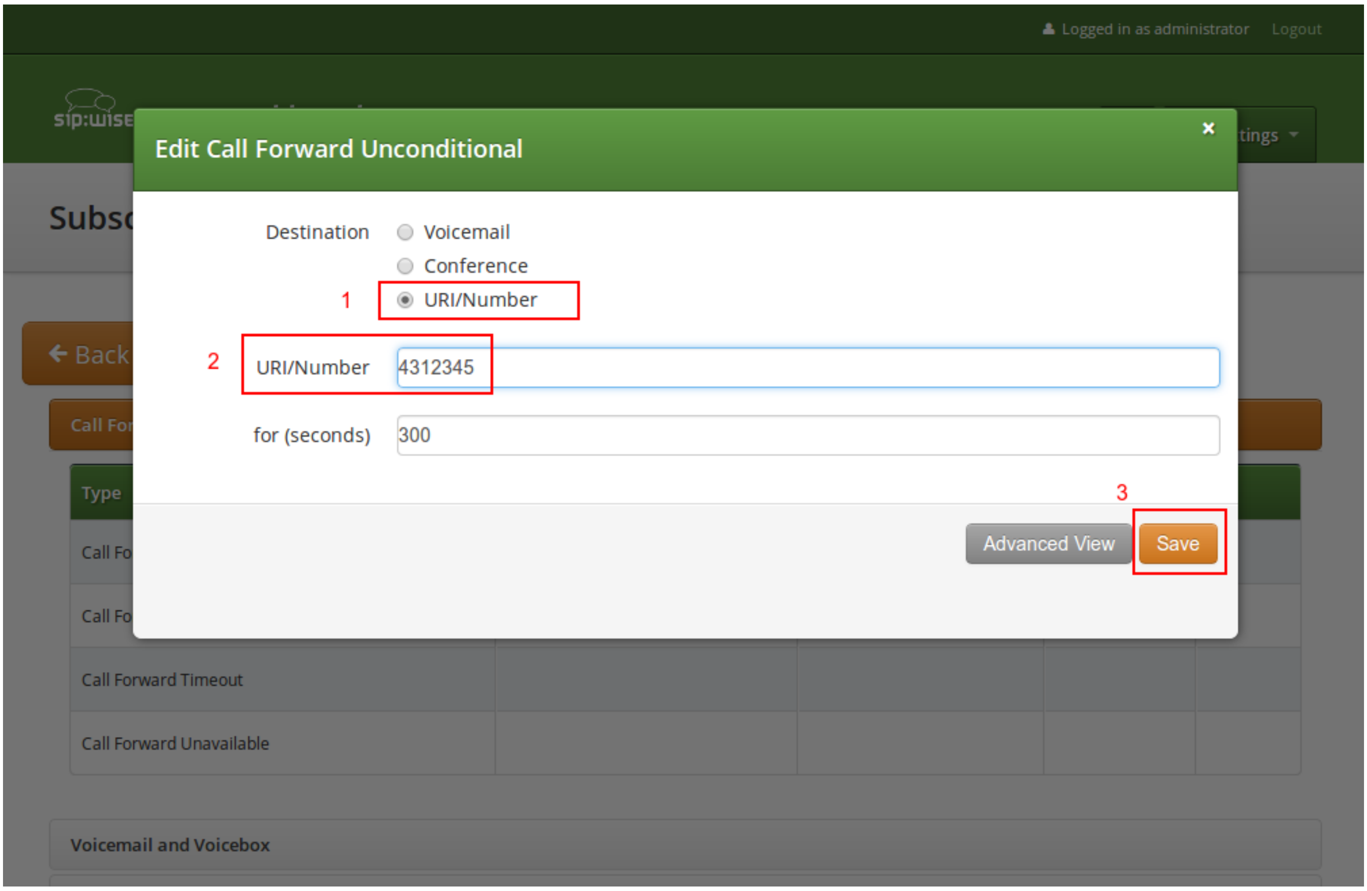
7.3 Call Forwarding and Call Hunting

The sip:provider CE provides the capabilities for normal *call forwarding* (deflecting a call for a local subscriber to another party immediately or based on events like the called party being busy or doesn't answer the phone for a certain number of seconds) and *serial call hunting* (sequentially executing a group of deflection targets until one of them succeeds). Targets can be stacked, which means if a target is also a local subscriber, it can have another call forward or hunt group which is executed accordingly.

Call Forwards and Call Hunting Groups can either be executed unconditionally or based on a *Time Set Definition*, so you can define deflections based on time period definitions (e.g. Monday to Friday 8am to 4pm etc).

7.3.1 Setting a simple Call Forward

Go to your *Subscriber Preferences* and click *Edit* on the Call Forward Type you want to set (e.g. *Call Forward Unconditional*).



If you select *URI/Number* in the *Destination* field, you also have to set a *URI/Number*. The timeout defines for how long this destination should be tried to ring.

7.3.2 Advanced Call Hunting

Beside call forwarding to a single destination, Sipwise NGCP offers the possibility to activate call forwarding in a more sophisticated way:

- to multiple destinations (→ *Destination Set*)
- only during a pre-defined time set (→ *Time Set*)
- only for specific callers (→ *Source Set*)

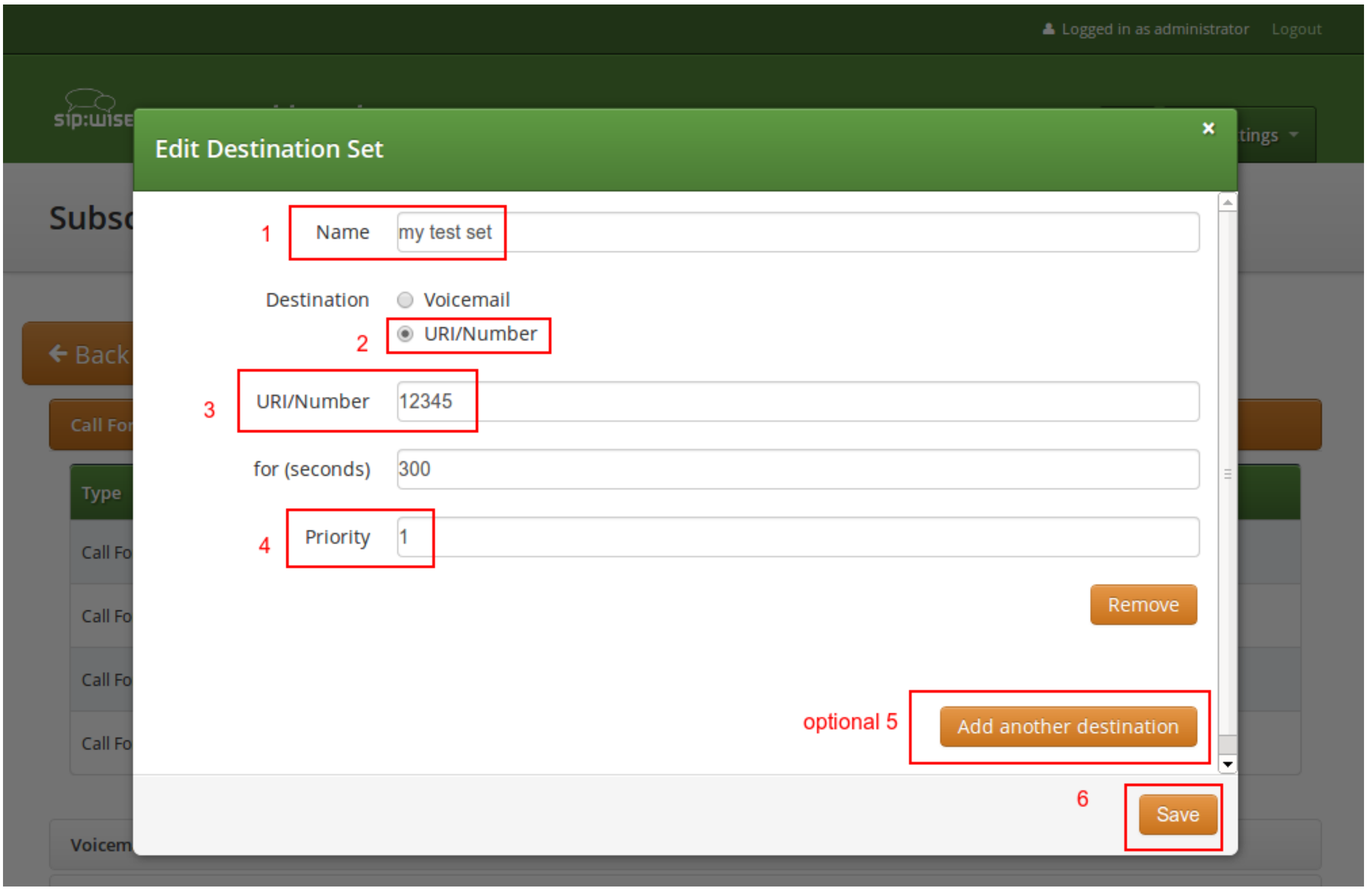
If you want to define such more detailed call forwarding rules, you need to change into the *Advanced View* when editing your call forward. There, you can select multiple *Destination Set - Time Set - Source Set* triples that determine all conditions under which the call will be forwarded.

Explanation of call forward parameters

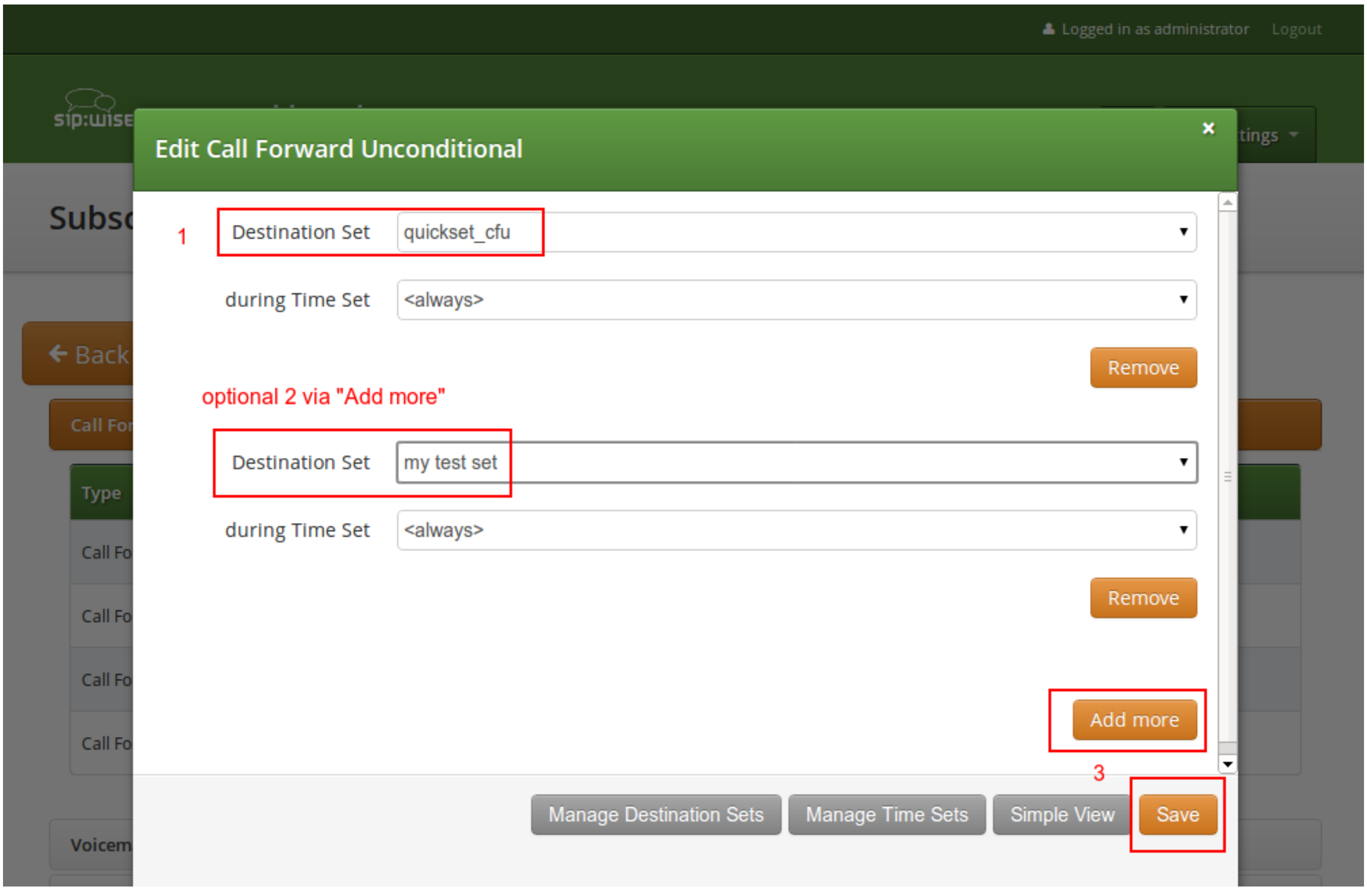
- A ***Destination Set*** is a list of destinations where the call will be routed to, one after another, according to the order of their assigned priorities. See the Destination Sets Section 7.3.2.1 subchapter for a detailed description.
- A ***Time Set*** is a time period definition, i.e. when the call forwarding has to be active. See the Time Sets Section 7.3.2.2 subchapter for a detailed description.
- A ***Source Set*** is a list of number patterns that will be matched against the calling party number; if the calling number matches the call forwarding will be executed. See the Source Sets Section 7.3.2.3 subchapter for a detailed description.

7.3.2.1 Configuring Destination Sets

Click on *Manage Destination Sets* to see a list of available sets. The *quickset_cfu* has been implicitly created during our creation of a simple call forward. You can edit it to add more destinations, or you can create a new destination set.



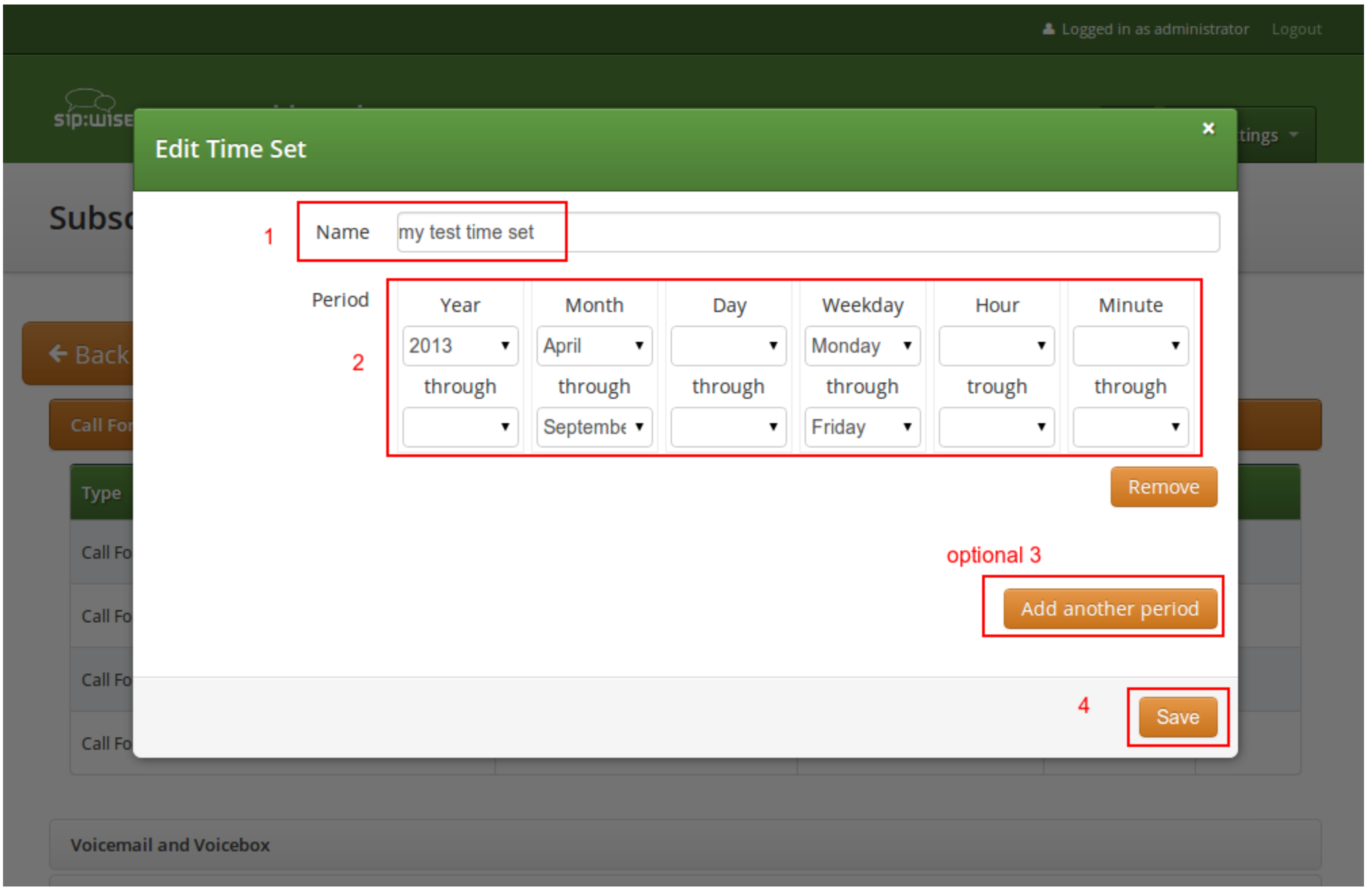
When you close the *Destination Set* Overview, you can now assign your new set in addition or instead of the *quickset_cfu* set.



Press *Save* to store your settings.

7.3.2.2 Configuring Time Sets

Click on *Manage Time Sets* in the advanced call-forward menu to see a list of available time sets. By default there are none, so you have to create one.



You need to provide a *Name*, and a list of *Periods* where this set is active. If you only set the top setting of a date field (like the *Year* setting in our example above), then it's valid for just this setting (like the full year of *2013* in our case). If you provide the bottom setting as well, it defines a period (like our *Month* setting, which means from beginning of April to end of September). For example, if a CF is set with the following timeset: "hour { 10-12 } minute { 20-30 }", the CF will be matched within the following time ranges:

- from 10.20am to 10:30am
- from 11.20am to 11:30am
- from 12.20am to 12:30am

Important

the period is a *through* definition, so it covers the full range. If you define an *Hour* definition *8-16*, then this means from *08:00* to *16:59:59* (unless you filter the *Minutes* down to something else).

If you close the *Time Sets* management, you can assign your new time set to the call forwards you're configuring.

7.3.2.3 Configuring Source Sets

Once the *Advanced View* of the call forward definition has been opened, you will need to press the *Manage Source Sets* button to start defining new Source Sets or managing an existing one. The following image shows the Source Set definition dialog:

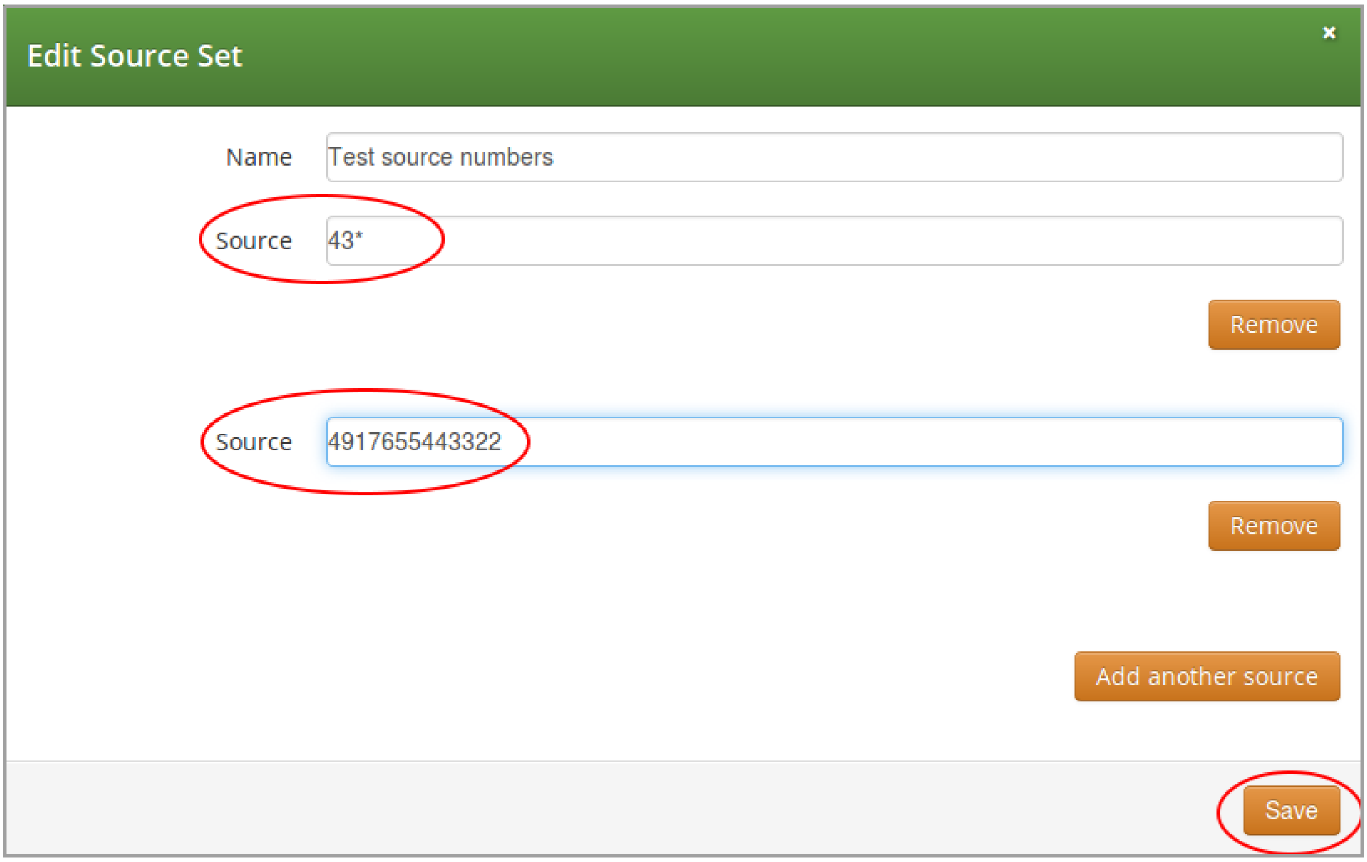


Figure 29: Creating a Call Forward Source Set

You will need to fill in the `Name` field first, then in `Source` field you can enter:

- A simple phone number in E.164 format
- A pattern, in order to define a range of numbers. You can use "*" (matches a string of 0 to any number of characters), "?" (matches any single character), "[abc]" (matches a single character that is part of the explicitly listed set: a, b or c) and "[0-9]" (matches a single character that falls in the range 0 to 9) as wildcards, as usual in shell patterns. Examples:
  - "`431*`" (all numbers from Vienna / Austria)
  - "`49176[0-5]77*`" (German numbers containing fixed digits and a variable digit in 0-5 range in position 6)
  - "`43130120??`" (numbers from Vienna with fixed prefix and 2 digits variable at the end)
- The constant string "anonymous" that indicates a suppressed calling number (CLIR)

You can add more patterns to the Source Set by pressing the *Add another source* button. When you finished adding all patterns, press the *Save* button. You will then see the below depicted list of Source Sets:

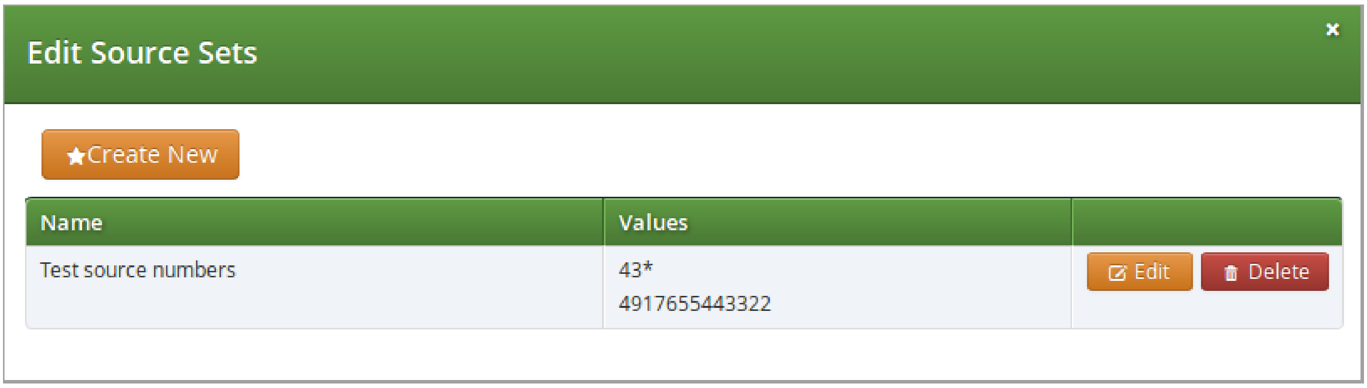


Figure 30: List of Call Forward Source Sets

As a next step you can define a Destination Set as described in Destination Sets Section 7.3.2.1 subchapter. For our example, we have defined the following Destination Set:

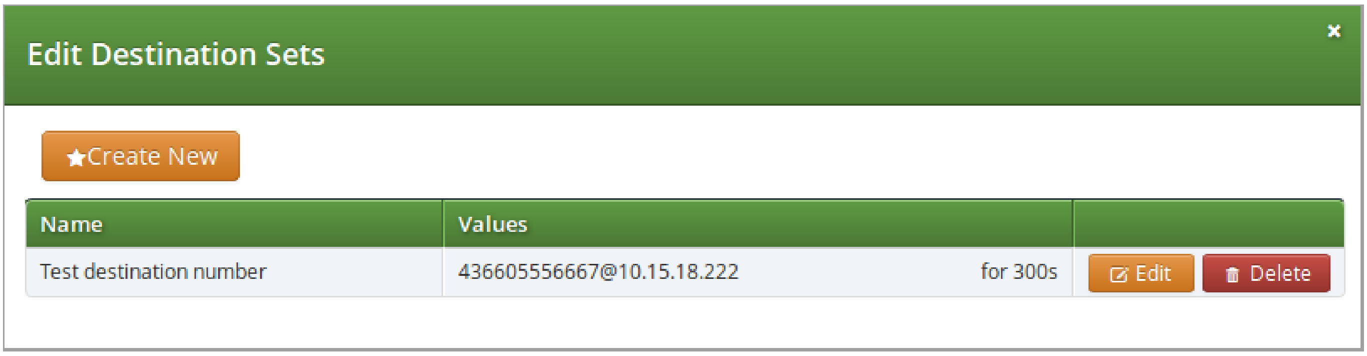


Figure 31: List of Call Forward Destination Sets

A final step of defining the call forward settings is selecting a Destination, a Time and a Source Set, as shown in the image below. *Please note* that there is no specific Time Set selected in our example, that means the call forward rule is valid (as shown) `<always>`.

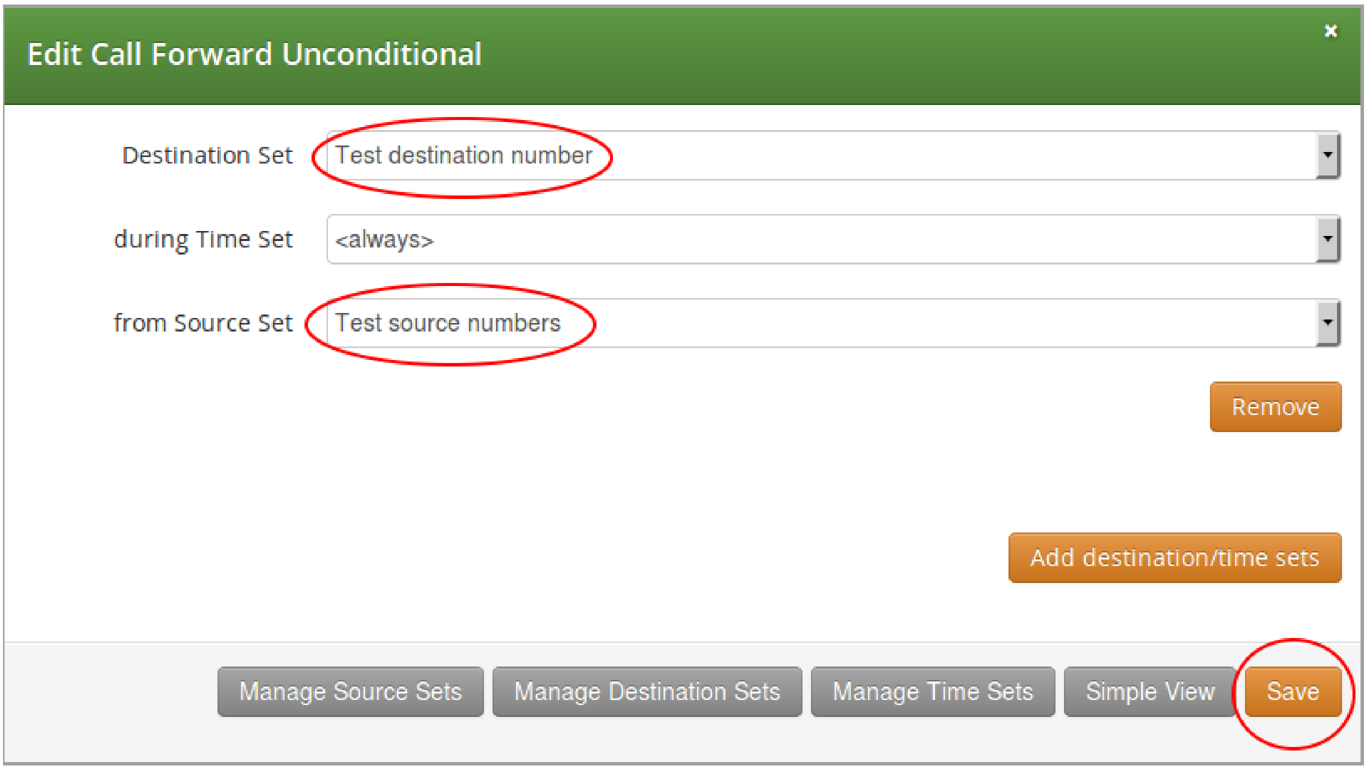


Figure 32: Definition of a Call Forward with Source and Destination Sets

Once all the settings have been defined and the changes are saved, you will see the call forward entry (in our example: *Call Forward Unconditional*), with the names of the selected Destination, Time and Source Sets provided, at *Subscriber Preferences* → *Call Forwards* location on the web interface:

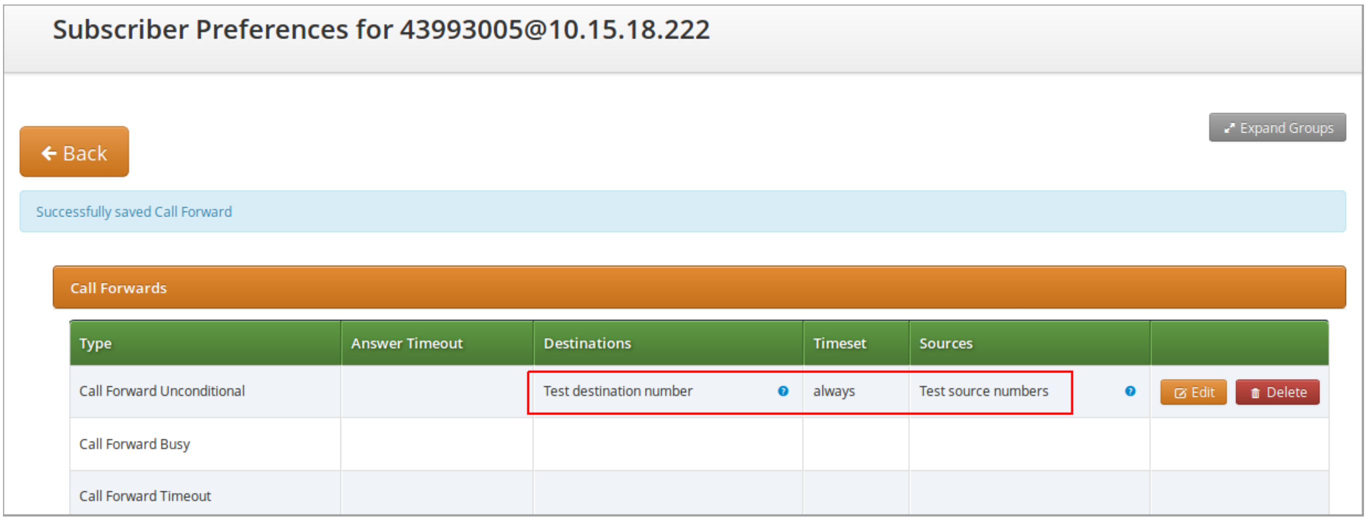


Figure 33: List of Call Forward with Source and Destination Sets

7.4 Local Number Porting

The Sipwise NGCP platform comes with two ways of accomplishing local number porting (LNP):

- one is populating the integrated LNP database with porting data,
- the other is accessing external LNP databases via the Sipwise LNP daemon using the LNP API.

> **Note**
> Accessing external LNP databases is available for PRO and CARRIER products only.

7.4.1 Local LNP Database

The local LNP database provides the possibility to define LNP Carriers (the owners of certain ported numbers or number blocks) and their corresponding LNP Numbers belonging to those carriers. It can be configured on the admin panel in *Settings→Number Porting* or via the API. The LNP configuration can be populated individually or via CSV import/export both on the panel and the API.

7.4.1.1 LNP Carriers

LNP Carriers are defined by an arbitrary *Name* for proper identification (e.g. *British Telecom*) and contain a *Prefix* which can be used as routing prefix in LNP Rewrite Rules and subsequently in Peering Rules to route calls to the proper carriers. The LNP prefix is written to CDRs to identify the selected carrier for post processing and analytics purposes of CDRs. LNP Carrier entries also have an *Authoritative* flag indicating that the numbers in this block belong to the carrier operating the sip:provider CE. This is useful to define your own number blocks, and in case of calls to those numbers reject the calls if the numbers are not assigned to local subscribers (otherwise they would be routed to a peer, which might cause call loops). Finally the *Skip Rewrite* flag skips executing of LNP Rewrite Rules if no number manipulation is desired for an LNP carrier.

7.4.1.2 LNP Numbers

LNP Carriers contain one or more LNP Numbers. Those LNP Numbers are defined by a *Number* entry in E164 format (*<cc><ac><sn>*) used to match a number against the LNP database. Number matching is performed on a longest match, so you can define number blocks without specifying the full subscriber number (e.g. a called party number *431999123* is going to match an entry *431999* in the LNP Numbers).

For an LNP Numbers entry, an optional *Routing Number* can be defined. This is useful to translate e.g. premium 900 or toll-free 800 numbers to actual routing numbers. If a Routing Number is defined, the called party number is implicitly replaced by the Routing Number and the call processing is continued with the latter. For external billing purposes, the optional *Type* tag of a matched LNP number is recorded in CDRs.

An optional *Start Date* and *End Date* allows to schedule porting work-flows up-front by populating the LNP database with certain dates, and the entries are only going to become active with those dates. Empty values for start indicate a start date in the past, while empty values for end indicate an end time in the future during processing of a call, allowing to define infinite date ranges. As intervals can overlap, the LNP number record with a start time closest to the current time is selected.

7.4.1.3 Enabling local LNP support

In order to activate Local LNP during routing, the feature must be activated in *config.yml*. Set *kamailio→proxy→lnp→enabled* to *yes* and *kamailio→proxy→lnp→type* to *local*.

7.4.1.4 LNP Routing Procedure

Calls to non-authoritative Carriers

When a call arrives at the system, the calling and called party numbers are first normalized using the *Inbound Rewrite Rules for Caller* and *Inbound Rewrite Rules for Callee* within the rewrite rule set assigned to the calling party (a local subscriber or a peer).

If the called party number is not assigned to a local subscriber, or if the called party is a local subscriber and has the subscriber/-domain preference *lnp_for_local_sub* set, the LNP lookup logic is engaged, otherwise the call proceeds without LNP lookup. The further steps assume that LNP is engaged.

If the call originated from a peer, and the peer preference *caller_lnp_lookup* is set for this peer, then an LNP lookup is performed using the normalized calling party number. The purpose for that is to find the LNP prefix of the calling peer, which is then stored as *source_lnp_prefix* in the CDR, together with the selected LNP number's *type* tag (*source_lnp_type*). If the LNP lookup does not return a result (e.g. the calling party number is not populated in the local LNP database), but the peer preference *default_lnp_prefix* is set for the originating peer, then the value of this preference is stored in *source_lnp_prefix* of the CDR.

Next, an LNP lookup is performed using the normalized called party number. If no number is found (using a longest match), no further manipulation is performed.

If an LNP number entry is found, and the *Routing Number* is set, the called party number is replaced by the routing number. Also, if the *Authoritative* flag is set in the corresponding LNP Carrier, and the called party number is not assigned to a local subscriber, the call is rejected. This ensures that numbers allocated to the system but not assigned to subscribers are dropped instead of routed to a peer.

> **Important**
> If the system is serving a local subscriber with only the routing number assigned (but not e.g. the premium number mapping to this routing number), the subscriber will not be found and the call will either be rejected if the called party premium number is within an authoritative carrier, or the call will be routed to a peer. This is due to the fact that the subscriber lookup is performed with the dialled number, but not the routing number fetched during LNP. So make sure to assign e.g. the premium number to the local subscriber (optionally in addition to the routing number if necessary using alias numbers) and do not use the LNP routing number mechanism for number mapping to local subscribers.

Next, if the the LNP carrier does not have the *Skip Rewriting* option set, the *LNP Rewrite Rules for Callee* are engaged. The rewrite rule set used is the one assigned to the originating peer or subscriber/domain via the *rewrite_rule_set* preference. The variables available in the match and replace part are, beside the standard variables for rewrite rules:

- `${callee_lnp_prefix}`: The prefix stored in the LNP Carrier
- `${callee_lnp_basenumber}`: The actual number entry causing the match (may be shorter than the called party number due to longest match)

Typically, you would create a rewrite rule to prefix the called party number with the *callee_lnp_prefix* by matching `^([0-9]+)$` and replacing it by `${callee_lnp_prefix}\1`.

Once the LNP processing is completed, the system checks for further preferences to finalize the number manipulation. If the originating local subscriber or peer has the preference *lnp_add_npdi* set, the Request URI user-part is suffixed with `;npdi`. Next, if the preference *lnp_to_rn* is set, the Request URI user-part is suffixed with `;rn=LNP_ROUTING_NUMBER`, where *LNP_ROUTING_NUMBER* is the *Routing Number* stored for the number entry in the LNP database, and the originally called number is kept in place. For example, if *lnp_to_rn* is set and the number *1800123* is called, and this number has a routing number *1555123* in the LNP database, the resulting Request-URI is `sip:1800123;rn=1555123@example.org`.

Finally, the *destination_lnp_prefix* in the CDR table is populated either by the prefix defined in the Carrier of the LNP database if a match was found, or by the *default_lnp_prefix* prefrence of the destination peer or subscriber/domain.

7.4.1.5 Blocking Calls Using LNP Data

The Sipwise NGCP provides means to allow or block calls towards ported numbers that are hosted by particular LNP carriers. Please visit Section 7.2.2.2 in the handbook to learn how this can be achieved.

7.4.1.6 Transit Calls using LNP

If a call originated from a peer and the peer preference *force_outbound_calls_to_peer* is set to *force_nonlocal_lnp* (the *if callee is not local and is ported* selection in the panel), the call is routed back to a peer selected via the peering rules.

This ensures that if a number once belonged to your system and is ported out, but other carriers are still sending calls to you (e.g. selecting you as an anchor network), the affected calls can be routed to the carrier the number got ported to.

7.4.1.7 CSV Format

The LNP database can be exported to CSV, and in the same format imported back to the system. On import, you can decide whether to drop existing data prior to applying the data from the CSV.

The CSV file format contains the fields in the following order:

carrier_name carrier_prefix number routing_number start end authoritative skip_rewrite

Table 3: LNP CSV Format

Name	Description
Carrier Name	The *Name* in the LNP Carriers table (string, e.g. *My Carrier*)
Carrier Prefix	The *Prefix* in the LNP Carriers table (string, e.g. *DD55*)
Number	The *Number* in the LNP Numbers table (E164 number, e.g. *1800666*)
Routing Number	The *Routing Number* in the LNP Numbers table (E164 number or empty, e.g. *1555666*)

Table 3: (continued)

Start	The *Start* in the LNP Numbers table (YYYY-MM-DD or empty, e.g. *2016-01-01*)
End	The *End* in the LNP Numbers table (YYYY-MM-DD or empty, e.g. *2016-12-30*)
Authoritative	The *Authoritative* flag in the LNP Carriers table (0 or 1)
Skip Rewrite	The *Skip Rewrite* flag in the LNP Carriers table (0 or 1)
Type	The *Type* tag in the LNP Numbers table (alphanumeric string, e.g. *mobile*)

7.5 Emergency Mapping

As opposed to the Simple Emergency Number Handling Section 6.7.5.1 solution, the Sipwise NGCP supports an advanced emergency call handling method, called *emergency mapping*. The main idea is: instead of obtaining a statically assigned emergency prefix / suffix from subscriber preferences, NGCP retrieves an emergency routing prefix from a central emergency call routing table, according to the current location of the calling subscriber.

The following figure shows the overview of emergency call processing when using *emergency mapping* feature:

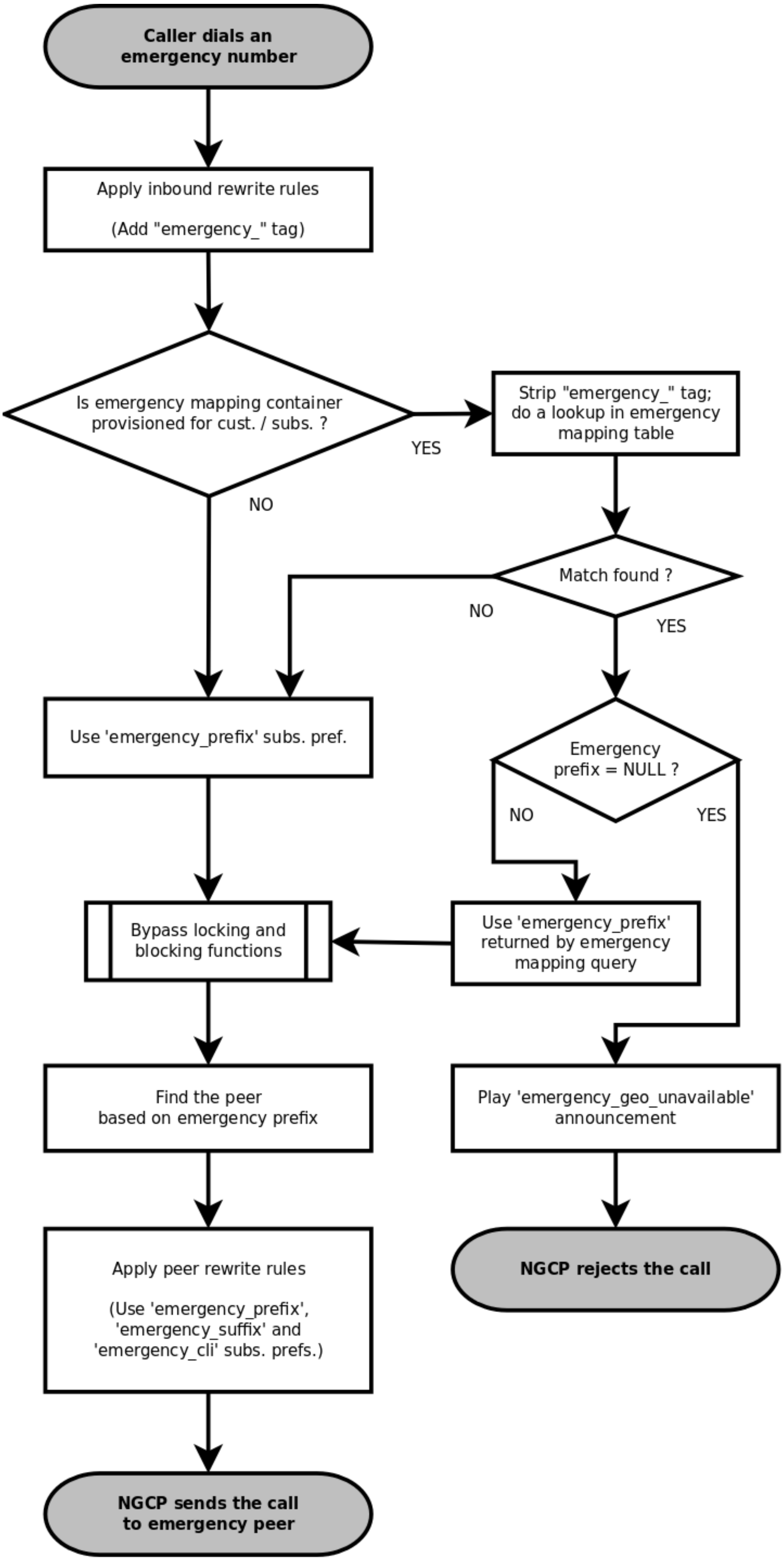


Figure 34: Emergency Call Handling with Mapping

7.5.1 Emergency Mapping Description

Emergency numbers per geographic location are mapped to different routing prefixes not deriveable from an area code or the emergency number itself. This is why a **global emergency mapping table** related to resellers is introduced, allowing to map emergency numbers to their geographically dependent routing numbers.

The geographic location is referenced by a location ID, which has to be populated by a north-bound provisioning system. No towns, areas or similar location data is stored on the NGCP platform. The locations are called *Emergency Containers* on NGCP.

The actual emergency number mapping is done per location (per *Emergency Container*), using the so-called *Emergency Mapping* entries. An *Emergency Mapping* entry assigns a routing prefix, valid only in a geographic area, to a generic emergency number (for example *112* in Europe, *911* in the U.S.A.) or a country specific one (for example *133*).

Note
As of mr4.5 version, the NGCP performs an exact match on the emergency number in the emergency routing table.

Emergency Containers may be assigned to various levels of the client hierarchy within NGCP. The following list shows such levels with each level overriding the settings of the previous one:

1. Customer or Domain
2. Customer Location, which is a territory representing a subset of the customer's subscribers, defined as one or more IP subnets.
3. Subscriber

Note
Please be aware that *Customer Location* is not necessarily identical to the "location" identified through an *Emergency Container*.

Once the emergency routing prefix has been retrieved from the emergency mapping table, call processing continues in the same way as in case of simple emergency call handling.

7.5.2 Emergency Mapping Configuration

The administrative web panel of NGCP provides the configuration interface for emergency mapping. Please navigate to *Settings* → *Emergency Mapping* menu item first, in order to start configuring the mapping.

An *Emergency Container* must be created, before the mapping entries can be defined. Press *Create Emergency Container* to start this. An example of a container is shown here:

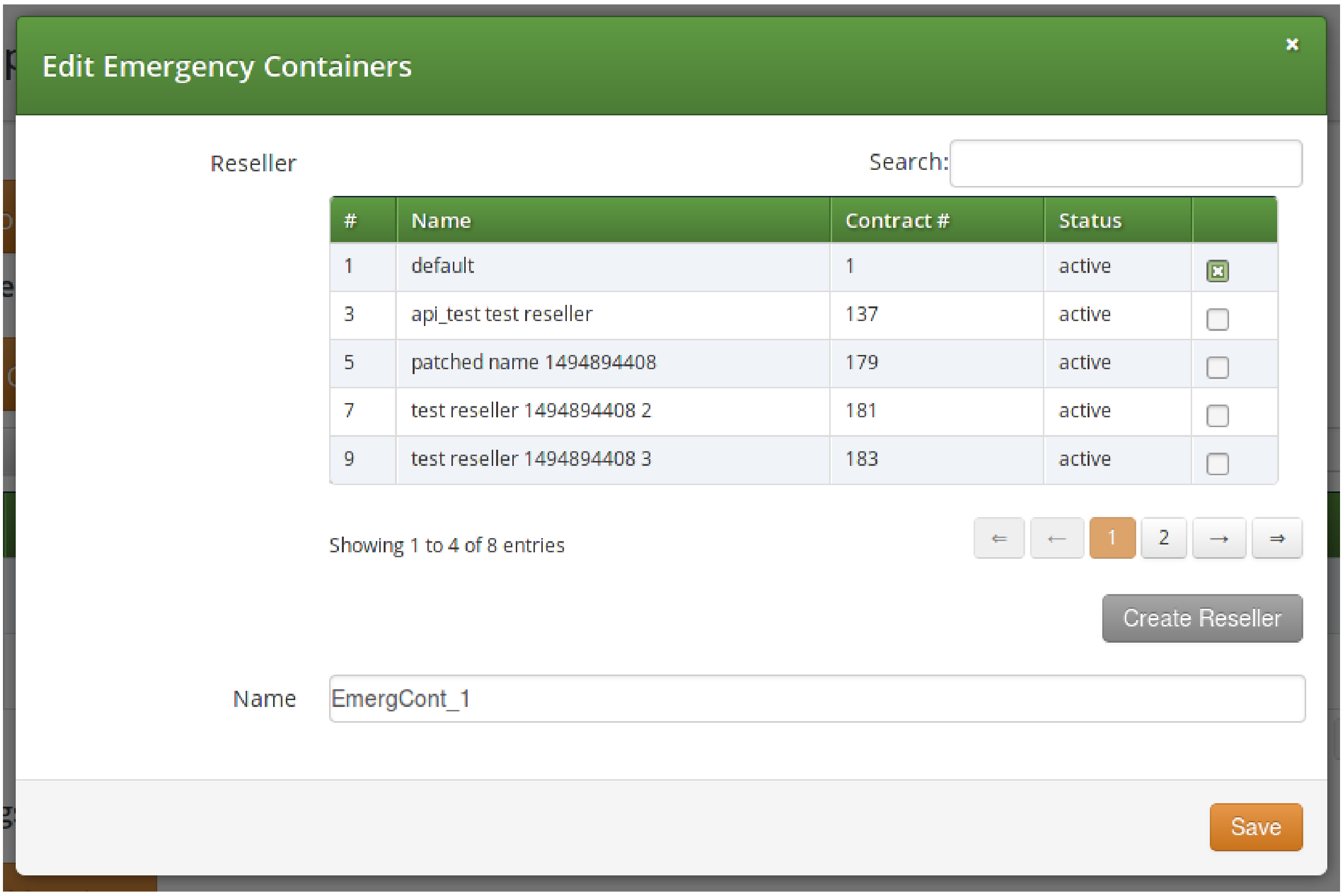


Figure 35: Creating an Emergency Container

You have to select a `Reseller` that this container belongs to, and enter a `Name` for the container, which is an arbitrary text.

Tip

The platform administrator has to create as many containers as the number of different geographic areas (locations) the subscribers are expected to be in.

As the second step of emergency mapping provisioning, the *Emergency Mapping* entries must be created. Press *Create Emergency Mapping* to start this step. An example is shown here:

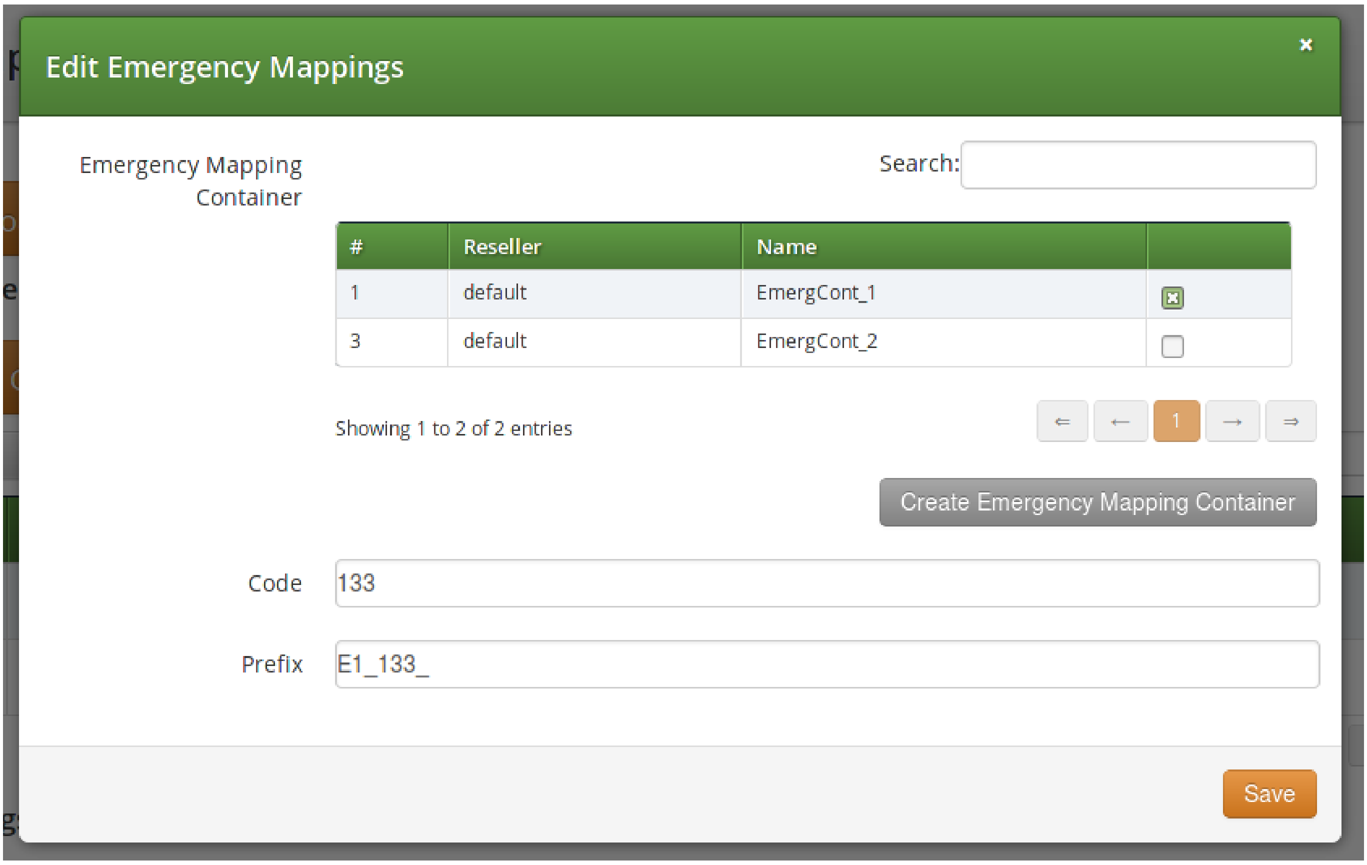


Figure 36: Creating an Emergency Mapping Entry

The following parameters must be set:

- `Container`: select an emergency mapping container (i.e. a location ID)
- `Code`: the emergency number that subscribers will dial
- `Prefix`: the routing prefix that belongs to the particular emergency service within the selected location

Once all the necessary emergency mappings have been defined, the platform administrator will see a list of containers and mapping entries:

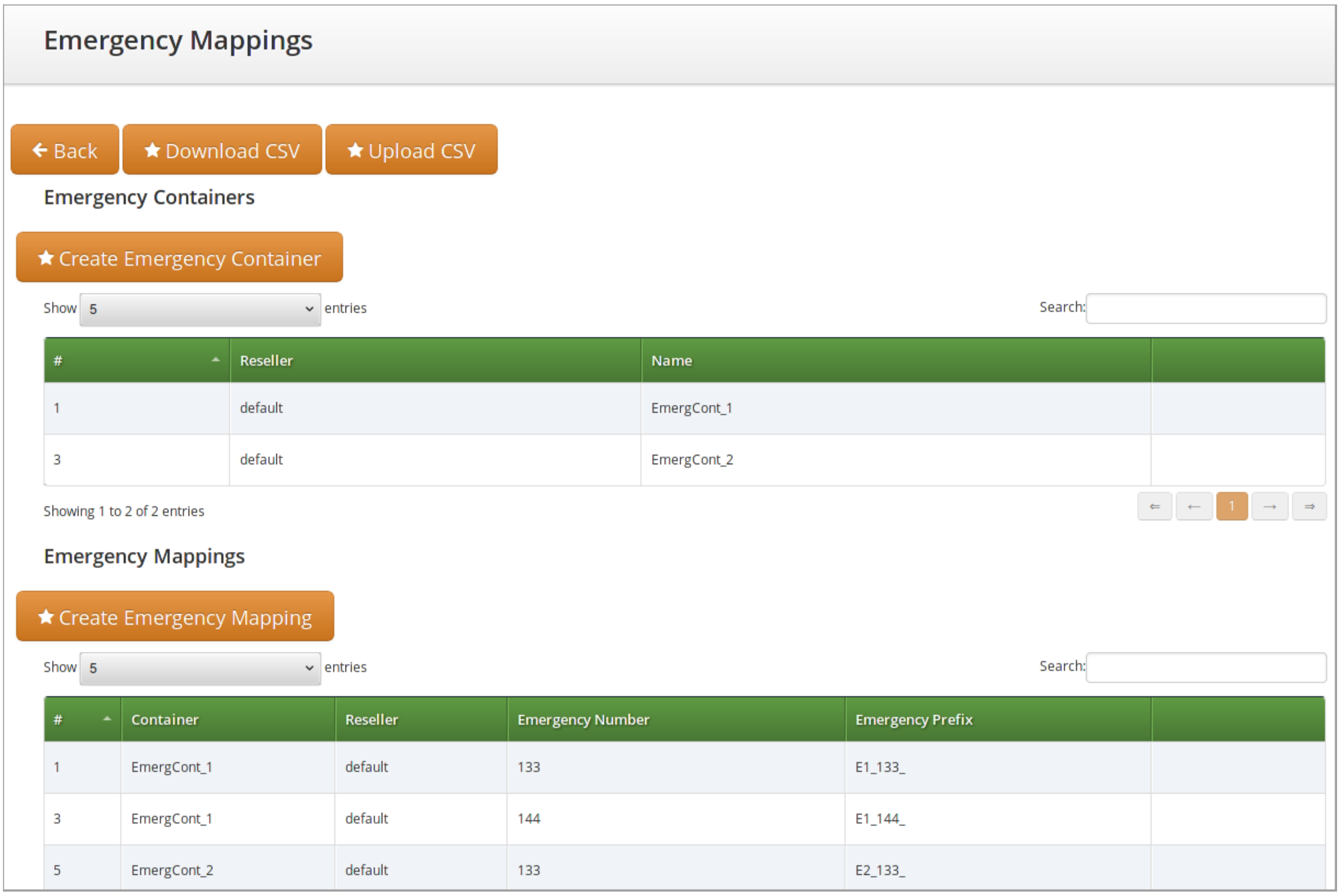


Figure 37: Emergency Mapping List

The emergency number mapping is now defined. As the next step, the platform administrator has to assign the emergency containers to *Customers / Domains / Customer Locations* or *Subscribers*. We'll take an example with a *Customer*: select the customer, then navigate to *Details* → *Preferences* → *Number Manipulations*. In order to assign a container, press the *Edit* button and then select one container from the drop-down list:

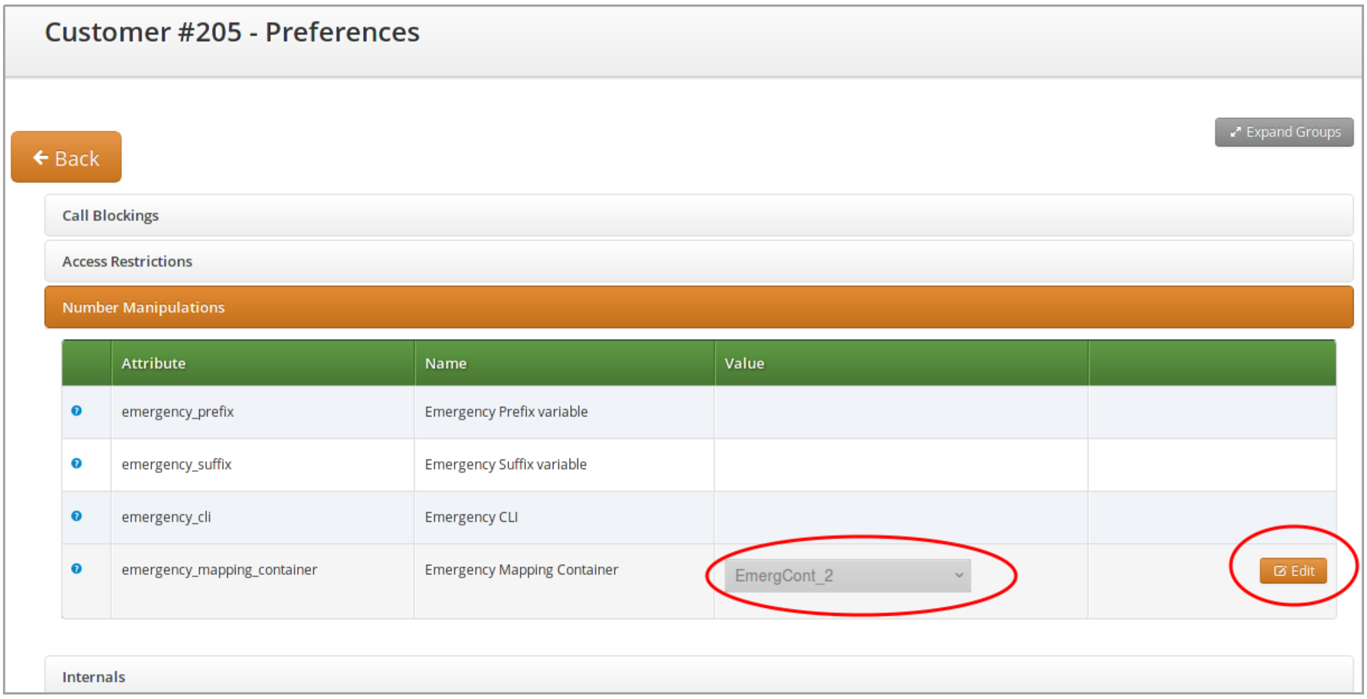


Figure 38: Assigning an Emergency Mapping Container

Rewrite Rules for Emergency Mapping

Once emergency containers and emergency mapping entries are defined, the NGCP administrator has to ensure that the proper number manipulation takes place, before initiating any emergency call towards peers.

Important

Please don't forget to define the rewrite rules for peers — particularly: *Outbound Rewrite Rules for Callee* — as described in Normalize Emergency Calls for Peers Section 6.7.5.3 section of the handbook.

7.5.2.1 Emergency Calls Not Allowed

There is a special case when the dialed number is recognized as an emergency number, but the emergency number is not available for the geographic area the calling party is located in.

In such a case the emergency mapping lookup will return an emergency prefix, but the value of this will be NULL. Therefore the call is rejected and an announcement is played. The announcement is a newly defined sound file referred as `emergency_geo_unavailable`.

It is possible to configure the rejection code and reason in `/etc/ngcp-config/config.yml` file, the parameters are: `kamailio.proxy.early_rejects.emergency_invalid.announce_code` and `kamailio.proxy.early_rejects.emergency_invalid.announce_reason`.

7.5.2.2 Bulk Upload or Download of Emergency Mapping Entries

The Sipwise NGCP offers the possibility to upload / download emergency mapping entries in form of CSV files. This operation is available for each reseller, and is very useful if a reseller has many mapping entries.

Downloading Emergency Mapping List

One has to navigate to *Settings* → *Emergency Mapping* menu and then press the *Download CSV* button to get the list of mapping entries in a CSV file. First the reseller must be selected, then the *Download* button must be pressed. As an example, the entries shown in "Emergency Mapping List" picture above would be written in the file like here below:

```
EmergCont_1,133,E1_133_
EmergCont_1,144,E1_144_
EmergCont_2,133,E2_133_
```

The **CSV file** has a plain text format, each line representing a mapping entry, and contains the following **fields**:

- Container name, as defined in *Emergency Containers*
- Emergency Number
- Emergency Prefix

Uploading Emergency Mapping List

Uploading a CSV file with emergency mapping entries may be started after pressing the *Upload CSV* button. The following data must be provided:

- `Reseller`: selected from the list
- `Upload mapping`: the CSV file must be selected after pressing the *Choose File* button
- `Purge existing`: an option to purge existing emergency mapping entries that belong to the selected reseller, before populating the new mapping data from the file

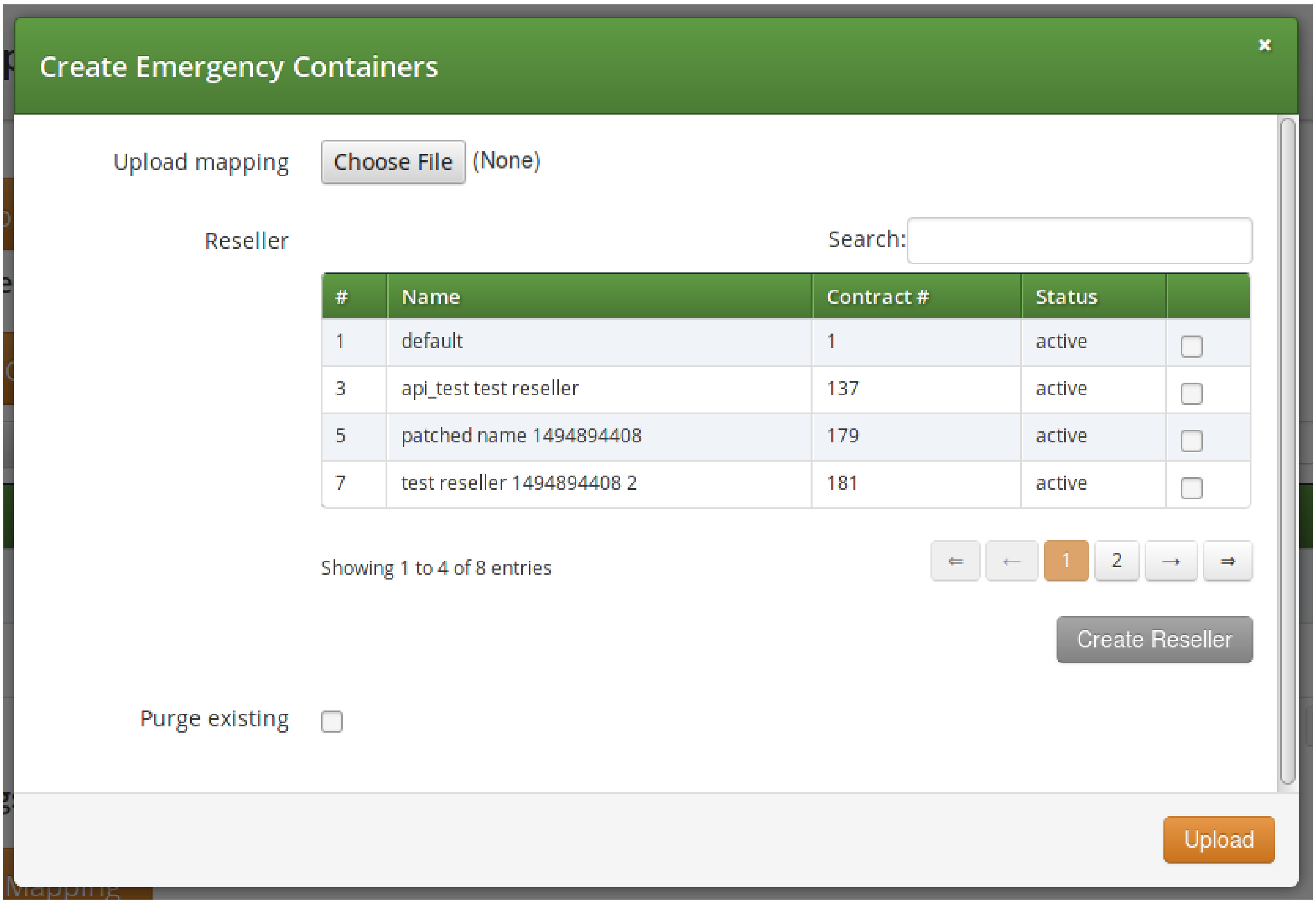


Figure 39: Uploading Emergency Mapping Data

The CSV file for the upload has the same format as the one used for download.

7.6 Emergency Priorization

The NGCP can potentially host *privileged subscribers* that offer emergency or at least prioritized services (civil defence, police etc.). In case of an emergency, the platform has to be free'd from any SIP flows (calls, registrations, presence events etc.) which do not involve those privileged subscribers.

Such an exceptional condition is called ***emergency mode*** and it can be activated for all domains on the system, or only for selected domains.

Once emergency mode is activated, Sipwise NGCP will immediately apply the following restrictions on new SIP requests or existing calls:

- Any SIP requests (calls, registrations etc.) from subscribers within the affected domains, who are not marked as privileged, are rejected.
- Any calls from peers not targeting privileged subscribers are rejected.

- Any active calls which do not have a privileged subscriber involved are terminated.

Calls from non-privileged subscribers to emergency numbers are still allowed.

7.6.1 Call-Flow with Emergency Mode Enabled

Typical call-flows of emergency mode will be shown in this section of the handbook. We have the following assumptions:

- Emergency priorization has been enabled on system-level
- There is a domain for which the emergency mode has been activated
- There is a privileged subscriber in that domain
- A generic peering connection has been configured for non-emergency calls
- A dedicated peering connection has been configured for emergency calls

The examples do not show details of SIP messages, but rather give a high-level overview of the call-flows.

1. A **non-privileged** subscriber makes a call **to another non-privileged subscriber**. Result: the call will be **rejected**.

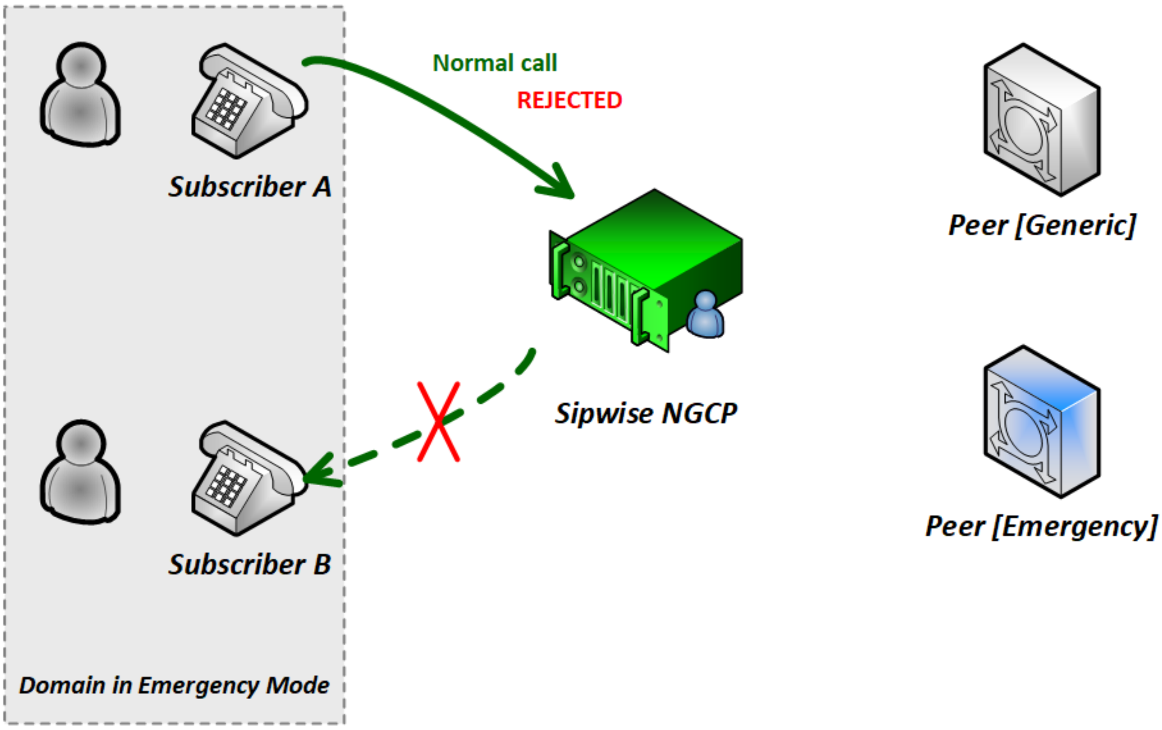


Figure 40: Call-flow in Emergency Mode 1. (Std to Std)

2. A **non-privileged** subscriber makes a call **to an external subscriber (via peer)**. Result: the call will be **rejected**.

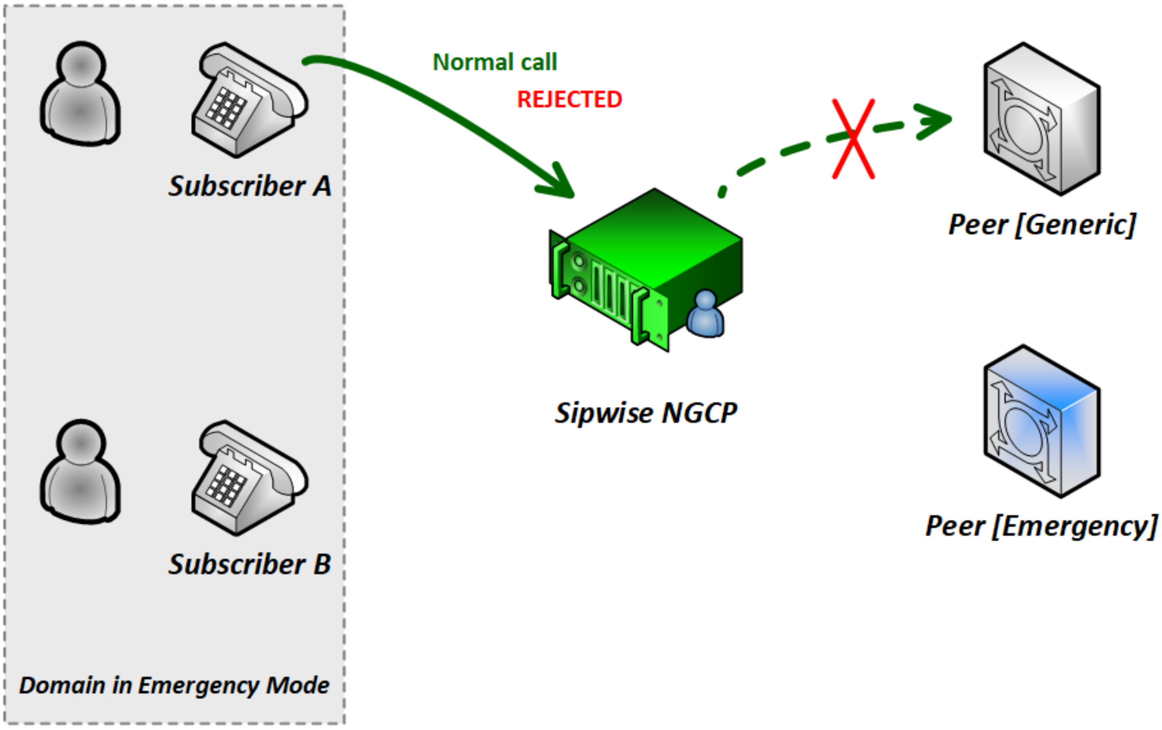


Figure 41: Call-flow in Emergency Mode 2. (Std to Peer)

3. A **non-privileged** subscriber makes a call **to a privileged subscriber**. Result: the call will be **accepted**.

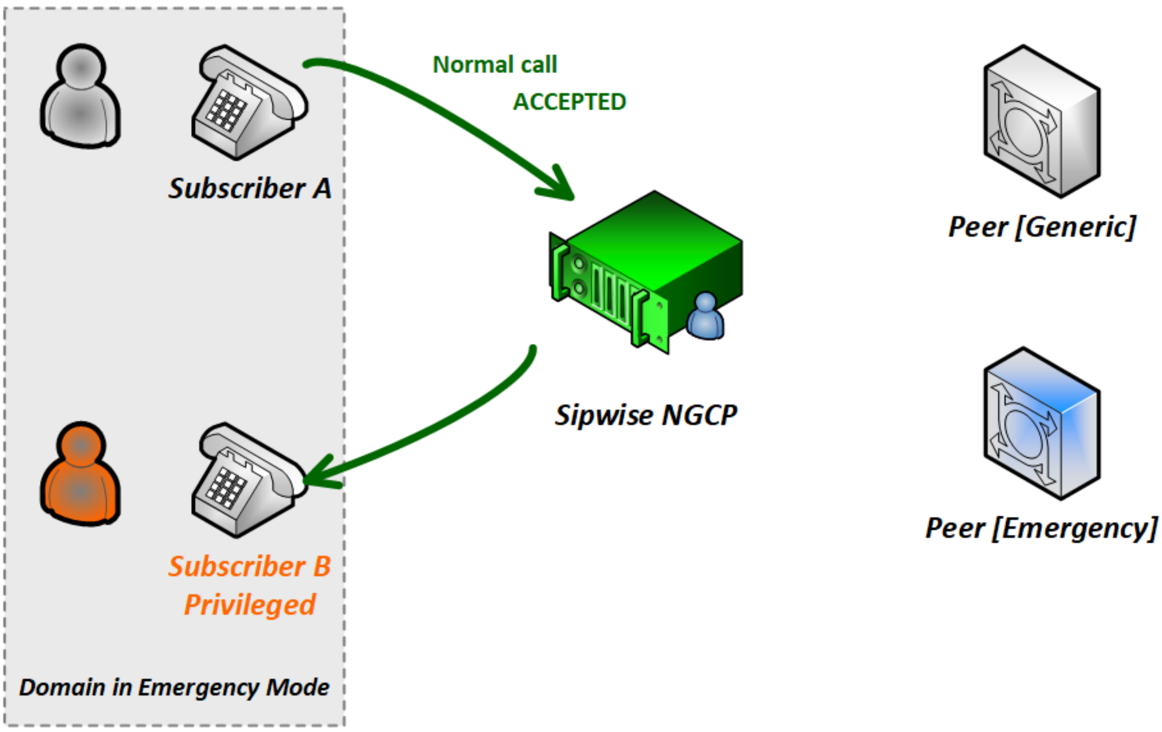


Figure 42: Call-flow in Emergency Mode 3. (Std to Priv)

4. A **non-privileged** subscriber makes a call **to an emergency number**. Result: the call will be **accepted**.

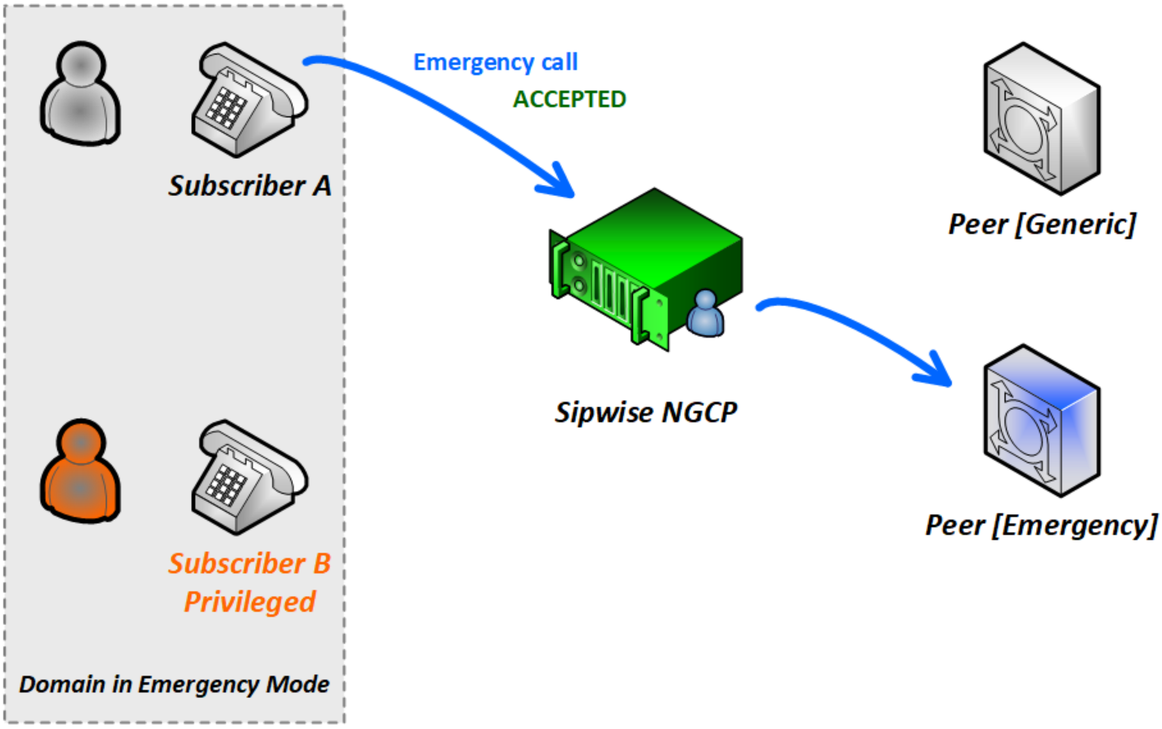


Figure 43: Call-flow in Emergency Mode 4. (Std to Emerg)

5. A **privileged** subscriber makes a call **to a non-privileged subscriber**. Result: the call will be **accepted**.

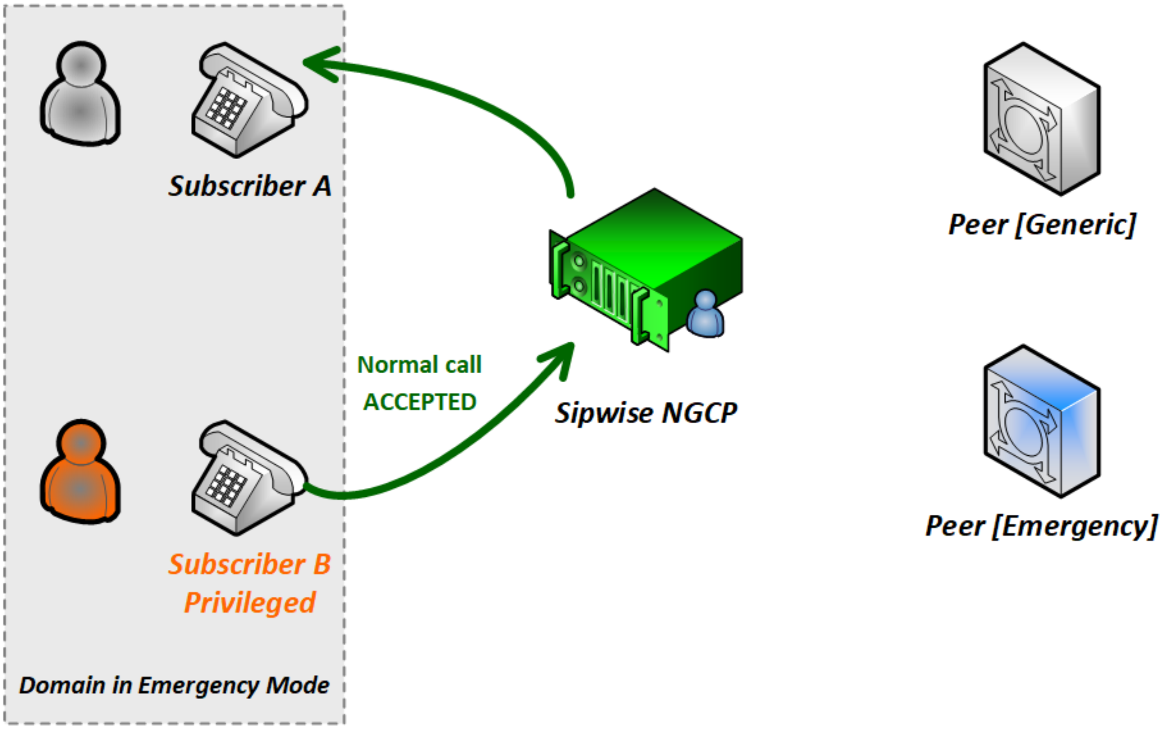


Figure 44: Call-flow in Emergency Mode 5. (Priv to Std)

6. A **privileged** subscriber makes a call **to an external subscriber (via peer)**. Result: the call will be **accepted**.

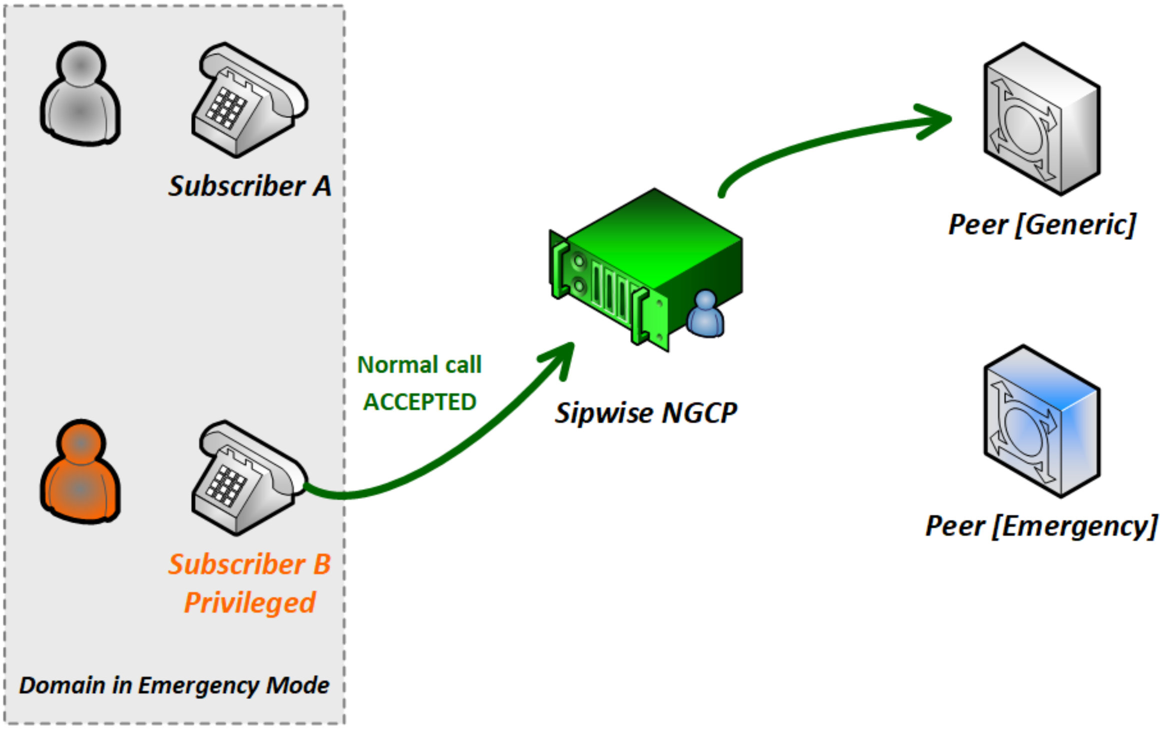


Figure 45: Call-flow in Emergency Mode 6. (Priv To Peer)

7.6.2 Configuration of Emergency Mode

The platform operator has to perform 2 steps of configuration so that the emergency mode can be activated. After the configuration is completed it is necessary to explicitly activate emergency mode, which can be accomplished as described in Section 7.6.3 later.

1. System-level Configuration

The emergency priorization function must be enabled for the whole system, otherwise emergency mode can not be activated. The platform operator has to set `kamailio.proxy.emergency_priorization.enabled` configuration parameter value to `"yes"` in the main configuration file `/etc/ngcp-config/config.yml`. Afterwards changes have to be applied in the usual way, with the command: `ngcpcfg apply "Enabled emergency priorization"`

In order to learn about other parameters related to emergency priorization please refer to Section B.1.14 part of the handbook.

2. Subscriber-level Configuration

The platform operator (or any administrator user) has the capability to declare a subscriber privileged, so that the subscriber can initiate and receive calls when emergency mode has been activated on the NGCP. In order to do that the administrator has to navigate to *Settings* → *Subscribers* → *select the subscriber* → *Details* → *Preferences* → *Internals* → *emergency_priorization* on the **administrative web interface**, and press the *Edit* button.

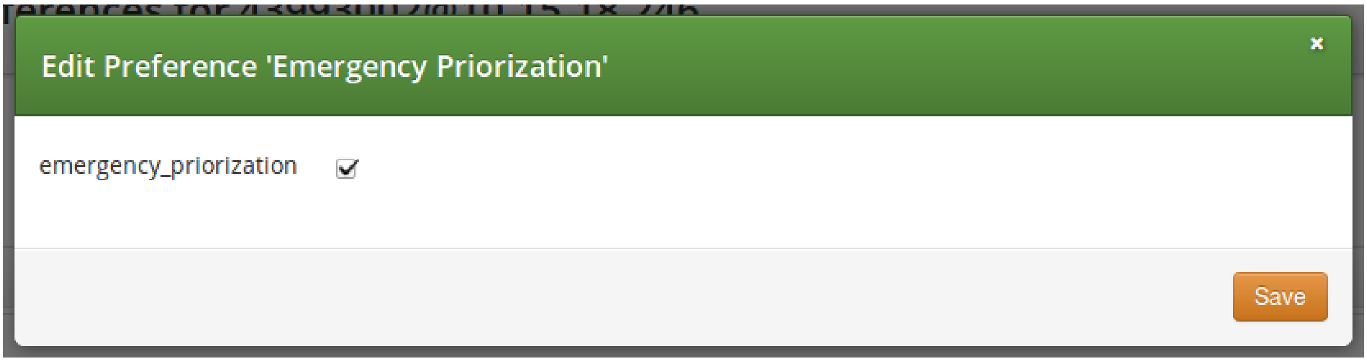


Figure 46: Emergency Priorization of Subscriber

The checkbox *emergency_priorization* has to be ticked and then press the *Save* button.

The same privilege can be added via the **REST API** for a subscriber: a HTTP PUT/PATCH request must be sent on `/api/subscriberpreferences/id` resource and the `emergency_priorization` property must be set to `"true"`.

7.6.3 Activating Emergency Mode

The platform operator can activate emergency mode for a single or multiple domains in 3 different ways:

- via the administrative web interface
- via the REST API
- via a command-line tool

Important
The interruption of ongoing calls is only possible with the command-line tool! Activating emergency mode for domains via the web interface or REST API will only affect upcoming calls.

1. Activate emergency mode via web interface: this way of activation is more appropriate if only a single (or just a few) domain is affected. Please navigate to *Settings* → *Domains* → *select a domain* → *Preferences* → *Internals* → *emergency_mode_enabled* → *Edit*.

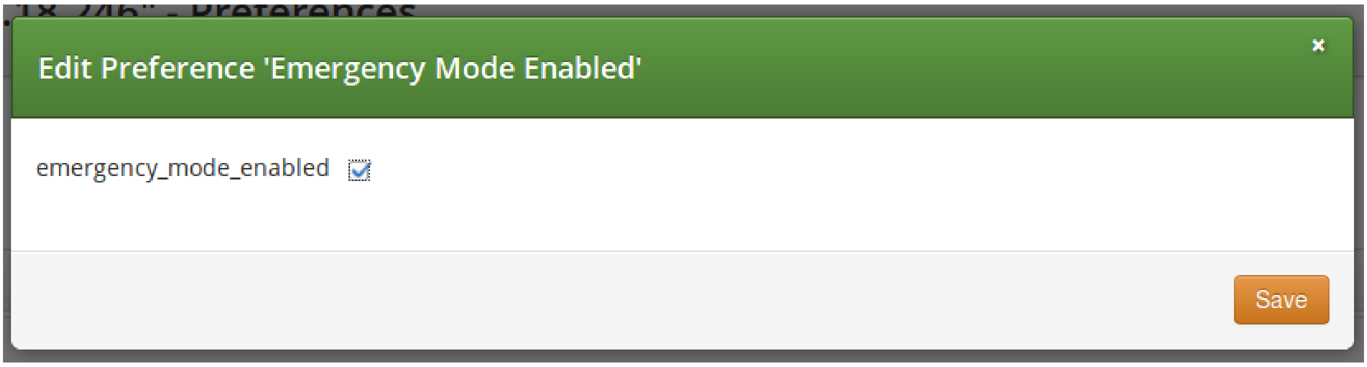


Figure 47: Activate Emergency Mode of Domain

The checkbox *emergency_mode_enabled* has to be ticked and then press the *Save* button.

2. Activate emergency mode via REST API: this way of activation is more appropriate if only a single (or just a few) domain is affected.

For that purpose a HTTP PUT/PATCH request must be sent on `/api/domainpreferences/id` resource and the `emergency_mode_enabled` property must be set to `"true"`.

3. Activate emergency mode using a command-line tool: the Sipwise NGCP provides a built-in script that may be used to enable/disable emergency mode for some particular or all domains.

- Enable emergency mode:

```
> ngcp-emergency-mode enable <all|[domain1 domain2 ...]>
```

- Disable emergency mode:

```
> ngcp-emergency-mode disable <all|[domain1 domain2 ...]>
```

- Query the status of emergency mode:

```
> ngcp-emergency-mode status <all|[domain1 domain2 ...]>
```

7.7 Header Manipulation

7.7.1 Header Filtering

Adding additional SIP headers to the initial INVITEs relayed to the callee (second leg) is possible by modifying the following template file: `/etc/ngcp-config/templates/etc/ngcp-sems/etc/ngcp.sbcprofile.conf.customtt.tt2`. The following section can be changed:

```
header_filter=whitelist
header_list=[%IF kamailio.proxy.debug == "yes"%]P-NGCP-CFGTEST,[%END%]
P-R-Uri,P-D-Uri,P-Preferred-Identity,P-Asserted-Identity,Diversion,Privacy,
```

```
Allow,Supported,Require,RAck,RSeq,Rseq,User-Agent,History-Info,Call-Info
[%IF kamailio.proxy.presence.enable == "yes"%],Event,Expires,
Subscription-State,Accept[%END%][%IF kamailio.proxy.allow_refer_method
== "yes"%],Referred-By,Refer-To,Replaces[%END%]
```

By default the system will remove from the second leg all the SIP headers which are not in the above list. If you want to keep some additional/custom SIP headers, coming from the first leg, into the second leg you just need to add them at the end of the *header_list=* list. After that, as usual, you need to apply the changes. In this way the system will keep your headers in the INVITE sent to the destination subscriber/peer.

Warning
DO NOT TOUCH the list if you don't know what you are doing.

7.7.2 Codec Filtering

Sometimes you may need to filter some audio CODEC from the SDP payload, for example if you want to force your subscribers to do not talk a certain codecs or force them to talk a particular one. To achieve that you just need to change the `/etc/ngcp-config/config.yml`, in the following section:

```
sdp_filter:
      codecs: PCMA,PCMU,telephone-event
      enable: yes
      mode: whitelist
```

In the example above, the system is removing all the audio CODECS from the initial INVITE except G711 alaw,ulaw and telephone-event. In this way the callee will be notified that the caller is able to talk only PCMA. Another example is the `blacklist` mode:

```
sdp_filter:
      codecs: G729,G722
      enable: yes
      mode: blacklist
```

In this way the G729 and G722 will be removed from the SDP payload. In order to apply the changes, as usual, you need to run `ngcpcfg apply` *`Enable CODEC filtering`*.

7.7.3 Enable History and Diversion Headers

It may be useful and mandatory - specially with NGN interconnection - to enable SIP History header and/or Diversion header for outbound requests to a peer or even for on-net calls. In order to do so, you should enable the following preferences in Domain's and Peer's Preferences:

- Domain's Prefererences: *inbound_uprn* = **Forwarder's NPN**

- Peer's Prefererences: *outbound_history_info* = **UPRN**
- Peer's Prefererences: *outbound_diversion* = **UPRN**
- Domain's Prefererences: *outbound_history_info* = **UPRN** (if you want to allow History Header for on-net call as well)
- Domain's Prefererences: *outbound_diversion* = **UPRN** (if you want to allow Diversion Header for on-net call as well)

7.8 SIP Trunking with SIPconnect

7.8.1 User provisioning

For the purpose of external SIP-PBX interconnect with sip:provider CE the platform admin should create a subscriber with multiple aliases representing the numbers and number ranges served by the SIP-PBX.

- Subscriber username - any SIP username that forms an "email-style" SIP URI.
- Subscriber Aliases - numbers in the global E.164 format without leading plus.

To configure the Subscriber, go to *Settings→Subscribers* and click *Details* on the row of your subscriber. There, click on the *Preferences* button on top.

You should look into the *Number Manipulations* and *Access Restrictions* sections in particular, which control the calling and called number presentation.

7.8.2 Inbound calls routing

Enable preference *Number Manipulations→e164_to_ruri* for routing inbound calls to SIP-PBX. This ensures that the Request-URI will comprise a SIP-URI containing the dialed alias-number as user-part, instead of the user-part of the registered AOR (which is normally a static value).

7.8.3 Number manipulations

The following sections describe the recommended configuration for correct call routing and CLI presentation according to the SIPconnect 1.1 recommendation.

7.8.3.1 Rewrite rules

The SIP PBX by default inherits the domain dialplan which usually has rewrite rules applied to normal Class 5 subscribers with inbound rewrite rules normalizing the dialed number to the E.164 standard. If most users of this domain are Class 5 subscribers the dialplan may supply calling number in national format - see Section 6.7. While the SIP-PBX trunk configuration can be sometimes amended it is a good idea in sense of SIPconnect recommendation to send only the global E.164 numbers.

Moreover, in mixed environments with the sip:provider CE Cloud PBX sharing the same domain with SIP trunking (SIP-PBX) customers the subscribers may have different rewrite rules sets assigned to them. The difference is caused by the fact that the dialplan for Cloud PBX is fundamentally different from the dialplan for SIP trunks due to extension dialing, where the Cloud PBX subscribers use the break-out code to dial numbers outside of this PBX.

The SIPconnect compliant numbering plan can be accommodated by assigning Rewrite Rules Set to the SIP-PBX subscriber. Below is a sample Rewrite Rule Set for using the global E.164 numbers with plus required for the calling and called number format compliant to the recommendation.

INBOUND REWRITE RULE FOR CALLER

- Match Pattern: `^(00|\+)([1-9][0-9]+)$`
- Replacement Pattern: `\2`
- Description: `International to E.164`
- Direction: `Inbound`
- Field: `Caller`

INBOUND REWRITE RULE FOR CALLEE

- Match Pattern: `^(00|\+)([1-9][0-9]+)$`
- Replacement Pattern: `\2`
- Description: `International to E.164`
- Direction: `Inbound`
- Field: `Callee`

OUTBOUND REWRITE RULE FOR CALLER

- Match Pattern: `^([1-9][0-9]+)$`
- Replacement Pattern: `+\1`
- Description: `For the calls to SIP-PBX add plus to E.164`
- Direction: `Outbound`
- Field: `Caller`

OUTBOUND REWRITE RULE FOR CALLEE

- Match Pattern: `^([1-9][0-9]+)$`
- Replacement Pattern: `+\1`
- Description: `For the calls to SIP-PBX add plus to E.164`
- Direction: `Outbound`

• Field: `Callee`

Assign the aforementioned Rewrite Rule Set to the SIP-PBX subscribers.

Warning
Outbound Rewrite Rules for Callee shall NOT be applied to the calls to normal SIP UAs like IP phones since the number with plus does not correspond to their SIP username.

7.8.3.2 User parameter

The following configuration is needed for your platform to populate the From and To headers and Request-URI of the INVITE request with "user=phone" parameter as per RFC 3261 Section 19.1.1 (if the user part of the URI contains telephone number formatted as a telephone-subscriber).

• Domain's Prefererences: *outbound_from_user_is_phone* = Y

• Domain's Prefererences: *outbound_to_user_is_phone* = Y

7.8.3.3 Forwarding number

The following is our common configuration that covers the calling number presentation in a variety of use-cases, including the incoming calls, on-net calls and Call Forward by the platform:

• Domain's Preferences: *inbound_uprn* = **Forwarder's NPN**

• Domain's Preferences: *outbound_from_user* = **UPRN (if set) or User-Provided Number**

• Domain's Preferences: *outbound_pai_user* = **UPRN (if set) or Network-Provided Number**

• Domain's Preferences: *outbound_history_info* = **UPRN** (if the called user expects History-Info header)

• Domain's Preferences: *outbound_diversion* = **UPRN** (if the called user expects Diversion header)

• Domain's Preferences: *outbound_to_user* = **Original (Forwarding) called user** if the callee expects the number of the subscriber forwarding the call, otherwise leave default.

The above parameters can be tuned to operator specifics as required. You can of course override these settings in the Subscriber Preferences if particular subscribers need special settings.

Tip
On outgoing call from SIP-PBX subscriber the Network-Provided Number (NPN) is set to the *cli* preference prefilled with main E.164 number. In order to have the full alias number as NPN on outgoing call set preference *extension_in_npn* = Y.

Externally forwarded call If the call forward takes place inside the SIP-PBX it can use one of the following specification for signaling the diversion number to the platform:

- using **Diversion** method (RFC 5806): configure Subscriber's Prefererences: *inbound_uprn* = **Forwarder's NPN / Received Diversion**
- using **History-Info** method (RFC 7044): NGCP platform extends the History-Info header received from the PBX by adding another level of indexing according to the specification RFC 7044.

7.8.3.4 Allowed CLIs

- For correct calling number presentation on outgoing calls, you should include the pattern matching all the alias numbers of SIP-PBX or each individual alias number under the *allowed_clis* preference.
- If the signalling calling number (usually taken from From user-part, see *inbound_upn* preferences) does not match the *allowed_clis* pattern, the *user_cli* or *cli* preference (Network-Provided Number) will be used for calling number presentation.

7.8.4 Registration

SIP-PBX can use either Static or Registration Mode. While SIPconnect 1.1 continues to require TLS support at MUST strength, one should note that using TLS for signaling does not require the use of the SIPS URI scheme. SIPS URI scheme is obsolete for this purpose.

Static Mode While SIPconnect 1.1 allows the use of Static mode, this poses additional maintenance overhead on the operator. The administrator should create a static registration for the SIP-PBX: go to Susbcribers, *Details→Registered Devices→Create Permanent Registration* and put address of the SIP-PBX in the following format: sip:username@ipaddress:5060 where username=username portion of SIP URI and ipaddress = IP address of the device.

Registration Mode It is recommended to use the Registration mode with SIP credentials defined for the SIP-PBX subscriber.

Important
The use of RFC 6140 style "bulk number registration" is discouraged. The SIP-PBX should register one AOR with email-style SIP URI. The sip:provider CE will take care of routing the aliases to the AOR with *e164_to_ruri* preference.

7.8.4.1 Trusted Sources

If a SIP-PBX cannot perform the digest authentication, you can authenticate it by its source IP address in sip:provider CE. To configure the IP-based authentication, go to the subscriber's preferences (*Details→ Preferences→Trusted Sources*) and specify the IP address of the SIP-PBX in the *Source IP* field.

To authenticate multiple subscribers from the same IP address, use the*From* field to distinguish these subscribers.

When this feature is configured for a subscriber, the sip:provider CE authenticates all calls that arrive from the specified IP address without challenging them.

> **Important**
>
> If the same IP address and the FROM field are mistakenly specified as trusted for different subscribers, the sip:provider CE will not know which subscriber to charge for the call and will randomly select one.

7.9 Trusted Subscribers

In some cases, when you have a device that cannot authenticate itself against sip:provider CE, you may need to create a *Trusted Subscriber*. Trusted Subscribers use IP-based authentication and they have a Permanent SIP Registration URI in order to receive messages from sip:provider CE.

In order to make a regular subscriber trusted, perform the following extra steps: * Create a permanent registration via (*Subscribers→Details→ Registered Devices→Create Permanent Registration*) * Add the IP address of the device as Trusted Source in your subscriber's preferences (*Details→Preferences→Trusted Sources*).

This way, all SIP messages coming from the device IP will be considered trusted (and get authenticated just by the source IP). All the SIP messages forwarded to the devices will be sent to the SIP URI specified in the subscriber's permanent registration.

7.10 Peer Probing

The basic way of selecting the appropriate peering server, where an outbound call can be routed to, has already been described in Section 6.6.2.3 of the handbook.

This chapter provides information on the *peer probing* feature of NGCP that is available since mr5.4.1 release.

7.10.1 Introduction to Peer Probing Feature

The Sipwise NGCP provides web admin panel and API capabilities to configure peering servers in order to terminate calls to non-local subscribers. Those peering servers may become *temporarily unavailable* due to overloading or networking issues. The NGCP will fail over to another peering server (matching the corresponding peering rules) after a timeout configured at system level (see `sems.sbc.outbound_timeout` configuration parameter; 6 sec by default), if no provisional response (a response with a code in the range of 100 to 199) is received for the outbound INVITE request.

Even if this timer is set much lower, like 3 sec, the call setup time is increased significantly. This is even more true if multiple peering servers fail at the same time, which will sum up the individual timeouts, finally *causing call setup times reach the order of tens of seconds*.

To optimize the call setup time in such scenarios, a new feature is implemented to *continuously probe peering servers* via SIP messages, and mark them as unavailable on timeout or when receiving unexpected response codes. Appropriate SIP response codes from the peering servers will mark them as available again.

Peering servers *marked as unavailable* are then *skipped during call routing* in the peering selection process, which significantly shortens the call setup times if peering servers fail.

7.10.2 Configuration of Peer Probing

The system administrator has to configure the peer probing feature in 2 steps:

1. System-level configuration enables the peer probing feature in general on the NGCP and determines the operational parameters, such as timeouts, the SIP method used for probing requests, etc.
2. Peering server configuration will add / remove a peering server to the list of probed endpoints.

7.10.2.1 System-level Configuration

The parameters of peer probing are found in the main system configuration file `/etc/ngcp-config/config.yml`. You can see the complete list of configuration parameters in Section B.1.14 of the handbook, while the most significant ones are discussed here.

Enabling peer probing system-wide happens through the `kamailio.proxy.peer_probe.enable` parameter. If it is set to *yes* (which is the default value) then NGCP will consider probing of individual peering servers based on their settings.

Timeout of a single probing request can be defined through `kamailio.proxy.peer_probe.timeout` parameter. This is a value interpreted as seconds while NGCP will wait for a SIP response from the peering server. Default is 5 seconds.

The **probing interval** can be set through the `kamailio.proxy.peer_probe.interval` parameter. This is the time period in seconds that determines how often a probing request is sent to the peering servers. Default is 10 seconds.

The **SIP method** used for probing requests can be defined through `kamailio.proxy.peer_probe.method` parameter. Allowed values are: OPTIONS (default) and INFO.

Tip

The system administrator, in most of the cases, will not need to modify the default configuration values other than that of timeout and interval.

If no available peering server is found, the call is rejected with the response code and reason configured in `kamailio.proxy.early_rejects.peering_unavailable.announce_code` and `kamailio.proxy.early_rejects.peering_unavailable.announce_reason`. If a sound file is configured within the *system sound set* assigned to the calling party, an announcement is played as early media before the rejection.

7.10.2.2 Individual Peering Server Configuration

When the peer probing feature is enabled on system-level, it is possible to add each individual peering server to the list of probed endpoints. You can change the probed status of a server in two ways:

Enable probing of a peering server via the admin web interface

1. Open the properties panel of a peering server: *Peerings* → *select a peering group* → *Details* → *select a peering server* → *Edit*

2. Tick the checkbox *Enable Probing*

3. *Save* changes

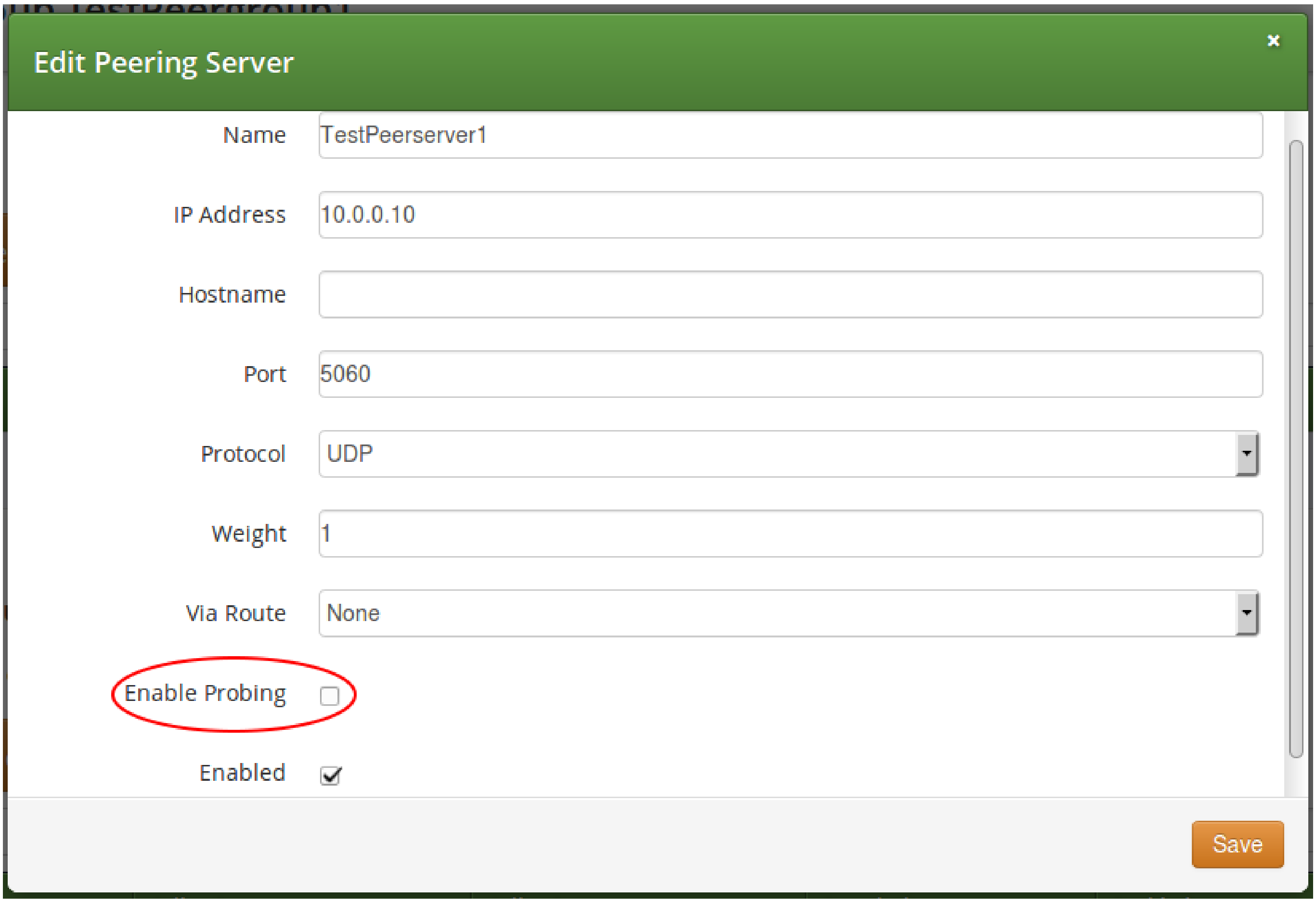


Figure 48: Enable Probing of Peering Server

Enable probing of a peering server via the REST API

- when you *create a new peering server* you will use an HTTP *POST* request and the target URL: https://<IP_of_NGCP>:1443/-api/peeringservers
- when you *update an existing peering server* you will use an HTTP *PUT* or *PATCH* request and the target URL: https://<IP_of_NGCP>:14 api/peeringservers/id

In all cases you have to set the *probe* property to `true` in order to enable probing, and to `false` in order to disable probing. Default value is `false` and this property may be omitted in a create/update request, which ensures backward compatibility of the `/api/peeringservers` API resource.

7.10.3 Monitoring of Peer Probing

Peering server states, such as "reachable" / "unreachable", are continuously stored in a time-series database (InfluxDB type) by NGCP Proxy nodes. It is possible to **graphically represent the state of peering servers** on NGCP's admin web interface, just like other system variables (like CPU and memory usage, number of registered subscribers, etc.). However this is not available by default and must be configured by Sipwise.

State changes of peering servers are also reported by means of **SNMP traps**. Each time the reachable state of one of the monitored peering servers changes, NGCP will send an SNMP trap, raising or clearing the alarm.

The Sipwise MIB is extended by a table of peers per proxy, containing the peer ID and the peer name, along with the peer probe status. An external monitoring system can **poll the peers table via SNMP** to gather the peer status from each proxy's point of view.

The peer status can be obtained through the following route / OID:

```
...enterprises.sipwise.ngcp.ngcpObjects.ngcpMonitor.ngcpMonitorPeering.psTable.psEntry. ↩
    psPeerStatus

.1.3.6.1.4.1.34274.1.1.2.40.2.1.7
```

Value of *psPeerStatus* can be:

- 0: unknown
- 1: administratively down
- 2: administratively up
- 3: probed, pending
- 4: probed, down
- 5: probed, up

7.10.4 Further Details for Advanced Users

Tip

This subchapter of the handbook is targeted on advanced system operators and Sipwise engineers and is not necessary to read in order to properly manage peer probing feature of NGCP.

7.10.4.1 Behaviour of Kamailio Proxy Instances

Each *kamailio-proxy* instance on the proxy nodes performs the probing individually for performance reasons. Each proxy holds its result in its cache to avoid central storage and replication of the probing results. Each proxy will send an SNMP trap if it detects a state change for a peering server, because proxies might be geographically distributed along with their load-balancers and can therefore experience different probing results.

Each peering server is cross-checked against the hash table filled during outbound probing requests and is skipped by call routing logic, if a match is found.

On start or restart of the *kamailio-proxy* instance, the probing will start after the first interval, and NOT immediately after start. In the first probing interval the proxy will always try to send call traffic to peering servers until the first probing round is finished, and will only then start to skip unavailable peering servers.

7.10.4.2 Changes to Kamailio Proxy Configuration

A new configuration template: `/etc/ngcp/config/templates/etc/kamailio/proxy/probe.cfg.tt2` is introduced to handle outbound probing requests.

7.10.4.3 Database Changes

A new DB column: `provisioning.voip_peer_hosts.probe` with type TINYINT(1) (boolean) is added to the DB schema.

A peer status change will populate the `kamailio.dispatcher` table, inserting the SIP URI in format `sip:$ip:$port;transport=$transport` in dispatcher group 100, which defines the probing group for peering servers.

Also the `kamailio.dispatcher.attrs` column is populated with a parameter `peerid=$id`. This ID is used during probing to load the peer preferences: `outbound_socket` and `lbrtp_set`, that are required to properly route the probing request.

7.11 Voicemail System

7.11.1 Accessing the IVR Menu

For a subscriber to manage his voicebox via IVR, there are two ways to access the voicebox. One is to call the URI `voicebox@yourdomain` from the subscriber itself, allowing password-less access to the IVR, as the authentication is already done on SIP level. The second is to call the URI `voiceboxpass@yourdomain` from any number, causing the system to prompt for a mailbox and the PIN. The PIN can be set in the *Voicemail and Voicebox* section of the *Subscriber Preferences*.

7.11.1.1 Mapping numbers and codes to IVR access

Since access might need to be provided from external networks like PSTN/Mobile, and since certain SIP phones do not support calling alphanumeric numbers to dial `voicebox`, you can map any number to the voicebox URIs using rewrite rules.

To do so, you can provision a match pattern e.g. `^(00|\+)12345$` with a replace pattern `voicebox` or `voiceboxpass` to map a number to either password-less or password-based IVR access respectively. Create a new rewrite rule with the `Inbound` direction and the `Callee` field in the corresponging rewrite rule set.

For inbound calls from external networks, assign this rewrite rule set to the corresponding incoming peer. If you also need to map numbers for on-net calls, assign the rewrite rule set to subscribers or the whole SIP domain.

7.11.1.2 External IVR access

When reaching `voiceboxpass`, the subscriber is prompted for her mailbox number and a password. All numbers assigned to a subscriber are valid input (primary number and any alias number). By default, the required format is in E.164, so the subscriber needs to enter the full number including country code, for example `4912345` if she got assigned a German number.

You can globally configure a rewrite rule in `config.yml` using `asterisk.voicemail.normalize_match` and `asterisk.voicemail.normalize_replace`, allowing you to customize the format a subscriber can enter, e.g. having `^0([1-9][0-9]+)$` as match part and `49$1` as replace part to accept German national format.

7.11.2 IVR Menu Structure

The following list shows you how the voicebox menu is structured.

- *1* Read voicemail messages
  - *3* Advanced options
    - *3* To Hear messages Envelope
    - *** Return to the main menu
  - *4* Play previous message
  - *5* Repeat current message
  - *6* Play next message
  - *7* Delete current message
  - *9* Save message in a folder
    - *0* Save in new Messages
    - *1* Save in old Messages
    - *2* Save in Work Messages
    - *3* Save in Family Messages
    - *4* Save in Friends Messages
    - *#* Return to the main menu
- *2* Change folders
  - *0* Switch to new Messages
  - *1* Switch to old Messages
  - *2* Switch to Work Messages
  - *3* Switch to Family Messages
  - *4* Switch to Friends Messages
  - *#* Get Back
- *3* Advanced Options

- *** To return to the main menu

- *0* Mailbox options
  - *1* Record your unavailable message
    - *1* accept it
    - *2* Listen to it
    - *3* Rerecord it
  - *2* Record your busy message
    - *1* accept it
    - *2* Listen to it
    - *3* Rerecord it
  - *3* Record your name
    - *1* accept it
    - *2* Listen to it
    - *3* Rerecord it
  - *4* Record your temporary greetings
    - *1* accept it / or re-record if one already exist
    - *2* Listen to it / or delete if one already exist
    - *3* Rerecord it
  - *5* Change your password
  - *** To return to the main menu
- *** Help
- *#* Exit

7.11.3 Type Of Messages

A message/greeting is a short message that plays before the caller is allowed to record a message. The message is intended to let the caller know that you are not able to answer their call. It can also be used to convey other information like when you will be available, other methods to contact you, or other options that the caller can use to receive assistance.

The IVR menu has three types of greetings.

7.11.3.1 Unavailable Message

The standard voice mail greeting is the "unavailable" greeting. This is used if you don't answer the phone and so the call is directed to your voice mailbox.

- You can record a custom unavailable greeting.

- If you have not recorded your unavailable greeting but have recorded your name, the system will play a generic message like: "`Recorded name` is unavailable."
- If you have not recorded your unavailable greeting, the phone system will play a generic message like: "`Digits-of-number-dialed` is unavailable".

7.11.3.2 Busy Message

If you wish, you can record a custom greeting used when someone calls you and you are currently on the phone. This is called your "Busy" greeting.

- You can record a custom busy greeting.
- If you have not recorded your busy greeting but have recorded your name, the phone system will play a generic message: "`Recorded name` is busy."
- If you have not recorded your busy greeting and have not recorded your name (see below), the phone system will play a generic message: "`Digits-of-number-dialed` is busy."

7.11.3.3 Temporary Greeting

You can also record a temporary greeting. If it exists, a temporary greeting will always be played instead of your "busy" or "unavailable" greetings. This could be used, for example, if you are going on vacation or will be out of the office for a while and want to inform people not to expect a return call anytime soon. Using a temporary greeting avoids having to change your normal unavailable greeting when you leave and when you come back.

7.11.4 Folders

The Voicemail system allows you to save and organize your messages into folders. There can be up to ten folders.

7.11.4.1 The Default Folder List

- 0 - New Messages
- 1 - Old Messages
- 2 - Work Messages
- 3 - Family Messages
- 4 - Friends Messages

When a caller leaves a message for you,the system will put the message into the "New Messages" folder. If you listen to the message, but do not delete the message or save the message to a different folder, it will automatically move the message to the "Old Messages" folder. When you first log into your mailbox, the Voicemail System will make the "New Messages" folder the current folder if you have any new messages. If you do not have any new messages the it will make the "Old Messages" folder the current folder.

7.11.5 Voicemail Languages Configuration

To add a new language or to change the pronunciation for an existing one, ensure that **mode=new** is defined in */etc/ngcp-config/templates/etc/asterisk/say.conf.tt2*. Adjust the configuration in the same file using the manual in the beginning. Then, as usual, make the new configuration active.

7.11.6 Flowcharts with Voice Prompts

This section shows flowcharts of calls to the voicemail system. Flowcharts contain the name of prompts as they are identified among *Asterisk* voice prompts.

7.11.6.1 Listening to New Messages

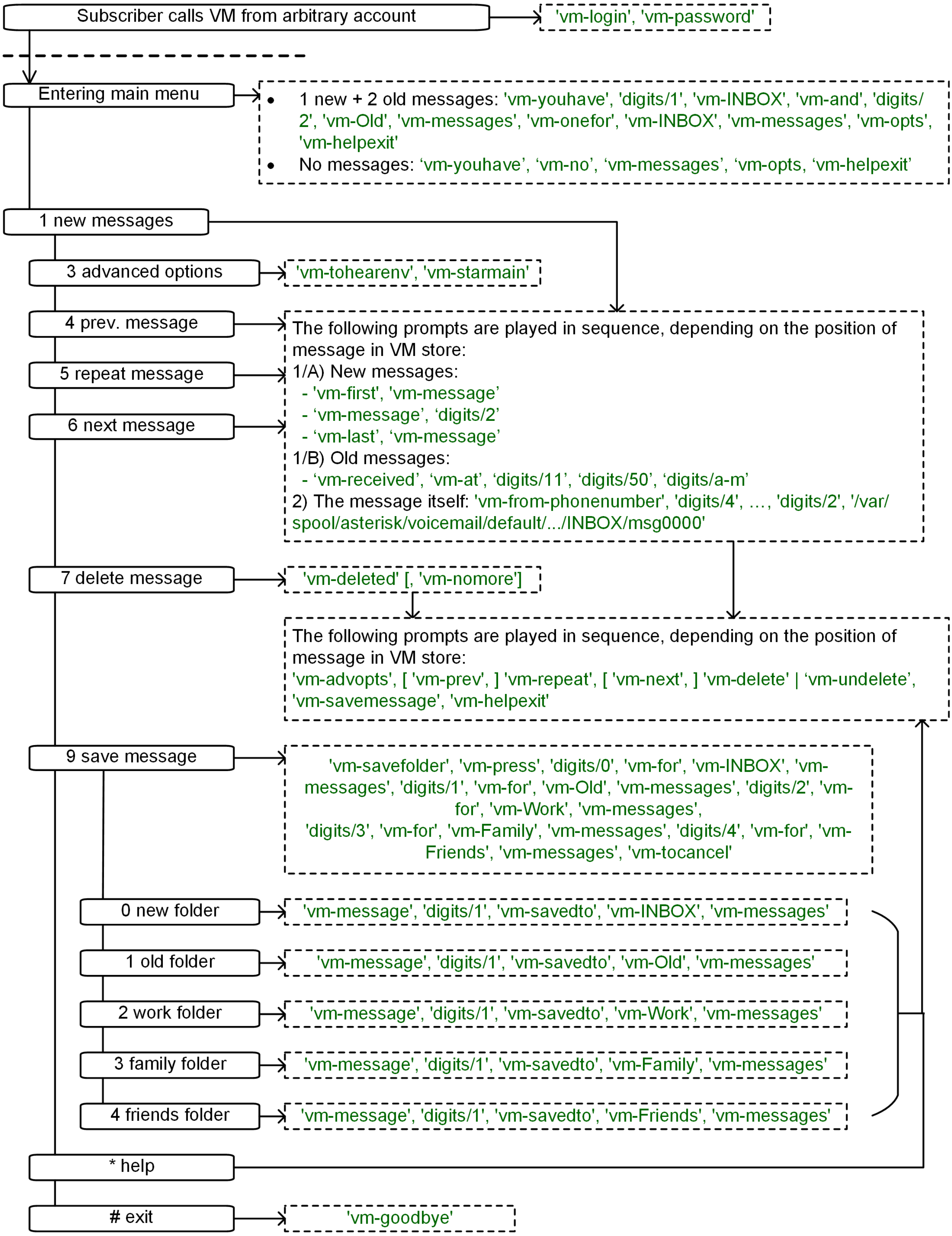


Figure 49: Flowchart of Listening to New Messages

7.11.6.2 Changing Voicemail Folders

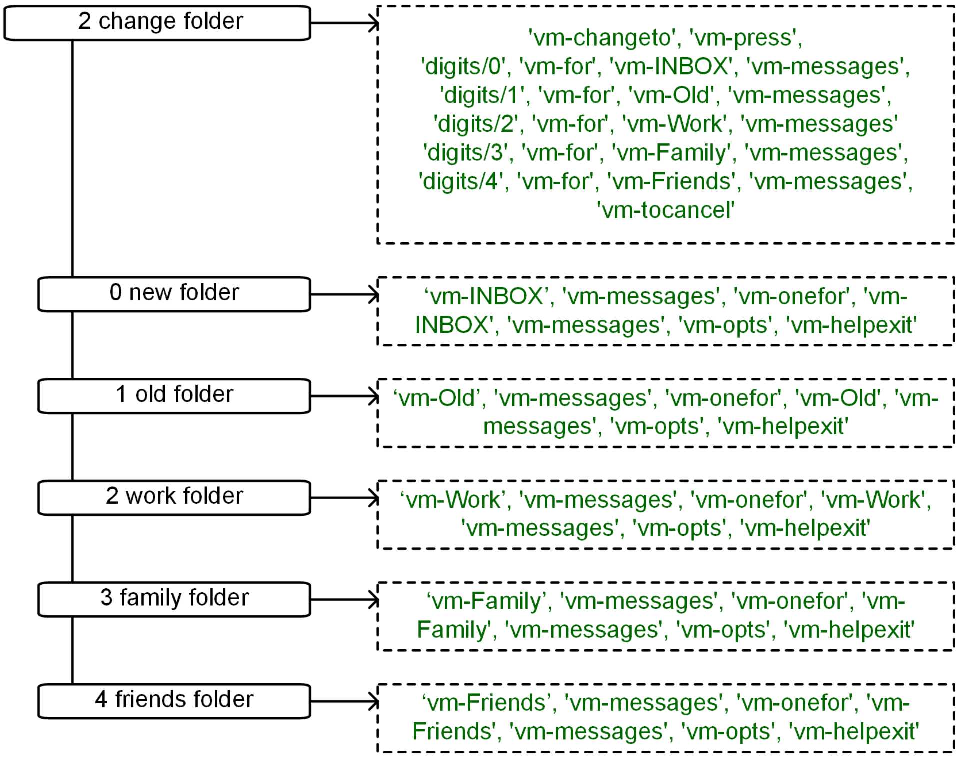


Figure 50: Flowchart of Changing Voicemail Folders

7.11.6.3 Mailbox Options

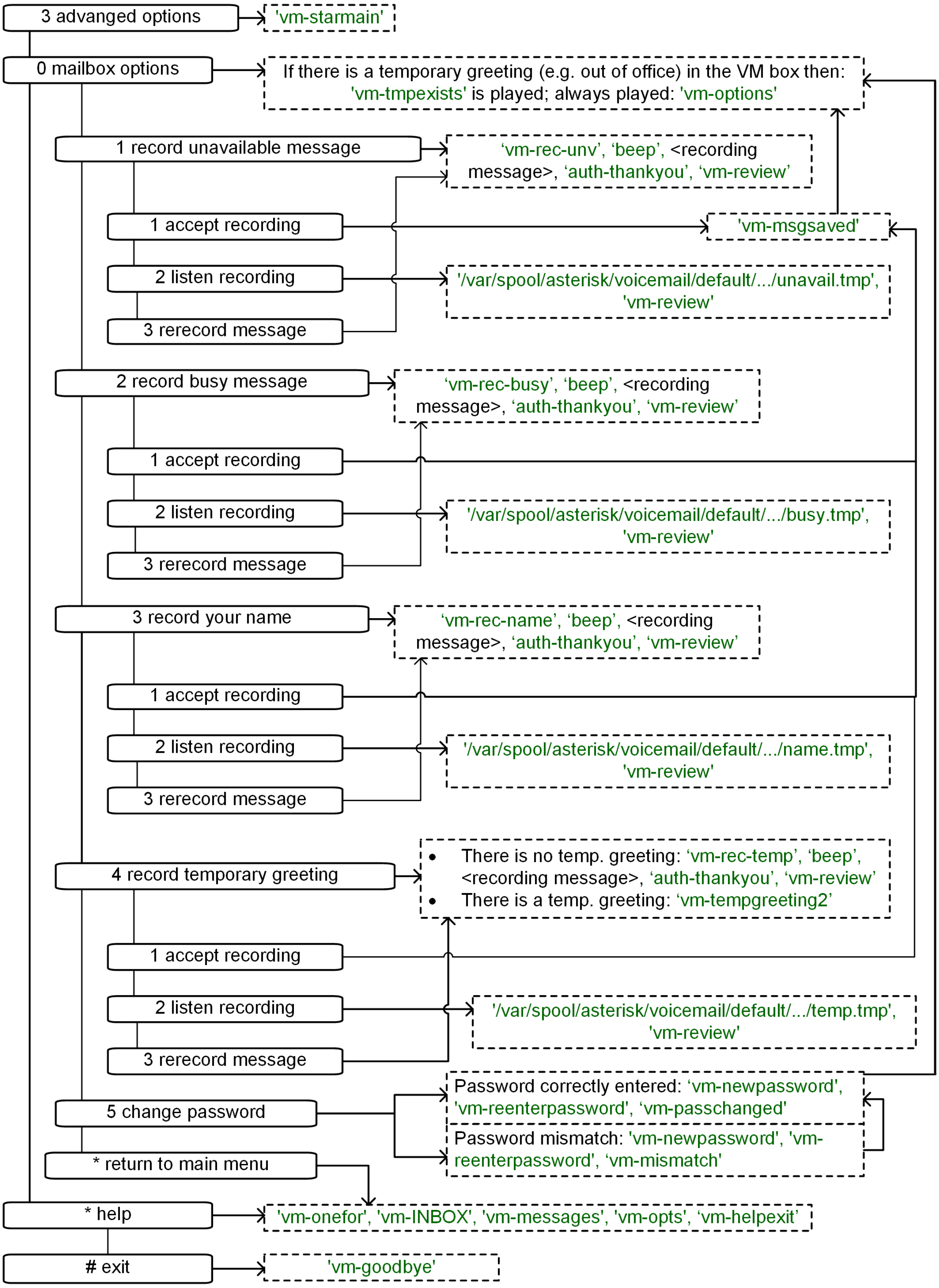


Figure 51: Flowchart of Changing Mailbox Options

7.11.6.4 Leaving a Message

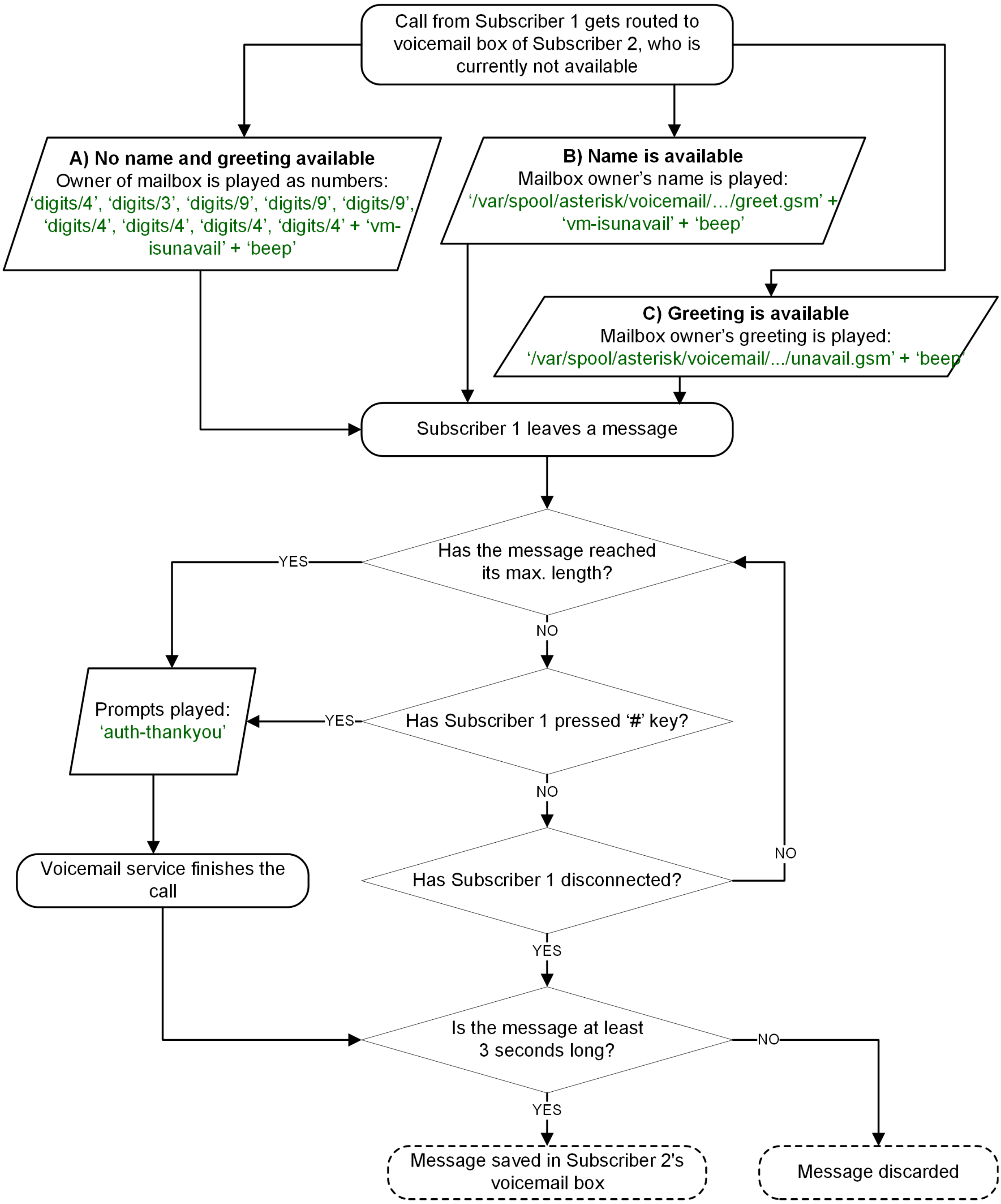
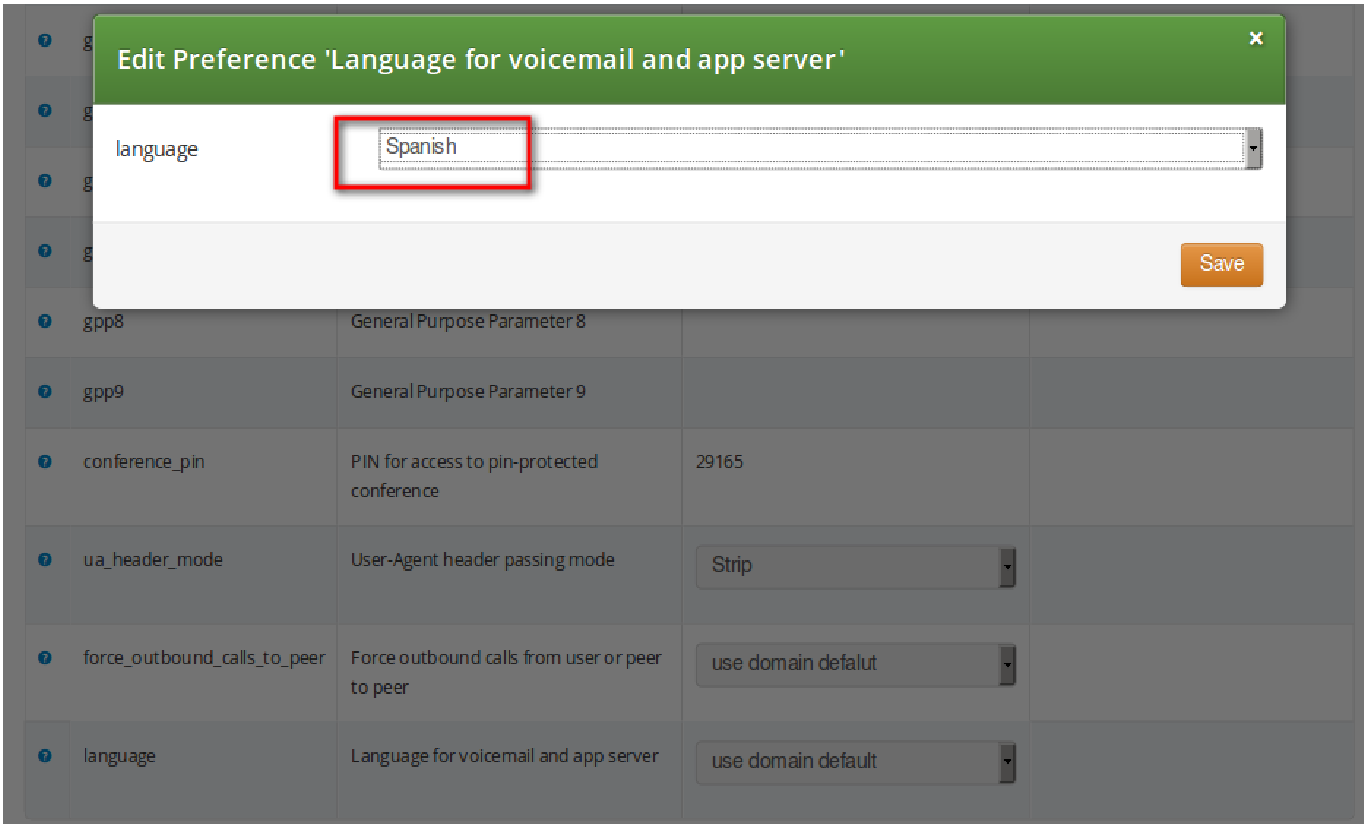


Figure 52: Flowchart of Leaving a Voice Message

7.12 Configuring Subscriber IVR Language

The language for the Voicemail system IVR or Vertical Service Codes (VSC) IVRs may be set using the subscriber or domain preference *language*.

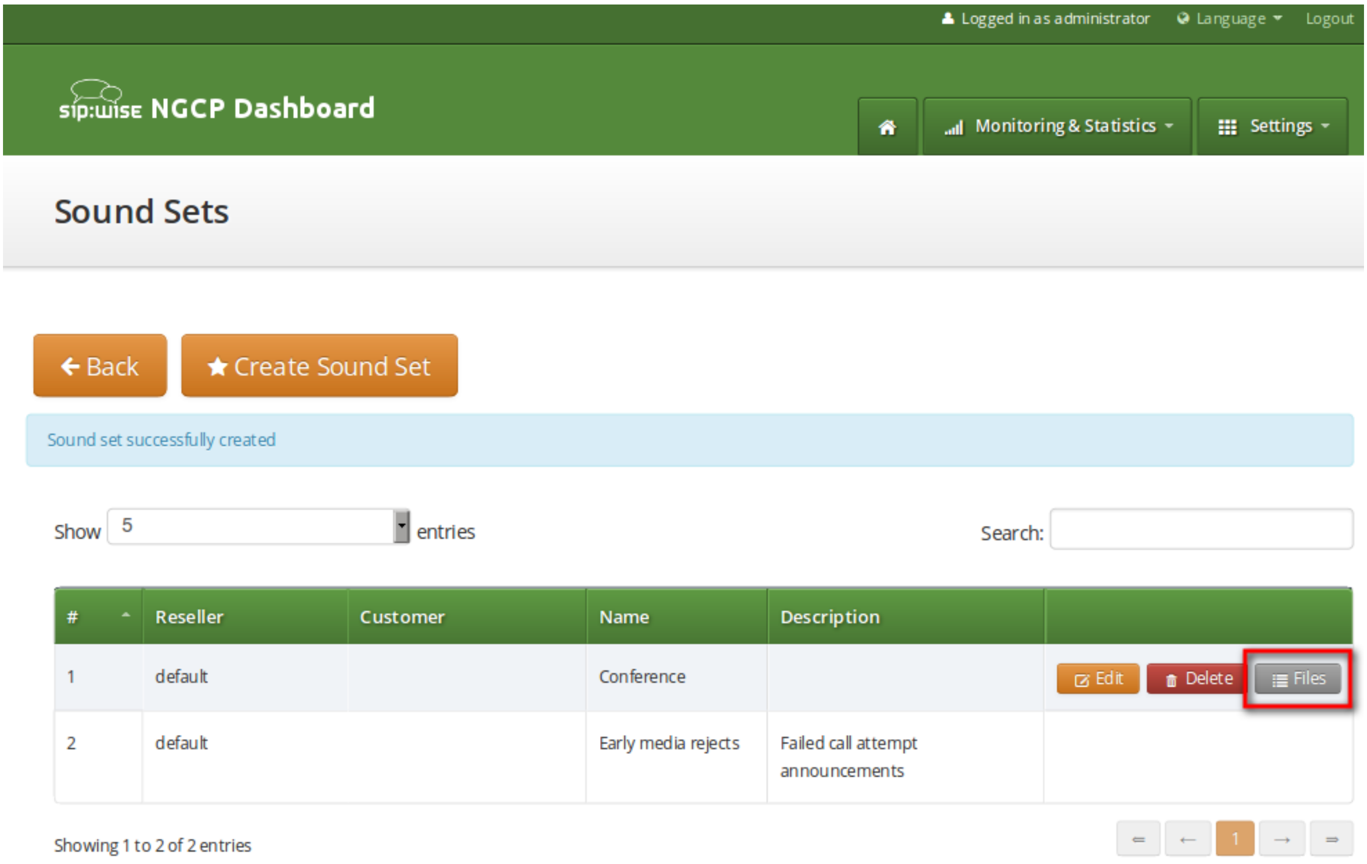


The sip:provider CE provides the pre-installed prompts for the Voicemail in the English, Spanish, French and Italian languages and the pre-installed prompts for the Vertical Service Codes IVRs in English only.

The other IVRs such as the Conference system and the error announcements use the Sound Sets configured in NGCP Panel and uploaded by the administrator in his language of choice.

7.13 Sound Sets

The sip:provider CE provides the administrator with ability to upload the voice prompts such as conference prompts or call error announcements on the *Sound Sets page*. There is a preference *sound_set* in the *NAT and Media Flow Control* section on Domain and Subscriber levels to link subscribers to the sound set that they should hear (as usual the subscriber preference overrides the domain one). Sound Sets can be defined in *Settings→Sound Sets*. To create a new Sound Set, click *Create Sound Set*. Then click the *Files* button.



Note

You may use 8 or 16 bit mono WAV audio files for all of the voice prompts.

7.13.1 Configuring Early Reject Sound Sets

The call error announcements are grouped under *Early Rejects* section. Unfold the section and click *Upload* next to the sound handles (Names) that you want to use. Choose a WAV file from your file system, and click the Loopplay setting if you want to play the file in a loop instead of just once. Click Save to upload the file.

The call error announcements are played to the user in early media hence the name "Early Reject". If you don't provide the sound files for any handles they will not be used and the sip:provider CE will fallback to sending the error response code back to the user.

The exact error status code and text are configurable in the `/etc/ngcp-config/config.yml` file, in `kamailio.proxy.early_rejects` section. Please look for the announcement handle listed in below table in order to find it in the configuration file.

Table 4: Early Reject Announcements

Handle	Description	Message played
`announce_before_cf`	This is an announcement that the calling party hears before the call is being forwarded (Unconditional and Not Available cases) to the destination. The feature can be activated with Applications / `play_announce_before_cf` domain or subscriber preference.	N/A (custom message, no default)
`block_in`	This is what the calling party hears when a call is made from a number that is blocked by the incoming block list (*adm_block_in_list*, *block_in_list* customer/subscriber preferences)	Your call is blocked by the number you are trying to reach.

Table 4: (continued)

Handle	Description	Message played
`block_out`	This is what the calling party hears when a call is made to a number that is blocked by the outgoing block list (*adm_block_out_list*, *block_out_list* customer/subscriber preferences)	Your call to the number you are trying to reach is blocked.
`block_ncos`	This is what the calling party hears when a call is made to a number that is blocked by the NCOS level assigned to the subscriber or domain (the NCOS level chosen in *ncos* and *adm_ncos* preferences). *PLEASE NOTE:* It is not possible to configure the status code and text.	Your call to the number you are trying to reach is not permitted.
`block_override_pin_wrong`	Announcement played to calling party if it used wrong PIN code to override the outgoing user block list or the NCOS level for this call (the PIN set by *block_out_override_pin* and *adm_block_out_override_pin* preferences)	The PIN code you have entered is not correct.
`callee_busy`	Announcement played on incoming call to the subscriber which is currently busy (486 response from the UAS)	The number you are trying to reach is currently busy. Please try again later.
`callee_offline`	Announcement played on incoming call to the subscriber which is currently not registered	The number you are trying to reach is currently not available. Please try again later.
`callee_tmp_unavailable`	Announcement played on incoming call to the subscriber which is currently unavailable (408, other 4xx or no response code or 30x with malformed contact)	The number you are trying to reach is currently not available. Please try again later.
`callee_unknown`	Announcement that is played on call to unknown or invalid number (not associated with any of our subscribers/hunt groups)	The number you are trying to reach is not in use.
`cf_loop`	Announcement played when the called subscriber has the call forwarding configured to itself	The number you are trying to reach is forwarded to an invalid destination.

Table 4: (continued)

Handle	Description	Message played
`emergency_geo_unavailable`	Announcement played when emergency destination is dialed but the destination is not provisioned for the location of the user. *PLEASE NOTE:* The configuration entry for this case in `/etc/ngcp-config/config.yml` file is `emergency_invalid`.	The emergency number you have dialed is not available in your region.
`emergency_unsupported`	Announcement played when emergency destination is dialed but the emergency calls are administratively prohibited for this user or domain (*reject_emergency* preference is enabled)	You are not allowed to place emergency calls from this line. Please use a different phone.
`error_please_try_later`	Announcement played when the call is handled by 3rd party call control (PCC) and there was an error during call processing. *PLEASE NOTE:* This announcement may be configured in the sound set in `voucher_recharge` section.	An error has occured. Please try again later.
`invalid_speeddial`	This is what the calling party hears when it calls an empty speed-dial slot	The speed dial slot you are trying to use is not available.
`locked_in`	Announcement played on incoming call to a subscriber that is locked for incoming calls	The number you are trying to reach is currently not permitted to receive calls.
`locked_out`	Announcement played on outgoing call to subscriber that is locked for outgoing calls	You are currently not allowed to place outbound calls.
`max_calls_in`	Announcement played on incoming call to a subscriber who has exceeded the *concurrent_max* limit by sum of incoming and outgoing calls or whose customer has exceeded the *concurrent_max_per_account* limit by sum of incoming and outgoing calls	The number you are trying to reach is currently busy. Please try again later.
`max_calls_out`	Announcement played on outgoing call to a subscriber who has exceeded the *concurrent_max* (total limit) or *concurrent_max_out* (limit on number of outbound calls) or whose customer has exceeded the *concurrent_max_per_account* or *concurrent_max_out_per_account* limit	All outgoing lines are currently in use. Please try again later.

Table 4: (continued)

Handle	Description	Message played
`max_calls_peer`	Announcement played on calls from the peering if that peer has reached the maximum number of concurrent calls (configured by admin in *concurrent_max* preference of peering server). *PLEASE NOTE:* There is no configuration option of the status code and text in `config.yml` file for this case.	The network you are trying to reach is currently busy. Please try again later.
`no_credit`	Announcement played when prepaid account has insufficient balance to make a call to this destination	You don't have sufficient credit balance for the number you are trying to reach.
`peering_unavailable`	Announcement played in case of outgoing off-net call when there is no peering rule matching this destination and/or source	The network you are trying to reach is not available.
`reject_vsc`	When the VSC (Vertical Service Code) service is disabled in domain or subscriber preferences (Access Restrictions / `reject_vsc` is set to TRUE) and a subscriber tries to make a call with VSC, an announcement is played.	N/A (custom message, no default)
`relaying_denied`	Announcement played on inbound call from trusted IP (e.g. external PBX) with non-local Request-URI domain	The network you are trying to reach is not available.
`unauth_caller_ip`	This is what the calling party hears when it tries to make a call from unauthorized IP address or network (*allowed_ips*, *man_allowed_ips* preferences)	You are not allowed to place calls from your current network location.
`voicebox_unavailable`	*PLEASE NOTE:* This announcement is already obsolete, as of NGCP version mr5.3	The voicemail of the number you are trying to reach is currently not available. Please try again later.

There are some early reject scenarios when either **no voice announcement is played, or a fixed announcement is played**. In either case a SIP error status message is sent from NGCP to the calling party. It is possible to configure the exact status code and text for such cases in the `/etc/ngcp-config/config.yml` file, in `kamailio.proxy.early_rejects` section. The below table gives an overview of those early reject cases.

Table 5: Additional Early Reject Reason Codes

Handle	Description
block_admin	Caller blocked by adm_block_in_list, adm_block_in_clir and callee blocked by adm_block_out_list (customer or subscriber preference)
block_callee	Callee blocked by subscriber preference block_out_list
block_caller	Caller blocked by subscriber preference block_in_list, block_in_clir
block_contract	Caller blocked by customer preference block_in_list, block_in_clir and callee blocked by customer preference block_out_list
callee_tmp_unavailable_gp	Callee is a PBX group with 0 members. Announcement callee_tmp_unavailable is played; status code and text can be configured.
callee_tmp_unavailable_tm	Callee is a PBX group and we have a timeout (i.e. no group member could be reached). Announcement callee_tmp_unavailable is played; status code and text can be configured.
emergency_invalid	*PLEASE NOTE:* This handle refers to the same early reject case as emergency_geo_unavailable, but is labeled differently in the configuration file.

7.14 Conference System

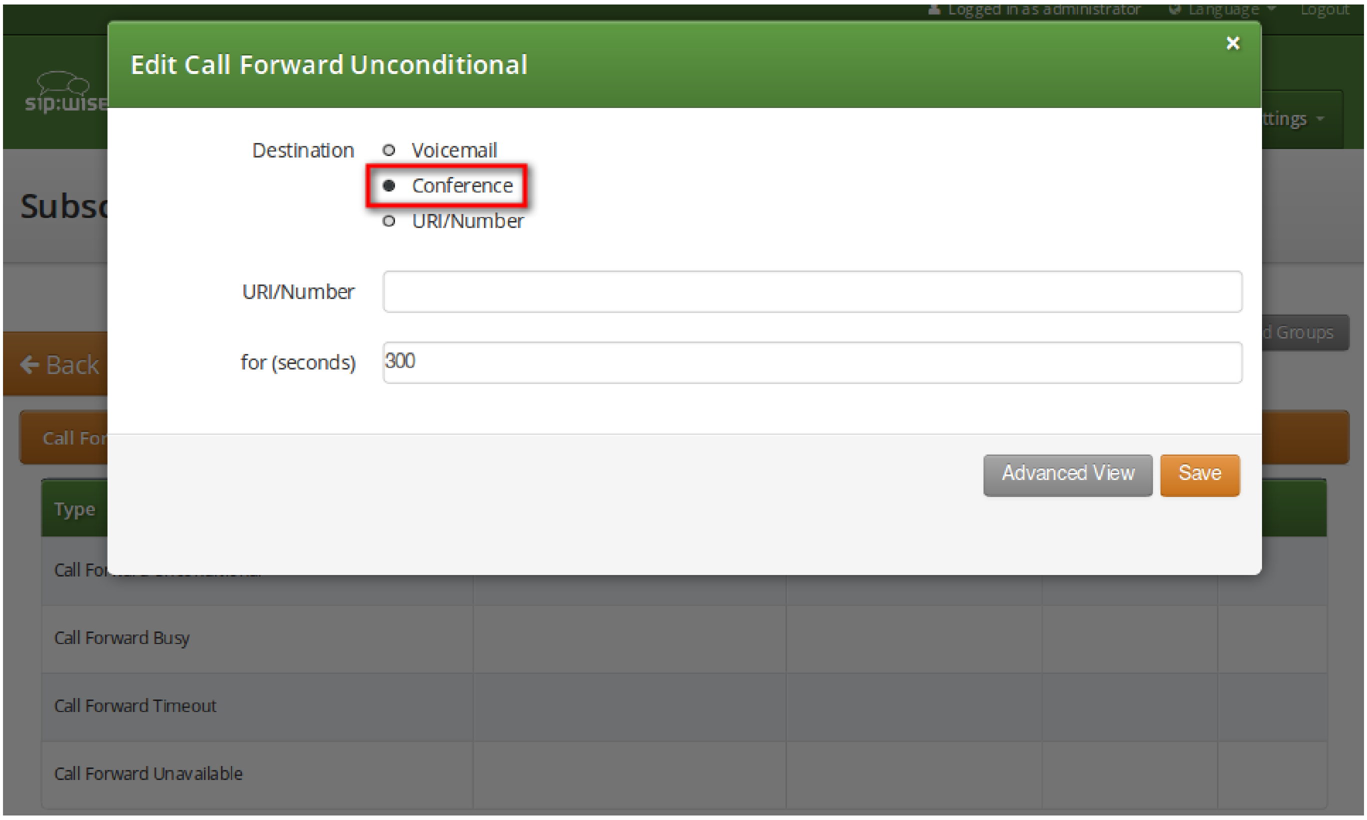
The sip:provider CE provides the simple pin-protected conferencing service built using the SEMS DSM scripting language. Hence it is open for all kinds of modifications and extensions.

Template files for the sems conference scripts stored in */etc/ngcp-config/templates/etc/ngcp-sems/*:

- IVR script: /etc/ngcp-config/templates/etc/ngcp-sems/dsm/confpin.dsm.tt2
- Config: /etc/ngcp-config/templates/etc/ngcp-sems/dsm/confpin.conf.tt2

7.14.1 Configuring Call Forward to Conference

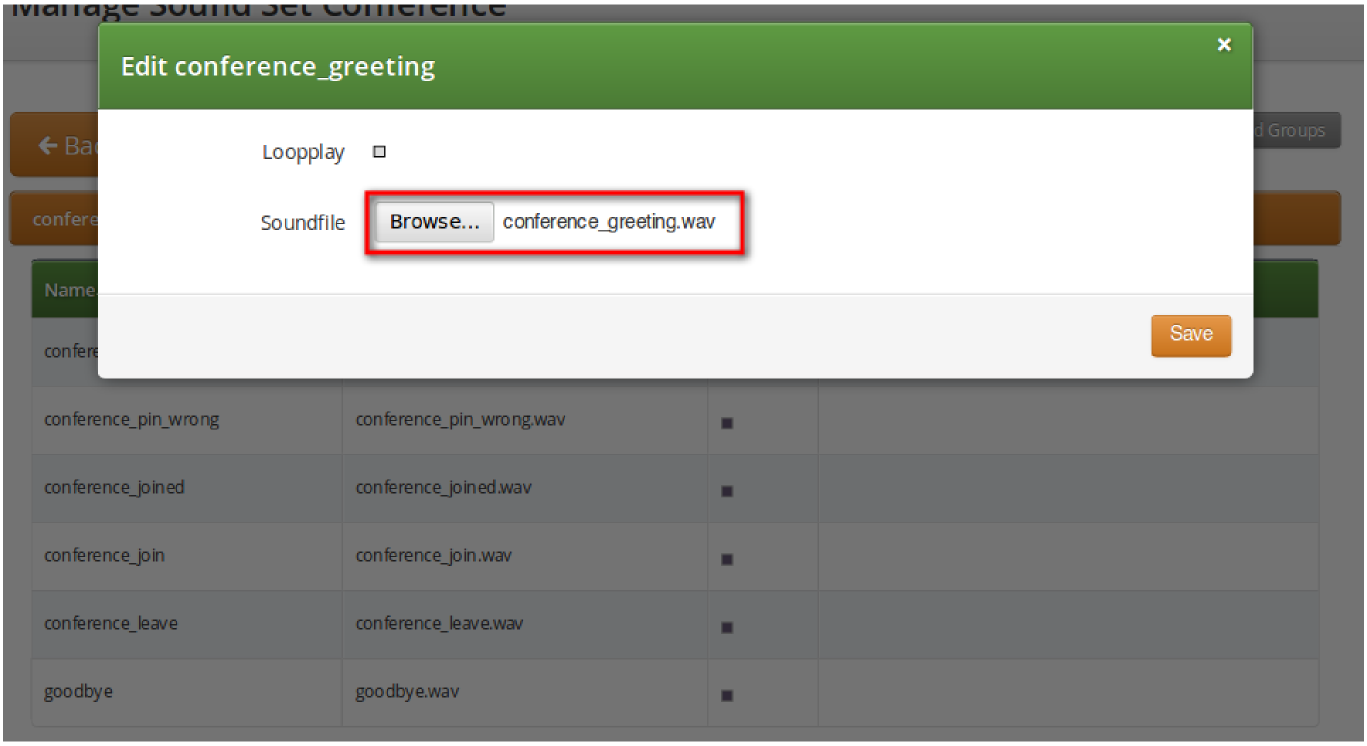
Go to your *Subscriber Preferences* and click *Edit* on the Call Forward Type you want to set (e.g. *Call Forward Unconditional*).



You should select *Conference* option in the *Destination* field and leave the *URI/Number* empty. The timeout defines for how long this destination should be tried to ring.

7.14.2 Configuring Conference Sound Sets

Sound Sets can be defined in *Settings→Sound Sets*. To create a new Sound Set, click *Create Sound Set*. Then click the *Files* button.



Upload the following files:

Table 6: Conference Sound Sets

Handle	Message played
conference_greeting	Welcome to the conferencing service.
conference_pin	Please enter your PIN, followed by the pound key.
conference_pin_wrong	You have entered an invalid PIN number. Please try again.
conference_joined	You will be placed into the conference.
conference_first	You are the first person in the conference.
conference_join	A person has joined the conference.
conference_leave	A person has left the conference.
conference_max_participants	All conference lines are currently in use. Please try again later.
conference_waiting_music	. . . waiting music. . .
goodbye	Goodbye.

Note

You may use 8 or 16 bit mono WAV audio files.

Then set the preference *sound_set* on the Domain or Subscriber level in order to assign the Sound Set you have just created to the subscriber (as usual the subscriber preference overrides the domain one).

7.14.3 Joining the Conference

There are 2 ways of joining a conference: with or without PIN code. The actual way of joining the conference depends on *Subscriber* settings. A subscriber who has activated the conference through call forwarding may set a PIN in order to protect the conference from unauthorized access. To activate the PIN one has to enter a value in *Subscriber* → *Details* → *Preferences* → *Internals* → *conference_pin* field.

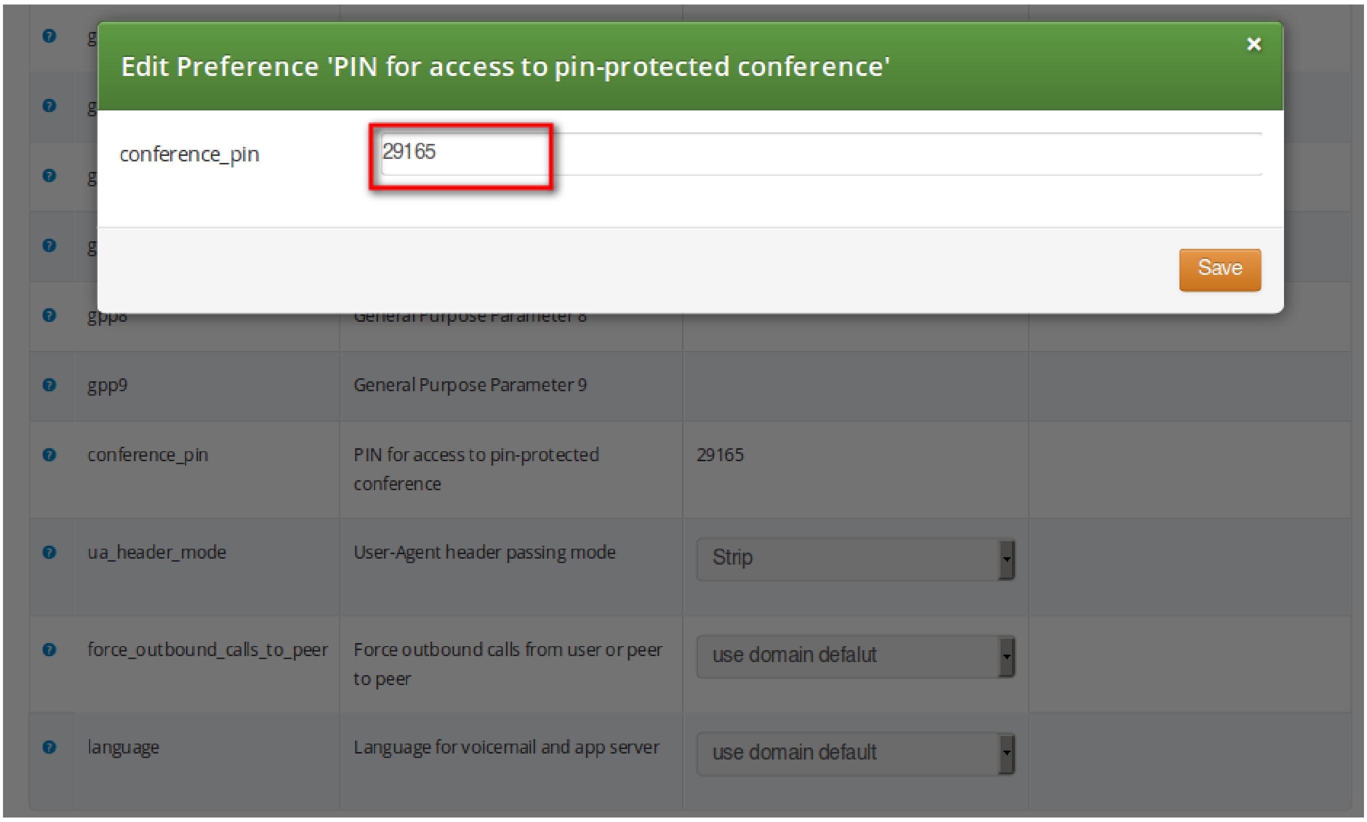


Figure 53: Setting Conference PIN

In case the PIN protection for the conference is activated, when someone calls the subscriber who has enabled the conference, the caller is prompted to enter the PIN of the conference. Upon the successful entry of the PIN the caller hears the announcement that he is going to be placed into the conference and at the same time this is announced to all participants already in the conference.

7.14.4 Conference Flowchart with Voice Prompts

The following 2 sections show flowcharts with voice prompts that are played to a caller when he dials the conference.

7.14.4.1 Conference Flowchart with PIN Validation

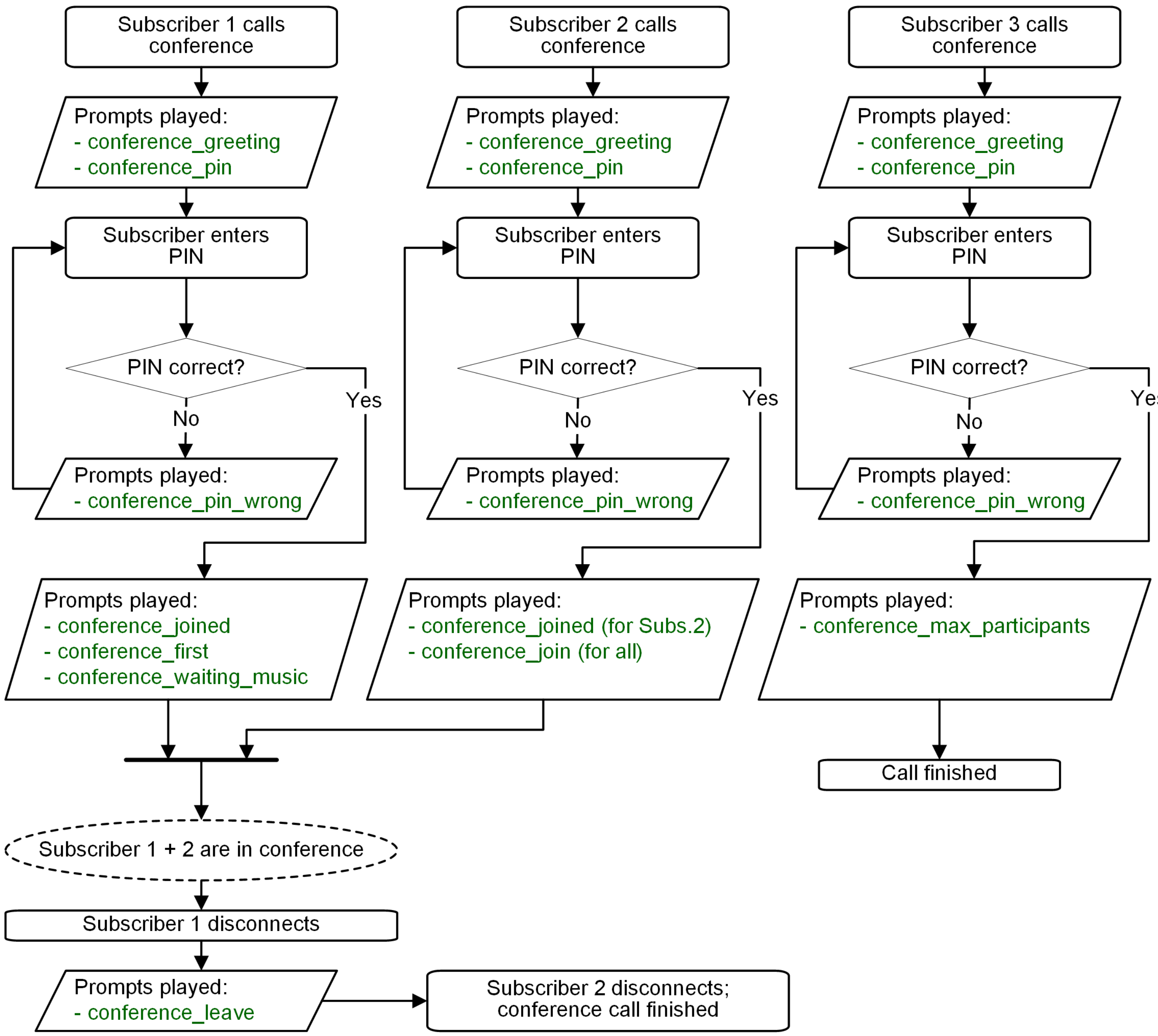


Figure 54: Flowchart of Conference with PIN Validation

7.14.4.2 Conference Flowchart without PIN

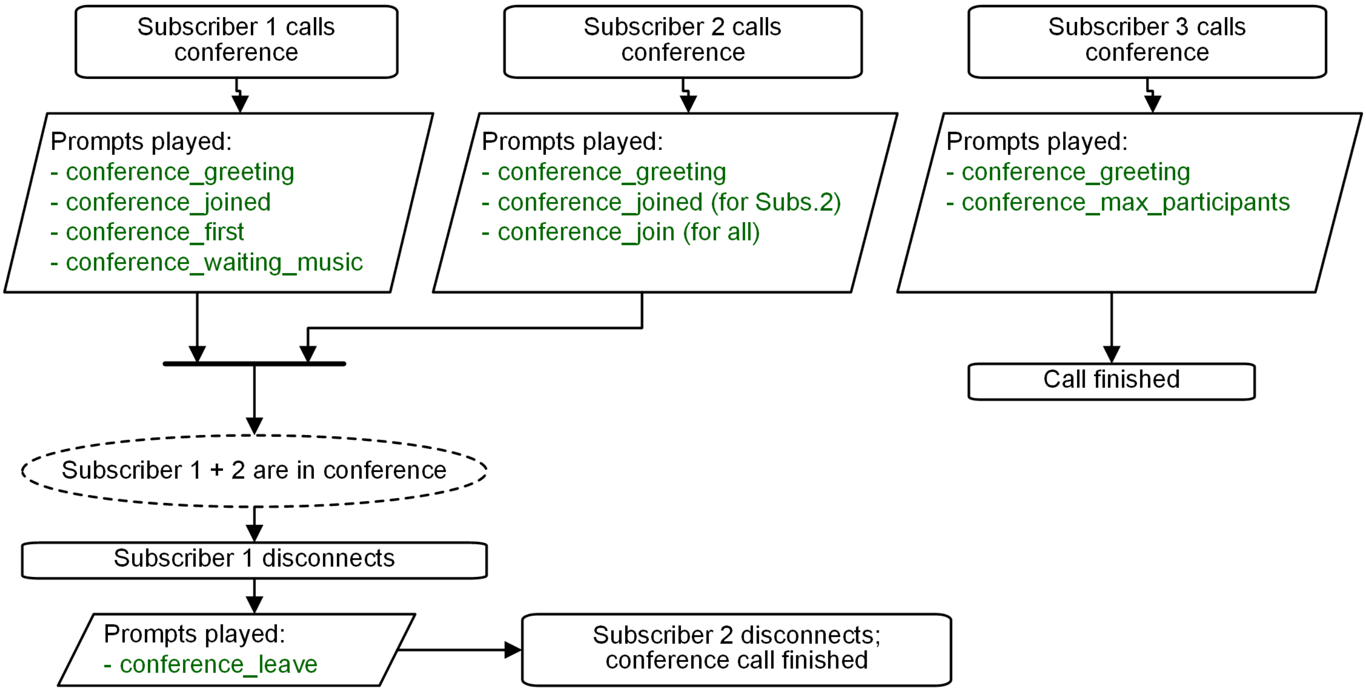


Figure 55: Flowchart of Conference without PIN

7.15 Malicious Call Identification (MCID)

MCID feature allows customers to report unwanted calls to the platform operator.

7.15.1 Setup

To enable the feature first edit `config.yml` and enable there `apps: malicious_call:` *yes* and `kamailio: store_recentcalls:` *yes*. The latter option enables kamailio to store recent calls per subscrbriber UUID in the redis DB (the amount of stored recent calls will not exceed the amount of provisionined subscribers).

Next step is to create a system sound set for the feature. In *Settings*→*Sound Sets* either use your already existing *Sound Set* or create a new *Sound Set* and then assign it to your domain or subscribers. In the *Sound Set* there is a fileset *malicious_call_identification*→ for that purpose.

Once the *Sound Set* is created the Subscriber's Preferences *Malicious Call Identification* must be enabled under *Subcriber* → *Preferences* → *Applications* menu. The same parameter can be set in the Customer's preferences to enable this feature for all its subscribers.

The final step is to create a new *Rewrite Rule* and to route calls to, for instance `*123 → MCID application`. For that you create a *Calee Inbound* rewrite rule `^(\*123)$` → `malicious_call`

Finaly you run `ngcpcfg apply` *`Enabling MCID`* to recreate the templates and automatically restart depended services.

7.15.2 Usage

As a subscriber, to report a malicious call you call to either *malicious_call* or to your custom number assigned for that purpose. Please note that you can report only your last received call. You will hear the media reply from the *Sound Set* you have previosuly configured.

To check reported malicious calls as the plafrom operator open *Settings→Malicious Calls* tab where you will see a list of registered calls. You can selectively delete records from the list and alternatively you can manage the reported calls by using the REST API.

7.15.3 Advanced configuration

By default the expiration time for the most recent call per subscriber is 3600 seconds (1 hour). If you wish to prolong or shorten the expiration time open `constants.yml` and set there `recentcalls: expire: 3600` to a new value, and issue `ngcpcfg apply` *`Enabling MCID`* afterwards.

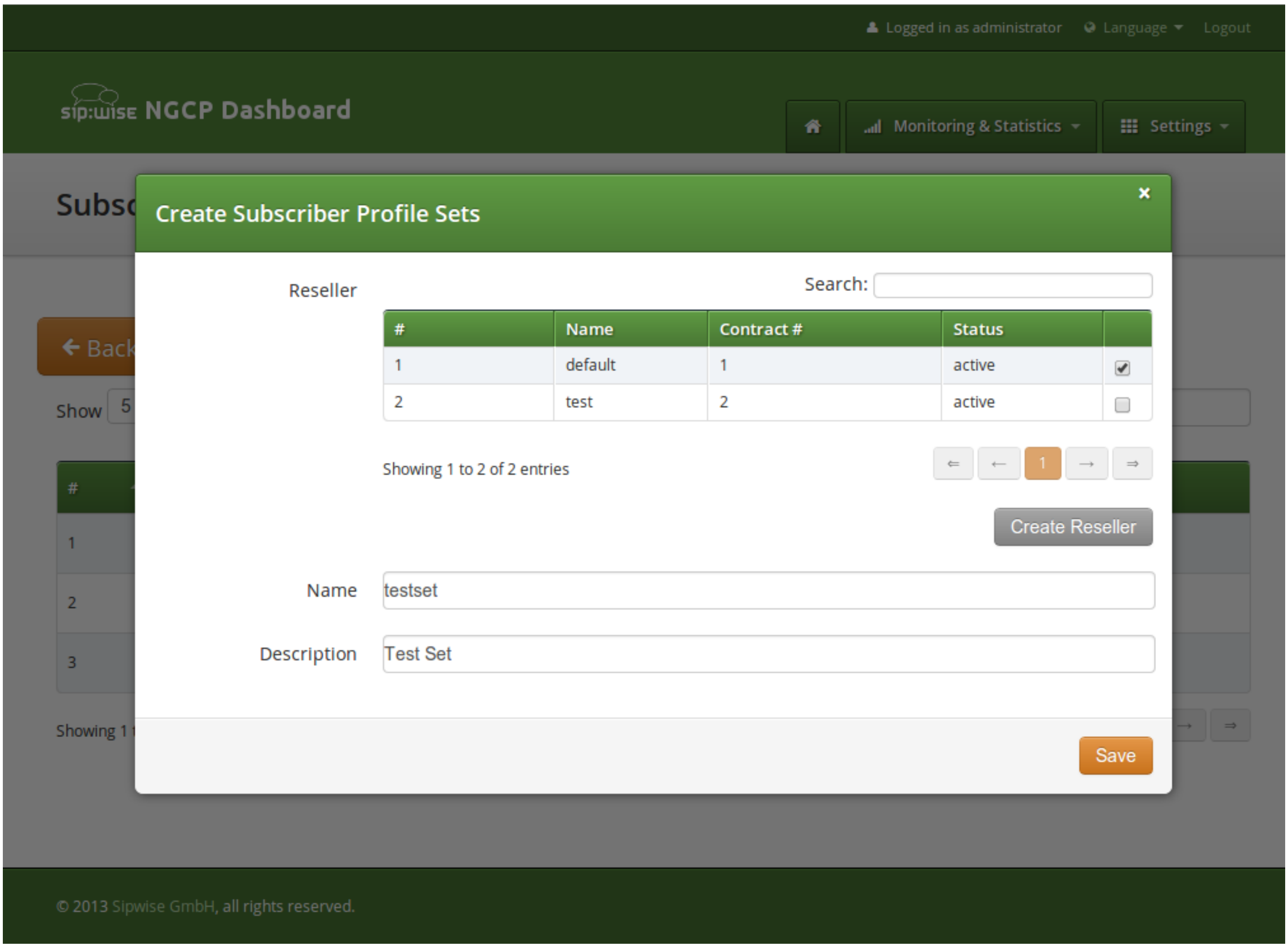
7.16 Subscriber Profiles

The preferences a subscriber can provision by himself via the CSC can be limited via profiles within profile sets assigned to subscribers.

7.16.1 Subscriber Profile Sets

Profile sets define containers for profiles. The idea is to define profile sets with different profiles by the administrator (or the reseller, if he is permitted to do so). Then, a subscriber with administrative privileges can re-assign profiles within his profile sets for the subscribers of his customer account.

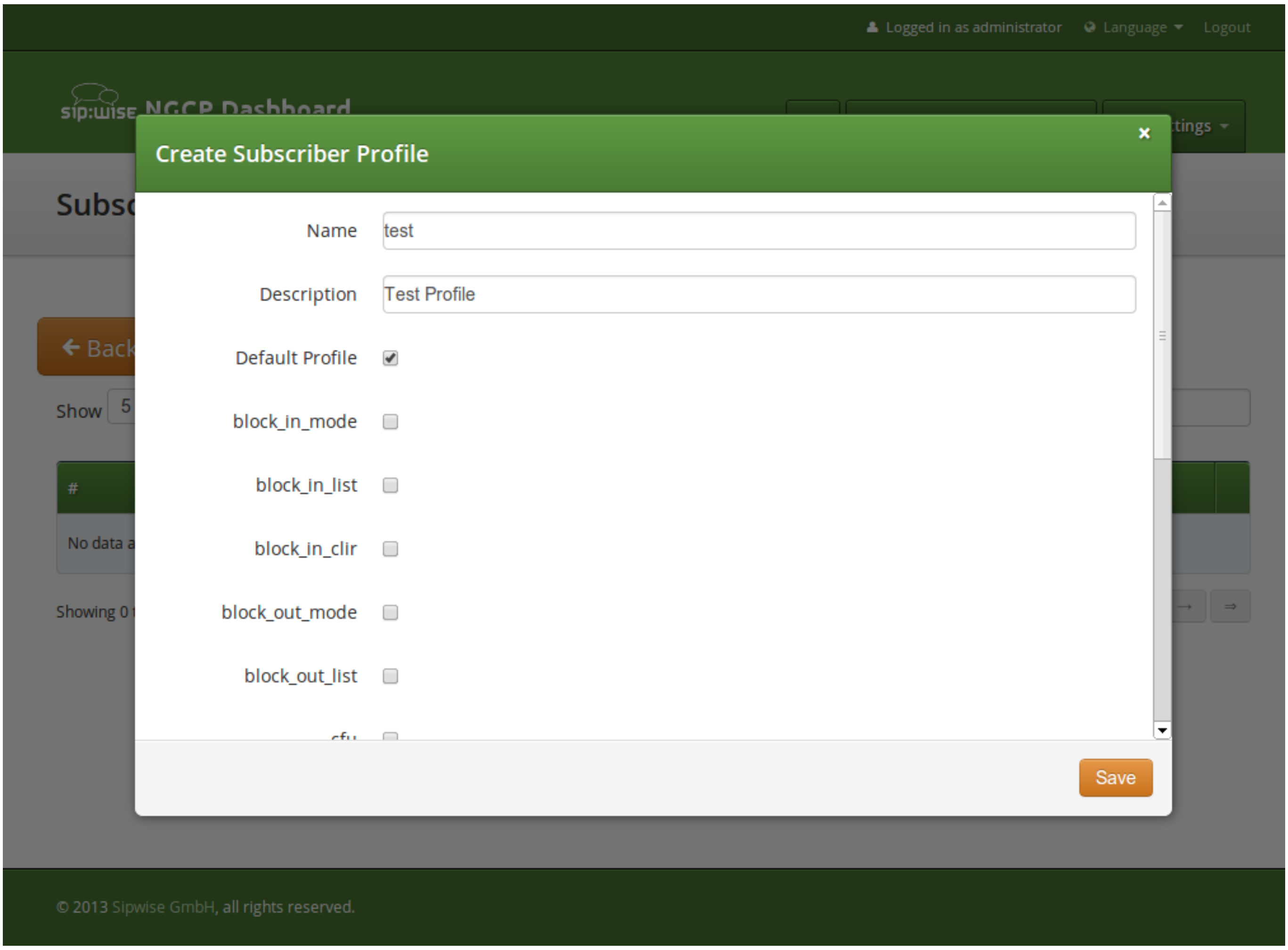
Profile Sets can be defined in *Settings→Subscriber Profiles*. To create a new Profile Set, click *Create Subscriber Profile Set*.



You need to provide a reseller, name and description.

To create Profiles within a Profile Set, hover over the Profile Set and click the *Profiles* button.

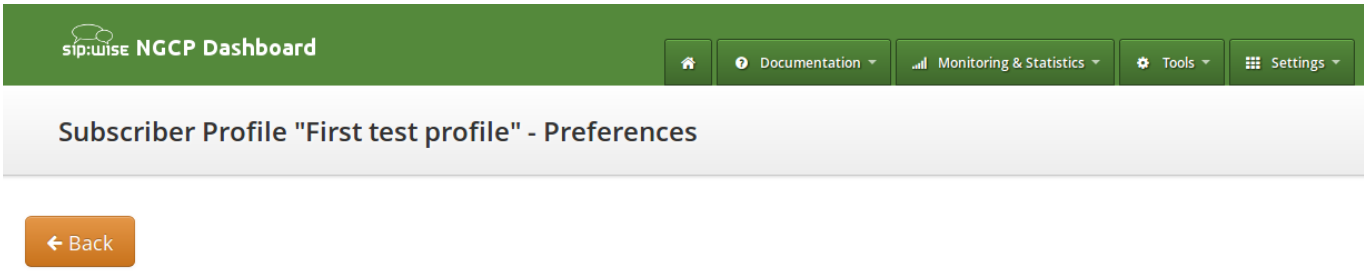
Profiles within a Profile Set can be created by clicking the *Create Subscriber Profile* button.



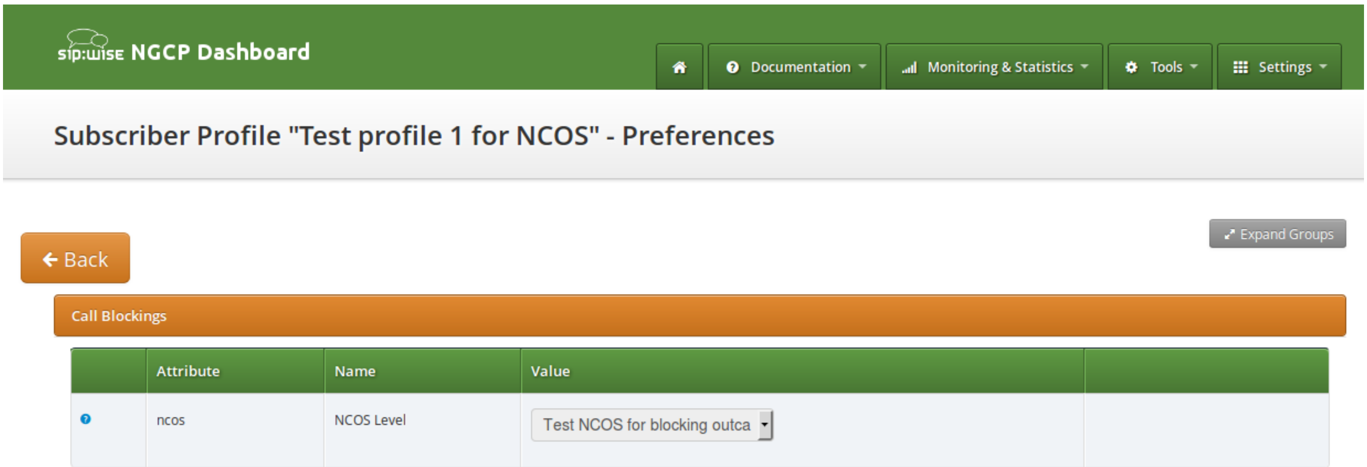
Checking the *Default Profile* option causes this profile to get assigned automatically to all subscribers, who have the profile set assigned. Other options define the user preferences which should be made available to the subscriber.

Note

When the platform administrator selects *Preferences* of the Subscriber Profile he will get an empty page like in the picture below, if none or only certain options are selected in the Subscriber Profile.



Some of the options, like `ncos` (NCOS level), will enable the definition of that preference within the Subscriber Profile Preferences. Thus all subscribers who have this profile assigned to will have the preference activated by default. The below picture shows the preferences linked to the sample Subscriber Profile:



7.17 SIP Loop Detection

In order to detect a SIP loop (incoming call as a response for a call request) sip:provider CE checks the combination of *SIP-URI*, *To* and *From* headers.

This check can be enabled in config.yml by setting kamailio.proxy.loop_detection.enable: 'yes'. The system tolerates kamailio.proxy.loop_d loops within kamailio.proxy.loop_detection.expire seconds. Higher occurrence of loops will be reported with a SIP 482 "Loop Detected" error message

.

7.18 Invoices and Invoice Templates

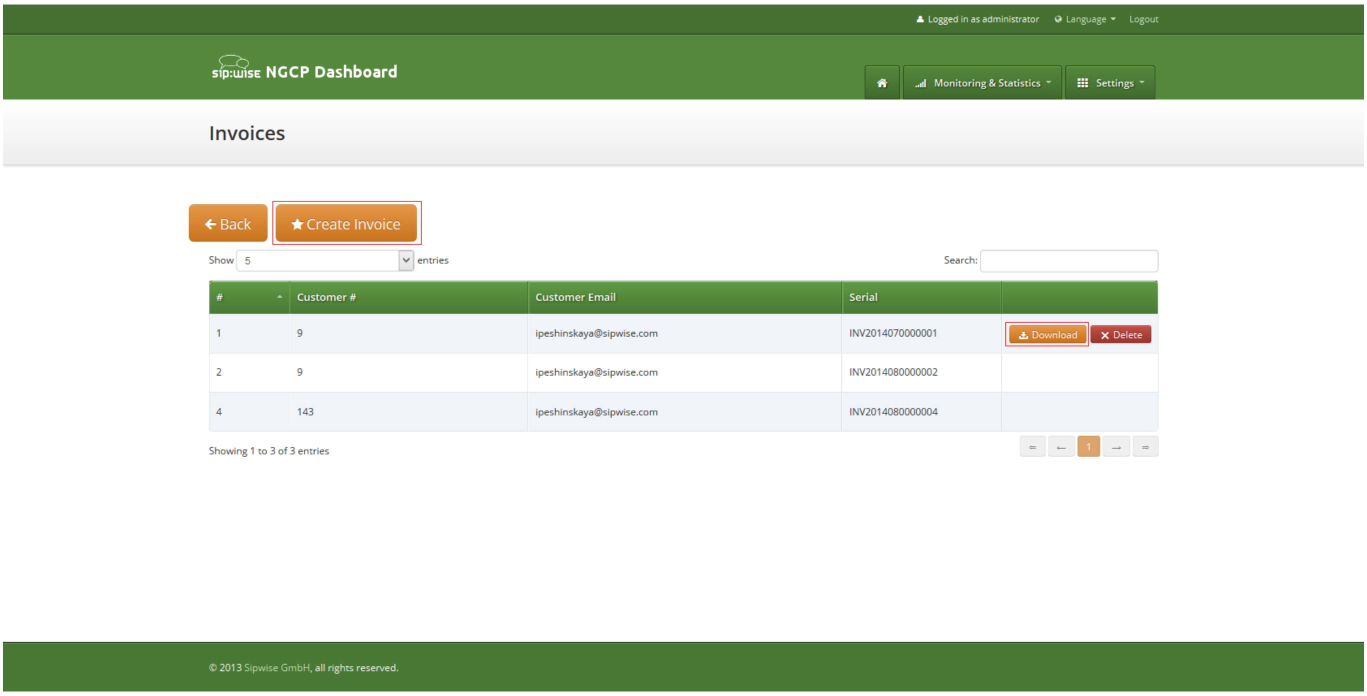
Content and vision of the invoices are customizable by invoice templates Section 7.18.2.

Note

The sip:provider CE generates invoices in pdf format.

7.18.1 Invoices Management

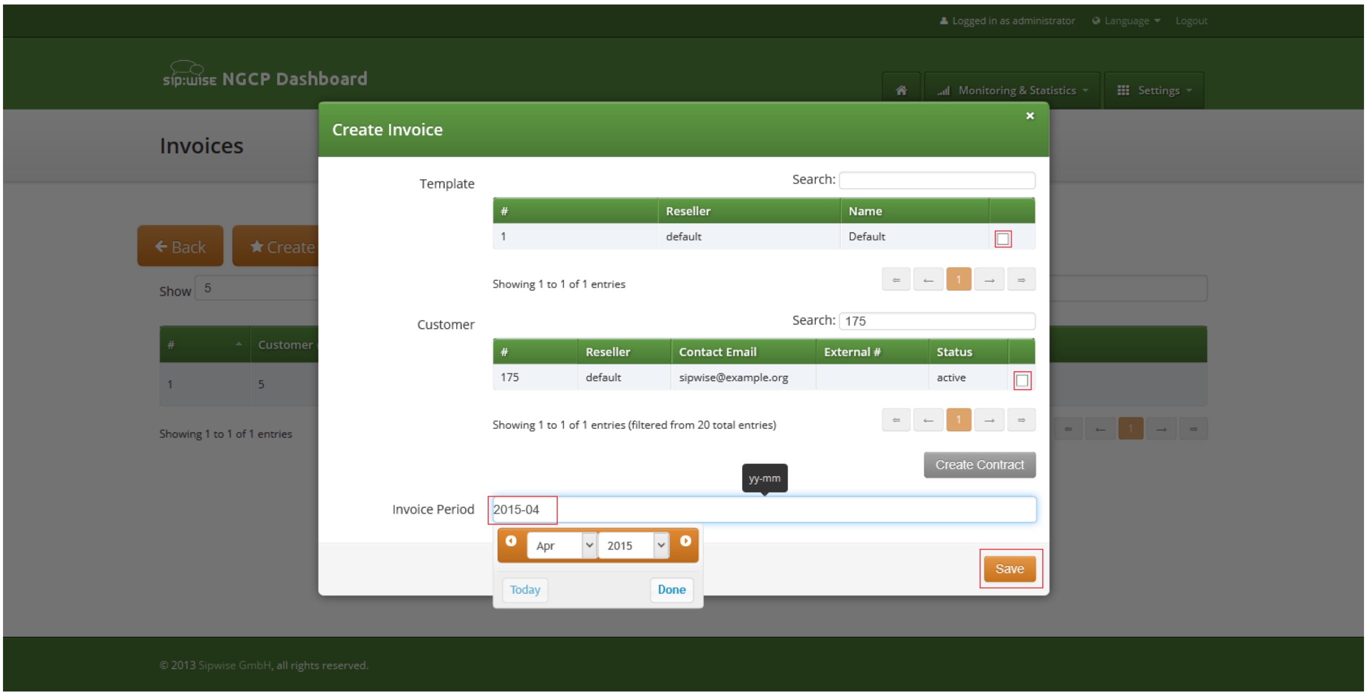
Invoices can be requested for generation, searched, downloaded and deleted in the invoices interface.



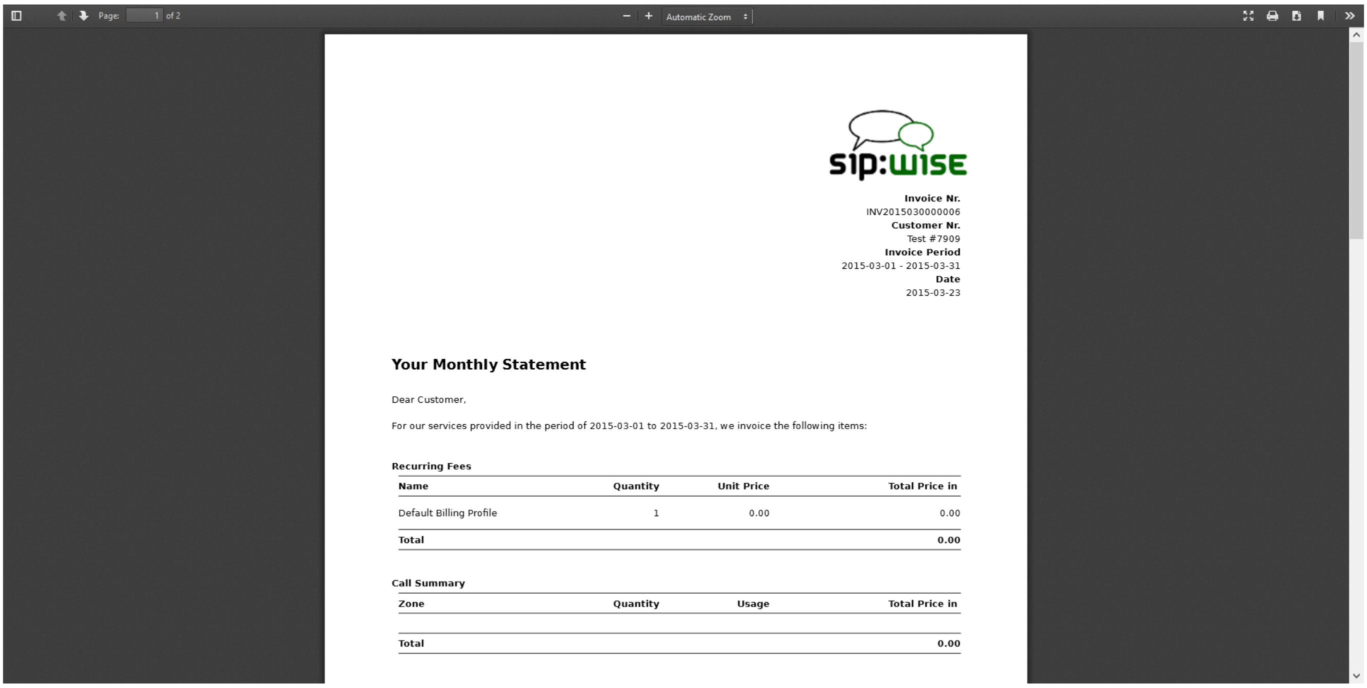
To request invoice generation for the particular customer and period press "Create invoice" button. On the invoice creation form following parameters are available for selection:

- **Template**: any of existent invoice template can be selected for the invoice generation.
- **Customer**: owner of the billing account, recipient of the invoice.
- **Invoice period**: billing period. Can be specified only as one calendar month. Calls with start time between first and last second of the period will be considered for the invoice

All form fields are mandatory.



Generated invoice can be downloaded as pdf file.



To do it press button "Download" against invoice in the invoice management interface.

Respectively press on the button "Delete" to delete invoice.

7.18.2 Invoice Templates

Invoice template defines structure and look of the generated invoices. The sip:provider CE makes it possible to create some invoice templates. Multiple invoice templates can be used to send invoices to the different customers using different languages.

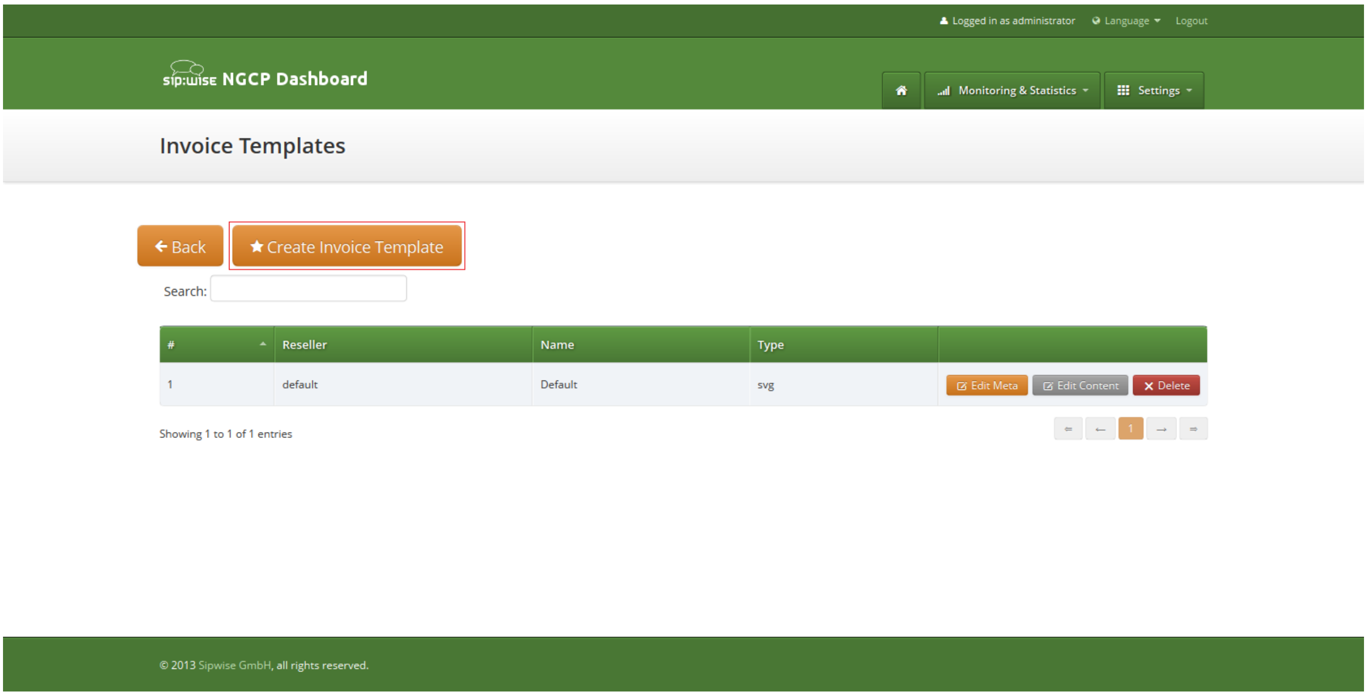
Important

At least one invoice template should be created to enable invoice generation. Each customer has to be associated to one of the existent invoice template, otherwise invoices will be not generated for this customer.

Customer can be linked to the invoice template in the customer interface.

7.18.2.1 Invoice Templates Management

Invoice templates can be searched, created, edited and deleted in the invoice templates management interface.



Invoice template creation is separated on two steps:

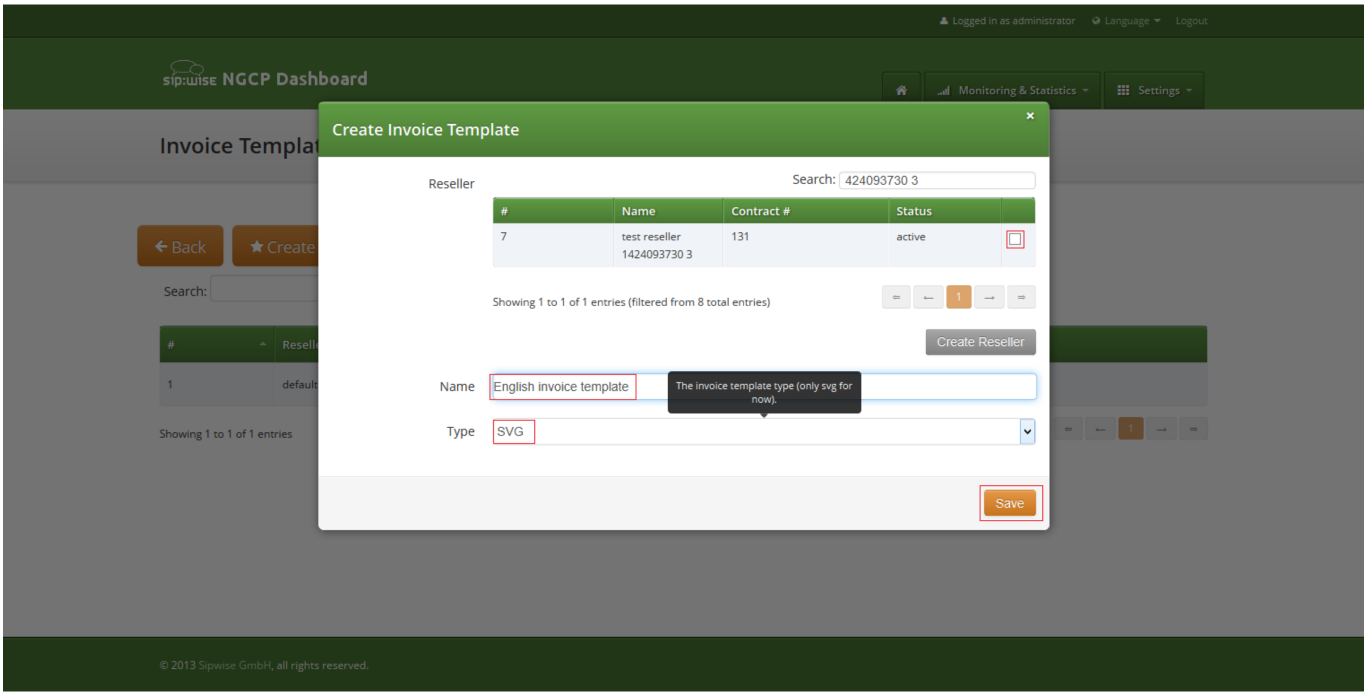
- Register new invoice template meta information.
- Edit content (template itself) of the invoice template.

To register new invoice template press "Create Invoice Template" button.

On the invoice template meta information form following parameters can be specified:

- **Reseller**: reseller who owns this invoice template. Please note, that it doesn't mean that the template will be used for the reseller customers by default. After creation, invoice template still need to be linked to the reseller customers.
- **Name**: unique invoice template name to differentiate invoice templates if there are some.
- **Type**: currently sip:provider CE supports only svg format of the invoice templates.

All form fields are mandatory.



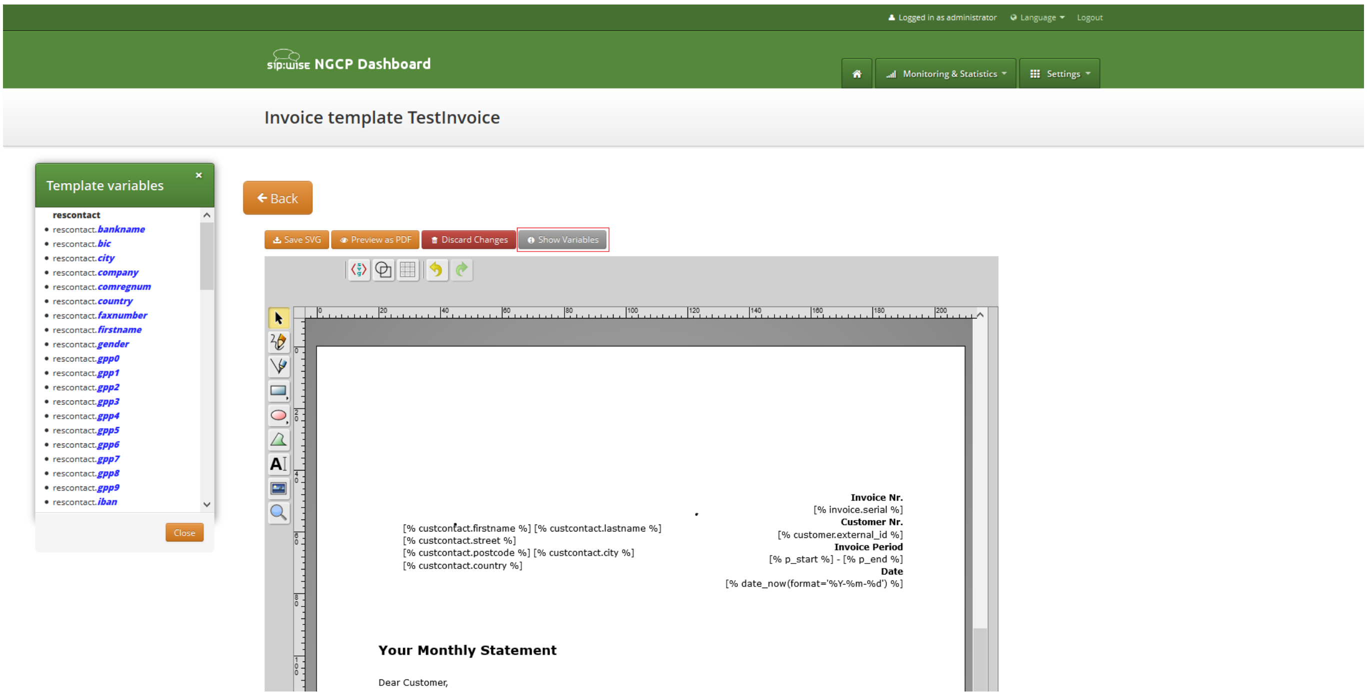
After registering new invoice template you can change invoice template structure in WYSIWYG SVG editor and preview result of the invoice generation based on the template.

7.18.2.2 Invoice Template Content

Invoice template is a XML SVG source, which describes content, look and position of the text lines, images or other invoice template elements. The sip:provider CE provides embedded WYSIWYG SVG editor svg-edit 2.6 to customize default template. The sip:provider CE svg-edit has some changes in layers management, image edit, user interface, but this basic introduction still may be useful.

Template refers to the owner reseller contact ("rescontact"), customer contract ("customer"), customer contact ("custcontact"), billing profile ("billprof"), invoice ("invoice") data as variables in the "[%%]" mark-up with detailed information accessed as field name after point e.g. [%invoice.serial%]. During invoice generation all variables or other special tokens in the "[% %]" mark-ups will be replaced by their database values.

Press on "Show variables" button on invoice template content page to see full list of variables with the fields:



You can add/change/remove embedded variables references directly in main svg-edit window. To edit text line in svg-edit main window double click on the text and place cursor on desired position in the text.

After implementation of the desired template changes, invoice template should be saved Section 7.18.2.3.

To return to the sip:provider CE invoice template **default** content you can press on the "Discard changes" button.

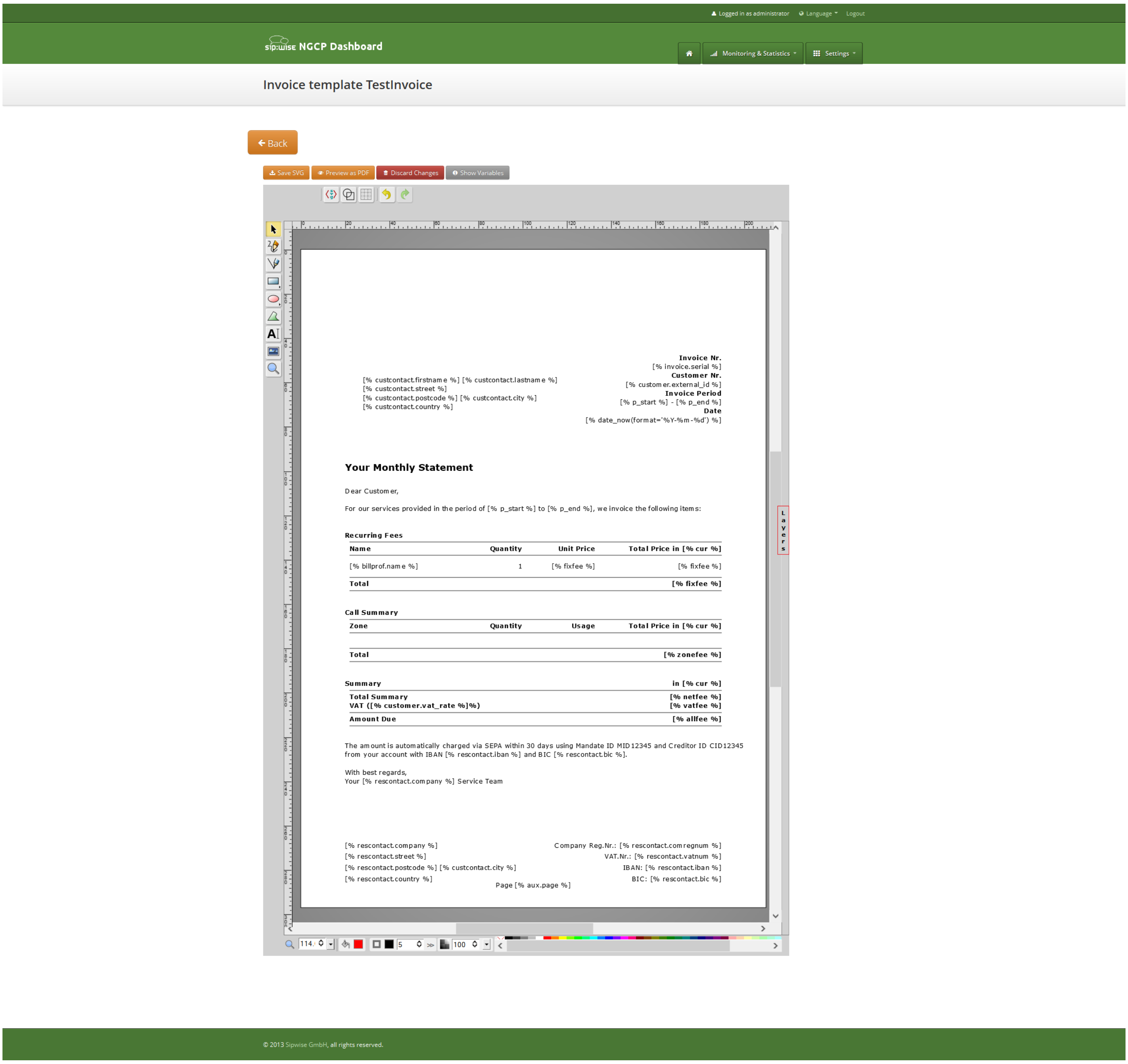
> **Important**
>
> "Discard changes" operation can't be undone.

Layers

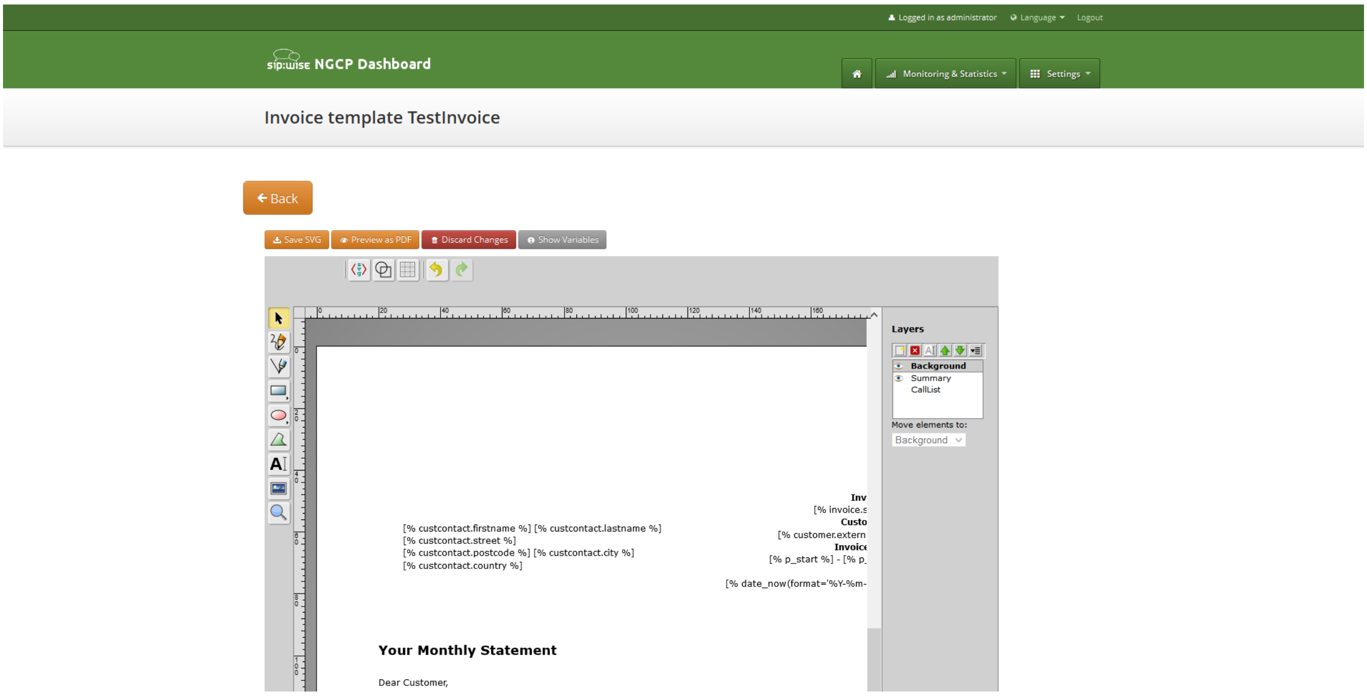
Default template contains three groups elements (<g/>), which can be thinked of as pages, or in terms of svg-edit - layers. Layers are:

- **Background**: special layer, which will be repeated as background for every other page of the invoice.
- **Summary**: page with a invoice summary.
- **CallList**: page with calls made in a invoice period. Is invisible by default.

To see all invoice template layers, press on "Layers" vertical sign on right side of the svg-edit interface:



Side panel with layers list will be shown.

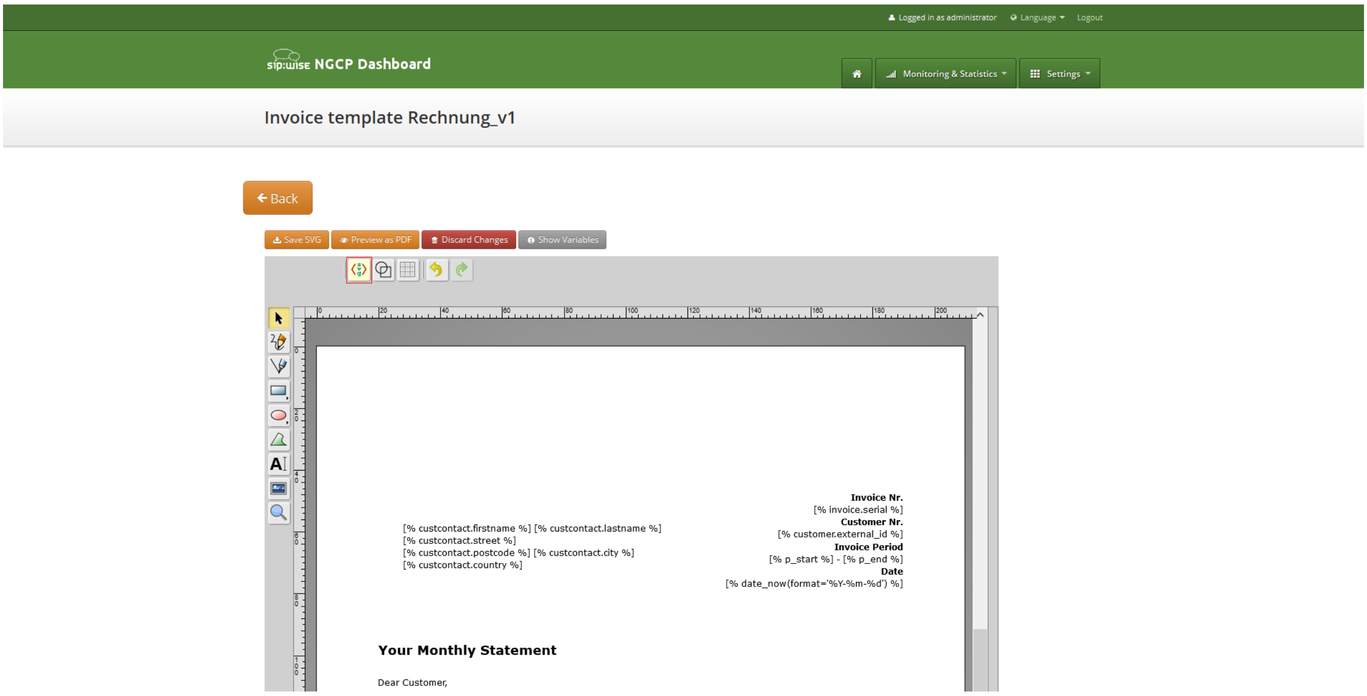


One of the layers is active, and its element can be edited in the main svg-edit window. Currently active layer's name is **bold** in the layers list. The layers may be visible or invisible. Visible layers have "eye" icon left of their names in the layers list.

To make a layer active, click on its name in the layers list. If the layer was invisible, its elements became visible on activation. Thus you can see mixed elements of some layers, then you can switch off visibility of other layers by click on their "eye" icons. It is good idea to keep visibility of the "Background" layer on, so look of the generated page will be seen.

Edit SVG XML source

Sometimes it may be convenient to edit svg source directly and svg-edit makes it possible to do it. After press on the <svg> icon in the top left corner of the svg-edit interface:



SVG XML source of the invoice template will be shown.

SVG source can be edited in place or just copy-pasted as usual text.

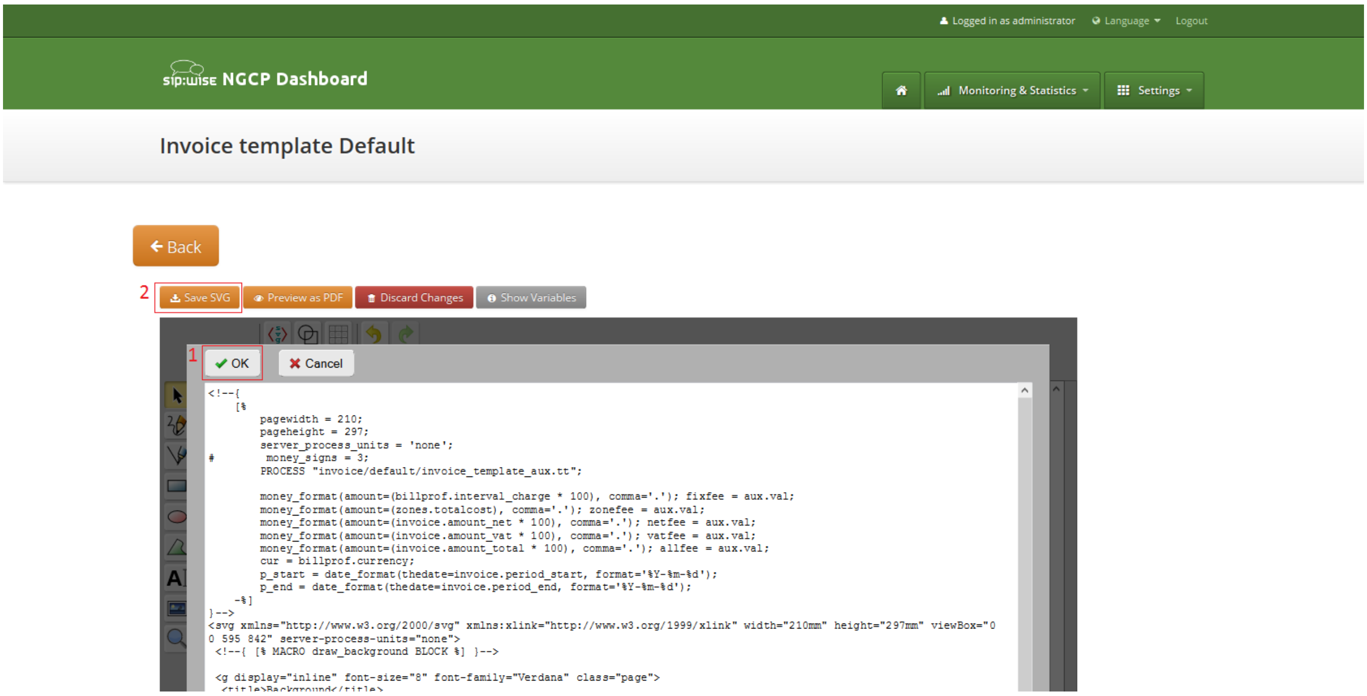
Note

Template keeps sizes and distances in pixels.

Important

When edit svg xml source, please change very carefully and thinkfully things inside special comment mark-up "<!--{ }-→". Otherwise invoice generation may be broken. Please be sure that document structure repeats default invoice template: has the same groups (<g/>) elements on the top level, text inside special comments mark-up "<!--{ }-→" preserved or changed appropriately, svg xml structure is correct.

To save your changes in the svg xml source, first press "OK" button on the top left corner of the source page:



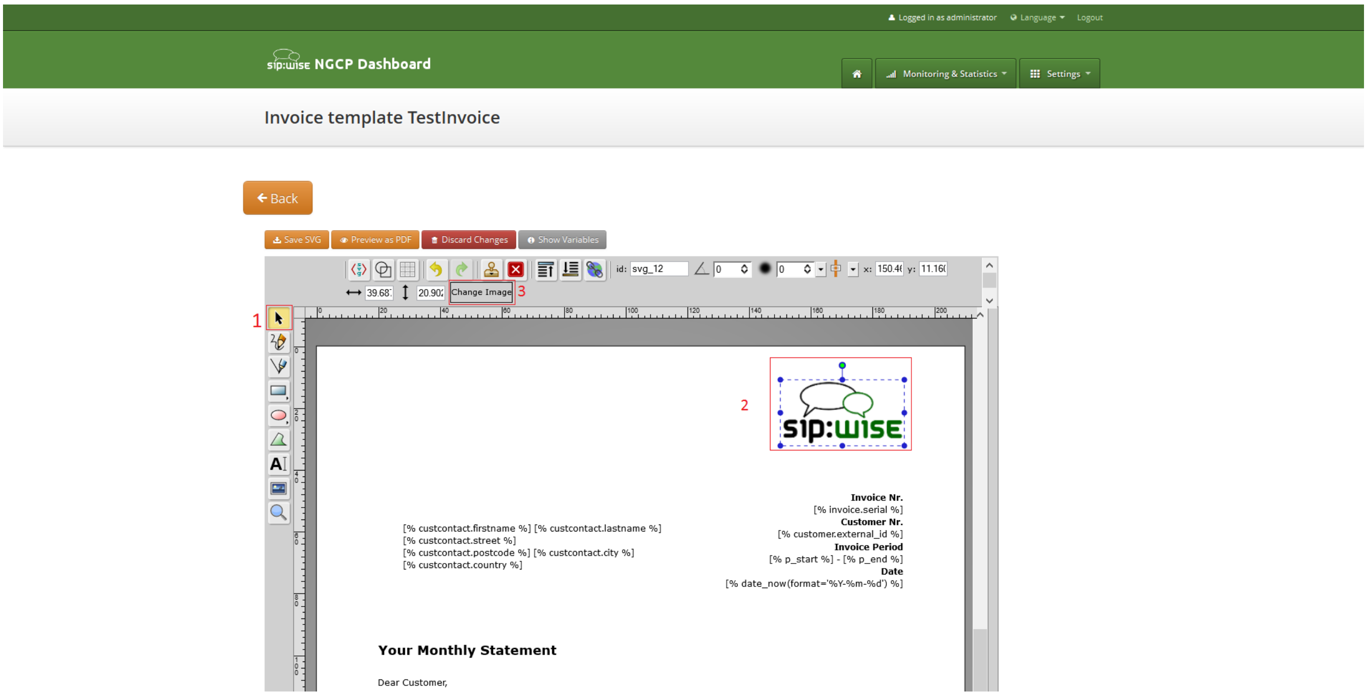
And then save invoice template changes Section 7.18.2.3.

Note

You can copy and keep the svg source of your template as a file on the disk before start experimenting with the template. Later you will be able to return to this version replacing svg source.

Change logo image

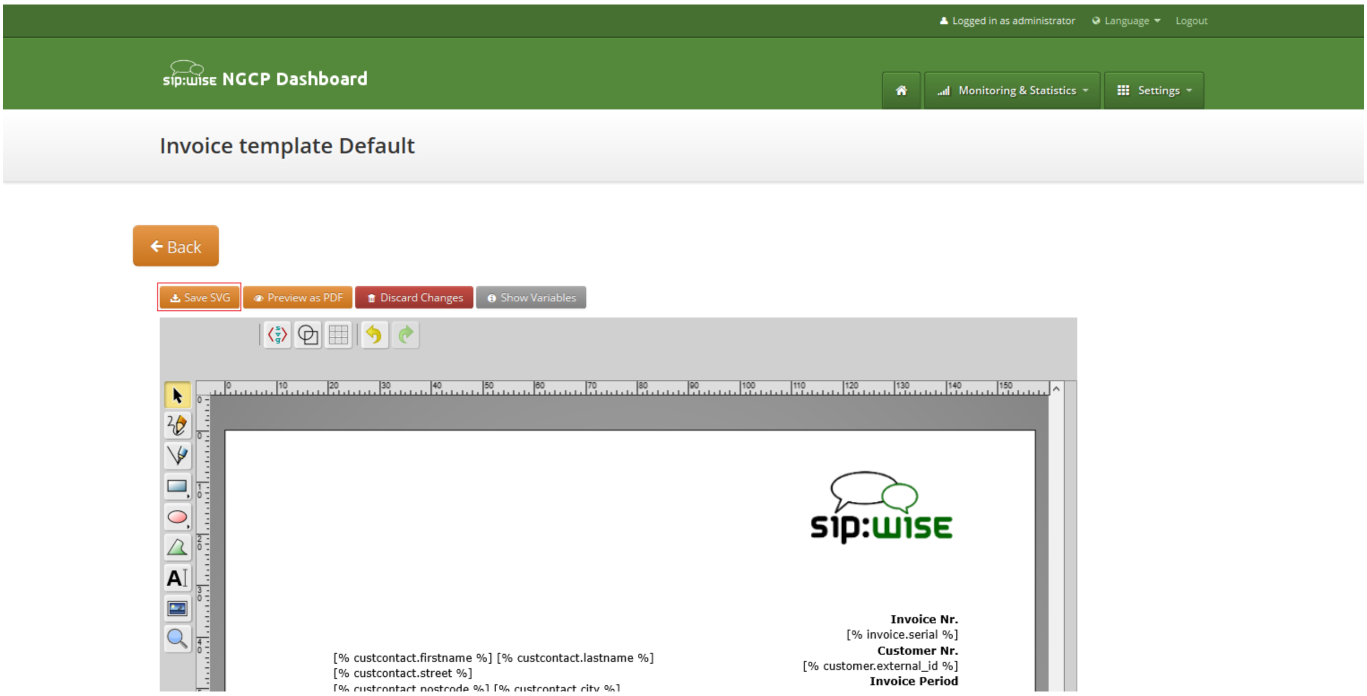
- Make sure that "Select tool" is active.
- Select default logo, clicking on the logo image.
- Press "Change image" button, which should appear on the top toolbar.



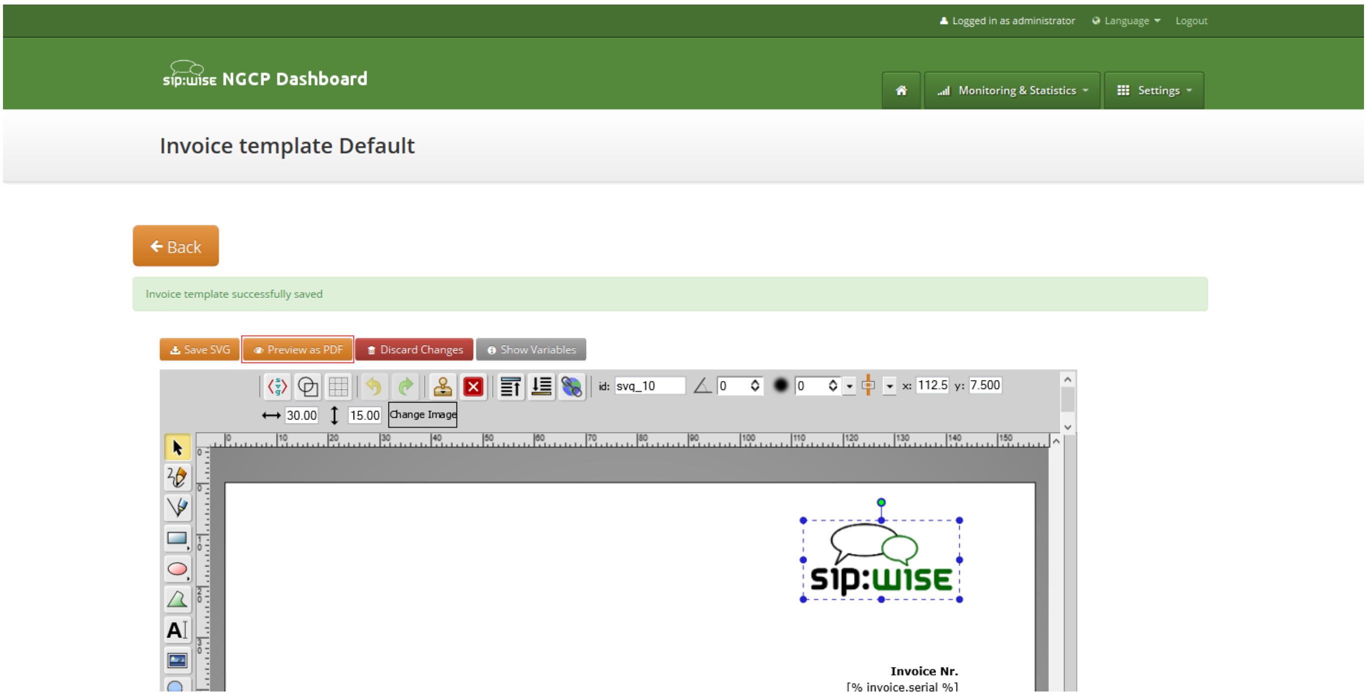
After image uploaded save invoice template changes Section 7.18.2.3.

7.18.2.3 Save and preview invoice template content

To save invoice template content changes press button "Save SVG".



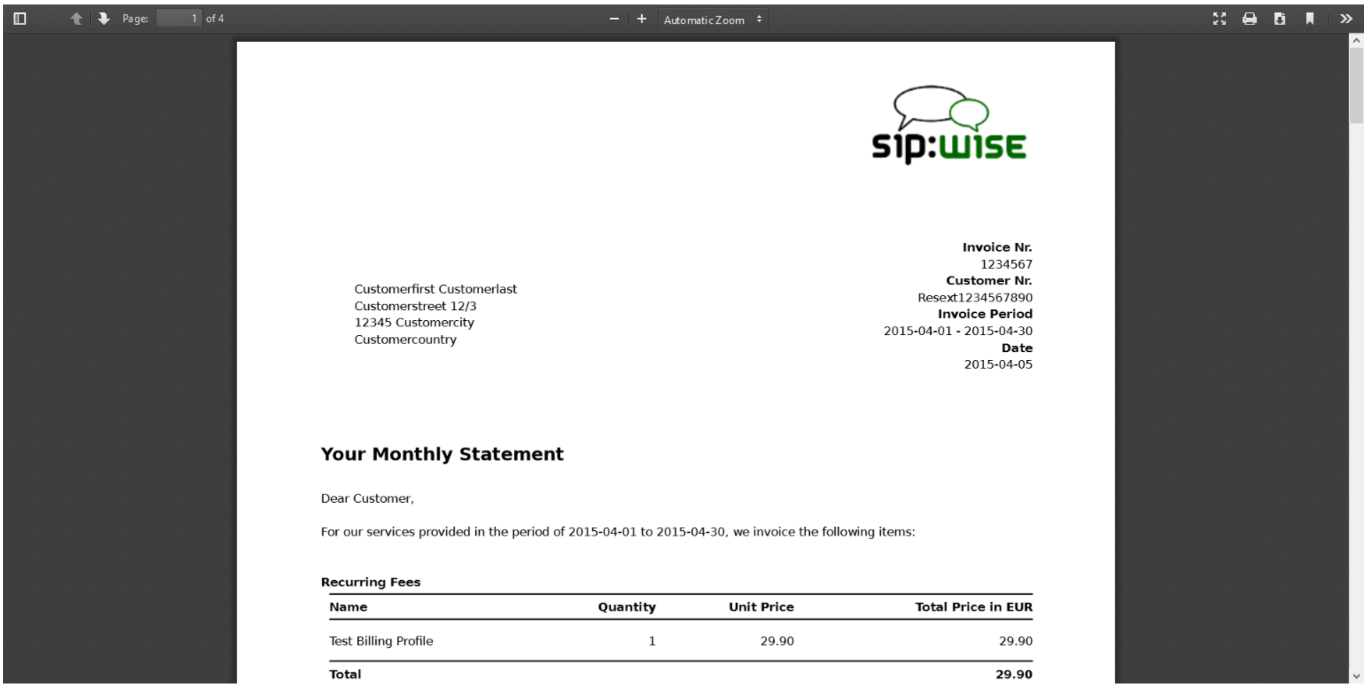
You will see message about successfully saved template. You can preview your invoice look in PDF format. Press on "Preview as PDF" button.



Invoice preview will be opened in the new window.

Note

Example fake data will be used for preview generation.



7.18.3 Invoices Generation

Besides generating invoices on demand using web interface, Sipwise NGCP contains an *invoice generator script* that allows for producing invoices automatically, at regular intervals, for all customers, using the *cron* system tool.

> **Warning**
> ***Automated invoice generation is deprecated since mr4.0 release of NGCP. The invoice generator script will damage billing records in the database.*** The rest of the description in "Invoices Generation" section is kept in the handbook for reference purposes only.

Script is located at: /usr/share/ngcp-panel/tools/generate_invoices.pl

In short:

- To generate and immediately send invoices for the previous month:

```
perl /usr/share/ngcp-panel/tools/generate_invoices.pl --send --prevmonth
```

- To generate invoices for the previous month without sending:

```
perl /usr/share/ngcp-panel/tools/generate_invoices.pl --prevmonth
```

- To send already generated invoices for the previous month:

```
perl /usr/share/ngcp-panel/tools/generate_invoices.pl --sendonly --prevmonth
```

- Regenerate invoices for the specified period:

```
perl /usr/share/ngcp-panel/tools/generate_invoices.pl --stime="2015-01-01  ↩
   00:00:00"  --etime="2015-01-31 00:00:00" --regenerate
```

Some not obvious options:

- **--allow_terminated* Generates invoices for the terminated contracts too.
- **--force_unrated* Generate invoices despite unrated calls existence in the specified generation period.
- **--no_empty* Skip invoices for the contracts without calls in the specified period and with null permanent fee for the billing profile.

To see all possible script options use --help or --man:

```
/usr/share/ngcp-panel/tools/generate_invoices.pl --man
```

Script will be run periodically as configured by the cron files. Cron files templates can be found at:

- /etc/ngcp-config/templates/etc/cron.d/ngcp-invoice-gen.tt2
- /etc/ngcp-config/templates/etc/cron.d/ngcp-invoice-gen.services

After applying your configuration cron file will be located at:

- /etc/cron.d/ngcp-invoice-gen

Script uses configuration file located at: /etc/ngcp-invoice-gen/invoice-gen.conf

Except common DB connection configuration following specific options can be defined in the config file:

- **RESELLER_ID** *1,2,3,. . . N*

Comma separated resellers id. Invoice generation will be performed only for the specified resellers.

- **CLIENT_CONTRACT_ID** *1,2,3,. . . N*

Comma separated customers id. Invoice generation will be performed only for the specified customers.

- **STIME** *YYYY-mm-DD HH:MM:SS*

Usually is not necessary. Script option --prevmonth will define correct start and end time for the previous month billing period. Generated invoices will include all calls with call start time more then STIME value and less the ETIME value.

- **ETIME** *YYYY-mm-DD HH:MM:SS*

Usually is not necessary. Script option --prevmonth will define correct start and end time for the previous month billing period. Generated invoices will include all calls with call start time more then STIME value and less the ETIME value.

- **SEND** *[0/1]*

Generated invoices will be immediately sent to the customers.

- **RESEND** *[0/1]*

Invoices, already sent to the customers, will be sent again.

- **REGENERATE** *[0/1]*

Already presented invoices files will be generated again. Otherwise they will stay intouched.

- **ALLOW_TERMINATED** *[0/1]*

Generate invoices for the already terminated customers too.

- **ADMIN_EMAIL** *your@email.com*

Purposed for notifications about invoices generation fails. Not in use now.

All generated invoices can be seen in the invoice management interface Section 7.18.1.

On request each invoice will be sent to the proper customer as e-mail with the invoice PDF in the attachment. Letter content is defined by the invoice email template.

7.19 Email Reports and Notifications

7.19.1 Email events

The sip:provider CE makes it possible to customize content of the emails sent on the following actions:

- Web password reset requested. Email will be sent to the subscriber, whom password was requested for resetting. If the subscriber doesn't have own email, letter will be sent to the customer, who owns the subscriber.
- New subscriber created. Email will be sent to the newly created subscriber or to the customer, who owns new subscriber.
- Letter with the invoice. Letter will be sent to the customer.

7.19.2 Initial template values and template variables

Default email templates for each of the email events are inserted on the initial sip:provider CE database creation. Content of the default template is described in the corresponding sections. Default email templates aren't linked to any reseller and can't be changed through sip:provider CE Panel. They will be used to initialize default templates for the newly created reseller.

Each email template refers to the values from the database using special mark-ups "[%" and "%]". Each email template has fixed set of the variables. Variables can't be added or changed without changes in the sip:provider CE Panel code.

7.19.3 Password reset email template

Email will be sent after subscriber or subscriber administrator requested password reset for the subscriber account. Letter will be sent to the subscriber. If subscriber doesn't have own email, letter will be sent to the customer owning the subscriber.

Default content of the password reset email template is:

Template name	**passreset_default_email**
From	default@sipwise.com
Subject	Password reset email

Body	<pre>Dear Customer,
	
	Please go to [%url%] to set your password and log into your self-care ↩
	    interface.
	Your faithful Sipwise system
	
	--
	This is an automatically generated message. Do not reply.</pre>

Following variables will be provided to the email template:

- [%**url**%]: specially generated url where subscriber can define his new password.
- [%**subscriber**%]: username@domain of the subscriber, which password was requested for reset.

7.19.4 New subscriber notification email template

Email will be sent on the new subscriber creation. Letter will be sent to the newly created subscriber if it has an email. Otherwise, letter will be sent to the customer who owns the subscriber.

> **Note**
> By default email content template is addressed to the customer. Please consider this when create the subscriber with an email.

Template name	**subscriber_default_email**
From	default@sipwise.com
Subject	Subscriber created

Body	<pre>Dear Customer,
	
	A new subscriber [%subscriber%] has been created for you.
	
	Your faithful Sipwise system
	
	--
	This is an automatically generated message. Do not reply.</pre>

Following variables will be provided to the email template:

- [%**url**%]: specially generated url where subscriber can define his new password.
- [%**subscriber**%]: username@domain of the subscriber, which password was requested for reset.

7.19.5 Invoice email template

Template name	**invoice_default_email**
From	default@sipwise.com
Subject	Invoice #[%invoice.serial%] from [%invoice.period_start_obj.ymd%] to [%invoice.period_end_obj.ymd%]

Body	`Dear Customer,`
	
	`Please find your invoice #[%invoice.serial%] for [%invoice.period_start_obj.month_name%], [%invoice.period_start_obj.year%] in attachment letter.`
	
	`Your faithful Sipwise system`
	
	`--`
	`This is an automatically generated message. Do not reply.`

Variables passed to the email template:

- [%**invoice**%]: container variable for the invoice information.

Invoice fields

- [%invoice.**serial**%]
- [%invoice.**amount_net**%]
- [%invoice.**amount_vat**%]
- [%invoice.**amount_total**%]
- [%invoice.**period_start_obj**%]
- [%invoice.**period_end_obj**%]

The fields [%invoice.period_start_obj%] and [%invoice.period_end_obj%] provide methods of the perl package DateTime for the invoice start date and end date. Further information about DateTime can be obtained from the package documentation: man DateTime

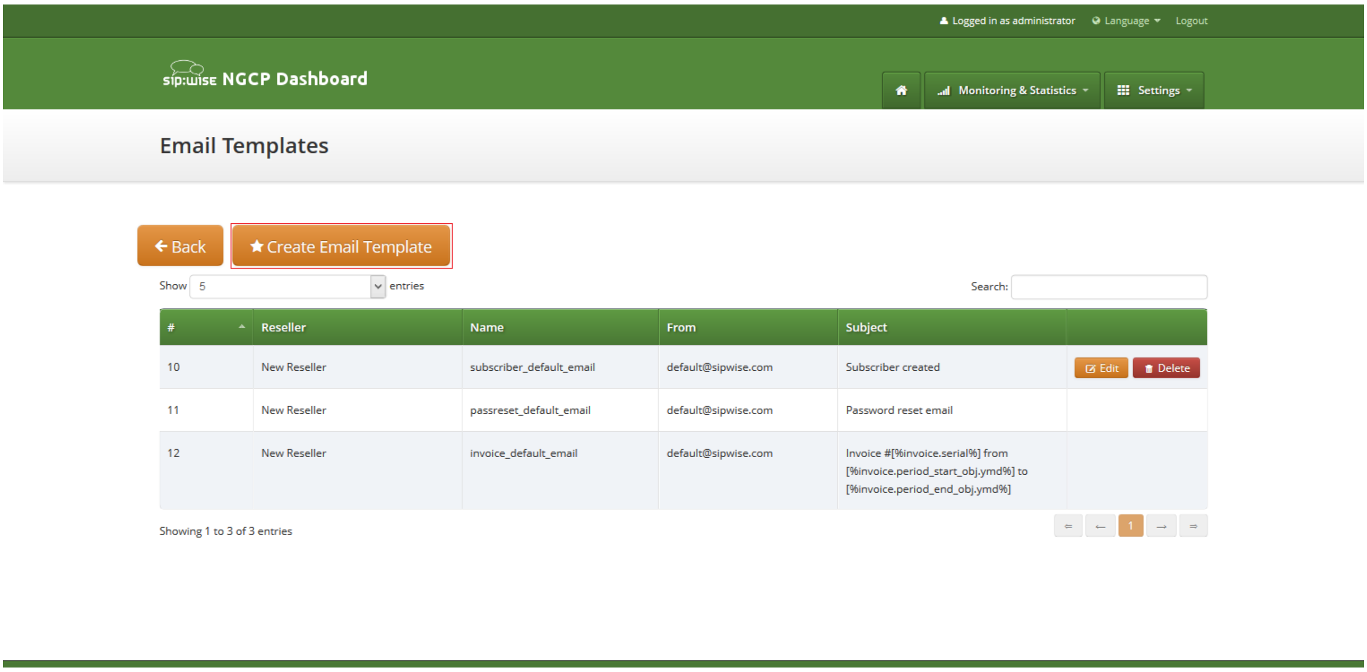
- [%**provider**%]: container variable for the reseller contact. All database contact values will be available.
- [%**client**%]: container variable for the customer contact.

Contact fields example for the "provider". Replace "provider" to client to access proper "customer" contact fields.

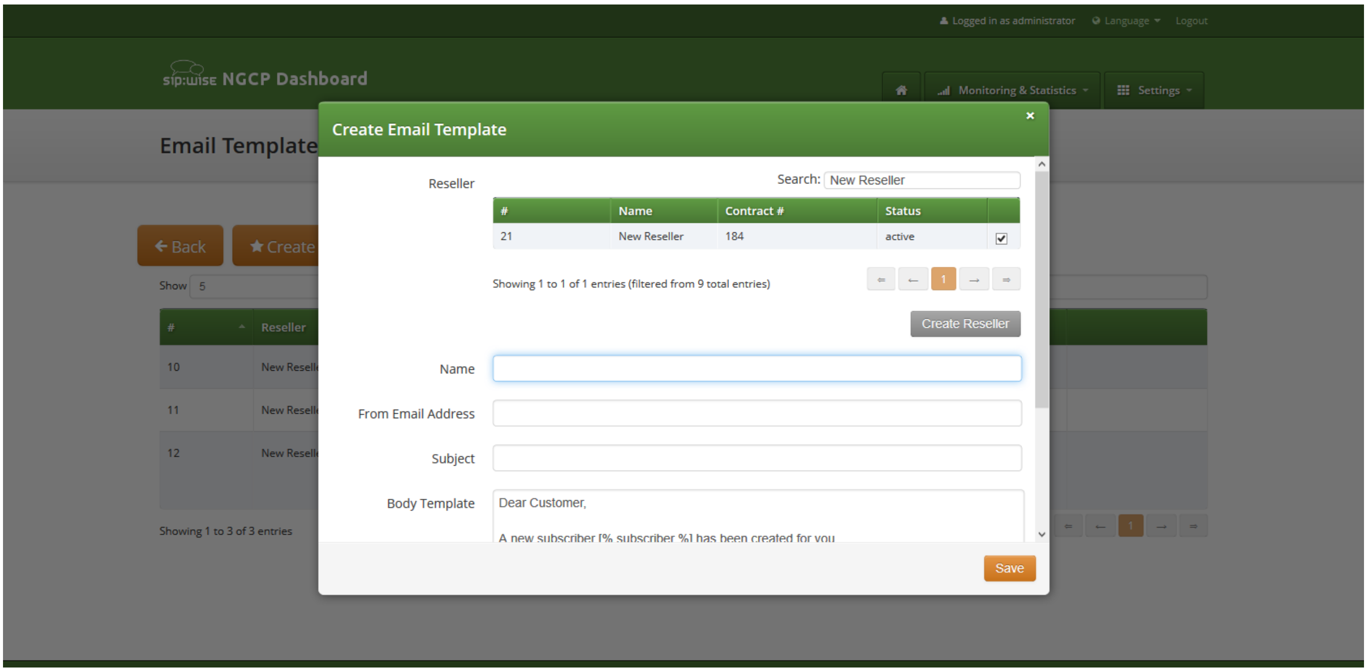
- [%provider.**gender**%]
- [%provider.**firstname**%]
- [%provider.**lastname**%]
- [%provider.**comregnum**%]
- [%provider.**company**%]
- [%provider.**street**%]
- [%provider.**postcode**%]
- [%provider.**city**%]
- [%provider.**country**%]
- [%provider.**phonenumber**%]
- [%provider.**mobilenumber**%]
- [%provider.**email**%]
- [%provider.**newsletter**%]
- [%provider.**faxnumber**%]
- [%provider.**iban**%]
- [%provider.**bic**%]
- [%provider.**vatnum**%]
- [%provider.**bankname**%]
- [%provider.**gpp0** - provider.**gpp9**%]

7.19.6 Email templates management

Email templates linked to the resellers can be customized in the email templates management interface. For the administrative account email templates of all the resellers will be shown. Respectively for the reseller account only owned email templates will be shown.



To create create new email template press button "Create Email Template".



On the email template form all fields are mandatory:

- **Reseller**: reseller who owns this email template.
- **Name**: currently only email template with the following names will be considered by the sip:provider CE on the appropriate event Section 7.19.1 :
  - passreset_default_email;
  - subscriber_default_email;

– invoice_default_email;

- **From Email Address**: email address which will be used in the From field in the letter sent by the sip:provider CE.
- **Subject**: Template of the email subject. Subject will be processed with the same template variables as the email body.
- **Body**: Email text template. Will be processed with appropriate template variables.

7.20 The Vertical Service Code Interface

Vertical Service Codes (VSC) are codes a user can dial on his phone to provision specific features for his subscriber account. The format is `*<code>*<value>` to activate a specific feature, and `#<code>` or `#<code>#` to deactivate it. The *code* parameter is a two-digit code, e.g. `72`. The *value* parameter is the value being set for the corresponding feature.

Important

The *value* user input is normalized using the Rewrite Rules Sets assigned to domain as described in Section 6.7.

By default, the following codes are configured for setting features. The examples below assume that there is a domain rewrite rule normalizing the number format *0<ac><sn>* to *<cc><ac><sn>* using *43* as country code.

- **72** - enable *Call Forward Unconditional* e.g. to 431000 by dialing `*72*01000`, and disable it by dialing `#72`.
- **90** - enable *Call Forward on Busy* e.g. to 431000 by dialing `*90*01000`, and disable it by dialing `#90`.
- **92** - enable *Call Forward on Timeout* e.g. after 30 seconds of ringing to 431000 by dialing `*92*30*01000`, and disable it by dialing `#92`.
- **93** - enable *Call Forward on Not Available* e.g. to 431000 by dialing `*93*01000`, and disable it by dialing `#93`.
- **50** - set *Speed Dial Slot*, e.g. set slot 1 to 431000 by dialing `*50*101000`, which then can be used by dialing `*1`.
- **55** - set *One-Shot Reminder Call* e.g. to 08:30 by dialing `*55*0830`.
- **31** - set *Calling Line Identification Restriction* for one call, e.g. to call 431000 anonymously dial `*31*01000`.
- **32** - enable *Block Incoming Anonymous Calls* by dialing `*32*`, and disable it by dialing `#32`.
- **80** - call using *Call Block Override PIN*, number should be prefixed with a block override PIN configured in admin panel to disable the outgoing user/admin block list and NCOS level for a call. For example, when override PIN is set to 7890, dial `*80*789001000` to call 431000 bypassing block lists.

7.20.1 Configuration of Vertical Service Codes

You can change any of the codes (but not the format) in */etc/ngcp-config/config.yml* in the section *sems→vsc*. After the changes, execute *ngcpcfg apply 'changed VSC codes'*.

Caution

If you have the EMTAs under your control, make sure that the specified VSCs don't overlap with EMTA-internal VSCs, because the VSC calls must be sent to the NGCP via SIP like normal telephone calls.

7.20.2 Voice Prompts for Vertical Service Code Configuration

Table 7: VSC Voice Prompts

Prompt Handle	Related VSC	Message
vsc_error	any	An error has occurred. Please try again later.
vsc_invalid	wrong code	Invalid feature code.
reject_vsc	any	Vertical service codes are disabled for this line.
vsc_cfu_on	72 (Call Forward Unconditional)	Your unconditional call forward has successfully been activated.
vsc_cfu_off	72 (Call Forward Unconditional)	Your unconditional call forward has successfully been deactivated.
vsc_cfb_on	90 (Call Forward Busy)	Your call forward on busy has successfully been activated.
vsc_cfb_off	90 (Call Forward Busy)	Your call forward on busy has successfully been deactivated.
vsc_cft_on	92 (Call Forward on Timeout)	Your call forward on ring timeout has successfully been activated.
vsc_cft_off	92 (Call Forward on Timeout)	Your call forward on ring timeout has successfully been deactivated.
vsc_cfna_on	93 (Call Forward on Not Available)	Your call forward while not reachable has successfully been activated.
vsc_cfna_off	93 (Call Forward on Not Available)	Your call forward while not reachable has successfully been deactivated.
vsc_speeddial	50 (Speed Dial Slot)	Your speed dial slot has successfully been stored.
vsc_reminder_on	55 (One-Shot Reminder Call)	Your reminder has successfully been activated.
vsc_reminder_off	55 (One-Shot Reminder Call)	Your reminder has successfully been deactivated.
vsc_blockinclir_on	32 (Block Incoming Anonymous Calls)	Your rejection of anonymous calls has successfully been activated.
vsc_blockinclir_off	32 (Block Incoming Anonymous Calls)	Your rejection of anonymous calls has successfully been deactivated.

7.21 Handling WebRTC Clients

WebRTC is an open project providing browsers and mobile applications with Real-Time Communications (RTC) capabilities. Configuring your platform to offer WebRTC is quite easy and straightforward. This allows you to have a SIP-WebRTC bridge in place and make audio/video call towards normal SIP users from WebRTC clients and vice versa. Sip Provider listens, by default, on the following WebSockets and WebSocket Secure: `ws://your-ip:5060/ws`, `wss://your-ip:5061/ws` and `wss://your-ip:1443/wss/sip/`.

The WebRTC subscriber is just a normal subscriber which has just a different configuration in his Preferences. You need to change the following preferences under *Subscribers→Details→Preferences→NAT and Media Flow Control*:

- **use_rtpproxy**: Always with rtpproxy as additional ICE candidate
- **transport_protocol**: RTP/SAVPF (encrypted SRTP with RTCP feedback)

The `transport_protocol` setting may change, depending on your WebRTC client/browser configuration. Supported protocols are the following:

- Transparent (Pass through using the client's transport protocol)
- RTP/AVP (Plain RTP)
- RTP/SAVP (encrypted SRTP)
- RTP/AVPF (RTP with RTCP feedback)
- RTP/SAVPF (encrypted SRTP with RTCP feedback)
- UDP/TLS/RTP/SAVP (Encrypted SRTP using DTLS)
- UDP/TLS/RTP/SAVPF (Encrypted SRTP using DTLS with RTCP feedback)

Warning
The below configuration is enough to handle a WebRTC client/browser. As mentioned, you may need to tune a little bit your `transport_protocol` configuration, depending on your client/browser settings.

In order to have a bridge between normal SIP clients (using plain RTP for example) and WebRTC client, the normal SIP clients' preferences have to have the following configuration:

transport_protocol: RTP/AVP (Plain RTP)

This will teach Sip Provider to translate between Plain RTP and RTP/SAVPF when you have calls between normal SIP clients and WebRTC clients.

7.22 XMPP and Instant Messaging

Instant Messaging (IM) based on XMPP comes with sip:provider CE out of the box. sip:provider CE uses `prosody` as internal XMPP server. Each subscriber created on the platform have assigned a XMPP user, reachable already - out of the box - by using the same SIP credentials. You can easily open an XMPP client (e.g. Pidgin) and login with your SIP `username@domain` and your SIP `password`. Then, using the XMPP client options, you can create your buddy list by adding your buddies in the format `user@domain`.

7.23 Call Recording

7.23.1 Introduction to Call Recording Function

Sipwise NGCP provides an opportunity to record call media content and store that in files. This function is available since mr5.3.1 version of the sip:provider CE.

Some characteristics of the Call Recording:

- Call Recording function can store both unidirectional (originating either from caller, or from callee) or bidirectional (combined) streams from calls, resulting in 1, 2 or 3 physical files as output
- The location and format of the files is configurable.
- File storage is planned to occur on an NFS shared folder.
- Activation of call recording may happen generally for a *Domain / Peer / Subscriber* through the NGCP admin web interface.

> **Important**
> NGCP's Call Recording function is not meant for individual call interception purpose! Sipwise provides its Lawful Interception solution for that use case.

- Querying or deletion of existing recordings may happen through the REST API.
- Listing recordings of a subscriber is possible on NGCP's admin web interface.

The Call Recording function is implemented using NGCP's ***rtpengine*** module.

> **Note**
> There are 2 *rtpengine* daemons employed when call recording is enabled and active. The *main rtpengine* takes care of forwarding media packets between caller and callee, as usual, while the *secondary rtpengine* (recording) daemon is responsible for storing call data streams in the file system.

Call Recording is disabled by default. Enabling and configuration of Call Recording takes place in 2 steps:

1. Enabling the feature on the NGCP by setting configuration parameters in the main `config.yml` configuration file.
2. Activating the feature for a *Domain / Peer / Subscriber.*

7.23.2 Information on Files and Directories

NGCP's Call Recording function uses an **NFS shared folder** to save recorded streams.

> **Important**
>
> Since call data amount may be huge (depending, of course, on the number and duration of calls), it is *strongly not recommended* to store recorded streams on NGCP's local disks. However if you *have to* store recorded streams as files in the local filesystem, please contact Sipwise Support team in order to get the proper configuration of Call Recording function.

The NFS share gets mounted during startup of the recording daemon. If the NFS share cannot be mounted for some reason, the recording daemon will not start.

Filenames have the format: `<call_ID>-<random>-<SSRC>.<extension>`, where:

- `call_ID`: SIP Call-ID of the call being recorded
- `random`: is a string of random characters, unique for each recorded call. It's purpose is to avoid possible filename collisions if a Call-ID ever gets reused.
- `SSRC`: is the RTP SSRC for unidirectional recordings, or "mix" for the bidirectional (combined) audio.
- `extension`: is either "mp3" or "wav", depending on the configuration (rtpproxy.recording.output_format)

There might be 1, 2 or 3 files produced as recorded streams. The **number of files** depends on the configuration:

1. `rtpproxy.recording.output_mixed = 'yes'` (combined stream required)

   `rtpproxy.recording.output_single = 'no'` (unidirectional streams not required)
2. `rtpproxy.recording.output_mixed = 'no'` (combined stream not required)

   `rtpproxy.recording.output_single = 'yes'` (unidirectional streams required)
3. `rtpproxy.recording.output_mixed = 'yes'` (combined stream required)

   `rtpproxy.recording.output_single = 'yes'` (unidirectional streams required)

7.23.3 Configuration

The Call Recording function can be enabled and configured on the NGCP by changing the following configuration parameters in `config.yml` file:

```
rtpproxy:
  ...
  recording:
    enabled: no
    mp3_bitrate: '48000'
    nfs_host: 192.168.1.1
```

```
    nfs_remote_path: /var/recordings
    output_dir: /var/lib/rtpengine-recording
    output_format: wav
    output_mixed: yes
    output_single: yes
    resample: no
    resample_to: '16000'
    spool_dir: /var/spool/rtpengine
```

7.23.3.1 Enabling Call Recording

Enabling the function requires changing the value of `rtpproxy.recording.enabled` parameter to “yes”. In order to make the new configuration active, it’s necessary to do:

```
ngcpcfg apply 'Activated call recording'
```

Description of configuration parameters:

- enabled: when set to “yes” Call Recording function is enabled; default: “no”
- mp3_bitrate: the bitrate used when recording happens in MP3 format; default: "48000"
- nfs_host: IP address of the NFS host that provides storage space for recorded streams; default: "192.168.1.1"
- nfs_remote_path: the remote path (folder) where files of recorded streams are stored on the NFS share; default: "/var/recordings"
- output_dir: is the local mount point for the NFS share, and thus where the final audio files will be written; default: "/var/lib/rtpengine-recording"

> **Caution**
> Normally you don’t need to change the default setting. If you do change the value, please be aware that recorded files will be written by *root* user in that directory.

- output_format: possible values are “wav” (Wave) or “mp3” (MP3); default: “wav”
- output_mixed: “yes” means that there is a file that contains a mixed stream of caller and callee voice data; default: "yes"
- output_single: “yes” means that there is a separate file for each stream direction, i.e. for the streams originating from caller and callee; default: "yes"
- resample: when set to “yes” the call data stream will be resampled before storing it in the file; default: “no”
- resample_to: the sample rate used for resampling output; default: "16000"
- spool_dir: is the place for temporary metadata files that are used by the recording daemon and the main rtpengine daemon for their communication; default: "/var/spool/rtpengine"

> **Caution**
>
> You should not change the default setting unless you have a good reason to do so! Sipwise has thoroughly tested the Call Recording function with the default setting.

If Call Recording is enabled you can see 2 *rtpengine* processes running when checking the NGCP system state with *monit* tool:

```
root@sp1:/etc/ngcp-config# monit summary
...
Process 'lb'                        Running
Process 'rtpengine'                 Running
Process 'rtpengine-recording'       Running
Process 'voisniff-ng'               Running
...
```

7.23.3.2 Activating Call Recording

Activating Call Recording for e.g. a *Subscriber:* please use NGCP's admin web interface for this purpose. On the web interface one has to navigate as follows: *Settings* → *Subscribers* → *select subscriber Details* → *Preferences* → *NAT and Media Flow Control*. Afterwards the `record_call` option has to be enabled by pressing the *Edit* button and ticking the checkbox.

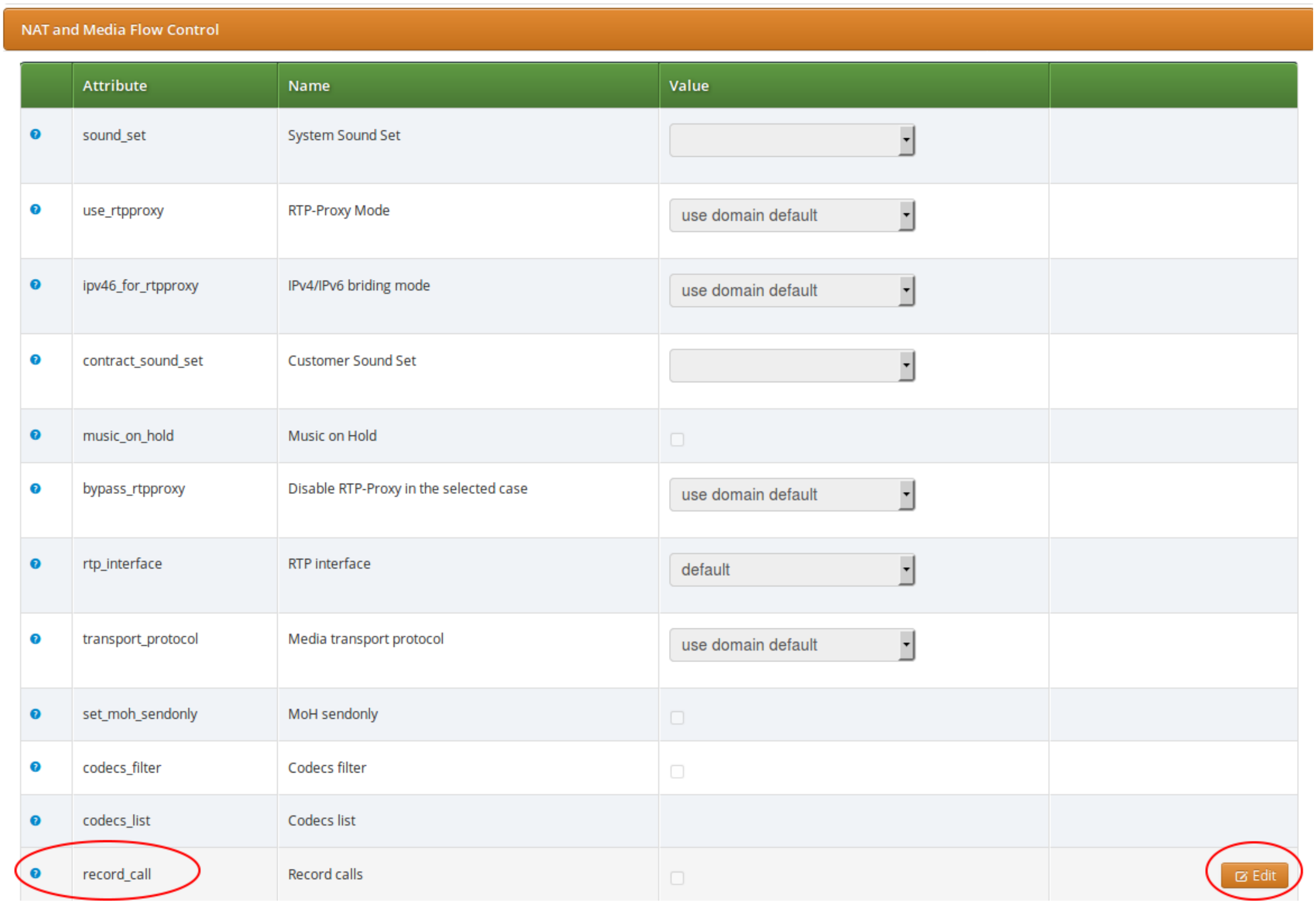


Figure 56: Activating Call Recording

Note

The call recording function may be activated for a single *Subscriber, a Domain and a Peer server* in the same way: *Preferences → NAT and Media Flow Control → record_call*. When activating call recording for a *Domain or Peer* this effectively activates the function for all subscribers that belong to the selected domain, and for all calls with a local endpoint going through the selected peer server, respectively.

It is possible to **list existing call recordings** of a *Subscriber* through the admin web interface of NGCP. In order to do so, please navigate to: *Settings → Subscribers → select subscriber Details → Call Recordings*

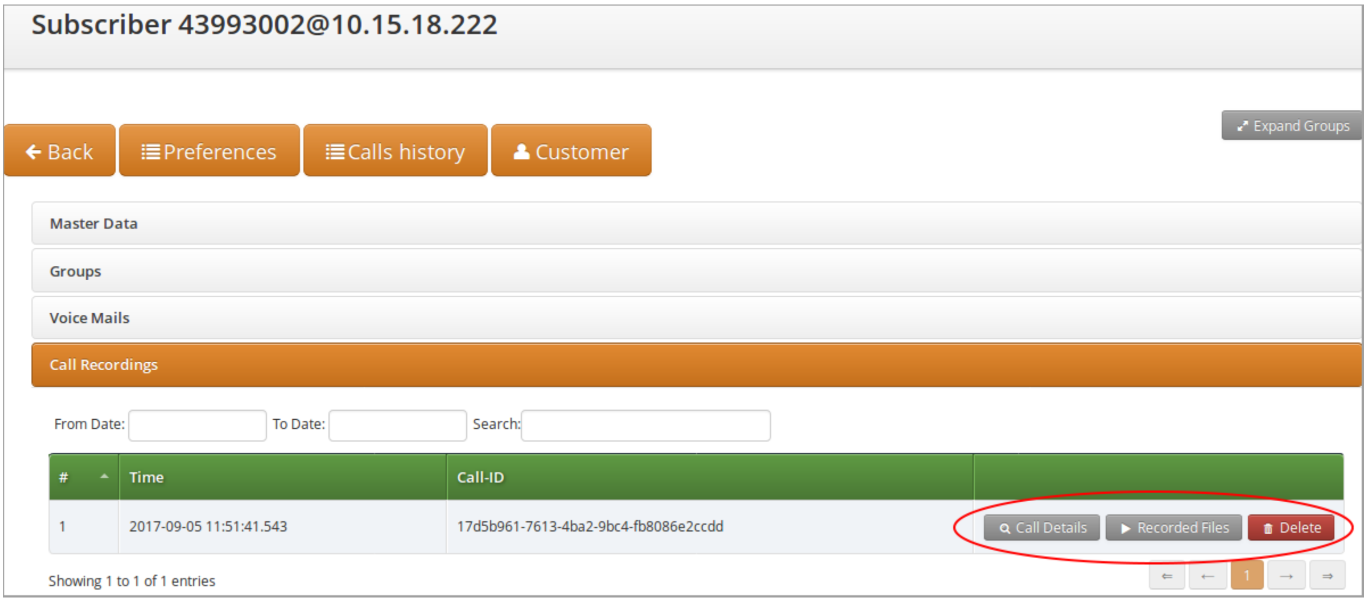


Figure 57: Listing Call Recordings

If you select an item in the list, besides the main properties such as the time of call and the SIP Call-ID, you can retrieve the details of the related call (press the *Call Details* button), get the list of recorded files (press the *Recorded Files* button) or *Delete* the recorded call.

When selecting *Call Details* you will see the most important accounting data of the call. Furthermore you can see the SIP *Call Flow* or the complete *Call Details* if you press the respective buttons.

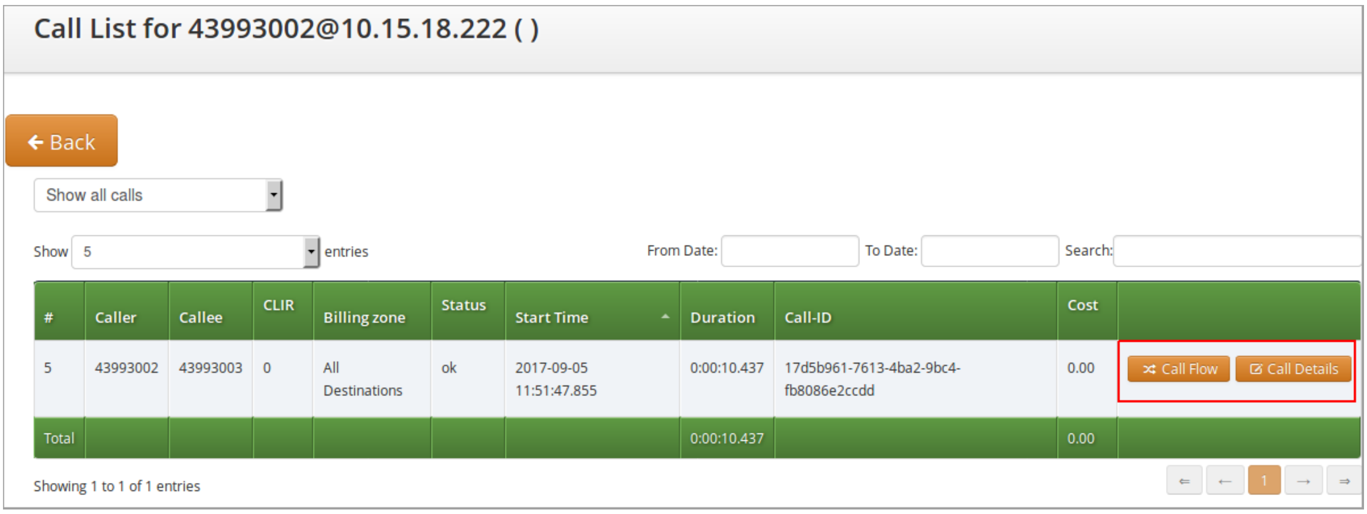


Figure 58: Listing Call Details for a Recording

When navigating to *Recorded Files* of a call you will be presented with a list of files. For each file item:

- `type` of stream is shown, that can be either "mixed" (combined voice data), or "single" (voice data of caller or callee)

- file `format` is shown, that can be either "wav", or "mp3"
- you can download the file by pressing the *Play* button

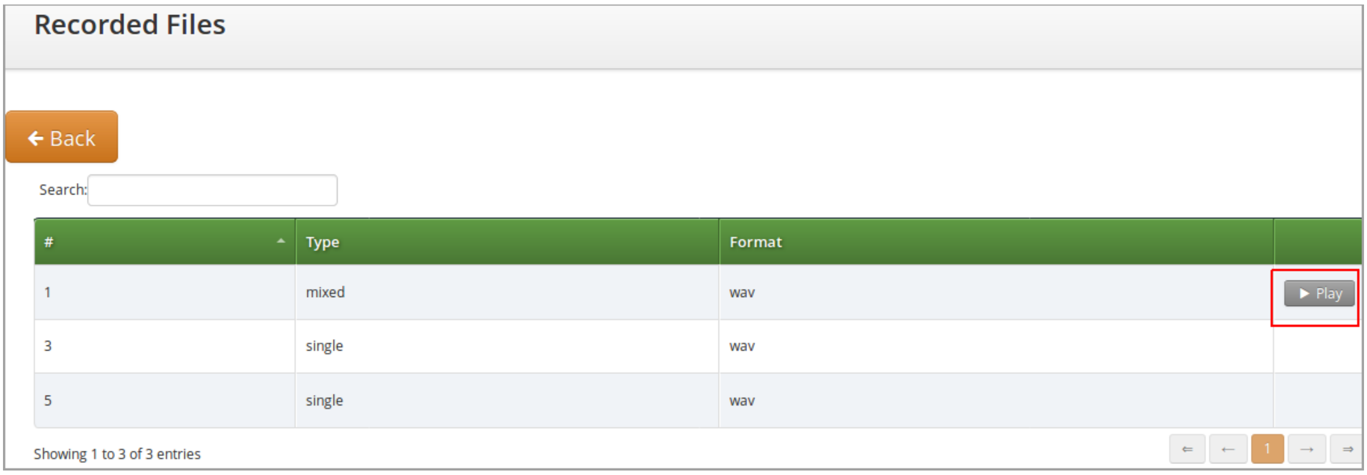


Figure 59: Listing Files for a Recording

7.23.4 REST API

The NGCP REST API provides methods for querying and deletion of existing recording data. The full documentation of the available API methods is available on the admin web interface of the NGCP, as usual.

The following API methods are provided for managing Call Recordings:

- CallRecordings:
  - Provides information about the calls recorded in the system; can also be used to delete a recording entry
  - accessible by the path: `/api/callrecordings` (collection) or `/api/callrecordings/id` (single item)
  - Supported HTTP methods: OPTIONS, GET, DELETE
- CallRecordingStreams:
  - Provides information about recorded streams, such as start time, end time, format, mixed/single type, etc.; can also be used to delete a recorded stream
  - accessible by the path: `/api/callrecordingstreams` (collection) or `/api/callrecordingstreams/id` (single item)
  - Supported HTTP methods: OPTIONS, GET, DELETE
- CallRecordingFiles:
  - Provides information about recorded streams, such as start time, end time, format, mixed/single type, etc.; additionally returns the file content too
  - accessible by the path: `/api/callrecordingfiles` (collection) or `/api/callrecordingfiles/id` (single item)
  - Supported HTTP methods: OPTIONS, GET

7.24 SMS (Short Message Service) on Sipwise NGCP

Starting with its mr5.0.1 release, Sipwise NGCP offers *short messaging service* to its local subscribers. The implementation is based on a widely used software module: *Kannel*, and it needs to interact with a mobile operator's SMSC in order to send and receive SMs for the local subscribers. The data exchange with SMSC uses *SMPP* (Short Message Peer-to-Peer) protocol.

SMS directions:

- incoming / received: the destination of the SM is a local subscriber on the NGCP
- outgoing / sent: the SM is submitted by a local subscriber

Note
The Sipwise NGCP behaves as a short message client towards the SMSC of a mobile operator. This means every outgoing SM will be forwarded to the SMSC, and every incoming SM will reach the NGCP through an SMSC.

The architecture of the SMS components of NGCP and their interactions to other elements is depicted below, on a *sip:carrier* system:

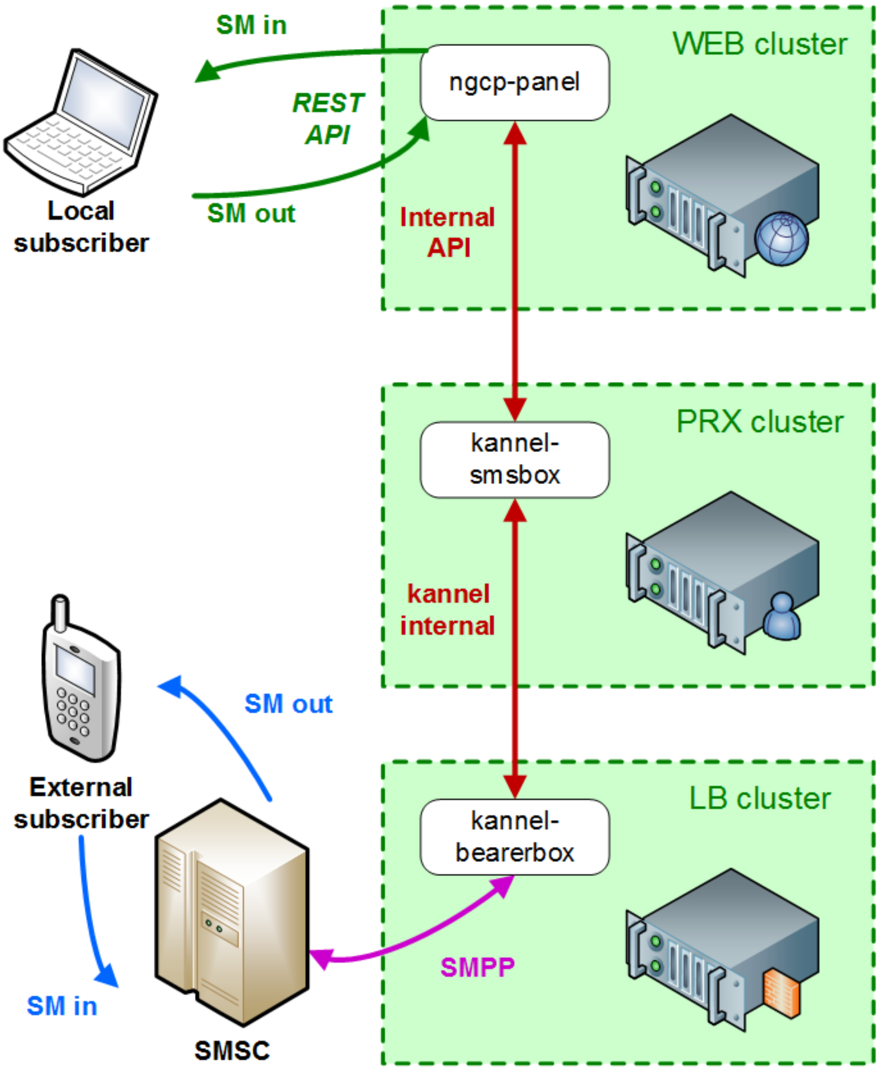


Figure 60: SMS Interactions on NGCP

Note

For the *sip:provider CE and PRO* NGCP installations: the *Kannel* components and the *ngcp-panel* all run on the same single node. The description of SMS module will continue referring to a *sip:carrier* installation in the handbook.

There are 2 components of the SMS module:

- **SMS Box**: this component takes care of handling the messages locally, that means:
  - delivering them to subscribers (writing into database for later retrieval)
  - picking up the submitted SMs from the database and forwarding them to the *Bearer Box* component
- **Bearer Box**: this component manages the transmission of SMs between Sipwise NGCP and the mobile operator's SMSC

7.24.1 Configuration

7.24.1.1 Main Parameters

The SMS function of NGCP is disabled by default. In order to **enable SMS** you have to change the value of configuration parameter `sms.enable` to `yes` in the main configuration file (`/etc/ngcp-config/config.yml`).

The second step of configuration is related to the **SMSC** where NGCP will connect to. You have to set the following parameters:

- `sms.smsc.host`: IP address of the SMSC
- `sms.smsc.port`: Port number of the SMSC
- `sms.smsc.username`: Username for authentication on the SMSC
- `sms.smsc.password`: Password for authentication on the SMSC

Other parameters of the SMSC connection may also need to be changed from the default values, but this is specific to each deployment.

Then, as usual, you have to make the new configuration active:

```
$ ngcpcfg apply 'Enabled SMS'
```

7.24.1.2 Configuration Files of Kannel

There are a few configuration files for the *Kannel* module, namely:

- `/etc/default/ngcp-kannel`: determines which components of *Kannel* will be started. This is auto-generated from `/etc/ngcp-config/templates/etc/default/ngcp-kannel.tt2` file when SMS is enabled.
- `/etc/kannel/kannel.conf`: contains detailed configuration of *Kannel* components. This is auto-generated from `/etc/ngcp-config/templates/etc/kannel/kannel.conf.tt2` file when SMS is enabled.
- `/etc/logrotate.d/ngcp-kannel.conf`: configuration of *logrotate* for *Kannel* log files. This is auto-generated from `/etc/ngcp-config/templates/etc/logrotate.d/ngcp-kannel.conf.tt2` file when SMS is enabled.

Caution
Please do not change settings in the above mentioned template files, unless you have to tailor *Kannel* settings to your specific needs!

Finally: see the description of each configuration parameter in the appendix Section B.1.31.

7.24.1.3 Call Forwarding for SMS (CFS)

Any subscriber registered on NGCP can apply a call forwarding setting for short messages, referred to as "CFS" (Call Forward - SMS). If the CFS feature is enabled, he can receive the SMs on his mobile phone, for example, instead of retrieving the SMs through the REST API. This is much more convenient for users if they do not have an application on their smartphone or computer that could manage the SMs through the REST API.

In order to enable CFS you have to set the forwarding as usual on the admin web interface, or through the REST API. Navigate to *Subscribers* → *select one* → *Details* → *Preferences* → *Call Forwards* and press the *Edit* button.

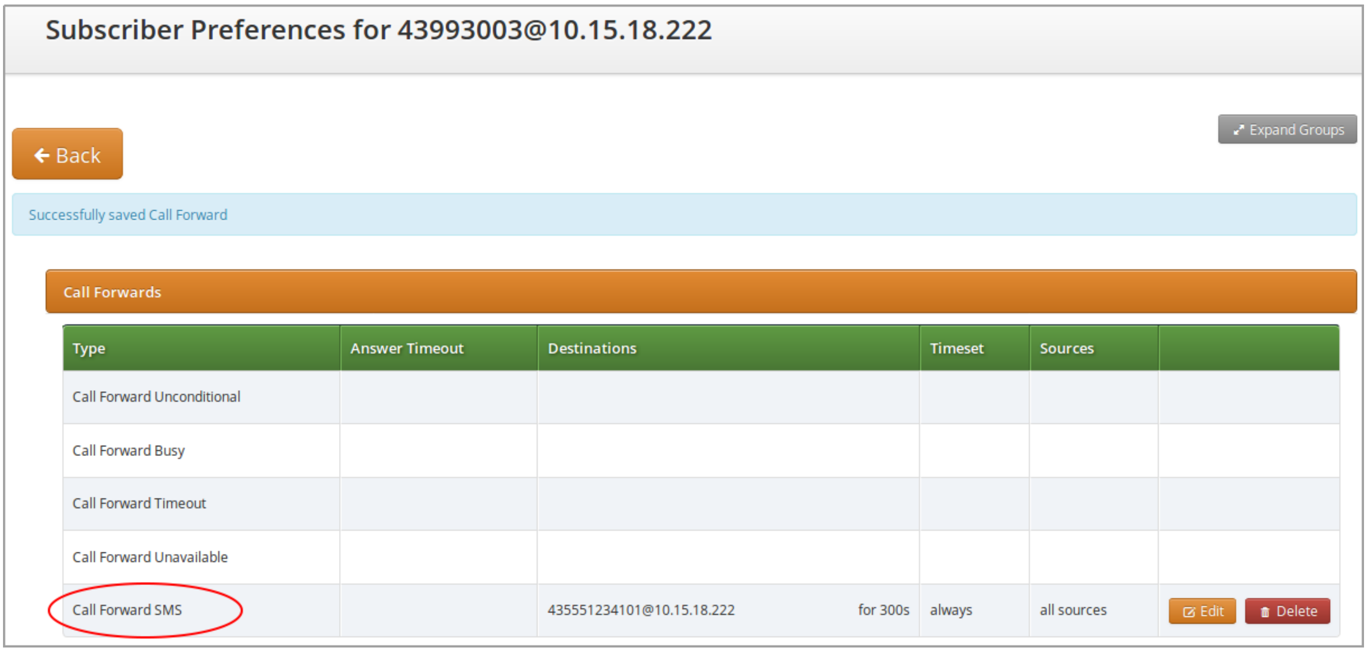


Figure 61: Call Forward for SMS

7.24.2 Monitoring, troubleshooting

7.24.2.1 Bearer Box (LB node of NGCP)

On the LB node you can see a **process** named **"bearerbox"**. This process has 2 **listening ports** assigned to it:

- 13000: this is the generic *Kannel* administration port, that belongs to the "core" component of Kannel.
- 13001: this is the communication port towards the *SMS Box* component running on PRX nodes of NGCP.

The *monit* tool also shows the *bearerbox* process in its status information:

```
$ monit summary
...
Process 'kannel-bearerbox'          Running
...
```

The following log files can provide information about the operation of *Bearer Box*:

- status messages and high level, short entries about sent and received messages: `/var/log/ngcp/kannel/kannel.log`

```
...
2017-09-26 08:57:32 [15922] [10] DEBUG: boxc_receiver: heartbeat with load value 0 ↩
    received
...
2017-09-26 11:12:06 [15922] [10] DEBUG: boxc_receiver: sms received
2017-09-26 11:12:06 [15922] [10] DEBUG: send_msg: sending msg to box: <192.168.1.4>
2017-09-26 11:12:06 [15922] [11] DEBUG: send_msg: sending msg to box: <192.168.1.4>
2017-09-26 11:12:06 [15922] [11] DEBUG: boxc_sender: sent message to <192.168.1.4>
2017-09-26 11:12:06 [15922] [10] DEBUG: boxc_receiver: got ack
...
```

- detailed information and message content of sent and received messages, link enquiries: `/var/log/kannel/smsc.log`

> **Note**
> Sent and received message examples shown here do not contain the full phone number and content for confidentiality reason.

  – Example received message:

```
...
2017-09-26 12:09:36 [15922] [6] DEBUG: SMPP[default_smsc]: Got PDU:
2017-09-26 12:09:36 [15922] [6] DEBUG: SMPP PDU 0x7f2274025070 dump:
2017-09-26 12:09:36 [15922] [6] DEBUG:   type_name: deliver_sm
2017-09-26 12:09:36 [15922] [6] DEBUG:   command_id: 5 = 0x00000005
2017-09-26 12:09:36 [15922] [6] DEBUG:   command_status: 0 = 0x00000000
2017-09-26 12:09:36 [15922] [6] DEBUG:   sequence_number: 11867393 = 0x00b51501
2017-09-26 12:09:36 [15922] [6] DEBUG:   service_type: NULL
2017-09-26 12:09:36 [15922] [6] DEBUG:   source_addr_ton: 2 = 0x00000002
2017-09-26 12:09:36 [15922] [6] DEBUG:   source_addr_npi: 1 = 0x00000001
2017-09-26 12:09:36 [15922] [6] DEBUG:   source_addr: "0660......."
2017-09-26 12:09:36 [15922] [6] DEBUG:   dest_addr_ton: 1 = 0x00000001
2017-09-26 12:09:36 [15922] [6] DEBUG:   dest_addr_npi: 1 = 0x00000001
2017-09-26 12:09:36 [15922] [6] DEBUG:   destination_addr: "43668......."
2017-09-26 12:09:36 [15922] [6] DEBUG:   esm_class: 0 = 0x00000000
2017-09-26 12:09:36 [15922] [6] DEBUG:   protocol_id: 0 = 0x00000000
2017-09-26 12:09:36 [15922] [6] DEBUG:   priority_flag: 0 = 0x00000000
2017-09-26 12:09:36 [15922] [6] DEBUG:   schedule_delivery_time: NULL
2017-09-26 12:09:36 [15922] [6] DEBUG:   validity_period: NULL
2017-09-26 12:09:36 [15922] [6] DEBUG:   registered_delivery: 0 = 0x00000000
2017-09-26 12:09:36 [15922] [6] DEBUG:   replace_if_present_flag: 0 = 0x00000000
2017-09-26 12:09:36 [15922] [6] DEBUG:   data_coding: 3 = 0x00000003
2017-09-26 12:09:36 [15922] [6] DEBUG:   sm_default_msg_id: 0 = 0x00000000
2017-09-26 12:09:36 [15922] [6] DEBUG:   sm_length: 158 = 0x0000009e
```

```
2017-09-26 12:09:36 [15922] [6] DEBUG:   short_message:
2017-09-26 12:09:36 [15922] [6] DEBUG:    Octet string at 0x7f2274000f80:
2017-09-26 12:09:36 [15922] [6] DEBUG:      len:  158
2017-09-26 12:09:36 [15922] [6] DEBUG:      size: 159
2017-09-26 12:09:36 [15922] [6] DEBUG:      immutable: 0
2017-09-26 12:09:36 [15922] [6] DEBUG:      data: 5a <14 bytes> 46
2017-09-26 12:09:36 [15922] [6] DEBUG:      data: 72 <14 bytes> 68
2017-09-26 12:09:36 [15922] [6] DEBUG:      data: 61 <14 bytes> 67
2017-09-26 12:09:36 [15922] [6] DEBUG:      data: 20 <14 bytes> 57
2017-09-26 12:09:36 [15922] [6] DEBUG:      data: 65 <14 bytes> 63
2017-09-26 12:09:36 [15922] [6] DEBUG:      data: 68 <14 bytes> 73
2017-09-26 12:09:36 [15922] [6] DEBUG:      data: 2e <14 bytes> 61
2017-09-26 12:09:36 [15922] [6] DEBUG:      data: 6c <14 bytes> 73
2017-09-26 12:09:36 [15922] [6] DEBUG:      data: 3a <14 bytes> 73
2017-09-26 12:09:36 [15922] [6] DEBUG:      data: 4d <14 bytes> 6e
2017-09-26 12:09:36 [15922] [6] DEBUG:    Octet string dump ends.
2017-09-26 12:09:36 [15922] [6] DEBUG: SMPP PDU dump ends.
2017-09-26 12:09:36 [15922] [6] DEBUG: SMPP[default_smsc]: Sending PDU:
2017-09-26 12:09:36 [15922] [6] DEBUG: SMPP PDU 0x7f2274020790 dump:
2017-09-26 12:09:36 [15922] [6] DEBUG:   type_name: deliver_sm_resp
2017-09-26 12:09:36 [15922] [6] DEBUG:   command_id: 2147483653 = 0x80000005
2017-09-26 12:09:36 [15922] [6] DEBUG:   command_status: 0 = 0x00000000
2017-09-26 12:09:36 [15922] [6] DEBUG:   sequence_number: 11867393 = 0x00b51501
2017-09-26 12:09:36 [15922] [6] DEBUG:   message_id: NULL
2017-09-26 12:09:36 [15922] [6] DEBUG: SMPP PDU dump ends.
2017-09-26 12:09:36 [15922] [6] DEBUG: SMPP[default_smsc]: throughput (0.00,5.00)
...
```

– Example sent message:

```
...
2017-09-26 12:04:08 [15922] [6] DEBUG: SMPP[default_smsc]: throughput (0.00,5.00)
2017-09-26 12:04:08 [15922] [6] DEBUG: SMPP[default_smsc]: Manually forced source addr ↩
    ton = 1, source add npi = 1
2017-09-26 12:04:08 [15922] [6] DEBUG: SMPP[default_smsc]: Manually forced dest addr ton ↩
     = 1, dest add npi = 1
2017-09-26 12:04:08 [15922] [6] DEBUG: SMPP[default_smsc]: Sending PDU:
2017-09-26 12:04:08 [15922] [6] DEBUG: SMPP PDU 0x7f2274025070 dump:
2017-09-26 12:04:08 [15922] [6] DEBUG:   type_name: submit_sm
2017-09-26 12:04:08 [15922] [6] DEBUG:   command_id: 4 = 0x00000004
2017-09-26 12:04:08 [15922] [6] DEBUG:   command_status: 0 = 0x00000000
2017-09-26 12:04:08 [15922] [6] DEBUG:   sequence_number: 98163 = 0x00017f73
2017-09-26 12:04:08 [15922] [6] DEBUG:   service_type: NULL
2017-09-26 12:04:08 [15922] [6] DEBUG:   source_addr_ton: 5 = 0x00000005
2017-09-26 12:04:08 [15922] [6] DEBUG:   source_addr_npi: 0 = 0x00000000
2017-09-26 12:04:08 [15922] [6] DEBUG:   source_addr: "any"
2017-09-26 12:04:08 [15922] [6] DEBUG:   dest_addr_ton: 1 = 0x00000001
2017-09-26 12:04:08 [15922] [6] DEBUG:   dest_addr_npi: 1 = 0x00000001
```

```
2017-09-26 12:04:08 [15922] [6] DEBUG:   destination_addr: "43676......."
2017-09-26 12:04:08 [15922] [6] DEBUG:   esm_class: 3 = 0x00000003
2017-09-26 12:04:08 [15922] [6] DEBUG:   protocol_id: 0 = 0x00000000
2017-09-26 12:04:08 [15922] [6] DEBUG:   priority_flag: 0 = 0x00000000
2017-09-26 12:04:08 [15922] [6] DEBUG:   schedule_delivery_time: NULL
2017-09-26 12:04:08 [15922] [6] DEBUG:   validity_period: NULL
2017-09-26 12:04:08 [15922] [6] DEBUG:   registered_delivery: 0 = 0x00000000
2017-09-26 12:04:08 [15922] [6] DEBUG:   replace_if_present_flag: 0 = 0x00000000
2017-09-26 12:04:08 [15922] [6] DEBUG:   data_coding: 0 = 0x00000000
2017-09-26 12:04:08 [15922] [6] DEBUG:   sm_default_msg_id: 0 = 0x00000000
2017-09-26 12:04:08 [15922] [6] DEBUG:   sm_length: 23 = 0x00000017
2017-09-26 12:04:08 [15922] [6] DEBUG:   short_message:
2017-09-26 12:04:08 [15922] [6] DEBUG:    Octet string at 0x7f227400c460:
2017-09-26 12:04:08 [15922] [6] DEBUG:      len:  23
2017-09-26 12:04:08 [15922] [6] DEBUG:      size: 24
2017-09-26 12:04:08 [15922] [6] DEBUG:      immutable: 0
2017-09-26 12:04:08 [15922] [6] DEBUG:      data: 44 <14 bytes> 73
2017-09-26 12:04:08 [15922] [6] DEBUG:      data: 74 <5 bytes> 39
2017-09-26 12:04:08 [15922] [6] DEBUG:    Octet string dump ends.
2017-09-26 12:04:08 [15922] [6] DEBUG: SMPP PDU dump ends.
2017-09-26 12:04:08 [15922] [6] DEBUG: SMPP[default_smsc]: throughput (1.00,5.00)
...
```

- Example link enquiry:

```
...
2017-09-26 12:13:38 [15922] [6] DEBUG: SMPP[default_smsc]: throughput (0.00,5.00)
2017-09-26 12:13:38 [15922] [6] DEBUG: SMPP[default_smsc]: Got PDU:
2017-09-26 12:13:38 [15922] [6] DEBUG: SMPP PDU 0x7f2274020790 dump:
2017-09-26 12:13:38 [15922] [6] DEBUG:   type_name: enquire_link
2017-09-26 12:13:38 [15922] [6] DEBUG:   command_id: 21 = 0x00000015
2017-09-26 12:13:38 [15922] [6] DEBUG:   command_status: 0 = 0x00000000
2017-09-26 12:13:38 [15922] [6] DEBUG:   sequence_number: 90764 = 0x0001628c
2017-09-26 12:13:38 [15922] [6] DEBUG: SMPP PDU dump ends.
2017-09-26 12:13:38 [15922] [6] DEBUG: SMPP[default_smsc]: Sending PDU:
2017-09-26 12:13:38 [15922] [6] DEBUG: SMPP PDU 0x7f2274025070 dump:
2017-09-26 12:13:38 [15922] [6] DEBUG:   type_name: enquire_link_resp
2017-09-26 12:13:38 [15922] [6] DEBUG:   command_id: 2147483669 = 0x80000015
2017-09-26 12:13:38 [15922] [6] DEBUG:   command_status: 0 = 0x00000000
2017-09-26 12:13:38 [15922] [6] DEBUG:   sequence_number: 90764 = 0x0001628c
2017-09-26 12:13:38 [15922] [6] DEBUG: SMPP PDU dump ends.
2017-09-26 12:13:38 [15922] [6] DEBUG: SMPP[default_smsc]: throughput (0.00,5.00)
...
```

7.24.2.2 SMS Box (PRX node of NGCP)

On the PRX node you can see a **process** named **"smsbox"**. This process has a **listening port** assigned to it: 13002, that is the communication port towards the *Bearer Box* component running on LB nodes.

The *monit* tool also shows the *smsbox* process in its status information:

```
$ monit summary
...
Process 'kannel-smsbox'              Running
...
```

The following log files can provide information about the operation of *SMS Box*:

- sent and received messages using the API of WEB node: `/var/log/kannel/smsbox.log`

> **Note**
> Sent and received message examples shown here do not contain the full phone number and content for confidentiality reason.

  – Example sent message:

```
...
2017-09-26 12:16:42 [22763] [2] DEBUG: HTTP: Creating HTTPClient for '192.168.1.2'.
2017-09-26 12:16:42 [22763] [2] DEBUG: HTTP: Created HTTPClient area 0x7f5dcc000ad0.
2017-09-26 12:16:42 [22763] [3] INFO: smsbox: Got HTTP request </cgi-bin/sendsms> from ↩
    <192.168.1.3>
2017-09-26 12:16:42 [22763] [3] INFO: sendsms used by <sipwise>
2017-09-26 12:16:42 [22763] [3] INFO: sendsms sender:<sipwise:43668.......> ↩
    (192.168.1.3) to:<43676.......> msg:<...>
2017-09-26 12:16:42 [22763] [3] DEBUG: Stored UUID ab95eb45-1ec0-4932-9863-1a95609a025f
2017-09-26 12:16:42 [22763] [3] DEBUG: message length 52, sending 1 messages
2017-09-26 12:16:42 [22763] [3] DEBUG: Status: 202 Answer: <Sent.>
2017-09-26 12:16:42 [22763] [3] DEBUG: Delayed reply - wait for bearerbox
2017-09-26 12:16:42 [22763] [0] DEBUG: Got ACK (0) of ab95eb45-1ec0-4932-9863-1 ↩
    a95609a025f
2017-09-26 12:16:42 [22763] [0] DEBUG: HTTP: Destroying HTTPClient area 0x7f5dcc000ad0.
2017-09-26 12:16:42 [22763] [0] DEBUG: HTTP: Destroying HTTPClient for '192.168.1.3'.
...
```

  – Example received message:

```
...
2017-09-26 11:59:45 [22763] [5] INFO: Starting to service <...message content...> from ↩
    <+43676-------> to <+43668------->
2017-09-26 11:59:45 [22763] [10] DEBUG: Queue contains 0 pending requests.
2017-09-26 11:59:45 [22763] [10] DEBUG: HTTPS URL; Using SSL for the connection
2017-09-26 11:59:45 [22763] [10] DEBUG: Parsing URL 'https://192.168.1.2:1443/ ↩
    internalsms/receive?auth_token=fNLosMgwdNUrKvEfFMm9
```

```
&timestamp=2017-09-26+09:59:45&from=%2B43676-------&to=%2B43668-------&charset=UTF-8& ↩
    coding=0&text=...':
2017-09-26 11:59:45 [22763] [10] DEBUG:   Scheme: https://
2017-09-26 11:59:45 [22763] [10] DEBUG:   Host: 192.168.1.2
2017-09-26 11:59:45 [22763] [10] DEBUG:   Port: 1443
2017-09-26 11:59:45 [22763] [10] DEBUG:   Username: (null)
2017-09-26 11:59:45 [22763] [10] DEBUG:   Password: (null)
2017-09-26 11:59:45 [22763] [10] DEBUG:   Path: /internalsms/receive
2017-09-26 11:59:45 [22763] [10] DEBUG:   Query: auth_token=fNLosMgwdNUrKvEfFMm9& ↩
    timestamp=2017-09-26+09:59:45&from=%2B43676-------
&to=%2B43668-------&charset=UTF-8&coding=0&text=...
2017-09-26 11:59:45 [22763] [10] DEBUG:   Fragment: (null)
2017-09-26 11:59:45 [22763] [10] DEBUG: Connecting nonblocking to <192.168.1.2>
2017-09-26 11:59:45 [22763] [10] DEBUG: HTTP: Opening connection to '192.168.1.2:1443' ( ↩
    fd=31).
2017-09-26 11:59:45 [22763] [10] DEBUG: Socket connecting
2017-09-26 11:59:45 [22763] [9] DEBUG: Get info about connecting socket
2017-09-26 11:59:45 [22763] [9] DEBUG: HTTP: Sending request:
2017-09-26 11:59:45 [22763] [9] DEBUG: Octet string at 0x7f5dbc00f470:
2017-09-26 11:59:45 [22763] [9] DEBUG:   len:  382
2017-09-26 11:59:45 [22763] [9] DEBUG:   size: 1024
2017-09-26 11:59:45 [22763] [9] DEBUG:   immutable: 0
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 47 45 54 20 2f 69 6e 74 65 72 6e 61 6c 73 ↩
     6d 73   GET /internalsms
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 2f 72 65 63 65 69 76 65 3f 61 75 74 68 5f ↩
     74 6f   /receive?auth_to
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 6b 65 6e 3d ...  ↩
                                      ken=
                                                         ... 20 48 54 54 50 2f 31 2e 31 ↩
                                                              0d 0a         HTTP/1.1..
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 43 6f 6e 6e 65 63 74 69 6f 6e 3a 20 6b 65 ↩
     65 70   Connection: keep
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 2d 61 6c 69 76 65 0d 0a 55 73 65 72 2d 41 ↩
     67 65   -alive..User-Age
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 6e 74 3a 20 4b 61 6e 6e 65 6c 2f 31 2e 34 ↩
     2e 34   nt: Kannel/1.4.4
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 0d 0a 48 6f 73 74 3a 20 31 39 32 2e 31 36 ↩
     38 2e   ..Host: 192.168.
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 31 2e 32 3a 31 34 34 33 0d 0a 0d 0a  ↩
                1.2:1443....
2017-09-26 11:59:45 [22763] [9] DEBUG: Octet string dump ends.
2017-09-26 11:59:45 [22763] [9] DEBUG: HTTP: Status line: <HTTP/1.1 200 OK>
2017-09-26 11:59:45 [22763] [9] DEBUG: HTTP: Received response:
2017-09-26 11:59:45 [22763] [9] DEBUG: Octet string at 0x7f5dbc006970:
2017-09-26 11:59:45 [22763] [9] DEBUG:   len:  333
2017-09-26 11:59:45 [22763] [9] DEBUG:   size: 1024
2017-09-26 11:59:45 [22763] [9] DEBUG:   immutable: 0
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 53 65 72 76 65 72 3a 20 6e 67 69 6e 78 0d ↩
```

```
    0a 44   Server: nginx..D
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 61 74 65 3a 20 54 75 65 2c 20 32 36 20 53 ←
    65 70   ate: Tue, 26 Sep
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 20 32 30 31 37 20 30 39 3a 35 39 3a 34 35 ←
    20 47    2017 09:59:45 G
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 4d 54 0d 0a 43 6f 6e 74 65 6e 74 2d 54 79 ←
    70 65   MT..Content-Type
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 3a 20 74 65 78 74 2f 68 74 6d 6c 3b 20 63 ←
    68 61   : text/html; cha
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 72 73 65 74 3d 75 74 66 2d 38 0d 0a 43 6f ←
    6e 74   rset=utf-8..Cont
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 65 6e 74 2d 4c 65 6e 67 74 68 3a 20 30 0d ←
    0a 43   ent-Length: 0..C
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 6f 6e 6e 65 63 74 69 6f 6e 3a 20 6b 65 65 ←
    70 2d   onnection: keep-
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 61 6c 69 76 65 0d 0a 53 65 74 2d 43 6f 6f ←
    6b 69   alive..Set-Cooki
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 65 3a 20 6e 67 63 70 5f 70 61 6e 65 6c 5f ←
    73 65   e: ngcp_panel_se
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 73 73 69 6f 6e 3d 34 35 30 32 64 64 66 65 ←
    31 62   ssion=4502ddfe1b
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 63 31 65 33 39 30 65 30 64 36 66 39 64 34 ←
    37 30   c1e390e0d6f9d470
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 35 30 37 62 64 64 33 61 65 32 36 62 64 63 ←
    3b 20   507bdd3ae26bdc;
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 70 61 74 68 3d 2f 3b 20 65 78 70 69 72 65 ←
    73 3d   path=/; expires=
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 54 75 65 2c 20 32 36 2d 53 65 70 2d 32 30 ←
    31 37   Tue, 26-Sep-2017
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 20 31 30 3a 35 39 3a 34 35 20 47 4d 54 3b ←
    20 48    10:59:45 GMT; H
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 74 74 70 4f 6e 6c 79 0d 0a 58 2d 43 61 74 ←
    61 6c   ttpOnly..X-Catal
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 79 73 74 3a 20 35 2e 39 30 30 37 35 0d 0a ←
    53 74   yst: 5.90075..St
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 72 69 63 74 2d 54 72 61 6e 73 70 6f 72 74 ←
    2d 53   rict-Transport-S
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 65 63 75 72 69 74 79 3a 20 6d 61 78 2d 61 ←
    67 65   ecurity: max-age
2017-09-26 11:59:45 [22763] [9] DEBUG:   data: 3d 31 35 37 36 38 30 30 30 0d 0a 0d 0a  ←
               =15768000....
2017-09-26 11:59:45 [22763] [9] DEBUG: Octet string dump ends.
2017-09-26 11:59:45 [22763] [6] WARNING: Tried to set Coding field, denied.
2017-09-26 11:59:45 [22763] [6] INFO: No reply sent, denied.
2017-09-26 11:59:55 [22763] [9] DEBUG: HTTP: Server closed connection, destroying it  ←
   <192.168.1.2:1443:1::><0x7f5db0000b20><fd:31>.
...
```

- short log of sent/received messages: `/var/log/kannel/smsbox-access.log`

```
...
2017-09-26 12:39:18 SMS HTTP-request sender:+43680------- request: '' url: 'https ↩
    ://192.168.1.2:1443/internalsms/receive?
auth_token=fNLosMgwdNUrKvEfFMm9&timestamp=2017-09-26+10:39:18&from=%2B43680-------&to=%2 ↩
    B43668-------&charset=UTF-8&coding=0
&text=<...message content...>' reply: 200 '<< successful >>'
...
2017-09-26 12:41:54 send-SMS request added - sender:sipwise:43668------- 192.168.1.3  ↩
    target:43680-------- request: '<...message content...>'
...
```

7.24.3 REST API

Handling of short messages from the user perspective happens with the help of NGCP's REST API. There is a dedicated resource: `https://<IP of WEB node>:1443/api/sms` that allows you to:

- Get a **list of sent and received messages**. This is achieved by sending a GET request on the `/api/sms` collection, as in the following example:

```
curl -i -X GET -H 'Connection: close' --cert NGCP-API-client-certificate.pem --cacert ca- ↩
    cert.pem \
  'https://example.org:1443/api/sms/?page=1&rows=10'
```

- **Retrieve an SM** (both sent and received). This is achieved by sending a GET request for a specific `/api/sms/id` item, as in the following example:

```
curl -i -X GET -H 'Connection: close' --cert NGCP-API-client-certificate.pem --cacert ca- ↩
    cert.pem 'https://example.org:1443/api/sms/1'
```

- **Send a new message** from a local subscriber. This is achieved by sending a POST request for the `/api/sms` collection, as in the following example:

```
curl -i -X POST -H 'Connection: close' -H 'Content-Type: application/json' --cert NGCP-API ↩
    -client-certificate.pem --cacert ca-cert.pem \
  'https://example.org:1443/api/sms/'  --data-binary '{"callee" : "43555666777", " ↩
      subscriber_id" : 4, "text" : "test"}'
```

As always, the full documentation of the REST API resources is available on the admin web interface of NGCP: `https://<IP of WEB node>:1443/api/#sms`

8 Customer Self-Care Interface and Menus

There are two ways for end users to maintain their subscriber settings: via the *Customer Self-Care Web Interface* and via *Vertical Service Codes* using their SIP phones.

8.1 The Customer Self-Care Web Interface

The NGCP provides a web panel for end users (CSC panel) to maintain their subscriber accounts, which is running on *https://<ngcp-ip>*. Every subscriber can log in there, change subscriber feature settings, view their call lists, retrieve voicemail messages and trigger calls using the click-to-dial feature.

8.1.1 Login Procedure

To log into the CSC panel, the end user has to provide his full web username (e.g. `user1@1.2.3.4`) and the web password defined in Section 6.3. Once logged in, he can change his web password in the *Account* section. This will NOT change his SIP password, so if you control the end user devices, you can auto-provision the SIP password into the device and keep it secret, and just hand over the web password to the customer. This way, the end user will only be able to place calls with this auto-provisioned device and not with an arbitrary soft-phone, but can nonetheless manage his account via the CSC panel.

8.1.2 Site Customization

As an operator (as well as a Reseller), you can change the branding logo of the Customer Self-Care (CSC) panel and the available languages on the CSC panel. This is possible via the admin web interface.

8.1.2.1 Changing the Logo

For changing the branding logo on a reseller's admin web page and on the CSC panel you just need to access the web interface **as Administrator** and navigate to *Reseller* menu. Once there click on the *Details* button for your selected reseller, finally select *Branding*.

In order to do the same **as Reseller**, login on the admin web interface with the reseller's web credentials, then access the *Panel Branding* menu.

The web panel customisation happens as follows:

1. Press the *Edit Branding* button to start the customisation process.
2. Press the *Browse* button to select an image for the new logo:

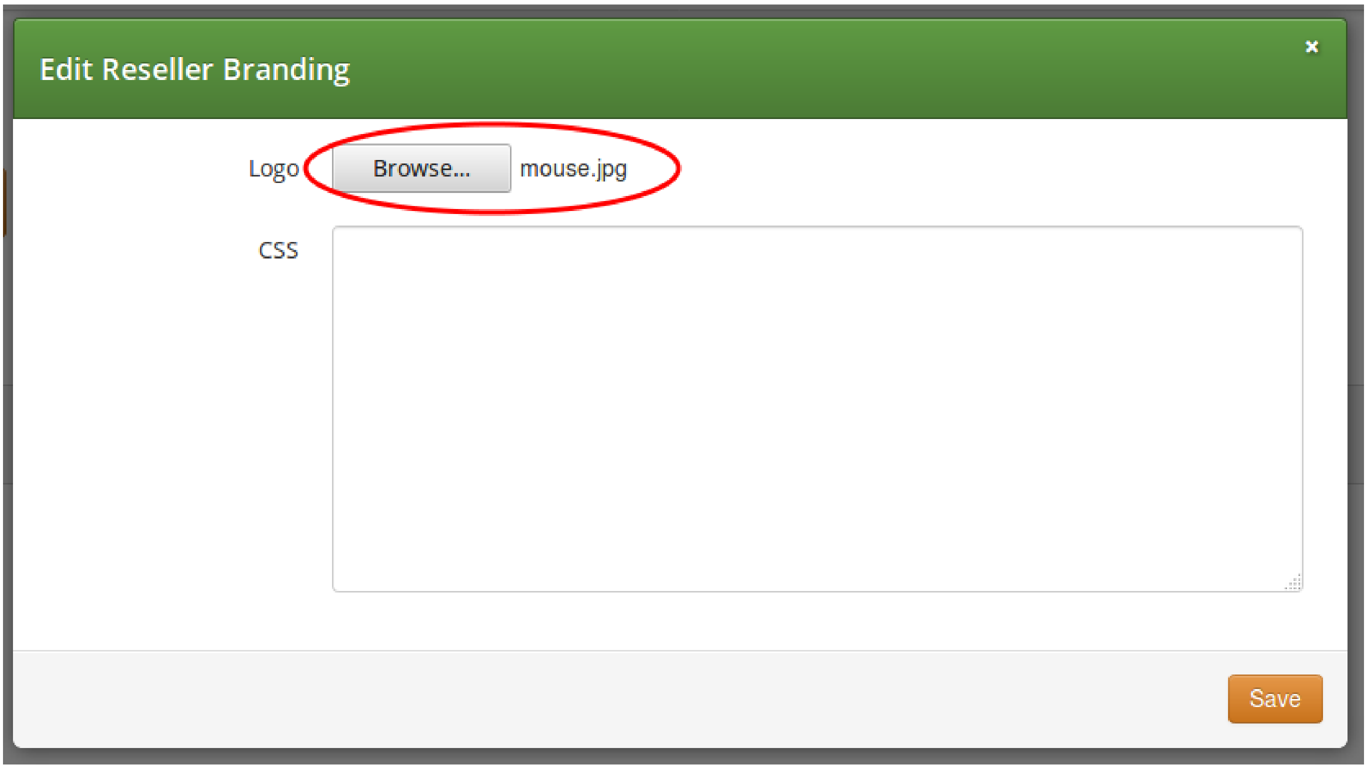


Figure 62: CSC Customisation Step 1: Select an image

3. Press the *Save* button to save changes.

4. Select and copy the auto-generated CSS code from the text box below the uploaded image:

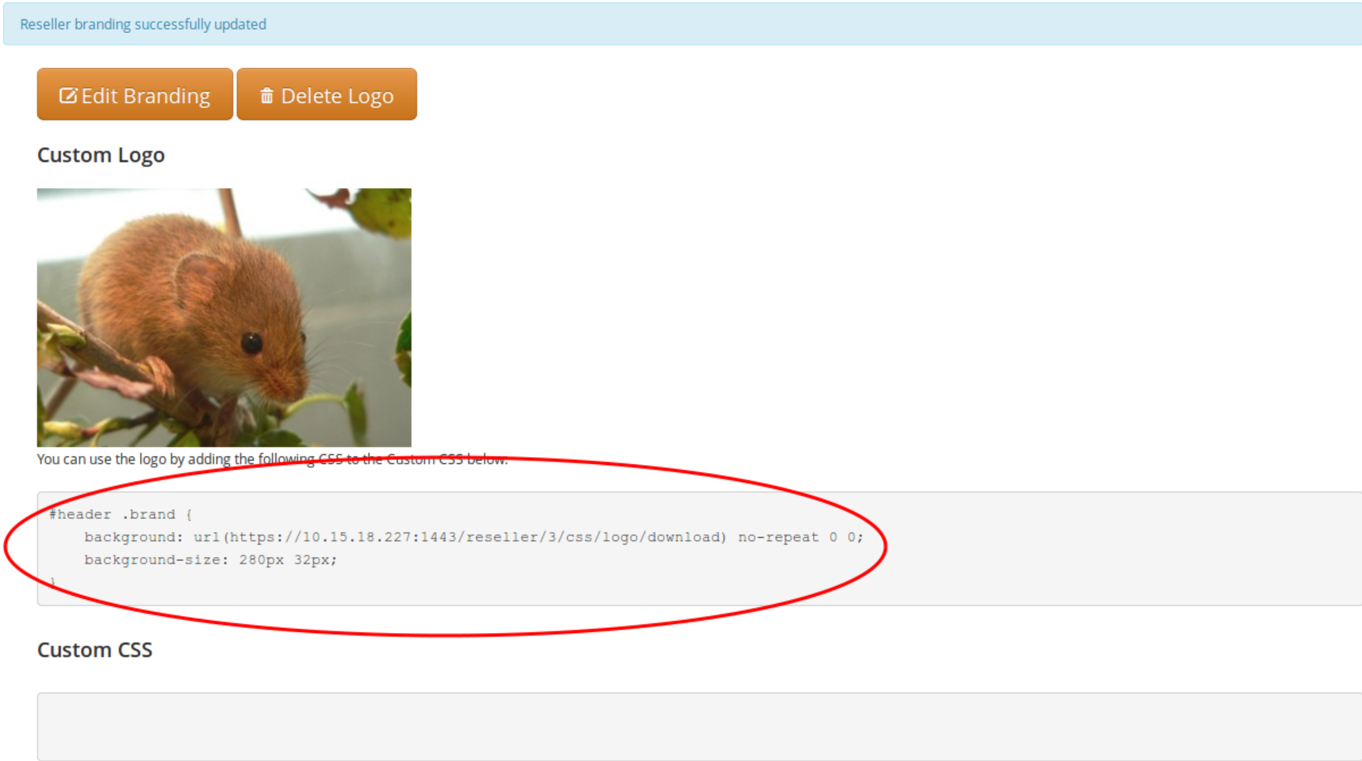


Figure 63: CSC Customisation Step 2: Copy CSS code

5. Press the *Edit Branding* button again.

6. Paste the CSS code into *CSS* text box and *Save* the changes:

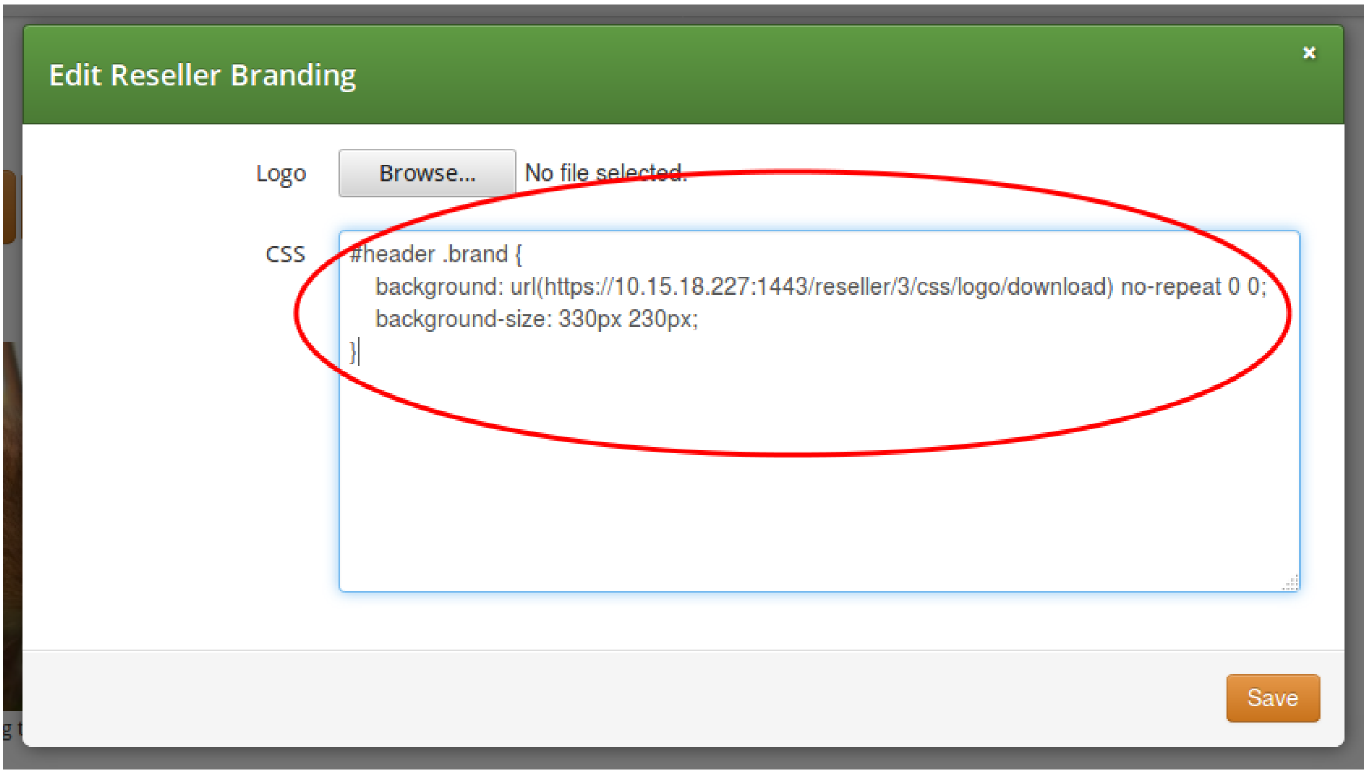


Figure 64: CSC Customisation Step 3: Paste CSS code

7. Now the new logo is already visible on the admin / CSC panel. If you want to hide the Sipwise copyright notice at the bottom of the web panels, add a line of CSS code as shown here:

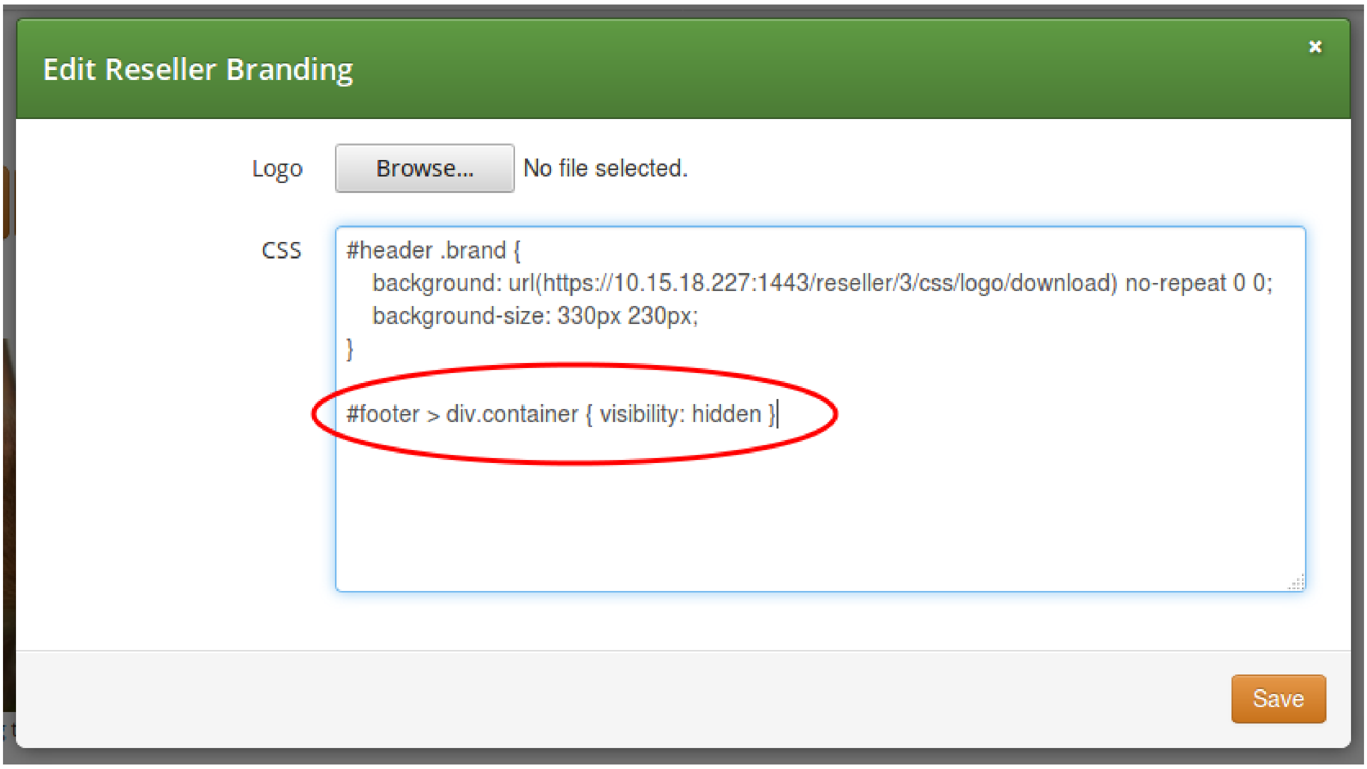


Figure 65: CSC Customisation: Hide copyright notice

8. The final branding data is shown on the admin web panel:

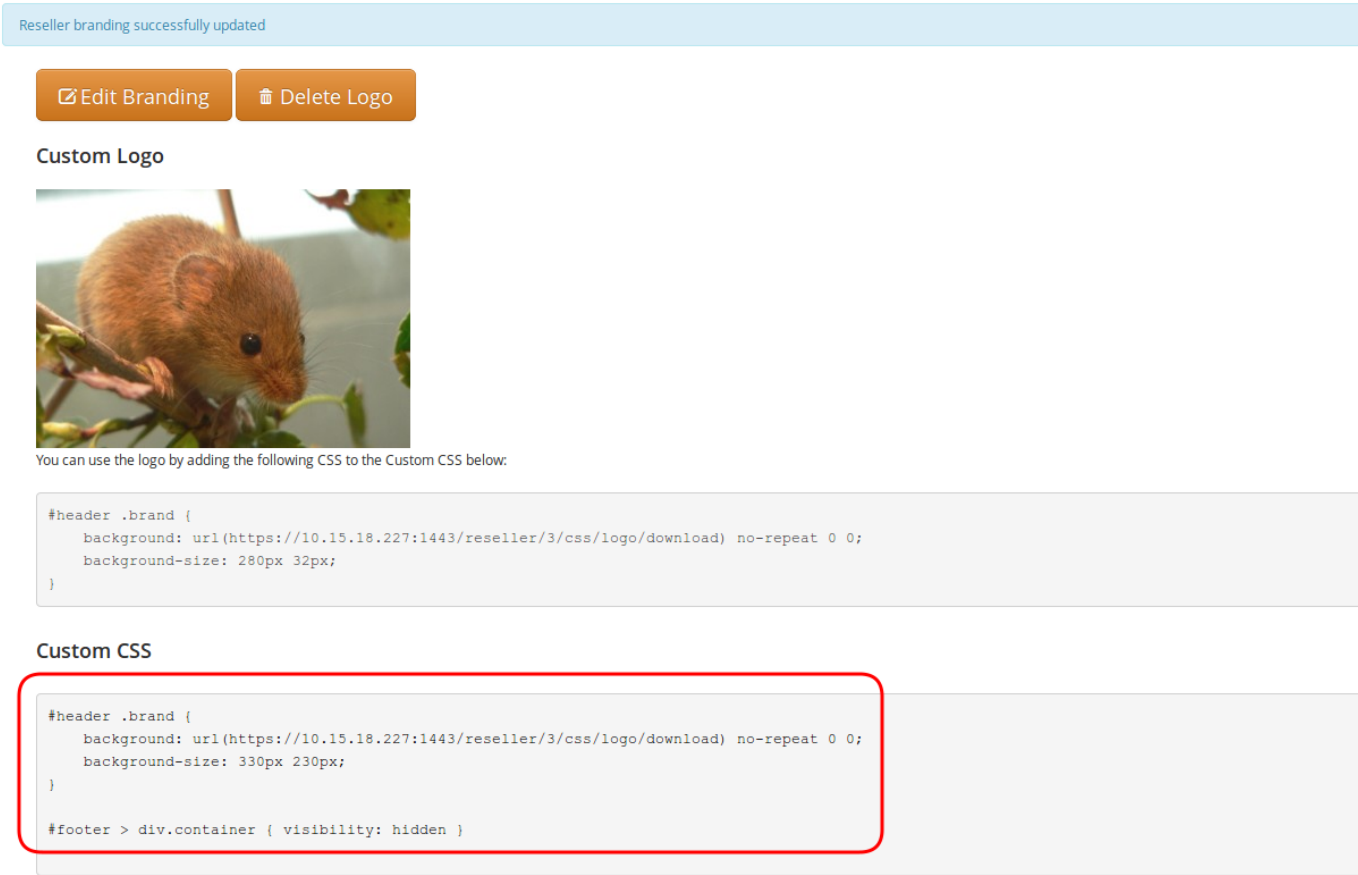


Figure 66: CSC Customisation: Custom data on panel

8.1.2.2 Other Website Customisations

The layout and style of NGCP's admin and CSC web panel is determined by a single CSS file: `/usr/share/ngcp-panel/static/css/application.css`

More complex changes, like replacing colour of some web panel components, is possible via the modification of the CSS file.

> **Warning**
> Only experienced users with profound CSS knowledge are advised to change web panel properties in the main CSS file. *Sipwise does not recommend and also does not support the modification of the main CSS file.*

8.1.2.3 Selecting Available Languages

You can also enable/disable specific languages a user can choose from in the CSC panel. Currently, English (`en`), German (`de`), Italian (`it`), Spanish (`es`) and Russian (`ru`) are supported, and the default language is the same as the browser's preferred one.

You can select the *default language* provided by CSC by changing the parameter `www_admin.force_language` in `/etc/ngcp-config/config.yml` file. An example to set the English language as default: `force_language: en`

8.2 The Voicemail Menu

NGCP offers several ways to access the Voicemail box.

The CSC panel allows your users to listen to voicemail messages from the web browser, delete them and call back the user who left the voice message. User can setup voicemail forwarding to the external email and the PIN code needed to access the voicebox from any telephone also from the CSC panel.

To manage the voice messages from SIP phone: simply dial internal voicemail access number `2000`.

To change the access number: look for the parameter *voicemail_number* in */etc/ngcp-config/config.yml* in the section *sems→vsc*. After the changes, execute *ngcpcfg apply 'changed voicebox number'*.

Tip

To let the callers leave a voice message when user is not available he should enable Call Forward to Voicebox. The Call Forward can be provisioned from the CSC panel as well as by dialing Call Forward VSC with the voicemail number. E.g. when parameter *voicemail_number* is set to *9999*, a Call Forward on Not Available to the Voicebox is set if the user dials `*93*9999`. As a result, all calls will be redirected to the Voicebox if SIP phone is not registered.

To manage the voice messages from any phone:

- As an operator, you can setup some DID number as external voicemail access number: for that, you should add a special rewrite rule (Inbound Rewrite Rule for Callee, see Section 6.7.) on the incoming peer, to rewrite that DID to "voiceboxpass". Now when user calls this number the call will be forwarded to the voicemail server and he will be prompted for mailbox and password. The mailbox is the full E.164 number of the subscriber account and the password is the PIN set in the CSC panel.
- The user can also dial his own number from PSTN, if he setup Call Forward on Not Available to the Voicebox, and when reaching the voicemail server he can interrupt the "user is unavailable" message by pressing * key and then be prompted for the PIN. After entering PIN and confirming with # key he will enter own voicemail menu. PIN is random by default and must be kept secret for that reason.

9 Billing Configuration

This chapter describes the steps necessary to rate calls and export rated CDRs (call detail records) to external systems.

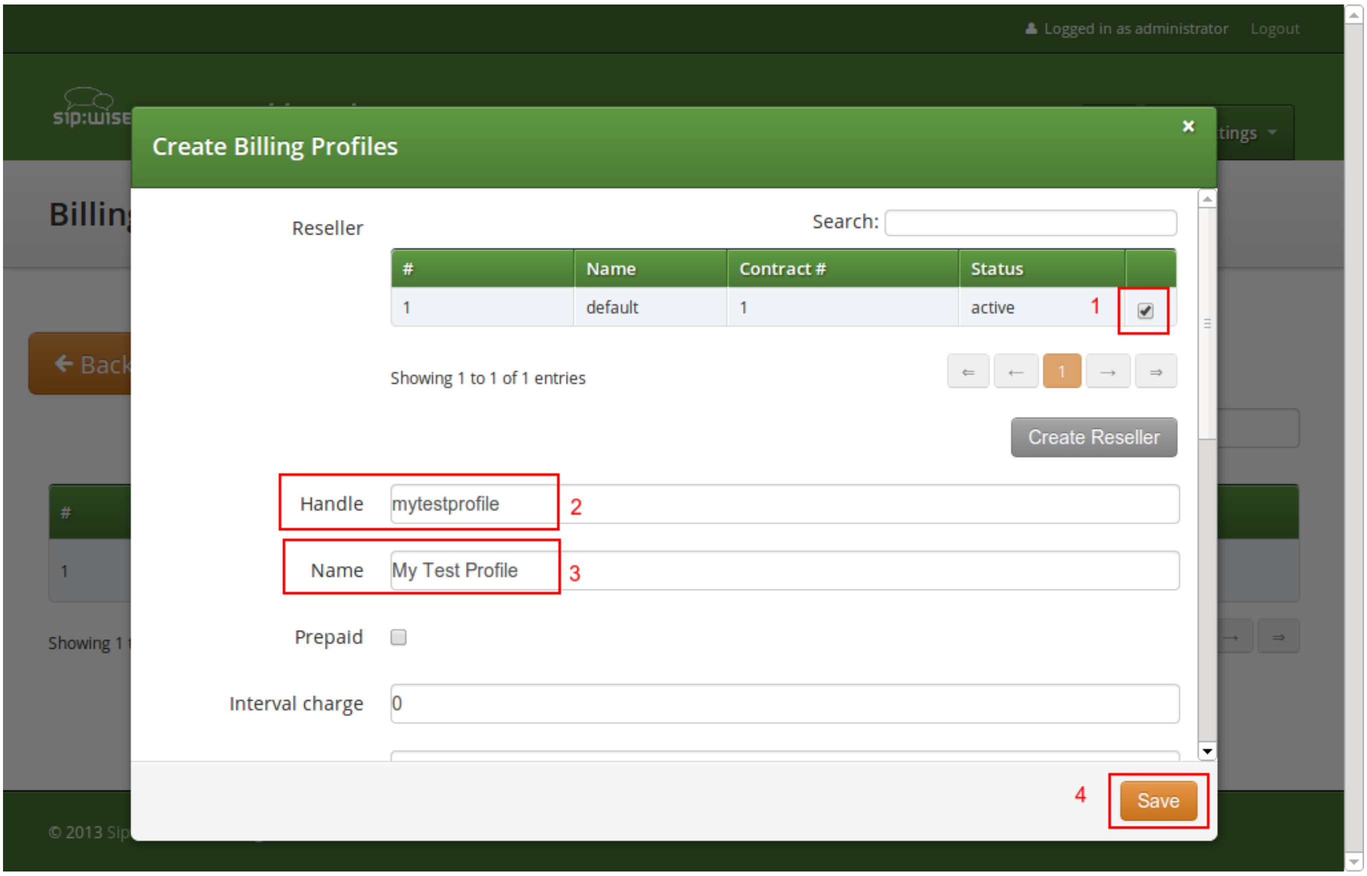
9.1 Billing Profiles

Service billing on the NGCP is based on billing profiles, which may be assigned to customers and SIP peerings. The design focuses on a simple, yet flexible approach, to support arbitrary dial-plans without introducing administrative overhead for the system administrators. The billing profiles may define a base fee and free time or free money per billing interval. Unused free time or money automatically expires at the end of the billing interval.

Each profile may have call destinations (usually based on E.164 number prefix matching) with configurable fees attached. Call destination fees each support individual intervals and rates, with a different duration and/or rate for the first interval. (e.g.: charge the first minute when the call is opened, then every 30 seconds, or make it independent of the duration at all) It is also possible to specify different durations and/or rates for peak and off-peak hours. Peak time may be specified based on weekdays, with additional support for manually managed dates based on calendar days. The call destinations can finally be grouped for an overview on user's invoices by specifying a zone in two detail levels. (E.g.: national landline, national mobile, foreign 1, foreign 2, etc.)

9.1.1 Creating Billing Profiles

The first step when setting up billing data is to create a billing profile, which will be the container for all other billing related data. Go to *Settings→Billing* and click on *Create Billing Profile*.



The fields *Reseller*, *Handle* and *Name* are mandatory.

- **Reseller**: The reseller this billing profile belongs to.
- **Handle**: A unique, permanently fixed string which is used to attach the billing profile to a customer or SIP peering contract.
- **Name**: A free form string used to identify the billing profile in the *Admin Panel*. This may be changed at any time.
- **Interval charge**: A base fee for the billing interval, specifying a monetary amount (represented as a floating point number) in whatever currency you want to use.
- **Interval free time**: If you want to include free calling time in your billing profile, you may specify the number of seconds that are available every billing interval. See *Creating Billing Fees* below on how to select destinations which may be called using the free time.
- **Interval free cash**: Same as for *interval free time* above, but specifies a monetary amount which may be spent on outgoing calls. This may be used for example to implement a minimum turnover for a contract, by setting the *interval charge* and *interval free cash* to the same values.
- **Fraud monthly limit**: The monthly fraud detection limit (in Cent) for accounts with this billing profile. If the call fees of an account reach this limit within a billing interval, an action can be triggered.
- **Fraud monthly lock**: a choice of *none, foreign, outgoing, incoming, global*. Specifies a lock level which will be used to lock the account and his subscribers when *fraud monthly limit* is exceeded.
- **Fraud monthly notify**: An email address or comma-separated list of email addresses that will receive notifications when *fraud monthly limit* is exceeded.

- **Fraud daily limit**: The fraud detection limit (in Cent) for accounts with this billing profile. If the call fees of an account reach this limit within a calendar day, an action can be triggered.
- **Fraud daily lock**: a choice of *none, foreign, outgoing, incoming, global*. Specifies a lock level which will be used to lock the account and his subscribers when *fraud daily limit* is exceeded.
- **Fraud daily notify**: An email address or comma-separated list of email addresses that will receive notifications when *fraud daily limit* is exceeded.
- **Currency**: The currency symbol for your currency. Any UTF-8 character may be used and will be printed in web interfaces.
- **VAT rate**: The percentage of value added tax for all fees in the billing profile. Currently for informational purpose only and not used further.
- **VAT included**: Whether VAT is included in the fees entered in web forms or uploaded to the platform. Currently for informational purpose only and not used further.

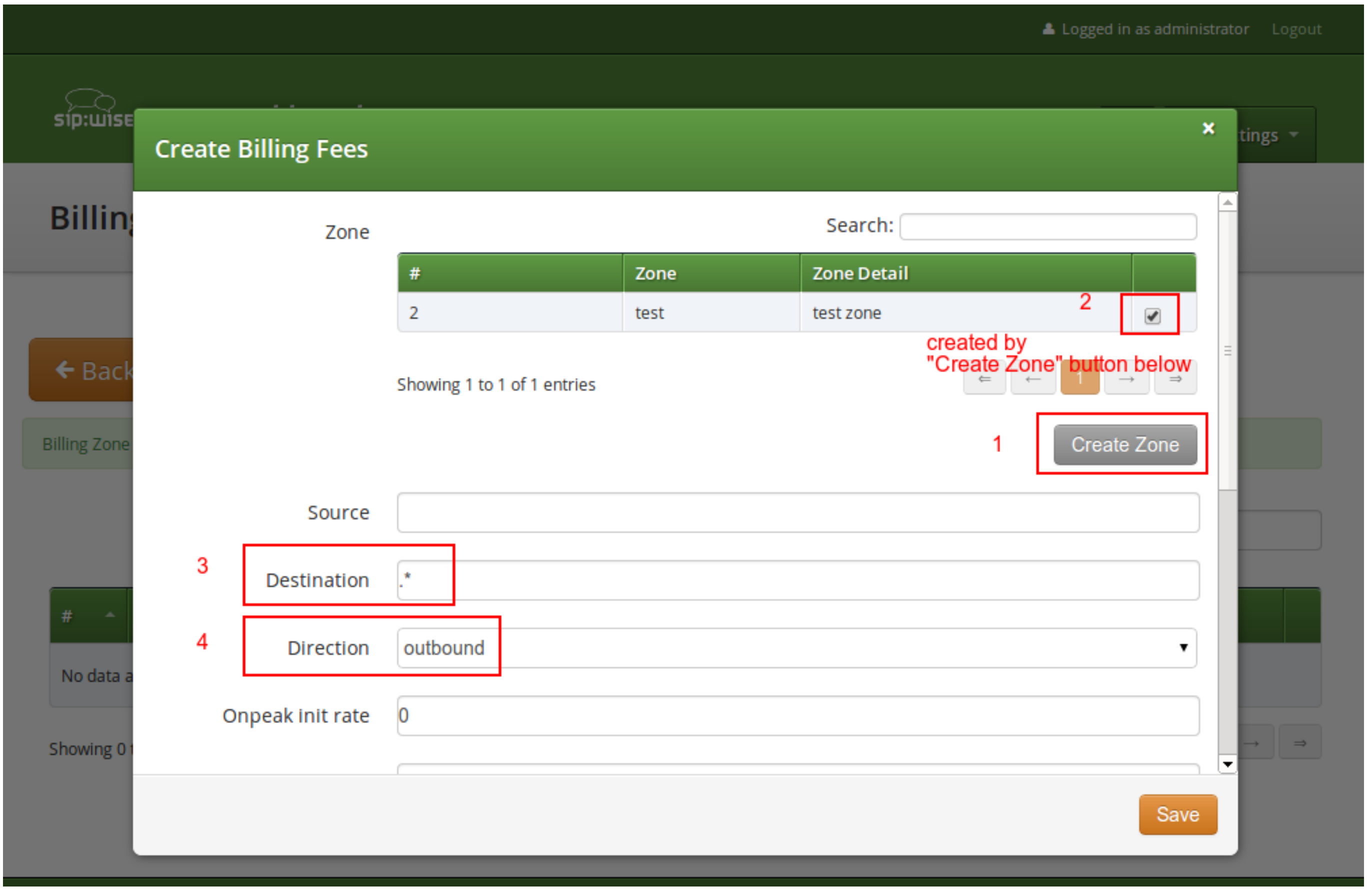
9.1.2 Creating Billing Fees

Each *Billing Profile* holds multiple *Billing Fees*.

To set up billing fees, click on the *Fees* button of the billing profile you want to configure. Billing fees may be uploaded using a configurable CSV file format, or entered directly via the web interface by clicking *Create Fee Entry*. To configure the CSV field order for the file upload, rearrange the entries in the *www_admin→fees_csv→element_order* array in */etc/ngcp-config/config.yml* and execute the command `ngcpcfg apply changed fees element order`. The following is an example of working CSV file to upload (pay attention to double quotes):

```
".","^1",out,"EU","ZONE EU",5.37,60,5.37,60,5.37,60,5.37,60,0,0
"^01.+$","^02145.+$",out,"AT","ZONE Test",0.06250,1,0.06250,1,0.01755,1,0.01733,1,0
```

For input via the web interface, just fill in the text fields accordingly.



In both cases, the following information may be specified independently for every destination:

- **Zone**: A zone for a group of destinations. May be used to group destinations for simplified display, e.g. on invoices. (e.g. `foreign zone 1`)
- **Source**: The source pattern. This is a POSIX regular expression matching the complete source URI (e.g. `^.*@sip\.example\.org$` or `^someone@sip\.sipwise\.com$` or just `.` to match everything). If you leave this field empty, the default pattern `.` matching everything will be set implicitly. Internally, this pattern will be matched against the `<source_cli>@<source_domain>` fields of the CDR.
- **Destination**: The destination pattern. This is a POSIX regular expression matching the complete destination URI (e.g. `someone@sip\.example\.org` or `^43`). This field must be set.
- **Direction**: `Outbound` for standard origination fees (applies to callers placing a call and getting billed for that) or `Inbound` for termination fees (applies to callees if you want to charge them for receiving various calls, e.g. for 800-numbers). *If in doubt, use Outbound.* If you upload fees via CSV files, use `out` or `in`, respectively.

> **Important**
> The {source, destination, direction} combination needs to be unique for a billing profile. The system will return an error if such a set is specified twice, both for the file upload and the input via the web interface.

Important

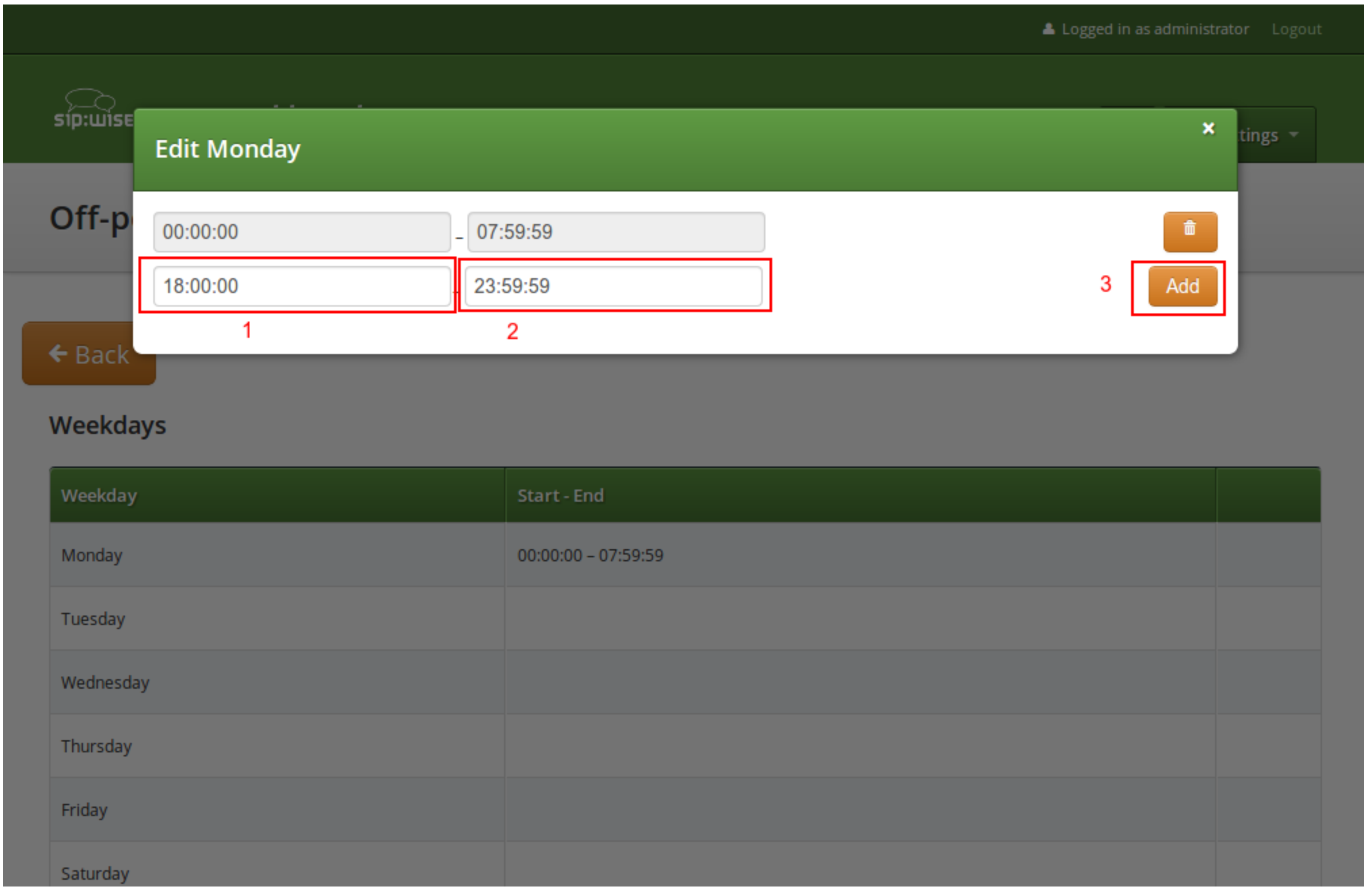
There are several internal services (vsc, conference, voicebox) which will need a specific destination entry with a domain-based destination. If you don't want to charge the same (or nothing) for those services, add a fee for destination `\.local$` there. If you want to charge different amounts for those services, break it down into separate fee entries for `@vsc\.local$`, `@conference\.local$` and `@voicebox\.local$` with the according fees. **NOT CREATING EITHER THE CATCH-ALL FEE OR THE SEPARATE FEES FOR THE `.local` DOMAIN WILL BREAK YOUR RATING PROCESS!**

- **Onpeak init rate**: The rate for the first rating interval in cent (of whatever currency, represented as a floating point number) per second. Applicable to calls during onpeak hours.
- **Onpeak init interval**: The duration of the first billing interval, in seconds. Applicable to calls during onpeak hours.
- **Onpeak follow rate**: The rate for subsequent rating intervals in cent (of whatever currency, represented as a floating point number) per second. Applicable to calls during onpeak hours. Defaults to *onpeak init rate*.
- **Onpeak follow interval**: The duration of subsequent billing intervals, in seconds. Applicable to calls during onpeak hours. Defaults to *onpeak init interval*.
- **Offpeak init rate**: The rate for the first rating interval in cent (of whatever currency, represented as a floating point number) per second. Applicable to calls during off-peak hours. Defaults to *onpeak init rate*.
- **Offpeak init interval**: The duration of the first billing interval, in seconds. Applicable to calls during off-peak hours. Defaults to *onpeak init interval*.
- **Offpeak follow rate**: The rate for subsequent rating intervals in cent (of whatever currency, represented as a floating point number) per second. Applicable to calls during off-peak hours. Defaults to *offpeak init rate* if that one is specified, or to *onpeak follow rate* otherwise.
- **Offpeak follow interval**: The duration of subsequent billing intervals, in seconds. Applicable to calls during off-peak hours. Defaults to *offpeak init interval* if that one is specified, or to *onpeak follow interval* otherwise.
- **Use free time**: Specifies whether free time minutes may be used when calling this destination. May be specified in the file upload as 0, n[o], f[alse] and 1, y[es], t[rue] respectively.

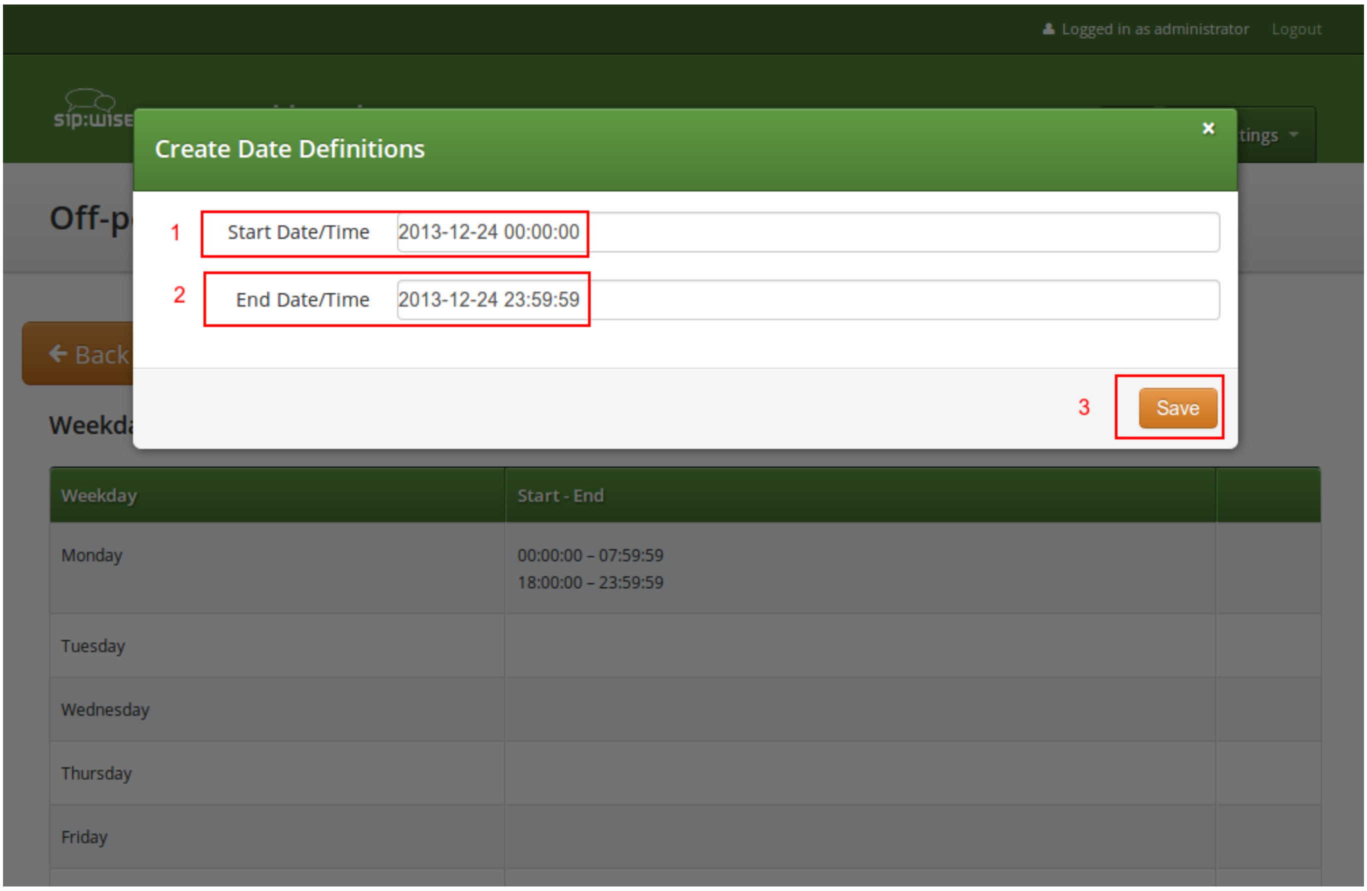
9.1.3 Creating Off-Peak Times

To be able to differentiate between on-peak and off-peak calls, the platform stores off-peak times for every billing profile based on weekdays and/or calendar days. To edit the settings for a billing profile, go to *Settings→Billing* and press the *Off-Peaktimes* button on the billing profile you want to configure.

To set off-peak times for a weekday, click on *Edit* next to the according weekday. You will be presented with two input fields which both receive a timestamp in the form of *hh:mm:ss* specifying a time of day for the start and end of the off-peak period. If any of the fields is left empty, the system will automatically insert `00:00:00` (*start* field) or *23:59:59* (*end* field). Click on *Add* to store the setting in the database. You may create more than one off-peak period per weekday. To delete a range, just click *Delete* next to the entry. Click the *close* icon when done.



To specify off-peak ranges based on calendar dates, click on *Create Special Off-Peak Date*. Enter a date in the form of *YYYY-MM-DD hh:mm:ss* into the *Start Date/Time* input field and *End Date/Time* input field to define a range for the off-peak period.



9.2 Fraud Detection and Locking

The NGCP supports a fraud detection feature, which is designed to detect accounts causing unusually high customer costs, and then to perform one of several actions upon those accounts. This feature can be enabled and configured through two sets of billing profile options described in Section 9.1.1, namely the monthly (*fraud monthly limit*, *fraud monthly lock* and *fraud monthly notify*) and daily limits (*fraud daily limit*, *fraud daily lock* and *fraud daily notify*). Either monthly/daily limits or both of them can be active at the same time.

Monthly fraud limit check runs once a day, shortly after midnight local time and daily fraud limit check runs every 30min. A background script (managed by cron daemon) automatically checks all accounts which are linked to a billing profile enabled for fraud detection, and selects those which have caused a higher cost than the *fraud monthly limit* configured in the billing profile, within the currently active billing interval (e.g. in the current month), or a higher cost than the *fraud daily limit* configured in the billing profile, within the calendar day. It then proceeds to perform at least one of the following actions on those accounts:

- If **fraud lock** is set to anything other than *none*, it will lock the account accordingly (e.g. if **fraud lock** is set to *outgoing*, the account will be locked for all outgoing calls).
- If anything is listed in **fraud notify**, an email will be sent to the email addresses configured. The email will contain information about which account is affected, which subscribers within that account are affected, the current account balance and the configured fraud limit, and also whether or not the account was locked in accordance with the **fraud lock** setting. It should be noted that this email is meant for the administrators or accountants etc., and not for the customer.

9.2.1 Fraud Lock Levels

Fraud lock levels are various protection (and notification) settings that are applied to subscribers of a *Customer*, if fraud detection is enabled in the currently active billing profile and the *Customer's* daily or monthly fraud limit has been exceeded.

The following lock levels are available:

- `none`: no account locking will happen
- `foreign calls`: only calls within the subscriber's own domain, and emergency calls, are allowed
- `all outgoing calls`: subscribers of the customer cannot place any calls, except calls to free and emergency destinations
- `incoming and outgoing`: subscribers of the customer cannot place and receive any calls, except calls to free and emergency destinations
- `global`: same restrictions as at `incoming and outgoing` level, additionally subscribers are not allowed to access the Customer Self Care (CSC) interface
- `ported`: only automatic call forwarding, due to number porting, is allowed

---

Important
You can override fraud detection and locking settings of a billing profile on a per-account basis via REST API or the Admin interface.

Caution
Accounts that were automatically locked by the fraud detection feature will **not** be automatically unlocked when the next billing interval starts. This has to be done manually through the administration panel or through the provisioning interface.

Important
If fraud detection is configured to only send an email and not lock the affected accounts, it will continue to do so for over-limit accounts every day. The accounts must either be locked in order to stop the emails (only currently active accounts are considered when the script looks for over-limit accounts) or some other action to resolve the conflict must be taken, such as disabling fraud detection for those accounts.

Note
It is possible to fetch the list of fraud events and thus get fraud status of *Customers* by using the REST API and referring to the resource: `/api/customerfraudevents`.

9.3 Notes on Billing and Call Rating

Cash balance with post-paid billing profile

Customers with a post-paid billing profile may have a positive account cash balance value. This is the regular case when using a post-paid billing profile showing a *free cash* greater than *0*.

Tip

You can set the free cash (and the free time) in the billing profile. The account balance will be set and managed (i.e. refilled or carried over) automatically for subsequent balance intervals.

In case the account has a positive cash balance, the cost of the call will be deducted from that balance and not considered as additional cost of that particular call for the customer.

Important

The rating engine (*rate-o-mat*) in sip:provider CE will write *0* instead of the real cost of a call in the CDR, if the source customer's (who initiated the call) account has a positive cash balance! The purpose of this is to reflect the usage of free cash in the CDR for the particular call.

Note

It might happen, for instance, that a customer's billing profile is changed from pre-paid to post-paid, and the customer already had a positive cash balance on his account. In that case the same call rating mechanism is involved as for the free cash.

9.4 Billing Data Export

Regular billing data export is done using CSV (*comma separated values*) files which may be downloaded from the platform using the *cdrexport* user which has been created during the installation.

There are two types of exports. One is *CDR* (Call Detail Records) used to charge for calls made by subscribers, and the other is *EDR* (Event Detail Records) used to charge for provisioning events like enabling certain features.

9.4.1 Glossary of Terms

Billing records contain fields that hold data of various entities that play a role in the phone service offered by Sipwise NGCP. For a better understanding of billing data please refer to the glossary provided here:

- **Account**: the customer's account that is charged for calls of its subscriber(s)
- **Carrier**: a SIP peer that sends incoming calls to, or receives outgoing calls from NGCP. A carrier may charge fees for the outgoing calls from NGCP (outbound billing fee), or for the incoming calls to NGCP (inbound billing fee).
- **Contract**: the service contract that represents a customer, a reseller or a SIP peer; a contract on NGCP contains the billing profile (billing fees) too
- **Customer**: the legal entity that represents any number of subscribers; this entity receives the bills for calls of its subscriber(s)
- **Provider**: either the reseller that holds a subscriber who is registered on NGCP, or the SIP peer that handles calls between an external subscriber and NGCP

- **Reseller**: the entity who is the direct, administrative service provider of a group of customers and subscribers registered on NGCP; the NGCP operator may also charge a reseller for the calls initiated or received by its subscribers
- **User**: the subscriber who either is registered on NGCP, or is an external call party

9.4.2 File Name Format

In order to be able to easily identify billing files, the file names are constructed by the following fixed-length fields:

```
<prefix><separator><version><separator><timestamp><separator><sequence number><suffix>
```

The definition of the specific fields is as follows:

Table 8: CDR/EDR export file name format

File name element	Length	Description
`<prefix>`	7	A fixed string. Always `sipwise`.
`<separator>`	1	A fixed character. Always _.
`<version>`	3	The format version, a three digit number. Currently `007`.
`<timestamp>`	14	The file creation timestamp in the format `YYYYMMDDhhmmss`.
`<sequence number>`	10	A unique 10-digit zero-padded sequence number for quick identification.
`<suffix>`	4	A fixed string. Always `.cdr` or `.edr`.

A valid example filename for a CDR billing file created at 2012-03-10 14:30:00 and being the 42nd file exported by the system, is:

```
sipwise_007_20130310143000_0000000042.cdr
```

9.4.3 File Format

Each billing file consists of three parts: one header line, zero to 5000 body lines and one trailer line.

9.4.3.1 File Header Format

The billing file header is one single line, which is constructed by the following fields:

```
<version>,<number of records>
```

The definition of the specific fields is as follows:

Table 9: CDR/EDR export file header line format

Body Element	Length	Type	Description
`<version>`	3	zero-padded uint	The format version. Currently `007`.
`<number of records>`	4	zero-padded uint	The number of body lines contained in the file.

A valid example for a Header is:

`007,0738`

9.4.3.2 File Body Format for Call Detail Records (CDR)

The body of a CDR consists of a minimum of zero and a default maximum of 5000 lines. The platform operator can configure the maximum number of lines kept in a file by updating the `cdrexport.max_rows_per_file` parameter in `/etc/ngcp-config/config.yml` file. Each line holds one call detail record in CSV format and is constructed by a configurable set of fields, all of them enclosed in single quotes.

The following table defines the **default set of fields** that are inserted into the CDR file, for exports related to *system* scope. The list of fields is defined in `/etc/ngcp-config/config.yml` file, `cdrexport.admin_export_fields` parameter.

Table 10: Default set of system CDR fields

Body Element	Length	Type	Description
`CDR_ID`	1-10	uint	Internal CDR ID.
`UPDATE_TIME`	19	timestamp	Timestamp of last modification, including date and time (with seconds precision).
`SOURCE_USER_ID`	36	string	Internal UUID of calling party subscriber. Value is `0` if calling party is external.
`SOURCE_PROVIDER_ID`	0-255	string	Internal ID of the contract of calling party provider (i.e. reseller or peer).
`SOURCE_EXTERNAL_SUBSCRIBER_ID`	0-255	string	External, arbitrary ID of calling party subscriber. (A string value shown as "External ID" property of an NGCP subscriber.)
`SOURCE_SUBSCRIBER_ID`	1-11	uint	Internal ID of calling party subscriber. Value is `0` if calling party is external.

Table 10: (continued)

Body Element	Length	Type	Description
SOURCE_EXTERNAL_CONTRACT_ID	0-255	string	External, arbitrary ID of calling party customer. (A string value shown as "External ID" property of an NGCP customer/peer.)
SOURCE_ACCOUNT_ID	1-11	uint	Internal ID of calling party customer.
SOURCE_USER	0-255	string	SIP username of calling party.
SOURCE_DOMAIN	0-255	string	SIP domain of calling party.
SOURCE_CLI	0-64	string	CLI of calling party in E.164 format.
SOURCE_CLIR	1	uint	1 for calls with CLIR, 0 otherwise.
SOURCE_IP	0-64	string	IP Address of the calling party.
DESTINATION_USER_ID	36	string	Internal UUID of called party subscriber. Value is 0 if called party is external.
DESTINATION_PROVIDER_ID	0-255	string	Internal ID of the contract of called party provider (i.e. reseller or peer).
DESTINATION_EXTERNAL_SUBSCRIBER_ID	0-255	string	External, arbitrary ID of called party subscriber. (A string value shown as "External ID" property of an NGCP subscriber.)
DESTINATION_SUBSCRIBER_ID	1-11	uint	Internal ID of called party subscriber. Value is 0 if calling party is external.
DESTINATION_EXTERNAL_CONTRACT_ID	0-255	string	External, arbitrary ID of called party customer. (A string value shown as "External ID" property of an NGCP customer/peer.)
DESTINATION_ACCOUNT_ID	1-11	uint	Internal ID of called party customer.
DESTINATION_USER	0-255	string	Final SIP username of called party.
DESTINATION_DOMAIN	0-255	string	Final SIP domain of called party.
DESTINATION_USER_IN	0-255	string	Incoming SIP username of called party, after applying inbound rewrite rules.
DESTINATION_DOMAIN_IN	0-255	string	Incoming SIP domain of called party, after applying inbound rewrite rules.
DESTINATION_USER_DIALED	0-255	string	The user-part of the SIP Request URI as received by NGCP.
PEER_AUTH_USER	0-255	string	Username used to authenticate towards peer.
PEER_AUTH_REALM	0-255	string	Realm used to authenticate towards peer.

Table 10: (continued)

Body Element	Length	Type	Description
`CALL_TYPE`	3-4	string	The type of the call - one of: `call`: normal call `cfu`: call forward unconditional `cfb`: call forward busy `cft`: call forward timeout `cfna`: call forward not available `cfs`: call forward for SMS
`CALL_STATUS`	2-8	string	The final call status - one of: `ok`: successful call `busy`: called party busy `noanswer`: no answer from called party `cancel`: cancel from caller `offline` called party offline `timeout`: no reply from called party `other`: unspecified, see `CALL_CODE` field for details
`CALL_CODE`	3	string	The final SIP status code.
`INIT_TIME`	23	timestamp	Timestamp of call initiation (SIP *INVITE* received from calling party). Includes date, time with milliseconds (3 decimals).
`START_TIME`	23	timestamp	Timestamp of call establishment (final SIP response received from called party). Includes date, time with milliseconds (3 decimals).
`DURATION`	4-13	fixed precision (3 decimals)	Length of call (calculated from `START_TIME`) including milliseconds (3 decimals).
`CALL_ID`	0-255	string	The SIP Call-ID.
`RATING_STATUS`	2-7	string	The internal rating status of the CDR - one of: `unrated`: not rated `ok`: successfully rated `failed`: error while rating Currently always `ok` or `unrated`, depending on whether rating is enabled or not.
`RATED_AT`	0-19	datetime	Time of rating, including date and time (with seconds precision). Empty if CDR is not rated.

Table 10: (continued)

Body Element	Length	Type	Description
SOURCE_CARRIER_COST	7-14	fixed precision (6 decimals)	The originating carrier cost that the carrier (i.e. SIP peer) charges for the calls routed to his network, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*
SOURCE_CUSTOMER_COST	7-14	fixed precision (6 decimals)	The originating customer cost, or empty if CDR is not rated.
SOURCE_CARRIER_ZONE	0-127	string	Name of the originating carrier billing zone, or `onnet` if data is not available. *PLEASE NOTE: Only available in system exports, not for resellers.*
SOURCE_CUSTOMER_ZONE	0-127	string	Name of the originating customer billing zone, or empty if CDR is not rated.
SOURCE_CARRIER_DETAIL	0-127	string	Description of the originating carrier billing zone, or `platform internal` if data is not available. *PLEASE NOTE: Only available in system exports, not for resellers.*
SOURCE_CUSTOMER_DETAIL	0-127	string	Description of the originating customer billing zone, or empty if CDR is not rated.
SOURCE_CARRIER_FREE_TIME	1-10	uint	The number of free time seconds used on originating carrier side, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*
SOURCE_CUSTOMER_FREE_TIME	1-10	uint	The number of free time seconds used from the originating customer's account balance, or empty if CDR is not rated.
DESTINATION_CARRIER_COST	7-14	fixed precision (6 decimals)	The terminating carrier cost, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*
DESTINATION_CUSTOMER_COST	7-14	fixed precision (6 decimals)	The terminating customer cost, or empty if CDR is not rated.

Table 10: (continued)

Body Element	Length	Type	Description
DESTINATION_CARRIER_ZONE	0-127	string	Name of the terminating carrier billing zone, or onnet if data is not available. *PLEASE NOTE: Only available in system exports, not for resellers.*
DESTINATION_CUSTOMER_ZONE	0-127	string	Name of the terminating customer billing zone, or empty if CDR is not rated.
DESTINATION_CARRIER_DETAIL	0-127	string	Description of the terminating carrier billing zone, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*
DESTINATION_CUSTOMER_DETAIL	0-127	string	Description of the terminating customer billing zone, or empty if CDR is not rated.
DESTINATION_CARRIER_FREE_TIME	1-10	uint	The number of free time seconds used on terminating carrier side, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*
DESTINATION_CUSTOMER_FREE_TIME	1-10	uint	The number of free time seconds used from the terminating customer's account balance, or empty if CDR is not rated.
SOURCE_RESELLER_COST	7-14	fixed precision (6 decimals)	The originating reseller cost, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*
SOURCE_RESELLER_ZONE	0-127	string	Name of the originating reseller billing zone, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*
SOURCE_RESELLER_DETAIL	0-127	string	Description of the originating reseller billing zone, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*

Table 10: (continued)

Body Element	Length	Type	Description
SOURCE_RESELLER_FREE_TIME	1-10	uint	The number of free time seconds used from the originating reseller's account balance, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*
DESTINATION_RESELLER_COST	7-14	fixed precision (6 decimals)	The terminating reseller cost, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*
DESTINATION_RESELLER_ZONE	0-127	string	Name of the terminating reseller billing zone, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*
DESTINATION_RESELLER_DETAIL	0-127	string	Description of the terminating reseller billing zone, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*
DESTINATION_RESELLER_FREE_TIME	1-10	uint	The number of free time seconds used from the terminating reseller's account balance, or empty if CDR is not rated. *PLEASE NOTE: Only available in system exports, not for resellers.*
<line_terminator>	1	string	Always `\n` (special char LF - ASCII `0x0A`).

A valid example of one body line of a rated CDR is (line breaks added for clarity):

```
'15','2013-03-26 22:09:11','a84508a8-d256-4c80-a84e-820099a827b0','1','','1','',
'2','testuser1','192.168.51.133','4311001','0','192.168.51.1',
'94d85b63-8f4b-43f0-b3b0-221c9e3373f2','1','','3','','4','testuser3',
'192.168.51.133','testuser3','192.168.51.133','testuser3','','','call','ok','200',
'2013-03-25 20:24:50.890','2013-03-25 20:24:51.460','10.880','44449842',
'ok','2013-03-25 20:25:27','0.00','24.00','onnet','testzone','platform internal',
'testzone','0','0','0.00','200.00','','foo','','foo','0','0',
'0.00','','','0','0.00','','','0'
```

The format of the **CDR export files generated for *resellers*** (as opposed to the complete system-wide export) is identical except for a few missing fields.

Note

Please check the description of fields in the table above, in order to see which fields are omitted for *reseller* related CDR exports.

The list of fields for *reseller* CDR export is defined in `/etc/ngcp-config/config.yml` file, `cdrexport.reseller_export_fields` parameter.

9.4.3.3 Extra fields that can be exported to CDRs

Supplementary Data

There are fields in CDR database that contain **supplementary data** related to subscribers. This data is not used by NGCP for CDR processing but rather provides the system administrator with a possibility to include supplementary information in CDRs.

Note

This informational section is meant for problem solving / debugging purpose: The supplementary data listed in following table is stored in `provisioning.voip_preferences` database table.

Table 11: Supplementary data in CDR fields

Body Element	Length	Type	Description
SOURCE_GPP0	0-255	string	Supplementary data field 0 of calling party.
SOURCE_GPP1	0-255	string	Supplementary data field 1 of calling party.
SOURCE_GPP2	0-255	string	Supplementary data field 2 of calling party.
SOURCE_GPP3	0-255	string	Supplementary data field 3 of calling party.
SOURCE_GPP4	0-255	string	Supplementary data field 4 of calling party.
SOURCE_GPP5	0-255	string	Supplementary data field 5 of calling party.
SOURCE_GPP6	0-255	string	Supplementary data field 6 of calling party.
SOURCE_GPP7	0-255	string	Supplementary data field 7 of calling party.
SOURCE_GPP8	0-255	string	Supplementary data field 8 of calling party.
SOURCE_GPP9	0-255	string	Supplementary data field 9 of calling party.
DESTINATION_GPP0	0-255	string	Supplementary data field 0 of called party.
DESTINATION_GPP1	0-255	string	Supplementary data field 1 of called party.
DESTINATION_GPP2	0-255	string	Supplementary data field 2 of called party.
DESTINATION_GPP3	0-255	string	Supplementary data field 3 of called party.
DESTINATION_GPP4	0-255	string	Supplementary data field 4 of called party.
DESTINATION_GPP5	0-255	string	Supplementary data field 5 of called party.
DESTINATION_GPP6	0-255	string	Supplementary data field 6 of called party.
DESTINATION_GPP7	0-255	string	Supplementary data field 7 of called party.
DESTINATION_GPP8	0-255	string	Supplementary data field 8 of called party.
DESTINATION_GPP9	0-255	string	Supplementary data field 9 of called party.

Account balance details (prepaid calls)

There are fields in CDR database that show **changes in cash or free time balance**. In addition to that, a history of billing packages / profiles may also be present, since the NGCP vouchers, that are used to top-up, may also be set up to cause a transition of profile packages. (Which in turn can result in changing the billing profile/applicable fees). Therefore the billing package and profile valid at the time of the CDR are recorded and exposed as fields for CDR export.

Tip

Such fields may also be required to integrate sip:provider CE with legacy billing systems.

Note

Please be aware that pre-paid billing functionality is only available in Sipwise *sip:provider PRO* and *sip:carrier* products.

The name of CDR data field consists of the elements listed below:

1. `source|destination`: decides if the data refers to calling (`source`) or called (`destination`) party
2. `carrier|reseller|customer`: the account owner, whose billing data is referred
3. data type:
   A. `cash_balance|free_time_balance _ before|after`: cash balance or free time balance, before or after the call
   B. `profile_package_id|contract_balance_id`: internal ID of the active pre-paid billing profile or the account balance

Examples:

- source_customer_cash_balance_before
- destination_customer_profile_package_id

Important

For calls spanning multiple balance intervals, the latter one will be selected, that is the balance interval where the call ended.

9.4.3.4 File Body Format for Event Detail Records (EDR)

The body of an EDR consists of a minimum of zero and a maximum of 5000 lines. The platform operator can configure the maximum number of lines kept in a file by updating the `eventexport.max_rows_per_file` parameter in `/etc/ngcp-`

config/config.yml file. Each line holds one call detail record in CSV format and is constructed by the fields as per the subsequent table.

The following table defines the **default set of fields** that are inserted into the EDR file, for exports related to *system* scope. The list of fields is defined in /etc/ngcp-config/config.yml file, eventexport.admin_export_fields parameter.

Table 12: Default set of system EDR fields

Body Element	Length	Type	Description
EVENT_ID	1-11	uint	Internal EDR ID.
TYPE	0-255	string	The type of the event - one of: start_profile: A subscriber profile has been newly assigned to a subscriber. end_profile: A subscriber profile has been removed from a subscriber. update_profile: A subscriber profile has been changed for a subscriber. start_huntgroup: A subscriber has been provisioned as PBX / hunting group. end_huntgroup: A subscriber has been deprovisioned as PBX / hunting group. start_ivr: A subscriber has a new call-forward to Auto-Attendant. end_ivr: A subscriber has removed a call-forward to Auto-Attendant.
CONTRACT_EXTERNAL_ID	0-255	string	The external ID of the customer. (A string value shown as "External ID" property of an NGCP customer.)
COMPANY	0-127	string	The company name of the customer's contact.
SUBSCRIBER_EXTERNAL_ID	0-255	string	The external ID of the subscriber. (A string value shown as "External ID" property of an NGCP subscriber.) *PLEASE NOTE: This field is empty in case of start_huntgroup and end_huntgroup events.*
PILOT_PRIMARY_NUMBER	0-64	string	The pilot subscriber's primary number (HPBX subscribers). *PLEASE NOTE: This is not included in default set of EDR fields from NGCP version mr5.0 upwards.*
PRIMARY_NUMBER	0-64	string	The VoIP number of the subscriber with the highest ID (DID or primary number).

Table 12: (continued)

Body Element	Length	Type	Description
OLD_PROFILE_NAME	0-255	string	The old status of the event. Depending on the event_type: start_profile: Empty. end_profile: The name of the subscriber profile which got removed from the subscriber. update_profile: The name of the former subscriber profile which got updated. start_huntgroup: Empty. end_huntgroup: Empty. start_ivr: Empty. end_ivr: Empty.
NEW_PROFILE_NAME	0-255	string	The new status of the event. Depending on the event_type: start_profile: The name of the subscriber profile which got assigned to the subscriber. end_profile: Empty. update_profile: The name of the new subscriber profile which got applied. start_huntgroup: Empty. end_huntgroup: Empty. start_ivr: Empty. end_ivr: Empty.
TIMESTAMP	23	timestamp	Timestamp of event. Includes date, time with milliseconds (3 decimals).
RESELLER_ID	1-11	uint	Internal ID of the reseller which the event belongs to. *PLEASE NOTE: Only available in system exports, not for resellers.*
<line_terminator>	1	string	A fixed character. Always \n (special char LF - ASCII 0x0A).

A valid example of one body line of an EDR is (line breaks added for clarity):

```
"1","start_profile","sipwise_ext_customer_id_4","Sipwise GmbH",
"sipwise_ext_subscriber_id_44","436667778","","1","2014-06-19 11:34:31","1"
```

The format of the **EDR export files generated for *resellers*** (as opposed to the complete system-wide export) is identical except for a few missing fields.

Note

Please check the description of fields in the table above, in order to see which fields are omitted for *reseller* related EDR exports.

The list of fields for *reseller* EDR export is defined in `/etc/ngcp-config/config.yml` file, `eventexport.reseller_export_fields` parameter.

9.4.3.5 Extra fields that can be exported to EDRs

There are fields in EDR database that contain **supplementary data** related to subscribers, for example subscriber phone numbers are such data.

Table 13: Supplementary data in EDR fields

Body Element	Length	Type	Description
SUBSCRIBER_PROFILE_SET_NAME	0-255	string	The subscriber's profile set name.
PILOT_SUBSCRIBER_PROFILE_SET_NAME	0-255	string	The profile set name of the subscriber's pilot subscriber.
PILOT_SUBSCRIBER_PROFILE_NAME	0-255	string	The profile name of the subscriber's pilot subscriber.
FIRST_NON_PRIMARY_ALIAS_USERNAME_BEFORE	0-255	string	The subscriber's non-primary alias with lowest ID, before number updates during the operation.
FIRST_NON_PRIMARY_ALIAS_USERNAME_AFTER	0-255	string	The subscriber's non-primary alias with lowest ID, after number updates during the operation.
PILOT_FIRST_NON_PRIMARY_ALIAS_USERNAME_BEFORE	0-255	string	The non-primary alias with lowest ID of the subscriber's pilot subscriber, before number updates during the operation.
PILOT_FIRST_NON_PRIMARY_ALIAS_USERNAME_AFTER	0-255	string	The non-primary alias with lowest ID of the subscriber's pilot subscriber, after number updates during the operation.
NON_PRIMARY_ALIAS_USERNAME	0-255	string	The non-primary alias of a subscriber affected by an `update_profile`, `start_profile` or `end_profile` event to track number changes.
PRIMARY_ALIAS_USERNAME_BEFORE	0-255	string	The subscriber's primary alias, before number updates during the operation.
PRIMARY_ALIAS_USERNAME_AFTER	0-255	string	The subscriber's primary alias, after number updates during the operation.
PILOT_PRIMARY_ALIAS_USERNAME_BEFORE	0-255	string	The primary alias of the subscriber's pilot subscriber, before number updates during the operation.
PILOT_PRIMARY_ALIAS_USERNAME_AFTER	0-255	string	The primary alias of the subscriber's pilot subscriber, after number updates during the operation.

Table 13: (continued)

Body Element	Length	Type	Description
FIRST_NON_PRIMARY_ALIAS_USERNAME_BEFORE_AFTER	0-255	string	Equals FIRST_NON_PRIMARY_ALIAS_USERNAME_BEFORE, if the value is not NULL, otherwise it's the same as FIRST_NON_PRIMARY_ALIAS_USERNAME_AFTER.
PILOT_FIRST_NON_PRIMARY_ALIAS_USERNAME_BEFORE_AFTER	0-255	string	Equals PILOT_FIRST_NON_PRIMARY_ALIAS_USERNAME_BEFORE, if the value is not NULL, otherwise it's the same as PILOT_FIRST_NON_PRIMARY_ALIAS_USERNAME_AFTER.

9.4.3.6 File Trailer Format

The billing file trailer is one single line, which is constructed by the following fields:

```
<md5 sum>
```

The <md5 sum> is a 32 character hexadecimal MD5 hash of the *Header* and *Body*.

To validate the billing file, one must remove the Trailer before computing the MD5 sum of the file. An example bash script to validate the integrity of the file is given below:

```
#!/bin/sh

error() { echo $@; exit 1; }
test -n "$1" || error "Usage: $0 <cdr-file>"
test -f "$1" || error "File '$1' not found"

TMPFILE="/tmp/$(basename "$1").$$"
MD5="$(sed -rn '$ s/^([a-z0-9]{32}).*$/\1/i p' "$1")  $TMPFILE"
sed '$d' "$1" > "$TMPFILE"
echo "$MD5" | md5sum -c -
rm -f "$TMPFILE"
```

Given the script is located in cdr-md5.sh and the CDR-file is sipwise_001_20071110123000_0000000004.cdr, the output of the integrity check for an intact CDR file would be:

```
$ ./cdr-md5.sh sipwise_001_20071110123000_0000000004.cdr
/tmp/sipwise_001_20071110123000_0000000004.cdr: OK
```

If the file has been altered during transmission, the output of the integrity check would be:

```
$ ./cdr-md5.sh sipwise_001_20071110123000_0000000004.cdr
/tmp/sipwise_001_20071110123000_0000000004.cdr: FAILED
md5sum: WARNING: 1 of 1 computed checksum did NOT match
```

9.4.4 File Transfer

Billing files are created twice per hour at minutes `25` and `55` and are stored in the home directory of the `cdrexport` user. If the amount of records within the transmission interval exceeds the threshold of 5000 records per file, multiple billing files are created. If no billing records are found for an interval, a billing file without body data is constructed for easy detection of lost billing files on the 3rd party side.

CDR and EDR files are fetched by a 3rd party billing system using SFTP or SCP with either public key or password authentication using the username cdrexport.
If public key authentication is chosen, the public key file has to be stored in the file `~/.ssh/authorized_keys2` below the home directory of the `cdrexport` user. Otherwise, a password has to be set for the user.

The 3rd party billing system is responsible for deleting CDR files after fetching them.

Note

The `cdrexport` user is kept in a jailed environment on the system, so it has only access to a very limited set of commandline utilities.

---

10 Provisioning REST API Interface

The sip:provider CE provides the REST API interface for interconnection with 3rd party tools.

The sip:provider CE provides a REST API to provision various functionality of the platform. The entry point - and at the same time the official documentation - is at *https://<your-ip>:1443/api*. It allows both administrators and resellers (in a limited scope) to manage the system.

You can either authenticate via username and password of your administrative account you're using to access the admin panel, or via SSL client certificates. Find out more about client certificate authentication in the online API documentation.

10.1 API Workflows for Customer and Subscriber Management

The typical tasks done on the API involve managing customers and subscribers. The following chapter focuses on creating, changing and deleting these resources.

The standard life cycle of a customer and subscriber is:

1. Create customer contact
2. Create customer
3. Create subscribers within customer
4. Modify subscribers
5. Modify subscriber preferences (features)
6. Terminate subscriber
7. Terminate customer

The boiler-plate to access the REST API is described in the online API documentation at */api/#auth*. A simple example in Perl using password authentication looks as follows:

```
#!/usr/bin/perl -w
use strict;
use v5.10;

use LWP::UserAgent;
use JSON qw();

my $uri = 'https://ngcp.example.com:1443';
my $ua = LWP::UserAgent->new;
my $user = 'myusername';
my $pass = 'mypassword';
$ua->credentials('ngcp.example.com:1443', 'api_admin_http', $user, $pass);
my ($req, $res);
```

For each customer you create, you need to assign a billing profile id. You either have the ID stored somewhere else, or you need to fetch it by searching for the billing profile handle.

```
my $billing_profile_handle = 'my_test_profile';
$req = HTTP::Request->new('GET', "$uri/api/billingprofiles/?handle=$billing_profile_handle" ↩
    );
$res = $ua->request($req);
if($res->code != 200) {
    die "Failed to fetch billing profile: ".$res->decoded_content."\n";
}
my $billing_profile = JSON::from_json($res->decoded_content);
my $billing_profile_id = $billing_profile->{_embedded}->{'ngcp:billingprofiles'}->{id};
say "Fetched billing profile, id is $billing_profile_id";
```

A customer is mainly a billing container for subscribers without a real identification other than the *external_id* property you might have stored somewhere else (e.g. the ID of the customer in your CRM). To still easily identify a customer, a customer contact is required. It is created using the */api/customercontacts/* resource.

```
$req = HTTP::Request->new('POST', "$uri/api/customercontacts/");
$req->header('Content-Type' => 'application/json');
$req->content(JSON::to_json({
    firstname => 'John',
    lastname => 'Doe',
    email => 'john.doe\@example.com'
}));
$res = $ua->request($req);
if($res->code != 201) {
    die "Failed to create customer contact: ".$res->decoded_content."\n";
}
my $contact_id = $res->header('Location');
$contact_id =~ s/^.+\/(\d+)$/$1/; # extract the ID from the Location header
say "Created customer contact, id is $contact_id";
```

Important
To get the ID of the recently created resource, you need to parse the *Location* header. In future, this approach will be changed for POST requests. The response will also optionally return the ID of the resource. It will be controlled via the *Prefer: return=representation* header as it is already the case for PUT and PATCH.

Warning
The example above implies the fact that you access the API via a reseller user. If you are accessing the API as the admin user, you also have to provide a *reseller_id* parameter defining the reseller this contact belongs to.

Once you have created the customer contact, you can create the actual customer.

```
$req = HTTP::Request->new('POST', "$uri/api/customers/");
```

```
$req->header('Content-Type' => 'application/json');
$req->content(JSON::to_json({
    status => 'active',
    contact_id => $contact_id,
    billing_profile_id => $billing_profile_id,
    type => 'sipaccount',
    external_id => undef, # can be set to your crm's customer id
}));
$res = $ua->request($req);
if($res->code != 201) {
    die "Failed to create customer: ".$res->decoded_content."\n";
}
my $customer_id = $res->header('Location');
$customer_id =~ s/^.+\/(\d+)$/$1/; # extract the ID from the Location header
say "Created customer, id is $customer_id";
```

Once you have created the customer, you can add subscribers to it. One customer can hold multiple subscribers, up to the *max_subscribers* property which can be set via */api/customers/*. If this property is not defined, a virtually unlimited number of subscribers can be added.

```
$req = HTTP::Request->new('POST', "$uri/api/subscribers/");
$req->header('Content-Type' => 'application/json');
$req->content(JSON::to_json({
    status => 'active',
    customer_id => $customer_id,
    primary_number => { cc => 43, ac => 9876, sn => 10001 }, # the main number
    alias_numbers => [ # as many alias numbers the subscriber can be reached at (or skip ↩
        param if none)
        { cc => 43, ac => 9877, sn => 10001 },
        { cc => 43, ac => 9878, sn => 10001 }
    ],
    username => 'test_10001'
    domain => 'ngcp.example.com',
    password => 'secret subscriber pass',
    webusername => 'test_10001',
    webpassword => undef, # set undef if subscriber shouldn't be able to log into sipwise ↩
        csc
    external_id => undef, # can be set to the operator crm's subscriber id
}));
$res = $ua->request($req);
if($res->code != 201) {
    die "Failed to create subscriber: ".$res->decoded_content."\n";
}
my $subscriber_id = $res->header('Location');
$subscriber_id =~ s/^.+\/(\d+)$/$1/; # extract the ID from the Location header
say "Created subscriber, id is $subscriber_id";
```

> **Important**
>
> A domain must exist before creating a subscriber. You can create the domain via */api/domains/*.

At that stage, the subscriber can connect both via SIP and XMPP, and can be reached via the primary number, all alias numbers, as well as via the SIP URI.

If you want to set call forwards for the subscribers, then perform an API call as follows.

```
$req = HTTP::Request->new('PUT', "$uri/api/callforwards/$subscriber_id");
$req->header('Content-Type' => 'application/json');
$req->header('Prefer' => "return=minimal"); # use return=representation to get full json ←
    response
$req->content(JSON::to_json({
    cfna => { # set a call-forward if subscriber is not registered
        destinations => [
            { destination => "4366610001", timeout => 10  }, # ring this for 10s
            { destination => "4366710001", timeout => 300 }, # if no answer, ring that for ←
                300s
        ],
        times => undef # no time-based call-forward, trigger cfna always
    }
}));
$res = $ua->request($req);
if($res->code != 204) { # if return=representation, it's 200
    die "Failed to set cfna for subscriber: ".$res->decoded_content."\n";
}
```

You can set cfu, cfna, cft and cft via this API call, also all at once. Destinations can be hunting lists as described above or just a single number. Also, a time set can be provided to trigger call forwards only during specific time periods.

To provision certain features of a subscriber, you can manipulate the subscriber preferences. You can find a full list of preferences available for a subscriber at */api/subscriberpreferencedefs/*.

```
$req = HTTP::Request->new('GET', "$uri/api/subscriberpreferences/$subscriber_id");
$res = $ua->request($req);
if($res->code != 200) {
    die "Failed to fetch subscriber preferences: ".$res->decoded_content."\n";
}
my $prefs = JSON::from_json($res->decoded_content);
delete $prefs->{_links}; # not needed in update

$prefs->{prepaid_library} = 'libinewrate'; # switch to inew billing
$prefs->{block_in_clir} = JSON::true; # reject incoming anonymous calls
$prefs->{block_in_list} = [ # reject calls from the following numbers:
    '4366412345', # this particular number
    '431*', # all vienna/austria numbers
```

```
];
$req = HTTP::Request->new('PUT', "$uri/api/subscriberpreferences/$subscriber_id");
$req->header('Content-Type' => 'application/json');
$req->header('Prefer' => "return=minimal"); # use return=representation to get full json ←
    response
$req->content(JSON::to_json($prefs));
$res = $ua->request($req);
if($res->code != 204) {
    die "Failed to update subscriber preferences: ".$res->decoded_content."\n";
}
say "Updated subscriber preferences";
```

Modifying numbers assigned to a subscriber, changing the password, locking a subscriber, etc. can be done directly on the subscriber resource.

```
$req = HTTP::Request->new('GET', "$uri/api/subscribers/$subscriber_id");
$res = $ua->request($req);
if($res->code != 200) {
    die "Failed to fetch subscriber: ".$res->decoded_content."\n";
}
my $sub = JSON::from_json($res->decoded_content);
delete $sub->{_links}; # not needed in update
push @{ $sub->{alias_numbers} }, { cc => 1, ac => 5432, sn => $t }; # add this number
push @{ $sub->{alias_numbers} }, { cc => 1, ac => 5433, sn => $t }; # add another number

$req = HTTP::Request->new('PUT', "$uri/api/subscribers/$subscriber_id");
$req->header('Content-Type' => 'application/json');
$req->header('Prefer' => "return=minimal"); # use return=representation to get full json ←
    response
$req->content(JSON::to_json($sub));
$res = $ua->request($req);
if($res->code != 204) {
    die "Failed to update subscriber: ".$res->decoded_content."\n";
}
say "Updated subscriber";
```

At the end of a subscriber life cycle, it can be terminated. Once terminated, you can NOT recover the subscriber anymore.

```
$req = HTTP::Request->new('DELETE', "$uri/api/subscribers/$subscriber_id");
$res = $ua->request($req);
if($res->code != 204) {
    die "Failed to terminate subscriber: ".$res->decoded_content."\n";
}
say "Terminated subscriber";
```

Note that certain information is still available in the internal database to perform billing/rating of calls done by this subscriber. Nevertheless, the data is removed from the operational tables of the database, so the subscriber is not able to connect to the system, login or make calls/chats.

Resources modification can be done via the GET/PUT combination. Alternatively, you can add, modify or delete single properties of a resource without actually fetching the whole resource. See an example below where we terminate the status of a customer using the PATCH method.

```
$req = HTTP::Request->new('PATCH', "$uri/api/customers/$customer_id");
$req->header('Content-Type' => 'application/json-patch+json');
$req->header('Prefer' => "return=minimal"); # use return=representation to get full json ←
    response
$req->content(JSON::to_json([
    { op => 'replace', path => '/status', value => 'terminated' }
]));
$res = $ua->request($req); # this will also terminate all still active subscribers
if($res->code != 204) {
    die "Failed to terminate customer: ".$res->decoded_content."\n";
}
say "Terminated customer";
```

10.2 API performance considerations

The REST API is designed with pagination support built-in. It is mandatory, to implement pagination in your API clients. If you circumvent pagination by setting the number of rows requested in one API call to a very high number the following side effects may appear:

1. An HTTP timeout at the gateway may appear. The default timeout limit is set to 60s. It can be modified via a customtt file: */etc/ngcp-config/templates/etc/nginx/sites-available/ngcp-panel_admin_api.customtt.tt2*.
2. Other parts of the system may become unresponsive due to mysql table locks. This especially applies to endpoints related to the Customers entity.

11 Configuration Framework

The sip:provider CE provides a configuration framework for consistent and easy to use low level settings management. A basic usage of the configuration framework only needs two actions already used in previous chapters:

- Edit */etc/ngcp-config/config.yml* file.
- Execute *ngcpcfg apply 'my commit message'* command.

Low level management of the configuration framework might be required by advanced users though. This chapter explains the architecture and usage of the NGCP configuration framework. If the basic usage explained above fits your needs, feel free to skip this chapter and return to it when your requirements change.

A more detailed workflow of the configuration framework for creating a configuration file consists of 7 steps:

- Generation or editing of configuration templates and/or configuration values.
- Generation of the configuration files based on configuration templates and configuration values defined in config.yml, constants.yml and network.yml files.
- Execution of *prebuild* commands if defined for a particular configuration file or configuration directory.
- Placement of the generated configuration file in the target directory. This step is called *build* in the configuration framework.
- Execution of *postbuild* commands if defined for that configuration file or configuration directory.
- Execution of *services* commands if defined for that configuration file or configuration directory. This step is called *services* in the configuration framework.
- Saving of the generated changes. This step is called *commit* in the configuration framework.

11.1 Configuration templates

The sip:provider CE provides configuration file templates for most of the services it runs. These templates are stored in the directory */etc/ngcp-config/templates*.

Example: Template files for */etc/ngcp-sems/sems.conf* are stored in */etc/ngcp-config/templates/etc/ngcp-sems/*.

There are different types of files in this template framework, which are described below.

11.1.1 .tt2 and .customtt.tt2 files

These files are the main template files that will be used to generate the final configuration file for the running service. They contain all the configuration options needed for a running sip:provider CE system. The configuration framework will combine these files with the values provided by *config.yml*, *constants.yml* and *network.yml* to generate the appropriate configuration file.

Example: Let's say we are changing the IP used by kamailio load balancer on interface *eth0* to IP 1.2.3.4. This will change kamailio's listen IP address, when the configuration file is generated. A quick look to the template file under */etc/ngcp-config/templates/etc/ka* will show a line like this:

```
listen=udp:[% ip %]:[% kamailio.lb.port %]
```

After applying the changes with the *ngcpcfg apply 'my commit message'* command, a new configuration file will be created under */etc/kamailio/lb/kamailio.cfg* with the proper values taken from the main configuration files (in this case *network.yml*):

```
listen=udp:1.2.3.4:5060
```

All the low-level configuration is provided by these .tt2 template files and the corresponding config.yml file. Anyway, advanced users might require a more particular configuration.

Instead of editing .tt2 files, the configuration framework recognises .customtt.tt2 files. These files are the same as .tt2, but they have higher priority when the configuration framework creates the final configuration files. An advanced user should create a .customtt.tt2 file from a copy of the corresponding .tt2 template and leave the .tt2 template untouched. This way, the user will have his personalized configuration and the system will continue providing a working, updated configuration template in .tt2 format.

Example: We'll create */etc/ngcp-config/templates/etc/lb/kamailio.cfg.customtt.tt2* and use it for our personalized configuration. In this example, we'll just append a comment at the end of the template.

```
cd /etc/ngcp-config/templates/etc/kamailio/lb
cp kamailio.cfg.tt2 kamailio.cfg.customtt.tt2
echo '# This is my last line comment' >> kamailio.cfg.customtt.tt2
ngcpcfg apply 'my commit message'
```

The ngcpcfg command will generate */etc/kamailio/kamailio.cfg* from our custom template instead of the general one.

```
tail -1 /etc/kamailio/kamailio.cfg
# This is my last line comment
```

Tip

The tt2 files use the Template Toolkit language. Therefore you can use all the feature this excellent toolkit provides within ngcpcfg's template files (all the ones with the .tt2 suffix).

11.1.2 .prebuild and .postbuild files

After creating the configuration files, the configuration framework can execute some commands before and after placing that file in its target directory. These commands usually are used for changing the file's owner, groups, or any other attributes. There are some rules these commands need to match:

- They have to be placed in a *.prebuild* or *.postbuild* file in the same path as the original *.tt2* file.
- The file name must be the same as the configuration file, but having the mentioned suffixes.
- The commands must be *bash* compatible.
- The commands must return 0 if successful.

- The target configuration file is matched by the environment variable *output_file*.

Example: We need *www-data* as owner of the configuration file */etc/ngcp-ossbss/provisioning.conf*. The configuration framework will by default create the configuration files with root:root as owner:group and with the same permissions (rwx) as the original template. For this particular example, we will change the owner of the generated file using the *.postbuild* mechanism.

```
echo 'chgrp www-data ${output_file}' \
  > /etc/ngcp-config/templates/etc/ngcp-ossbss/provisioning.conf.postbuild
```

11.1.3 .services files

.services files are pretty similar and might contain commands that will be executed after the *build* process. There are two types of *.services* files:

- The particular one, with the same name as the configuration file it is associated to.
  Example: */etc/ngcp-config/templates/etc/asterisk/sip.conf.services* is associated to */etc/asterisk/sip.conf*
- The general one, named *ngcpcfg.services* wich is associated to every file in its target directory.
  Example: */etc/ngcp-config/templates/etc/asterisk/ngcpcfg.services* is associated to every file under */etc/asterisk/*

When the *services* step is triggered all *.services* files associated to a changed configuration file will be executed. In case of the general file, any change to any of the configuration files in the directory will trigger the execution of the commands.

Tip

If the service script has the execute flags set (*chmod +x $file*) it will be invoked directly. If it doesn't have execute flags set it will be invoked under bash. Make sure the script is bash compatible if you do not set execute permissions on the service file.

These commands are usually service reload/restarts to ensure the new configuration has been loaded by running services.

Note

The configuration files mentioned in the following example usually already exist on the platform. Please make sure you don't overwrite any existing files if following this example.

Example:

```
echo '/etc/init.d/mysql restart' \
  > /etc/ngcpcfg-config/templates/etc/mysql/my.cnf.services
echo '/etc/init.d/asterisk restart' \
  > /etc/ngcpcfg-config/templates/etc/asterisk/ngcpcfg.services
```

In this example we created two *.services* files. Now, each time we trigger a change to */etc/mysql.my.cnf* or to */etc/asterisk/** we'll see that MySQL or Asterisk services will be restarted by the ngcpcfg system.

11.2 config.yml, constants.yml and network.yml files

The */etc/ngcp-config/config.yml* file contains all the user-configurable options, using the YAML (YAML Ain't Markup Language) syntax.

The */etc/ngcp-config/constants.yml* file provides configuration options for the platform that aren't supposed to be edited by the user. Do not manually edit this file unless you really know what you're doing.

The */etc/ngcp-config/network.yml* file provides configuration options for all interfaces and IP addresses on those interfaces. You can use the *ngcp-network* tool for conveniently change settings without having to manually edit this file.

The */etc/ngcp-config/ngcpcfg.cfg* file is the main configuration file for ngcpcfg itself. Do not manually edit this file unless you really know what you're doing.

11.3 ngcpcfg and its command line options

The ngcpcfg utility supports the following command line options:

11.3.1 apply

The *apply* option is a short-cut for the options "check && build && services && commit" and also executes *etckeeper* to record any modified files inside /etc. It is the recommended option to use the ngcpcfg framework unless you want to execute any specific commands as documented below.

11.3.2 build

The *build* option generates (and therefore also updates) configuration files based on their configuration (config.yml) and template files (.tt2). Before the configuration file is generated a present .prebuild will be executed, after generation of the configuration file the according .postbuild script (if present) will be executed. If a *file* or *directory* is specified as argument the build will generate only the specified configuration file/directory instead of running through all present templates.

Example: to generate only the file /etc/nginx/sites-available/ngcp-panel you can execute:

```
ngcpcfg build /etc/nginx/sites-available/ngcp-panel
```

Example: to generate all the files located inside the directory /etc/nginx/ you can execute:

```
ngcpcfg build /etc/nginx/
```

11.3.3 commit

The *commit* option records any changes done to the configuration tree inside */etc/ngcp-config*. The commit option should be executed when you've modified anything inside the configuration tree.

11.3.4 decrypt

Decrypt /etc/ngcp-config-crypted.tgz.gpg and restore configuration files, doing the reverse operation of the *encrypt* option. Note: This feature is only available if the ngcp-ngcpcfg-locker package is installed.

11.3.5 diff

Show uncommited changes between ngcpcfg's Git repository and the working tree inside */etc/ngcp-config*. Iff the tool doesn't report anything it means that there are no uncommited changes. If the *--addremove* option is specified then new and removed files (iff present) that are not yet (un)registered to the repository will be reported, no further diff actions will be executed then. Note: This option is available since ngcp-ngcpcfg version 0.11.0.

11.3.6 encrypt

Encrypt /etc/ngcp-config and all resulting configuration files with a user defined password and save the result as /etc/ngcp-config-crypted.tgz.gpg. Note: This feature is only available if the ngcp-ngcpcfg-locker package is installed.

11.3.7 help

The *help* options displays ngcpcfg's help screen and then exits without any further actions.

11.3.8 initialise

The *initialise* option sets up the ngcpcfg framework. This option is automatically executed by the installer for you, so you shouldn't have to use this option in normal operations mode.

11.3.9 pull

Retrieve modifications from shared storage. Note: This option is available in the High Availability setup only.

11.3.10 push

Push modifications to shared storage and remote systems. After changes have been pushed to the nodes the *build* option will be executed on each remote system to rebuild the configuration files (unless the --nobuild has been specified, then the build step will be skipped). If hostname(s) or IP address(es) is given as argument then the changes will be pushed to the shared storage and to the given hosts only. If no host has been specified then the hosts specified in */etc/ngcp-config/systems.cfg* are used. Note: This option is available in the High Availability setup only.

11.3.11 services

The *services* option executes the service handlers for any modified configuration file(s)/directory.

11.3.12 status

The *status* option provides a human readable interface to check the state of the configuration tree. If you are unsure what should be done as next step or if want to check the current state of the configuration tree just invoke *ngcpcfg status*.

If everything is OK and nothing needs to be done the output should look like:

```
# ngcpcfg status
Checking state of ngcpcfg:
OK:   has been initialised already (without shared storage)
Checking state of configuration files:
OK:   nothing to commit.
Checking state of /etc files
OK:   nothing to commit.
```

If the output doesn't say "OK" just follow the instructions provided by the output of *ngcpcfg status*.

Further details regarding the ngcpcfg tool are available through *man ngcpcfg* on the Sipwise Next Generation Platform.

12 Network Configuration

Starting with version 2.7, the sip:provider CE uses a dedicated *network.yml* file to configure the IP addresses of the system. The reason for this is to be able to access all IPs of all nodes for all services from any particular node in case of a distributed system on one hand, and in order to be able the generate */etc/network/interfaces* automatically for all nodes based on this central configuration file.

12.1 General Structure

The basic structure of the file looks like this:

```
hosts:
  self:
    role:
      - proxy
      - lb
      - mgmt
    interfaces:
      - eth0
      - lo
    eth0:
      ip: 192.168.51.213
      netmask: 255.255.255.0
      type:
        - sip_ext
        - rtp_ext
        - web_ext
        - web_int
    lo:
      ip: 127.0.0.1
      netmask: 255.255.255.0
      type:
        - sip_int
        - ha_int
```

Some more complete, sample configuration is shown in network.yml Overview Section B.3 section of the handbook.

The file contains all configuration parameters under the main key: `hosts`

In sip:provider CE systems there is only one host entry in the file, and it's always named *self*.

12.1.1 Available Host Options

There are three different main sections for a host in the config file, which are *role*, *interfaces* and the actual interface definitions.

- *role*: The role setting is an array defining which logical roles a node will act as. Possible entries for this setting are:

- *mgmt*: This entry means the host is acting as management node for the platform. In a sip:provider CE system this option must always be set. The management node exposes the admin and CSC panels to the users and the APIs to external applications and is used to export CDRs.
- *lb*: This entry means the host is acting as SIP load-balancer for the platform. In a sip:provider CE system this option must always be set. The SIP load-balancer acts as an ingress and egress point for all SIP traffic to and from the platform.
- *proxy*: This entry means the host is acting as SIP proxy for the platform. In a sip:provider CE system this option must always be set. The SIP proxy acts as registrar, proxy and application server and media relay, and is responsible for providing the features for all subscribers provisioned on it.
- *db*: This entry means the host is acting as the database node for the platform. In a sip:provider CE system this option must always be set. The database node exposes the MySQL and Redis databases.
- *rtp*: This entry means the host is acting as the RTP relay node for the platform. In a sip:provider CE system this option must always be set. The RTP relay node runs the *rtpengine* NGCP component.

- *interfaces*: The interfaces setting is an array defining all interface names in the system. The actual interface details are set in the actual interface settings below. It typically includes `lo, eth0, eth1` physical and a number of virtual interfaces, like: `bond0, vlanXXX`
- *<interface name>*: After the interfaces are defined in the *interfaces* setting, each of those interfaces needs to be specified as a separate set of parameters.

Addtional main parameters of a node:

- *dbnode*: the sequence number (unique ID) of the node in the database cluster; not used in sip:provider CE system
- *status*: one of *online*, *offline*, *inactive*. *inactive* means that the node is up but is not ready to work in the cluster (installing process). *offline* means that the node is not reachable. *online* is a normal working node.

12.1.2 Interface Parameters

- `hwaddr`: MAC address of the interface
- `ip`: IPv4 address of the node
- `v6ip`: IPv6 address of the node; optional
- `netmask`: IPv4 netmask
- `advertised_ip`: the IP address that is used in SIP messages when the NGCP system is behind NAT/SBC. An example of such a deployment is *Amazon AMI*, where the server doesn't have a public IP, so *load-balancer* component of NGCP needs to know what his public domain is (→ `advertised_ip`).
- `type`: type of services that the node provides; these are usually the VLANs defined for a particular NGCP system.

Note

You can assign a type only once per node.

Available types are:

- `api_int`: internal, API-based communication interface. It is used for the internal communication of such services as faxserver, fraud detection and others.
- `aux_ext`: interface for potentially insecure external components like remote system log collection service.
- `mon_ext`: remote monitoring interface (e.g. SNMP)
- `rtp_ext`: main (external) interface for media traffic
- `sip_ext`: main (external) interface for SIP signalling traffic between NGCP and other SIP endpoints
- `sip_ext_incoming`: additional, optional interface for incoming SIP signalling traffic
- `sip_int`: internal SIP interface used by NGCP components (*lb, proxy, etc.*)
- `ssh_ext`: command line (SSH) remote access interface
- `web_ext`: interface for web-based or API-based provisioning and administration
- `web_int`: interface for the administrator's web panel, his API and generic internal API communication

Note

Please note that, apart from the standard ones described so far, there might be other *types* defined for a particular NGCP system.

- `vlan_raw_device`: tells which physical interface is used by the particular VLAN
- `post_up`: routes can be defined here (interface-based routing)
- `bond_XY`: specific to "bond0" interface only; these contain Ethernet bonding properties

12.2 Advanced Network Configuration

You have a typical deployment now and you are good to go, however you may need to do extra configuration depending on the devices you are using and functionality you want to achieve.

12.2.1 Extra SIP Sockets

By default, the load-balancer listens on the UDP and TCP ports 5060 (*kamailio→lb→port*) and TLS port 5061 (*kamailio→lb→tls→port*). If you need to setup one or more extra SIP listening ports or IP addresses in addition to those standard ports, please edit the *kamailio→lb→extra_sockets* option in your */etc/ngcp-config/config.yml* file.

The correct format consists of a label and value like this:

```
extra_sockets:
  port_5064: udp:10.15.20.108:5064
  test: udp:10.15.20.108:6060
```

The label is shown in the `outbound_socket` peer preference (if you want to route calls to the specific peer out via specific socket); the value must contain a transport specification as in example above (udp, tcp or tls). After adding execute `ngcpcfg apply`:

```
ngcpcfg apply 'added extra socket'
```

The direction of communication through this SIP extra socket is incoming+outgoing. The sip:provider CE will answer the incoming client registrations and other methods sent to the extra socket. For such incoming communication no configuration is needed. For the outgoing communication the new socket must be selected in the `outbound_socket` peer preference. For more details read the next section Section 12.2.2 that covers peer configuration for SIP and RTP in greater detail.

> **Important**
> In this section you have just added an extra SIP socket. RTP traffic will still use your *rtp_ext* IP address.

12.2.2 Extra SIP and RTP Sockets

If you want to use an additional interface (with a different IP address) for SIP signalling and RTP traffic you need to add your new interface in the */etc/network/interfaces* file. Also the interface must be declared in */etc/ngcp-config/network.yml*.

Suppose we need to add a new SIP socket and a new RTP socket on VLAN 100. You can use the *ngcp-network* tool for adding interfaces without having to manually edit this file:

```
ngcp-network --set-interface=eth0.100 --ip=auto --netmask=auto --hwaddr=auto --type= ↩
    sip_ext_incoming --type=rtp_int_100
```

The generated file should look like the following:

```
..
..
    eth0.100:
      hwaddr: ff:ff:ff:ff:ff:ff
      ip: 192.168.1.3
      netmask: 255.255.255.0
      type:
        - sip_ext_incoming
        - rtp_int_100
..
..
    interfaces:
      - lo
      - eth0
      - eth0.100
      - eth1
..
..
```

As you can see from the above example, extra SIP interfaces must have type *sip_ext_incoming*. While *sip_ext* should be listed only once per host, there can be multiple *sip_ext_incoming* interfaces. The direction of communication through this SIP interface is incoming only. The sip:provider CE will answer the incoming client registrations and other methods sent to this address and remember the interfaces used for clients' registrations to be able to send incoming calls to him from the same interface.

In order to use the interface for the outbound SIP communication it is necessary to add it to extra_sockets section in */etc/ngcp-config/config.yml* and select in the `outbound_socket` peer preference. So if using the above example we want to use the vlan100 IP as source interface towards a peer, the corresponding section may look like the following:

```
    extra_sockets:
      port_5064: udp:10.15.20.108:5064
      test: udp:10.15.20.108:6060
      int_100: udp:192.168.1.3:5060
```

The changes have to be applied:

```
ngcpcfg apply 'added extra SIP and RTP socket'
```

After applying the changes, a new SIP socket will listen on IP `192.168.1.3` and this socket can now be used as source socket to send SIP messages to your peer for example. In above example we used label *int_100*. So the new label "int_100" is now shown in the `outbound_socket` peer preference.

Also, RTP socket is now listening on `192.168.1.3` and you can choose the new RTP socket to use by setting parameter `rtp_interface` to the Label "int_100" in your Domain/Subscriber/Peer preferences.

13 Licenses

The Sipwise NGCP — starting from mr5.5.1 release — implements *software licensing* primarily for the commercial products PRO and CARRIER. However as a CE platform operator you may also see a new process running on the system: `"licensed"`. The only purpose of this software module is to collect anonymous statistics about the system usage, namely the following performance indicators are recorded:

- number of provisioned subscribers
- number of registered subscribers
- number of concurrent calls

The **anonymous usage statistics is enabled by default but you can disable it**. In order to do that you have to edit the main configuration file */etc/ngcp-config/config.yml* and set `general.anonymous_usage_statistics` parameter to `no`. Then apply the new configuration with the usual command: `ngcpcfg apply "Disabled anon.  usage stat."`

Tip

If the NGCP operator does not want to have the license client package (`ngcp-license-client` and `ngcp-license-module`) on his system at all, it is possible to replace it with a dummy package: `ngcp-license-client-dummy`. This dummy package does not contain any licensing software, and is available from mr5.5.2 NGCP release.

---

14 Software Upgrade

14.1 Release Notes

The sip:provider CE version mr5.5.10 has the following important changes:

- Upgrade Debian from Debian 8 (jessie) to Debian 9 (stretch). SysV is still the init system in use. [TT#21905]
- Migration from MariaDB 10.0 to MariaDB 10.1 [TT#21824]
- [PRO/Carrier] Upgrade GlusterFS from 3.8.4 to 3.12.1 [TT#21896]
- [PRO/Carrier] Remove now unused ngcp-glusterfs-config package. [TT#23433]
- The monitoring backend has been migrated away completely from Redis to InfluxDB [TT#18250]
- [PRO/Carrier] ngcp-collective-check will now check the free swap space. [TT#22752]
- [PRO/Carrier] The ngcp-snmp-agent supports logging to syslog. [TT#21756]
- Add ability to configure supported SSL protocols/ciphers in config.yml [TT#12195]
- [PRO/Carrier] Discontinued and removed sems-app and sems-ha (sems-pbx takes over their tasks) [TT#22069] [TT#22070]
- [PRO/Carrier] Presence event as-feature-event is now supported in kamailio with optional CloudPBX module [TT#18851]
- [PRO/Carrier] Extension dialing within Auto-Attendant is now possible with CloudPBX module [TT#18736]
- Implemented request and response processing runtime logging in proxy [TT#22659]
- Updated asterisk to 13.14.1 [TT#20081]
- Updated kamailio to 4.4.6 [TT#18950]
- Implemented ability to check History-Info against allowed_clis or aliases per preference [TT#19559]
- [PRO/Carrier] Add support for the new HTTP/2 and JWT based protocol for sending push messages to Apple/iOS devices (APNs) [TT#14952]
- [PRO/Carrier] Different SIP domains can now be configured with distinct authentication data for sending push messages (Google GCM and Apple/iOS APNs) [TT#18841]
- Add support for dynamically adding iptables firewall rules for RTP proxy media ports as they are opened and closed [TT#19350]
- [PRO/Carrier] Add license management and enforcement system [TT#23200]
- Optionally send anonymous usage statistics [TT#23200]
- Adressed an issue in ngcp-credit-warning where it reported only 10 customers [TT#10964]
- Adressed an issue with reseller termination via the API and his related contract remained active [TT#21106]
- Adressed issues with customer and susbcriber lock level settings [TT#21271, TT#14285, TT#19656, TT#18765]

- Addressed an issue with the fraud notification script not enlisting all the automatically locked subscribers [TT#18755]
- Improved NGCP Panel UI and the API performance [TT#22827]
- Added partitioning support for accounting.cdr [TT#3668]
- Optimised performance, addressed several minor issues and introduced partitioning support in the acc-cleanup tool [TT#3668]
- Added a "Manager Secretary" feature, currently only supports Polycom phones [TT#20302]
- Addressed several minor issues in the rewrite rules data representation [TT#23426]
- Added POST method support in the API to generate invoices [TT#17848]
- libmyodbc is no longer supported in Debian and replaced by the MariaDB odbc connector library [TT#21307]
- [PRO/Carrier] - Addressed an issue with reseller manipulation and the rtcengine provisioning when the rtcengine access was not configured [TT#20690]
- [PRO/Carrier] Added Polycom phones provisioning support [TT#7515]
- [PRO/Carrier] Improved Yealink phones provisioning support [TT#18335]
- [PRO/Carrier] Improved error handling for the SMS delivery [TT#20095]
- [PRO/Carrier] Improved Glusterfs server ngcp volume control [TT#22435]
- [PRO/Carrier] Improved ngcp-sync-db to handle large blobs of data [TT#22806]
- Performance tuning:
  - Disable Transparent Huge Pages (THP) support [TT#22160] (Redis performance tuning)
  - Linux sysctl changes:
    - net.core.somaxconn = 512 [TT#22160] (Redis performance tuning)
    - vm.overcommit_memory = 1 [TT#22160] (Redis performance tuning)
    - vm.swappiness = 10 [TT#22407] (prevent unnecessary swapping with available RAM)
    - sunrpc.min_resvport = 700 [TT#22658] (prevent heartbeat and NFS ports collisions)

Please find the complete changelog in our release notes on our WEB site.

14.2 Upgrade from previous versions to mr5.5.10

> **Warning**
> Debian 9 (stretch) provides OpenSSH 7 with the ssh-dss (DSA) public key algorithm disabled (see more information here). Please ensure you are NOT using ssh-dss (DSA) based SSH keys, otherwise migrate to modern ssh-ed25519 (ED25519) algorithm BEFORE the upgrade.

The sip:provider CE system upgrade to mr5.5.10 will be performed in several steps:

- Verify APT source lists
- Upgrade the base system from Debian 8 (jessie) to Debian 9 (stretch)
- Upgrade the NGCP software packages
- Upgrade the NGCP configuration templates
- Upgrade the NGCP DB schema
- Upgrade the NGCP configuration schema
- Upgrade the base system within Debian 9 (stretch) to the latest package versions

14.2.1 Preparing for maintenance mode

Sipwise NGCP introduced **Maintenance Mode** with its mr5.4.1 release. The maintenance mode of NGCP will disable some background services (for instance, *mediator*) during the software upgrade. It thus prevents the system from getting into an inconsistent state while the upgrade is being performed. You can activate maintenance mode by applying a simple configuration change as described later.

> **Important**
> *For sip:provider CE systems running NGCP release older than mr5.4.1:* As upgrading from an earlier release to mr5.4.1 or later will result in a system being aware and making use of the maintenance mode, it is necessary to prepare the NGCP for it *before* the software upgrade.

Enable maintenance mode:

- Edit */etc/ngcp-config/config.yml* file:
  - For systems running NGCP release older than mr5.4.1: insert a new line: "`maintenance: yes`" in the `general` section of the configuration file
  - For systems running NGCP release mr5.4.1 or later: set the `general.maintenance` parameter to `yes`

```
general:
  maintenance: yes
```

Disabling background services

These tasks are for systems currently running NGCP release older than mr5.4.1. The *maintenance mode* of NGCP is not available in your system prior to the software upgrade. It is therefore necessary to manually disable the following background services to avoid a potential system inconsistency:

- Mediator
- Rate-o-mat

- CDR Exporter

Execute the following:

1. **Stop *mediator* and *rate-o-mat* services**:

```
service mediator stop
service ngcp-rate-o-mat stop
```

2. **Prevent background services from (re)starting**. You need to change configuration in the `/etc/ngcp-config/config.yml` file: set the `enable` parameter to "no":

```
mediator:
  enabled: no
...
rateomat:
  enable: no
```

3. In order to **disable the *CDR exporter* process**, add a new custom template file because there is no such file by default:

```
config="/etc/cron.d/ngcp-cdr-exporter"
customtt="/etc/ngcp-config/templates/etc/cron.d/ngcp-cdr-exporter.customtt.tt2"
[ -f "${customtt}" ] || cp "${config}" "${customtt}"
vim "${customtt}"
```

Comment out this single line in the file:

```
# 25,55 * * * *   root    . /etc/default/ngcp-roles; if ...
```

Please don't forget to reverse the manual changes after the software upgrade as described in Enabling background services section of the handbook!

For all NGCP systems, regardless of their version:

Apply configuration changes by executing :

```
ngcpcfg apply 'Enabling maintenance mode before the upgrade to mr5.5.10'
```

14.2.2 Set the proper software repositories

Warning
Ensure you are using the Sipwise APT repositories. Public Debian mirrors may not provide packages for old Debian releases anymore. Also, they might be outdated. Consider using Sipwise repositories for the time of the upgrade.

These commands are for systems currently running previous LTS release mr4.5.*

Execute the following commands as *root*:

```
echo "# Please visit /etc/apt/sources.list.d/ instead." > /etc/apt/sources.list

mkdir -p /etc/apt/sources.list.d
for file in /etc/apt/sources.list.d/*.list ; do mv "${file}" "${file}.DISABLED" ; done

NGCP_CURRENT_VERSION=$(cat /etc/ngcp_version)
cat > /etc/apt/sources.list.d/debian.list << EOF
# Debian repositories, deployed via upgrade ${NGCP_CURRENT_VERSION}->mr5.5.10
deb https://debian.sipwise.com/debian/ jessie main contrib non-free
#deb-src https://debian.sipwise.com/debian/ jessie main contrib non-free
#
deb https://debian.sipwise.com/debian-security/ jessie-security main contrib non-free
#deb-src https://debian.sipwise.com/debian-security/ jessie-security main contrib non-free
#
deb https://debian.sipwise.com/debian/ jessie-updates main contrib non-free
#deb-src https://debian.sipwise.com/debian/ jessie-updates main contrib non-free
EOF

NGCP_CURRENT_VERSION=$(cat /etc/ngcp_version)
cat > /etc/apt/sources.list.d/sipwise.list << EOF
# NGCP_MANAGED_FILE
# Sipwise repository, deployed via upgrade ${NGCP_CURRENT_VERSION}->mr5.5.10
deb https://deb.sipwise.com/spce/${NGCP_CURRENT_VERSION}/ jessie main
#deb-src https://deb.sipwise.com/spce/${NGCP_CURRENT_VERSION}/ jessie main
EOF
```

These commands are for systems currently running previous build mr5.5.*

Execute the following commands as *root*:

```
echo "# Please visit /etc/apt/sources.list.d/ instead." > /etc/apt/sources.list

mkdir -p /etc/apt/sources.list.d
for file in /etc/apt/sources.list.d/*.list ; do mv "${file}" "${file}.DISABLED" ; done

NGCP_CURRENT_VERSION=$(cat /etc/ngcp_version)
cat > /etc/apt/sources.list.d/debian.list << EOF
# Debian repositories, deployed via upgrade ${NGCP_CURRENT_VERSION}->mr5.5.10
deb https://debian.sipwise.com/debian/ stretch main contrib non-free
#deb-src https://debian.sipwise.com/debian/ stretch main contrib non-free
#
deb https://debian.sipwise.com/debian-security/ stretch-security main contrib non-free
#deb-src https://debian.sipwise.com/debian-security/ stretch-security main contrib non-free
#
deb https://debian.sipwise.com/debian/ stretch-updates main contrib non-free
#deb-src https://debian.sipwise.com/debian/ stretch-updates main contrib non-free

deb https://debian.sipwise.com/debian-debug/ stretch-debug main contrib non-free
```

```
#deb-src https://debian.sipwise.com/debian-debug/ stretch-debug main contrib non-free
EOF

NGCP_CURRENT_VERSION=$(cat /etc/ngcp_version)
cat > /etc/apt/sources.list.d/sipwise.list << EOF
# NGCP_MANAGED_FILE
# Sipwise repository, deployed via upgrade ${NGCP_CURRENT_VERSION}->mr5.5.10
deb https://deb.sipwise.com/spce/${NGCP_CURRENT_VERSION}/ stretch main
#deb-src https://deb.sipwise.com/spce/${NGCP_CURRENT_VERSION}/ stretch main
EOF
```

Warning

Do not use "ngcpcfg apply/build" after executing the steps from the above block, as otherwise the changes will be overwritten and you will have to redo these steps.

Run "apt-get update" and ensure you have no warnings/errors here.

14.2.3 Switch to new repositories

To upgrade the sip:provider CE to release mr5.5.10, execute the following commands:

```
NGCP_CURRENT_VERSION=$(cat /etc/ngcp_version)
sed -i "s/$NGCP_CURRENT_VERSION/mr5.5.10/" /etc/apt/sources.list.d/sipwise.list
sed -i "s/jessie/stretch/g" /etc/apt/sources.list.d/sipwise.list /etc/apt/sources.list.d/ ↩
    debian.list
grep -q "debian-debug" /etc/apt/sources.list.d/debian.list || \
  echo "deb https://debian.sipwise.com/debian-debug/ stretch-debug main contrib non-free"  ↩
      >> /etc/apt/sources.list.d/debian.list

apt-get update
apt-get install ngcp-upgrade-ce
```

14.2.4 Upgrade the sip:provider CE

Run the upgrade script as *root* like this:

```
ngcp-upgrade
```

Note

sip:provider CE can be upgraded to mr5.5.10 from previous release or previous build only. The script ngcp-upgrade will find all the possible destination releases for the upgrade and allow to choose the proper one.

Note
If there is an error during the upgrade, the ngcp-upgrade script will request you to solve it. Once you've fixed the problem, just execute `ngcp-upgrade` again and it will continue from the previous step.

The upgrade script will ask you to confirm that you want to start. Read the given information **carefully**, and if you agree, proceed with *y*.

The upgrade process will take several minutes, depending on your network connection and server performance. After everything has been updated successfully, it will finally ask you to reboot your system. Confirm to let the system reboot (it will boot with an updated kernel).

Once up again, double-check your config file */etc/ngcp-config/config.yml* (sections will be rearranged now and will contain more parameters) and your domain/subscriber/peer configuration and test the setup.

14.2.5 `ngcp-upgrade` options

The following options in `ngcp-upgrade` can be specially useful in some instances of upgrade:

- **`--step-by-step`**: confirm before proceeding to next step. With this option the upgrade operation is performed confirming every step before execution, with the possibility to instruct to continue without confirming further steps until the end (if confirmation is only needed for some steps at the beginning).
- **`--pause-before-step STEP_NAME`**: pause execution before step, given by the name of the script (e.g. "backup_mysql_db"). This option can be useful in several scenarios, for example:
  - to help to debug problems or work around known problems during upgrades. In this case the operator can pause at a given step known to be problematic or just before a problematic set, perform some manual checks or changes, then continue the upgrade until another step (with confirmation like with the recent option `--step-by-step`), or just continue without stop until the end
  - another use might be to help to speed up upgrades when it involves several nodes: they can all proceed in parallel when it's known to be safe to do so; then perform some parts in lock-step (some nodes waiting until others finish with some stage); then continue in parallel until the end
- **`--skip-db-backup`**: This will speed-up the process in cases where it's deemed unnecessary, and this is very likely in the upgrade of nodes other than the first.

14.3 Post-upgrade tasks

14.3.1 Disabling maintenance mode

In order to disable the *maintenance mode*, do the following:

- Disable the maintenance mode:

```
ngcpcfg set /etc/ngcp-config/config.yml "general.maintenance=no"
```

Enabling background services

If you upgraded from NGCP release earlier than mr5.4.1, you have to manually reverse the changes you applied before the upgrade in the (Disabling background services) section.

1. **Enable background services**:

```
ngcpcfg set /etc/ngcp-config/config.yml "mediator.enabled=yes"
ngcpcfg set /etc/ngcp-config/config.yml "rateomat.enable=yes"
```

2. In order to **enable the *CDR exporter* process**, remove the corresponding customtt file:

```
rm /etc/ngcp-config/templates/etc/cron.d/ngcp-cdr-exporter.customtt.tt2
```

For all NGCP systems, regardless of their previous version:

- Execute the command:

```
ngcpcfg apply 'Disable the maintenance mode after the upgrade to mr5.5.10'
```

14.3.2 Post-upgrade checks

When everything has finished successfully, check that replication is running. Check `ngcp-status`. Finally, do a basic functionality test. Check the web interface, register two test subscribers and perform a test call between them to ensure call routing works.

Note
You can find a backup of some important configuration files of your existing installation under */var/backup/ngcp-mr5.5.10-** (where * is a place holder for a timestamp) in case you need to roll back something at any time. A log file of the upgrade procedure is available at */var/backup/ngcp-mr5.5.10-*/upgrade.log*.

15 Backup, Recovery and Database Maintenance

15.1 sip:provider CE Backup

For any service provider it is important to maintain a reliable backup policy as it enables prompt services restoration after any force majeure event. Hence, we strongly suggest you to configure a backup procedure. The sip:provider CE can be integrated with any Debian compatible backup software.

15.1.1 What data to back up

- The database

This is the most important data in the system. All subscriber and billing information, CDRs, user preferences, etc. are stored in the MySQL server. It is strongly recommended to have up-to-date dumps of all the databases.

- System configuration

The system configuration files such as */etc/mysql/sipwise.cnf* and the */etc/ngcp-config/* directory should be included in the backup as well. We suggest backing up the whole */etc* folder.

- Exported CDRs (optional)

The */home/jail/home/cdrexport* directory contains the exported CDRs. It depends on your call data retention policy whether or not to remove these files after exporting them to an external system.

15.2 Recovery

In the worst case scenario, when the system needs to be recovered from a total loss, you only need 4 steps to get the services back online:

- Install the sip:provider CE as explained in chapter 2.
- Restore the */etc/ngcp-config/* directory and the */etc/mysql/sipwise.cnf* file from the backup, overwriting your local files.
- Restore the database from the latest MySQL dump.
- Apply the changes to bring the original configuration into effect:

```
ngcpcfg apply 'restored the system from the backup'
```

15.3 Reset Database

Important

All existing data will be wiped out! Use this script only if you want to clear all previously configured services and start configuration from scratch.

To reset database to its original state you can use a script provided by CE: * Execute *ngcp-reset-db*. It will assign new unique passwords for the NGCP services and reset all services. The script will also create dumps for all NGCP databases.

15.4 Accounting Data (CDR) Cleanup

Sipwise sip:provider CE offers an easy way to cleanup, backup or archive old accounting data — i.e. CDRs — that is not necessary for further processing any more, or must be deleted according to the law. There are some NGCP components designed for this purpose and they are commonly called *cleanuptools*. These are basically configurable scripts that interact with NGCP's `accounting` and `kamailio` databases, or remove exported CDR files in order to clean or archive the unnecessary data.

15.4.1 Cleanuptools Configuration

The configuration parameters of *cleanuptools* are located in the main NGCP configuration file: `/etc/ngcp-config/config.yml`. Please refer to the `config.yml` file description: Cleanuptools Configuration Data Section B.1.7 for configuration parameter details.

In case the system administrator needs to modify some configuration value, the new configuration must be activated in the usual way, by running the following commands:

```
> ngcpcfg apply 'Modified cleanuptools config'
```

As a result new configuration files will be generated for the accounting database and the exported CDR cleanup tools. Please read detailed description of those tools in subsequent sections of the handbook.

The NGCP system administrator can also select the time when cleanup scripts are run, by modifying the schedule here: `/etc/cron.d/cleanup-tools`

15.4.2 Accounting Database Cleanup

The script responsible for cleaning up the database is: `/usr/sbin/acc-cleanup.pl`

The configuration file used by the script is: `/etc/ngcp-cleanup-tools/acc-cleanup.conf`

An extract from a sample configuration file is provided here:

```
############

batch = 10000
```

```
archive-target = /var/backup/cdr
compress = gzip

username = dbcleaner
password = rcKamRdHhx7saYRbkJfP
host = localhost


connect accounting
time-column = from_unixtime(start_time)
backup-months = 2
backup-retro = 2
backup cdr

connect accounting
archive-months = 2
archive cdr

connect kamailio
time-column = time
cleanup-days = 90
cleanup acc

# Clean up after mediator by deleting old leftover acc entries and deleting
# old entries out of acc_trash and acc_backup
connect kamailio
time-column = time
cleanup-days = 30
cleanup acc_trash
cleanup acc_backup
```

The configuration file itself contains a detailed description of how database cleanup script works. It consists of a series of statements, one per line, which are going to be executed in sequence. A statement can either just set a variable to some value, or perform an action.

There are 3 types of actions the database cleanup script can take:

- backup CDRs
- archive CDRs
- cleanup CDRs

These actions are discussed in following sections.

A generic action is connecting to the proper database: `connect <database name>`

15.4.2.1 Backup CDRs

The database cleanup tool can create *monthly backups* of CDRs in the `accounting` database and store those data records in separate tables named: `cdr_YYYYMM`. The instruction in the configuration file looks like: `backup <table name>`, by default and typically it is: `backup cdr`

Configuration values that govern the backup procedure are:

- `time-column`: Which column in *cdr* table shows the month which a CDR belongs to.
- `batch`: How many records to process within a single SQL statement. If unset, less than or equals 0, all of them are processed at once.
- `backup-months`: How many months worth of records to keep in the *cdr* table — where current CDRs are stored — and not move into the monthly backup tables.

> **Important**
> Months are always processed as a whole, thus the value specifies how many months to keep AT MOST. In other words, if the script is started on December 15th and this value is set to "2", then all of December and November is kept, and all of October will be backed up.

- `backup-retro`: How many months to process for backups, going backwards in time. Using the example above, with this value set to "3", the months October, September and August would be backed up, while any older records would be left untouched.

15.4.2.2 Archive CDRs

The database cleanup tool can archive (dump) old monthly backup tables. The statement used for this purpose is: `archive <table name>`, by default and typically it is: `archive cdr`

This creates an SQL dump out of too old tables created by the `backup` statement and drop them afterwards from database. Archiving uses the following configuration values:

- `archive-months`: Uses the same logic as the `backup-months` variable above. If set to "12" and the script was started on December 15th, it will start archiving with the December table of the previous year.

> **Important**
> Note that the sum of `backup-retro` + `backup-months` values cannot be larger than `archive-months` value for the same table. Otherwise you end up creating empty monthly backup tables, only to dump and delete them right afterwards.

- `archive-target`: Target directory for writing the SQL dump files into. If explicitly specified as "/dev/null", then no actual archiving will be performed, but instead the tables will only be dropped from database.
- `compress`: If set to "gzip", then gzip the dump files after creation. If unset, do not compress.
- `host, username` and `password`: As dumping is performed by an external command, those variables are reused from the `connect` statement.

15.4.2.3 Cleanup CDRs

The database cleanup tool may do database table cleanup without performing backup. In order to do that, the statement: `cleanup <table name>` is used. Typically this has to be done in `kamailio` database, examples:

- `cleanup acc`
- `cleanup acc_trash`
- `cleanup acc_backup`

Basically the `cleanup` statement works just like the `backup` statement, but doesn't actually backup anything, but rather just deletes old records. Configuration values used by the procedure:

- `time-column`: Gives the database column name that shows the time of CDR creation.
- `batch`: The same as with `backup` statement.
- `cleanup-days`: Any record older than this many days will be deleted.

15.4.3 Exported CDR Cleanup

The script responsible for cleaning up exported CDR files is: `/usr/sbin/cleanup-old-cdr-files.pl`

The configuration file used by exported CDR cleanup script is: `/etc/ngcp-cleanup-tools/cdr-files-cleanup.yml`

A sample configuration file is provided here:

```
enabled: no
max_age_days: 30
paths:
  -
    path: /home/jail/home/*/20[0-9][0-9][0-9][0-9]/[0-9][0-9]
    wildcard: yes
    remove_empty_directories: yes
    max_age_days: ~
  -
    path: /home/jail/home/cdrexport/resellers/*/20[0-9][0-9][0-9][0-9]/[0-9][0-9]
    wildcard: yes
    remove_empty_directories: yes
    max_age_days: ~
  -
    path: /home/jail/home/cdrexport/system/20[0-9][0-9][0-9][0-9]/[0-9][0-9]
    wildcard: yes
    remove_empty_directories: yes
    max_age_days: ~
```

The exported CDR cleanup tool simply deletes CDR files in the directories provided in the configuration file, if those have already expired.

Configuration values that define the files to be deleted:

- `enabled`: Enable (`yes`) or disable (`no`) exported CDR cleanup.
- `max_age_days`: Gives the expiration time of the exported CDR files in days. There is a general value which may be overridden by a local value provided at a specific path. The local value is valid for the particular path only.
- `paths`: an array of path definitions
  - `path`: a path where CDR files are to be found and deleted; this may contain wildcard characters
  - `wildcard`: Enable (`yes`) or disable (`no`) using wildcards in the `path`
  - `remove_empty_directories`: Enable (`yes`) or disable (`no`) removing empty directories if those are found in the given `path`
  - `max_age_days`: the local expiration time value for files in the particular `path`

16 Platform Security, Performance and Troubleshooting

Once the sip:provider CE is in production, security and maintenance becomes really important. In this chapter, we'll go through a set of best practices for any production system.

16.1 Sipwise SSH access to sip:provider CE

The sip:provider CE provides SSH access to the system for Sipwise operational team for debugging and final tuning. Operational team uses user *sipwise* which can be logged in through SSH key only (password access is disabled) from dedicated access server *jump.sipwise.com* only.

To completely remove Sipwise access to your system, please execute as user root:

```
root@myserver:~# ngcp-support-access --disable && apt-get install ngcp-support-noaccess
```

Note

you have to execute the command above on each node of your sip:provider CE system!

Warning

please ensure that the script complete successfully:

```
 * Support access successfully disabled.
```

If you need to restore Sipwise access to the system, please execute as user root:

```
root@myserver:~# apt-get install ngcp-support-access && ngcp-support-access --enable
```

Warning

please ensure that the script complete successfully:

```
 * Support access successfully enabled.
```

16.2 Firewalling

16.2.1 Firewall framework

The sip:provider CE runs a wide range of services. In order to secure the platfrom while allowing access to the sip:provider CE, the NGCP configuration framework provides a set of predefined network zones. Services are aggregated into appropriate zones by default. Zones are assigned to network interfaces (and VLANs if applicable) in /etc/ngcp-config/network.yml.

Caution

Though the default firewall setup provided by the NGCP configuration framework provides a safe setup for sip:provider CE, security audits of the platform performed by qualified engineers before commissioning the platform into service are strongly recommended. Customization of the setup requires in-depth knowledge of firewalling principles in general and the *netfilter* facility in particular.

Table 14: NGCP network zones

Zone name	Description
ha_int	Internal cluster interface providing internal cluster communications between cluster pairs (heartbeat) and synchronization of data and configuration
mon_ext	Interface to conect external monitoring appliances (SNMP)
rtp_ext	Interface for external RTP media relay between sip:provider CE and endpoints (e.g. user agents, peers)
sip_ext	Interface for external SIP signalling between sip:provider CE and endpoints (e.g. user agents, peers)
sip_int	Interface for internal signalling, e.g. between load-balancers, proxies and applications servers
ssh_ext	Interface providing external access to the sip:provider CE command line interface
web_ext	Interface providing access to the customers' self-care Web panel
web_int	Interface for access to the administrative Web panel, its REST APIs and internal API communications

Note

Additional custom zones may be configured, but will not be automatically integrated into the firewall configuration.

To facilitate firewall functionality, sip:provider CE uses the Kernel's *netfilter* facility and *iptables-persistent* as an interface to *netfilter*. *Netfilter* is using *tables* and within that *chains* to store rules in this hierarchy: *table* → *chain* → *rule*. Default firewall setups of sip:provider CE do not use netfilter tables *nat* and *raw*, but only default table *filter*.

Note

Custom *nat* rules for IPv4 and IPv6 may be added in file /etc/ngcp-config/config.yml in sections *security→firewall→nat_rules4* and *security→firewall→nat_rules6*.

Each *chain* deploys a *default policy* handling packets which did not trigger and rule in a prticular *chain*.

Table 15: NGCP *netfilter* default policies

Chain	Default policy	Description
INPUT	DROP	Handling all packets directly destined for a sip:provider CE node (only packets matching a rule are allowed)
FORWARD	DROP	Handling all packets received by a sip:provider CE node and destined for another, non-local IP destination (no default rules added)
OUTPUT	ACCEPT	Handling all packets originating on a sip:provider CE node (no default rules added)
rtpengine	N/A	Container for rptengine rule to allow the rule to persist even when the Kernel module is unloaded (e.g. during upgrades)

The default firewall setup provided by sip:provider CE:

- adds rules to INPUT to secure access to platform and services
- blocks all traffic from and to FORWARD
- allows all OUTPUT traffic

16.2.2 NGCP firewall configuration

The sip:provider CE comes with a preconfigured set of firewall rules, which can be enabled and configured in `/etc/ngcp-config/config.yml` in section `security→firewall`. Refer to Section B.1.29 for available configuration options.

Firewall configuration is applied by running `ngcpcfg apply`. However, this will not activate new rules automatically to avoid inadvertent self-lockout. To finally activate new firewall rules run `iptables-apply`. This will prompt for another system logon to verify access remains available. If the prompt is not confirmed, firewall rules will automatically be reverted to the previous state re-enabling access to the command line.

Caution

The NGCP firewall subsystem by default is disabled in */etc/ngcp-config/config.yml* key `security.firewall.enable: no`. This is to avoid blocking any traffic inadvertently during installation. After the firewall subsystem has been configured appropriately, it needs to be enabled by setting `security.firewall.enable: yes` in */etc/ngcp-config/config.yml*.

16.2.3 IPv4 System rules

The following set of rules is added by the system upon activation of the firewall subsystem. Individual system rules are configured in */etc/ngcp-config/templates/etc/iptables/rules.v4.tt2* and */etc/ngcp-config/templates/etc/iptables/rules.v6.tt2*

Table 16: Firewall system rules

Zone	Chain	Target	Rule	Description
all	INPUT	rtpengine	`-p udp -j rtpengine`	Redirects all incoming UDP packets to chain *rtpengine* (putting RTPENGINE rule into a dedicated chain allows for the rule to persist even when the Kernel module gets unloaded, e.g. during upgrades)
all	rtpengine	RTPENGINE	`-p udp -j RTPENGINE --id 0`	Feeds all RTP packets to RTPENGINE Kernel module
n/a	INPUT	ACCEPT	`-i lo -j ACCEPT`	Accept all packets received by local loopback interface
all	INPUT	ACCEPT	`-m state --state RELATED,ESTABLISHED -j ACCEPT`	Accept all incoming packets tied to *related* or *established* connections
all	INPUT (IPv4)	ACCEPT	`-p icmp -m icmp --icmp-type 8 -j ACCEPT`	Accept all ICMP *echo* messages
all	INPUT (IPv4)	ACCEPT	`-p icmp -m icmp --icmp-type 0 -j ACCEPT`	Accept all ICMP *echo reply* messages
all	INPUT (IPv6)	ACCEPT	`-A INPUT -p ipv6-icmp -j ACCEPT`	Accept all ICMPv6 messages
all	INPUT	cluster	`-j cluster`	Divert all incoming packets to the *cluster* chain
all	cluster	ACCEPT	`-s <node_ip> -j ACCEPT`	Set of rules white-listing all IP-addresses owned by the NGCP platform for incoming traffic
api_int	INPUT	ACCEPT	`-p tcp --dport <ossbss.port> -j ACCEPT`	Set of rules for all *api_int* interfaces accepting all incoming packets for API port defined in */etc/ngcp-config/config.yml* with key *ossbss.port*
mon_ext	INPUT	ACCEPT	+-p udp -s *<snmpclient_ip>* --dport 161 -j ACCEPT	Set of rules for all *mon_ext* interfaces based on a list of IPs for all SNMP communities configured in *checktools.snmpd.communities*
rtp_ext	INPUT	ACCEPT/*name*	`-p udp --dport <rtpproxy.minport>: '<rtpproxy.maxport>' -j ACCEPT/name`	Set of rules for all *rtp_ext* interfaces accepting all incoming packets for RTP port range defined in */etc/ngcp-config/config.yml* with keys *rtpproxy.minport* and *rtpproxy.maxport* (see note below for custom options)

Table 16: (continued)

Zone	Chain	Target	Rule	Description
sip_ext	INPUT	ACCEPT	`-p udp --dport <kamailio.lb.port> -j ACCEPT`	Set of rules for all *sip_ext* interfaces accepting all packets on the loda balancer's SIP signalling port defined in */etc/ngcp-config/config.yml* with key *kamailio.lb.port* (UDP)
sip_ext	INPUT	ACCEPT	`-p tcp --dport <kamailio.lb.port> -j ACCEPT`	Set of rules for all *sip_ext* interfaces accepting all packets on the loda balancer's SIP signalling port defined in */etc/ngcp-config/config.yml* with key *kamailio.lb.port* (TCP)
sip_ext	INPUT	ACCEPT	`-p tcp --dport <kamailio.lb.tls.port> -j ACCEPT`	Set of rules for all *sip_ext* interfaces accepting all packets on the loda balancer's SIP signalling port defined in */etc/ngcp-config/config.yml* with key *kamailio.lb.tls.port* (TCP/TLS)
sip_ext	INPUT	ACCEPT	`-p tcp --dport 5222 -j ACCEPT`	Set of rules for all *sip_ext* interfaces accepting all packets on TCP port 5222 (XMPP client)
sip_ext	INPUT	ACCEPT	`-p tcp --dport 5269 -j ACCEPT`	Set of rules for all *sip_ext* interfaces accepting all packets on TCP port 5269 (XMPP server)
sip_ext	INPUT	ACCEPT	`-p tcp --dport <pushd.port> -j ACCEPT`	Set of rules for all *sip_ext* interfaces accepting all packets incoming for the *pushd* server port configured in */etc/ngcp-config/config.yml* with key *pushd.port*
ssh_ext	INPUT	ACCEPT	`-A INPUT -i <ssh_ext_interface> -p tcp -s <sshd.permit_support_from> --dport sshd.port -j ACCEPT`	List of rules to accept incoming packets for SSH on all *ssh_ext* interfaces from hosts configured in */etc/ngcp-config/config.yml* with key *sshd.permit_support_from*

Table 16: (continued)

Zone	Chain	Target	Rule	Description
web_ext	INPUT	ACCEPT	`-p tcp --dport <www_admin.http_csc.port> -j ACCEPT`	List of rules to accept incoming packets for the *Customer Self Care* interface defined in */etc/ngcp-config/config.yml* with key *www_admin.http_csc.port* on all *web_ext* interfaces
web_int	INPUT	ACCEPT	`-p tcp --dport <www_admin.http_admin.port> -j ACCEPT`	List of rules to accept incoming packets for the *Admin Panel* interface defined in */etc/ngcp-config/config.yml* with key *www_admin.http_admin.port* on all *web_int* interfaces

Caution

To function correctly, the *rtpengine* requires an additional *iptables* rule installed. This rule (with a target of `RTPENGINE`) is automatically installed and removed when the rtpengine starts and stops, so normally you don't need to worry about it. However, any 3rd party firewall solution can potentially flush out all existing iptables rules before installing its own, which would leave the system without the required `RTPENGINE` rule and this would lead to decreased performance. It is imperative that any 3rd party firewall solution either leaves this rule untouched, or installs it back into place after flushing all rules out. The complete parameters to install this rule (which needs to go into the `INPUT` chain of the `filter` table) are: `-p udp -j RTPENGINE --id 0`

Note

Some of the parameters used to populate the firewall rules automatically may contain hostnames instead of IP addresses. Since firewall rules need to be configured based on IP addresses by design, the NGCP configuration framework will lookup such hostnames during *ngcpcfg apply* and expand them to the IP addresses as returnd by *gethostbyname*. If DNS resolving changes for such hostnames due to changes to DNS the rules will not update automatically. Another run of *ngcpcfg apply* will be needed to reperform the lookup and update the rules to reflect chages in DNS. If this step is omitted, clients may be locked out of the system.

Note

By default, the rules for the *rtp_ext* zone are created with a target of `ACCEPT`. It is optionally possible to create these rules with another *iptables* chain as target, and instruct the RTP proxy to dynamically manage individual rules for each running call in this chain. If this is enabled, the chain with the name given in the `/etc/ngcp-config/config.yml` key `rtpproxy→firewall_iptables_chain` will be created as empty, leaving the effective target for UDP packets within the RTP port range as the table's default policy (normally `DROP`). The RTP proxy will then dynamically created one `ACCEPT` rule for each open RTP media port in the given chain when a call starts, and delete it when the call is finished. It should be noted that dynamically creating and deleting iptables rules can incur a singificant performance overhead, especially in scenarios with high call volumes, and it is therefore not recommended to enable this feature in such cases.

16.2.4 Custom rules

The NGCP configuration framework allows to add custom rules to the firewall setup in */etc/ngcp-config/config.yml*. The custom rules are added after the system rules. Hence, they apply for packets not matched by the systems rules only.

Example custom rule to whitelist all IPv4 traffic from network interface eth1.301 effectively making VLAN 301 a trusted network:

```
rules4:
  - '-A INPUT -i eth1.301 -j ACCEPT'
```

Example custom rule to accept incoming traffic from monitoring station 203.0.113.93 for an optionally installed check_mk agent:

```
rules4:
  - '-A INPUT -p tcp -s 203.0.113.93 --dport 6556 -j ACCEPT'
```

To add hosts or networks to the SSH whitelist they can be either added to key *sshd.permit_support_from* in */etc/ngcp-config/config.yml* or a custom rule may be used:

```
rules4:
  - '-A INPUT -s 198.51.100.0/24 --dport 22 - j ACCEPT'
  - '-A INPUT -s 203.0.113.93 --dport 22 -j ACCEPT'
```

Note

In custom rules keys from */etc/ngcp-config/config.yml* cannot be referenced. Thus, the values need to be manually looked up, hard coded, and kept in sync manually. This is by design of YAML.

16.2.5 Example firewall configuration section

An example for NGCP firewall configuration in */etc/ngcp-config/config.yml* enabling both the firewall subsystem and the logging facility may look like:

```
security:
  firewall:
    enable: 'yes'
```

```
    logging:
      enable: 'yes'
      file: '/var/log/firewall.log'
      tag: 'NGCPFW'
    policies:
      input: 'DROP'
      forward: 'DROP'
      output: 'ACCEPT'
    rules4:
      - '-A INPUT -i eth0 -j ACCEPT'
```

16.3 Password management

The sip:provider CE comes with some default passwords the user should change during the deployment of the system. They have been explained in the previous chapters of this handbook.

> **Important**
> Many NGCP services use MySQL backend. Users and passwords for these services are created during the installation. These passwords are unique for each installation, and the connections are restricted to localhost. You should not change these users and passwords.

16.3.1 The "root" account

The sip:provider CE's super-user account comes with a preconfigured password. It is imperative that this password is changed by the operator immediately after the sip:provider CE is shipped and before it is connected to any potentially unsecure public or private network using a secure password in compliance with existing password policies of the operator. The "root" password must not be shared outside of the operator's organization including Sipwise engineers. The "root" password must not be shared in any publicly accessible communications including e-mail or ticketing systems.

To change the root password log into the freshly deployed system as "root" using the preconfigured password and execute:

```
root@myserver:~# passwd
```

Then follow the prompts to change the password.

The Vagrant/VirtualBox/VMWare sip:provider CE images come with more default credentials which should be changed immediately:

- The default password of the system account *root* is *sipwise*. A password must be changed immediately using command *passwd root*.
- SSH authorized_keys for users *root* and *sipwise* should be wiped out using command *rm ~root/.ssh/sipwise_vagrant_key ~sipwise/.ssh/sipwise_vagrant_key* for VirtualBox/VMWare images (skip the step if you use Vagrant).

16.3.2 The "administrator" account

The sip:provider CE Web-interface comes with a preconfigured "administrator" account deployed with a default password. This account can be considered the NGCP application super-user and has far-reaching access to application specific settings via the Web-interface. It is imperative that the password for this account is changed by the operator immediately after the sip:provider CE is shipped and before it is connected to any potentially unsecure public or private network using a secure password in compliance with existing password policies of the operator. The "administrator" password must not be shared outside of the operator's organization including Sipwise engineers. The "administrator" password must not be shared in any publicly accessible communications including e-mail or ticketing systems.

The password for the "administrator" account can be changed via the Web-interface.

16.3.3 The "cdrexport" account

The login for the system account *cdrexport* is disabled by default. Although this is a jailed account, it has access to sensitive information, namely the Call Detail Records of all calls. SSH keys should be used to login this user, or alternatively a really strong password should be used when setting the password via *passwd cdrexport*.

16.3.4 The MySQL "root" user

The *root* user in MySQL has no default password. A password should be set using the *mysqladmin password* command.

16.3.5 The "ngcpsoap" account

Generate new password for user *ngcpsoap* to access the provisioning interfaces, see the details in Section 10.

16.4 SSL certificates.

The sip:provider CE provides default, self-signed SSL certificates for SSL connections. These certificates are common for every installation. Before going to production state, the system administrator should provide SSL certificates for the web services. These certificates can either be shared by all web interfaces (*provisioning*, *administrator interface* and *customer self care interface*), or separate ones for each them can be used.

- Generate the certificates. The *customer self care interface* certificate should be signed by a certification authority to avoid browser warnings.
- Upload the certificates to the system
- Set the path to the new certificates in */etc/ngcp-config/config.yml*:
  - *ossbss→apache→autoprov→sslcertfile* and *ossbss→apache→autoprov→sslcertkeyfile* for the *provisioning interface*.
  - *ossbss→apache→restapi→sslcertfile* and *ossbss→apache→restapi→sslcertkeyfile* for the *REST interface*.
  - *www_admin→http_admin→sslcertfile* and *www_admin→http_admin→sslcertkeyfile* for the *admin interface*.

- *www_admin→http_csc→sslcertfile* and *www_admin→http_csc→sslcertkeyfile* for the *customer self care interface*.

- Apply the configuration changes with *ngcpcfg apply 'added web ssl certs'*.

The sip:provider CE also provides the self-signed SSL certificates for SIP over TLS services. The system administrator should replace them with certificates signed by a trusted certificate authority if he is going to enable it for the production usage (*kamailio→lb→tls→enable* (disabled by default)).

- Generate the certificates.
- Upload the certificates to the system
- Set the path to the new certificates in */etc/ngcp-config/config.yml*:
  - *kamailio→lb→tls→sslcertfile* and *kamailio→lb→tls→sslcertkeyfile* .
- Apply the configuration changes with *ngcpcfg apply 'added kamailio certs'*.

16.5 Securing your sip:provider CE against SIP attacks

The sip:provider CE allows you to protect your VoIP system against SIP attacks, in particular **Denial of Service** and **brute-force attacks**. Let's go through each of those attacks and let's see how to configure your system in order to face such situations and react against them.

16.5.1 Denial of Service

As soon as you have packets arriving on your sip:provider CE server, it will require a bit of time of your CPU. Denial of Service attacks are aimed to break down your system by sending floods of SIP messages in a very short period of time and keep your system busy to handle such huge amount of requests. sip:provider CE allows you to block such kind of attacks quite easily, by configuring the following section in your `/etc/ngcp-config/config.yml` :

```
security:
  dos_ban_enable: 'yes'
  dos_ban_time: 3600
  dos_reqs_density_per_unit: 50
  dos_sampling_time_unit: 2
  dos_whitelisted_ips: []
  dos_whitelisted_subnets: []
```

Basically, as soon as sip:provider CE receives more than 50 messages from the same IP in a time window of 2 seconds, that IP will be blocked for 3600 sec, and you will see in the the `kamailio-lb.log` a line saying:

```
Nov 9 00:11:53 sp1 lb[41958]: WARNING: <script>: IP '1.2.3.4' is blocked and banned - R=< ↩
    null> ID=304153-3624477113-19168@tedadg.testlab.local
```

The banned IP will be stored in kamailio memory, you can check the list via web interface or via the following command:

```
# ngcp-kamctl lb fifo sht_dump ipban
```

Excluding SIP endpoints from banning

There may be some SIP endpoints that send a huge traffic towards NGCP from a specific IP address. A typical example is a *SIP Peering Server*.

Caution

sip:provider CE supports handling such situations by excluding all defined *SIP Peering Servers* from DoS protection mechanism.

The NGCP platform administrator may also add whitelisted IP addresses manually in `/etc/ngcp-config/config.yml` at `kamailio.lb.security.dos_whitelisted_ips` and `kamailio.lb.security.dos_whitelisted_subnets` parameters.

16.5.2 Bruteforcing SIP credentials

This is a very common attack you can easily detect checking your */var/log/ngcp/kamailio-proxy.log*. You will see INVITE/REGISTER messages coming in with strange usernames. Attackers is trying to spoof/guess subscriber's credentials, which allow them to call out. The very first protection against these attacks is: **ALWAYS USE STRONG PASSWORD**. Nevertheless sip:provider CE allow you to detect and block such attacks quite easily, by configuring the following */etc/ngcp-config/config.yml* section:

```
failed_auth_attempts: 3
failed_auth_ban_enable: 'yes'
failed_auth_ban_time: 3600
```

You may increase the number of failed attempt if you want (in same cases it's better to be safed, some users can be banned accidentally because they are not writing the right password) and adjust the ban time. If a user try to authenticate an INVITE (or REGISTER) for example and it fails more then 3 times, the "user@domain" (not the IP as for Denial of Service attack) will be block for 3600 seconds. In this case you will see in your */var/log/ngcp/kamailio-lb.log* the following lines:

```
Nov 9 13:31:56 sp1 lb[41952]: WARNING: <script>: Consecutive Authentication Failure for ' ↩
    sipvicous@mydomain.com' UA='sipvicous-client' IP='1.2.3.4' - R=<null> ID ↩
    =313793-3624525116-589163@testlab.local
```

Both the banned IPs and banned users are shown in the Admin web interface, you can check them by accessing the **Security Bans** section in the main menu. You can check the banned user as well by retrieving the same info directly from kamailio memory, using the following commands:

```
# ngcp-kamctl lb fifo sht_dump auth
```

16.6 Topology Hiding

16.6.1 Introduction to Topology Hiding on NGCP

The term "topology hiding" in SIP is used to describe the measures taken by typically an SBC (Session Border Controller) to hide detailed information of the internal network at the border of which it is located. Pieces of information such as IP addresses and port numbers used by SIP endpoints and intermediaries within the network are considered sensitive, as these can give some hints to potential attackers about the topology of the network.

In a typical SIP session the mandatory headers may carry that sensitive information, for example: *Contact, Via, Record-Route, To, From, Call-ID*. An SBC applying topology hiding will mangle the content of those headers.

Concealment of sensitive information is achieved through encoding the original content of selected SIP headers. Then NGCP will create a new SIP URI using a preselected IP address and the encoded content as URI parameter, finally re-assembling the SIP header.

Examples for encoded SIP headers:

```
Record-Route: <sip:127.0.0.8;line=sr-NvaAlWtecghucEhu6WtAcu...>
Contact: <sip:127.0.0.8;line=sr-NvaAli-1VeL.kRxLcbN86W...>
```

The *load-balancer* element of the Sipwise NGCP has an SBC role, from the SIP peers point of view. The *LB* offers topology hiding function that can be simply activated through a configuration change. By default the function is disabled.

16.6.2 Configuration of Topology Hiding

Activating topology hiding function is possible through the modification of the following configuration parameters in `/etc/ngcp-config/config.yml` file (shown below with default values of parameters):

```
kamailio:
  lb:
    security:
      topoh:
        enable: no
        mask_callid: no
        mask_ip: 127.0.0.8
```

Meaning of the configuration parameters:

- `enable`: if set to `yes`, the topology hiding will be activated
- `mask_callid`: if set to `yes`, the SIP Call-ID header will also be encoded
- `mask_ip`: an IP address that will be used to create valid SIP URIs, after encoding the real/original header content.

> **Tip**
> Any valid, preferably private network address can be used. The suggestion is however to use an address that is not used by any other SIP endpoint or intermediary element in the network.

16.6.3 Considerations for Topology Hiding

Although hiding sensitive information about a SIP provider's network is desired, there are some potential side effects caused by topology hiding.

The most common example is the consequence that **SIP message size may grow** when applying topology hiding. The fact that SIP messages become larger may even prevent NGCP from communicating successfully with another SIP entity (a peer SBC, for example). This can be expected under following circumstances:

- SIP transport protocol is UDP
- SIP messages have more *Via and Record-Route* headers
- IP packets of SIP messages without the topology hiding feature already have a size close to the MTU

In such a case the IP packets carrying SIP messages with encoded headers will have a size exceeding the MTU, that will cause loss of data.

The recommended solution in such a case is to use TCP transport for SIP messages.

16.7 System Requirements and Performance

The sip:provider CE is a very flexible system, capable of serving from hundreds to several tens of thousands of subscribers in a single node. The system comes with a default configuration, capable of serving up to 50.000 subscribers in a *normal* environment. But there is no such thing as a *normal* environment. And the sip:provider CE has sometimes to be tunned for special environments, special hardware requirements or just growing traffic.

> **Note**
> If you have performance issues with regards to disk I/O please consider enabling the *noatime* mount option for the root filesystem. Sipwise recommends the usage of *noatime*, though remove it if you use software which conflicts with its presence.

In this section some parameters will be explained to allow the sip:provider CE administrator tune the system requirements for optimum performance.

Table 17: Requirement_options

Option	Default value	Requirement impact
cleanuptools→binlog_days	15	Heavy impact on the harddisk storage needed for mysql logs. It can help to restore the database from backups or restore broken replication.
database→bufferpoolsize	64MB	For test systems or low RAM systems, lowering this setting is one of the most effective ways of releasing RAM. The administrator can check the innodb buffer hit rate on production systems; a hit rate over 99% is desired to avoid bottlenecks.

Table 17: (continued)

Option	Default value	Requirement impact
kamailio→lb→pkg_mem	16	This setting affects the amount of RAM the system will use. Each kamailio-lb worker will have this amount of RAM reserved. Lowering this setting up to 8 will help to release some memory depending on the number of kamailio-lb workers running. This can be a dangerous setting as the lb process could run out of memory. Use with caution.
kamailio→lb→shm_mem	1/16 * Total System RAM	The installer will set this value to 1/16 of the total system RAM. This setting does not change even if the system RAM does so it's up to the administrator to tune it. It has been calculated that 1024 (1GB) is a good value for 50K subscriber environment. For a test environment, setting the value to 64 should be enough. "Out of memory" messages in the kamailio log can indicate that this value needs to be raised.
kamailio→lb→tcp_children	8	Number of TCP workers kamailio-lb will spawn per listening socket. The value should be fine for a mixed UDP-TCP 50K subscriber system. Lowering this setting can free some RAM as the number of kamailio processes would decrease. For a test system or a pure UDP subscriber system 2 is a good value. 1 or 2 TCP workers are always needed.
kamailio→lb→tls→enable	yes	Enable or not TLS signaling on the system. Setting this value to "no" will prevent kamailio to spawn TLS listening workers and free some RAM.
kamailio→lb→udp_children	8	See *kamailio→lb→tcp_children* explanation
kamailio→proxy→children	8	See *kamailio→lb→tcp_children* explanation. In this case the proxy only listens udp so these children should be enough to handle all the traffic. It could be set to 2 for test systems to lower the requirements.
kamailio→proxy→*_expires		Set the default and the max and min registration interval. The lower it is more REGISTER requests will be handled by the lb and the proxy. It can impact in the network traffic, RAM and CPU usage.
kamailio→proxy→natping_interval	30	Interval for the proxy to send a NAT keepalive OPTIONS message to the nated subscriber. If decreased, this setting will increase the number of OPTIONS requests the proxy needs to send and can impact in the network traffic and the number of natping processes the system needs to run. See *kamailio→proxy→natping_processes* explanation.
kamailio→proxy→natping_processes	7	Kamailio-proxy will spawn this number of processes to send keepalive OPTIONS to the nated subscribers. Each worker can handle about 250 messages/second (depends on the hardware). Depending the number of nated subscribers and the *kamailio→proxy→natping_interval* parameter the number of workers may need to be adjusted. The number can be calculated like nated_subscribers/natping_interval/pings_per_second_per_process. For the default options, assuming 50K nated subscribers in the system the parameter value would be 50.000/30/250 = (6,66) 7 workers. 7 is the maximum number of processes kamailio will accept. Raising this value will cause kamailio not to start.

Table 17: (continued)

Option	Default value	Requirement impact
kamailio→proxy→shm_mem	1/16 * Total System RAM	See *kamailio→lb→shm_mem* explanation.
rateomat→enable	yes	Set this to no if the system shouldn't perform rating on the CDRs. This will save CPU usage.
rsyslog→external_log	0	If enabled, the system will send the log messages to an external server. Depending on the *rsyslog→external_loglevel* parameter this can increase dramatically the network traffic.
rsyslog→ngcp_logs_preserve_days	93	This setting will set the number of days ngcp logs under */var/log/ngcp* will be kept in disk. Lowering this setting will free a high amount of disk space.

Tip

In case of using virtualized environment with limited amount of hardware resources, you can use the script *ngcp-toggle-performance-config* to adjust sip:provider CE configuration for high/low performance:

```
root@spce:~# /usr/sbin/ngcp-toggle-performance-config
/usr/sbin/ngcp-toggle-performance-config - tool to adjust sip:provider configuration for ↩
    low/high performance

  --help                Display this usage information
  --high-performance    Adjust configuration for system with normal/high performance
  --low-performance     Adjust configuration for system with low performance (e.g. VMs)

root@spce:~#
```

16.8 Troubleshooting

The sip:provider CE platform provides detailed logging and log files for each component included in the system via rsyslog. The main folder for log files is */var/log/ngcp/*, it contains a list of self explanatory log files named by component name.

The sip:provider CE is a high performance system which requires compromise between traceability (maximum amount of debug information being written to hard drive) and productivity (minimum load on IO subsystem). This is the reason why different log levels are configured for the provided components by default.

Most log files are designed for debugging sip:provider CE by Sipwise operational team while main log files for daily routine usage are:

Log file	Content	Estimated size
/var/log/ngcp/api.log	API logs providing type and content of API requests and responses as well as potential errors	medium
/var/log/ngcp/panel.log /var/log/ngcp/panel-debug.log	Admin Web UI logs when performing operational tasks on the ngcp-panel	medium
/var/log/ngcp/cdr.log	mediation and rating logs, e.g. how many CDRs have been generated and potential errors in case of CDR generation or rating fails for particular accounting data	medium
/var/log/ngcp/kamailio-proxy.log	Overview of SIP requests and replies between lb, proxy and sems processes. It's the main log file for SIP overview	huge

Log file	Content	Estimated size
/var/log/ngcp/kamailio-lb.log	Overview of SIP requests and replies along with network source and destination information flowing through the platform	huge
/var/log/ngcp/sems.log	Overview of SIP requests and replies between lb, proxy and sems processes	small
/var/log/ngcp/rtp.log	rtpengine related log, showing information about RTP communication	small

Warning

it is highly NOT recommended to change default log levels as it can cause system IO overloading which will affect call processing.

Note

the exact size of log files depend on system type, system load, system health status and system configuration, so cannot be estimated with high precision. Additionally operational network parameters like ASR and ALOC may impact the log files' size significantly.

16.8.1 Collecting call information from logs

The easiest way to fetch information about a single call among the log files is the search for the SIP CallID (a unique identifier for a SIP dialog). The call ID is used as call marker in almost all the voip related log file, such as */var/log/ngcp/kamailio-lb.log* , */var/log/ngcp/kamailio-proxy.log* , */var/log/ngcp/sems.log* or */var/log/ngcp/rtp.log*. Example of kamailio-proxy.log line:

```
Nov 19 00:35:56 sp1 proxy[7475]: NOTICE: <script>: New request on proxy - M=REGISTER R=sip: ↩
    sipwise.local
F=sip:jdoe@sipwise.local T=sip:jdoe@sipwise.local IP=10.10.1.10:5060 (127.0.0.1:5060) ID ↩
    =364e4676776621034977934e055d19ea@127.0.0.1 UA='SIP-UA 1.2.3.4'
```

The above line shows the SIP information you can find in a general line contained in */var/log/ngcp/kamailio-**:

- M=REGISTER : The SIP Method
- R=sip:sipwise.local : The SIP Request URI
- F=sip:jdoe@sipwise.local : The SIP From header
- T=sip:jdoe@sipwise.local : The SIP To header
- IP=10.10.1.10:5060 (127.0.0.1:5060) : The source IP where the message is coming from. Between brackets it is shown the local internal IP where the message come from (in this case Load Balancer)
- ID=364e4676776621034977934e055d19ea@127.0.0.1 : The SIP CallID.
- UAIP=10.10.1.10 : The User Agent source IP
- UA=*SIP-UA 1.2.3.4* : The SIP User Agent header

In order to collect the full log related to a single call, it's necessary to "grep" the */var/log/ngcp/kamailio-proxy.log* using the **ID=** string, for example:

```
# grep "364e4676776621034977934e055d19ea@127.0.0.1" /var/log/ngcp/kamailio-proxy.log
```

16.8.2 Collecting SIP traces

The sip:provider CE platform provides several tools to collect SIP traces. It can be used the sip:provider CE *ngrep-sip* tool to collect SIP traces, for example to fetch traffic in text format from outbound and among load balancer, proxy and sems :

```
# ngrep-sip b
```

see the manual to know all the options:

```
# man ngrep-sip
```

The *ngrep* debian tool can be used in order to make a SIP trace and save it into a *.pcap* file :

```
# ngrep -s0 -Wbyline -d any -O /tmp/SIP_trace_file_name.pcap port 5062 or port 5060
```

The *sngrep* debian graphic tool as well can be used to visualize SIP trace and save them in a *.pcap* file :

```
# sngrep
```

17 Monitoring and Alerting

17.1 Internal Monitoring

17.1.1 System monitoring via Telegraf

The platform uses the internal *telegraf* service to monitor many aspects of the system, including CPU, memory, swap, disk, filesystem, network, processes, NTP, Nginx, Redis and MySQL.

The gathered information is stored in *InfluxDB*, in the *telegraf* database.

17.1.2 NGCP-specific monitoring via ngcp-witnessd

The platform uses the internal *ngcp-witnessd* service to monitor NGCP-specific metrics or system metrics currently not tracked by *telegraf*, including memory, process count, Heartbeat, MTA, Kamailio, SIP and MySQL.

The gathered information is stored in *InfluxDB*, in the *ngcp* database.

17.1.3 Monitoring data in InfluxDB

The platform uses *InfluxDB* as a time series database, to store most of the metrics collected in the system.

The monitoring data is used by various components of the platform, including *ngcp-collective-check*, *ngcp-snmp-agent* and by the statistics dashboard powered by *Grafana*.

The monitoring data can also be accessed directly by various means; by using the *influx* command-line tool in CLI or TUI modes; by using the *ngcp-influxdb-extract* wrapper which provides two convenience commands to run arbitrary queries or to fetch the last value for a measurement's field; or by using the HTTP API with *curl* (or other HTTP fetchers), or with the *Sipwise::InfluxDB::HTTP* perl module.

See https://docs.influxdata.com/influxdb/v1.1/query_language/spec/ for information about InfluxQL, the query language used by *InfluxDB*.

Tip

To get the list of all measurements for a specific database the following query can be used `SHOW MEASUREMENTS`.

Tip

To get the list of fields for a specific measurement the following query can be used `SELECT LAST(*) FROM "measurement"`.

Tip

To get the list of tags for a specific measurement the following query can be used `SHOW TAG KEYS FROM "measurement"`, and for all the current tag values for a tag `SHOW TAG VALUES FROM "measurement" WITH KEY = "tag"`.

See Section C.2.1 for detailed information about the list of data currently stored in the *InfluxDB ngcp* monitoring database.

17.2 Statistics Dashboard

The platform's administration interface (described in Section 6) provides a graphical overview based on *Grafana* of the most important system health indicators, such as memory usage, load averages and disk usage. VoIP statistics, such as the number of concurrent active calls, the number of provisioned and registered subscribers, etc. is also present.

A Basic Call Flows

A.1 General Call Setup

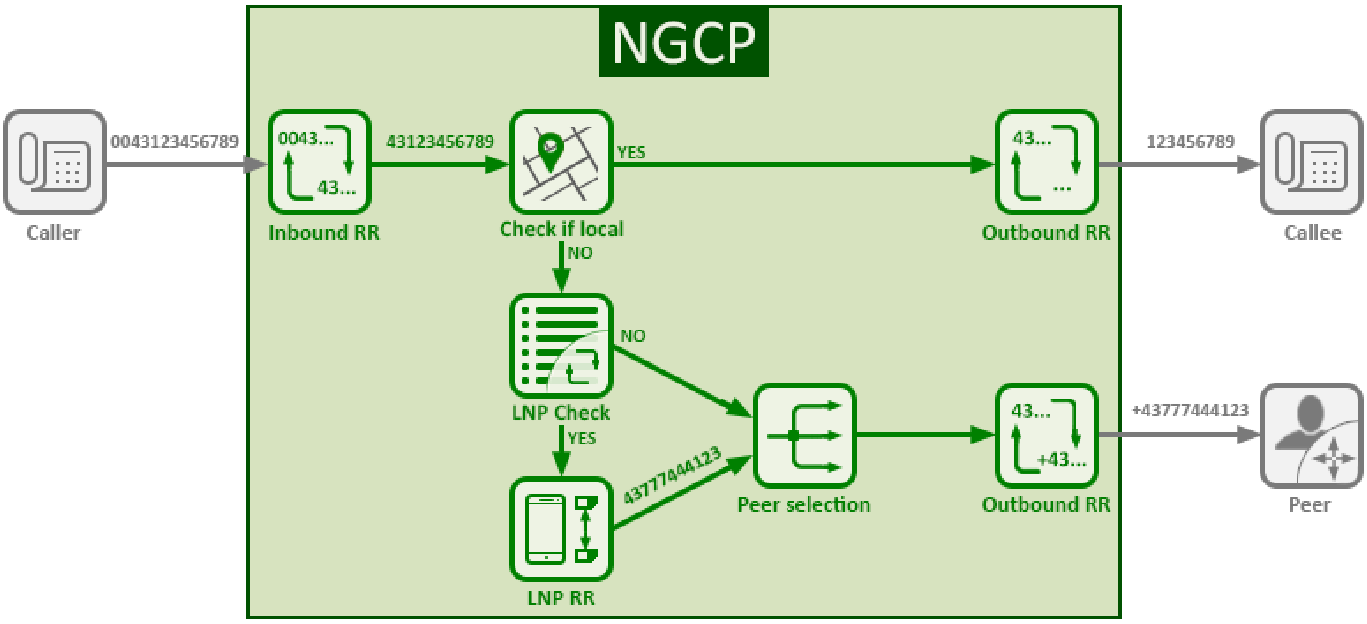


Figure 67: General Call Setup

NGCP performs the following checks when processing a call coming from a subscriber and terminated at a peer:

- Checks if the IP address where the request came from is in the list of trusted IP addresses. If yes, this IP address is taken as the identity for authentication. Otherwise, NGCP performs the digest authentication.
- When the subscriber is authorized to make the call, NGCP applies the Inbound Rewrite Rules for the caller and the callee assigned to the subscriber (if any). If there are no Rewrite Rules assigned to the subscriber, the ones assigned to the subscriber's domain are applied. On this stage the platform normalises the numbers from the subscriber's format to E.164.
- Matches the callee (called number) with local subscribers.
  - If it finds a matching subscriber, the call is routed internally. In this case, NGCP applies the Outbound Rewrite Rules associated with the callee (if any). If there are no Rewrite Rules assigned to the callee, the ones assigned to the callee's domain are applied.
  - If it does not find a matching subscriber, the call goes to a peer as described below.
- Queries the LNP database to find out if the number was ported or not.For details of LNP queries refer to the Local Number Porting Section 7.4 chapter.
  - If it was ported, NGCP applies the LNP Rewrite Rules to the called number.
- Based on the priorities of peering groups and peering rules (see Section 6.6.2.3 for details), NGCP selects peering groups for call termination and defines their precedence.

- Within every peering group the weight of a peering server defines its probability to receive the call for termination. Thus, the bigger the weight of a server, the higher the probability that NGCP will send the call to it.
- Applies the Outbound Rewrite Rules for the caller and the callee assigned to a peering server when sending the call to it.

A.2 Endpoint Registration

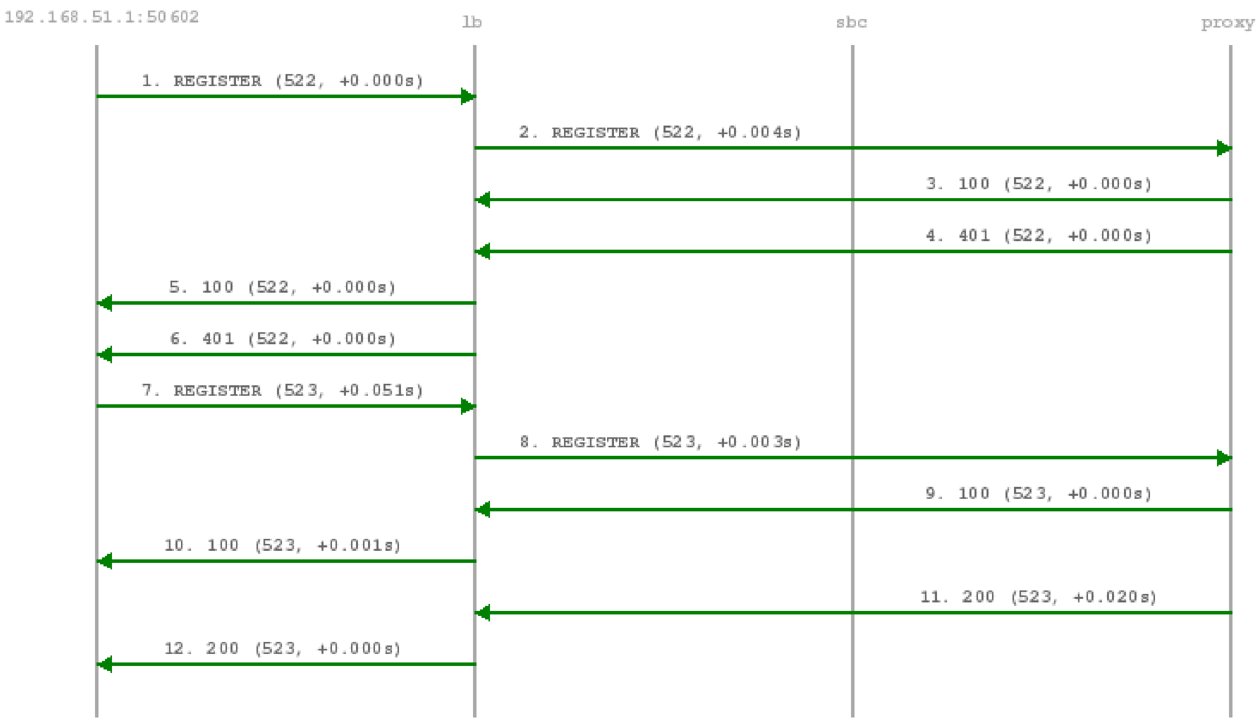


Figure 68: Registration Call-Flow

The subscriber endpoint starts sending a REGISTER request, which gets challenged by a 401. After calculating the response of the authentication challenge, it sends the REGISTER again, including the authentication response. The SIP proxy looks up the credentials of the subscriber in the database, does the same calculation, and if the result matches the one from the subscriber, the registration is granted.

The SIP proxy writes the content of the `Contact` header (e.g. `sip:me@1.2.3.4:1234;transport=UDP`) into its location table (in case of NAT the content is changed by the SIP load-balancer to the IP/port from where the request was received), so it knows where the reach a subscriber in case on an inbound call to this subscriber (e.g. `sip:someuser@example.org` is mapped to `sip:me@1.2.3.4:1234;transport=UDP` and sent out to this address).

If NAT is detected, the SIP proxy sends a OPTION message to the registered contact every 30 seconds, in order to keep the NAT binding on the NAT device open. Otherwise, for subsequent calls to this contact, the sip:provider PRO wouldn't be able to reach the endpoint behind NAT (NAT devices usually drop a UDP binding after not receiving any traffic for ~30-60 seconds).

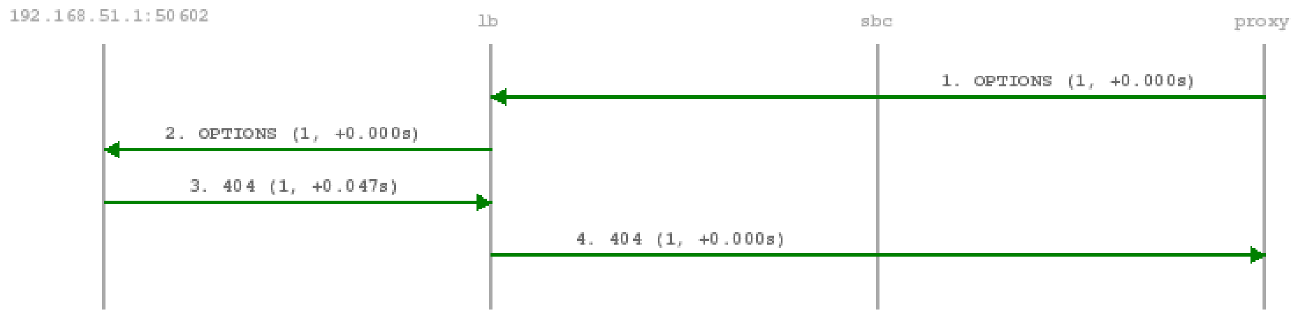


Figure 69: NAT-Ping Call-Flow

By default, a subscriber can register 5 contacts for an Address of Record (AoR, e.g. `sip:someuser@example.org`).

A.3 Basic Call

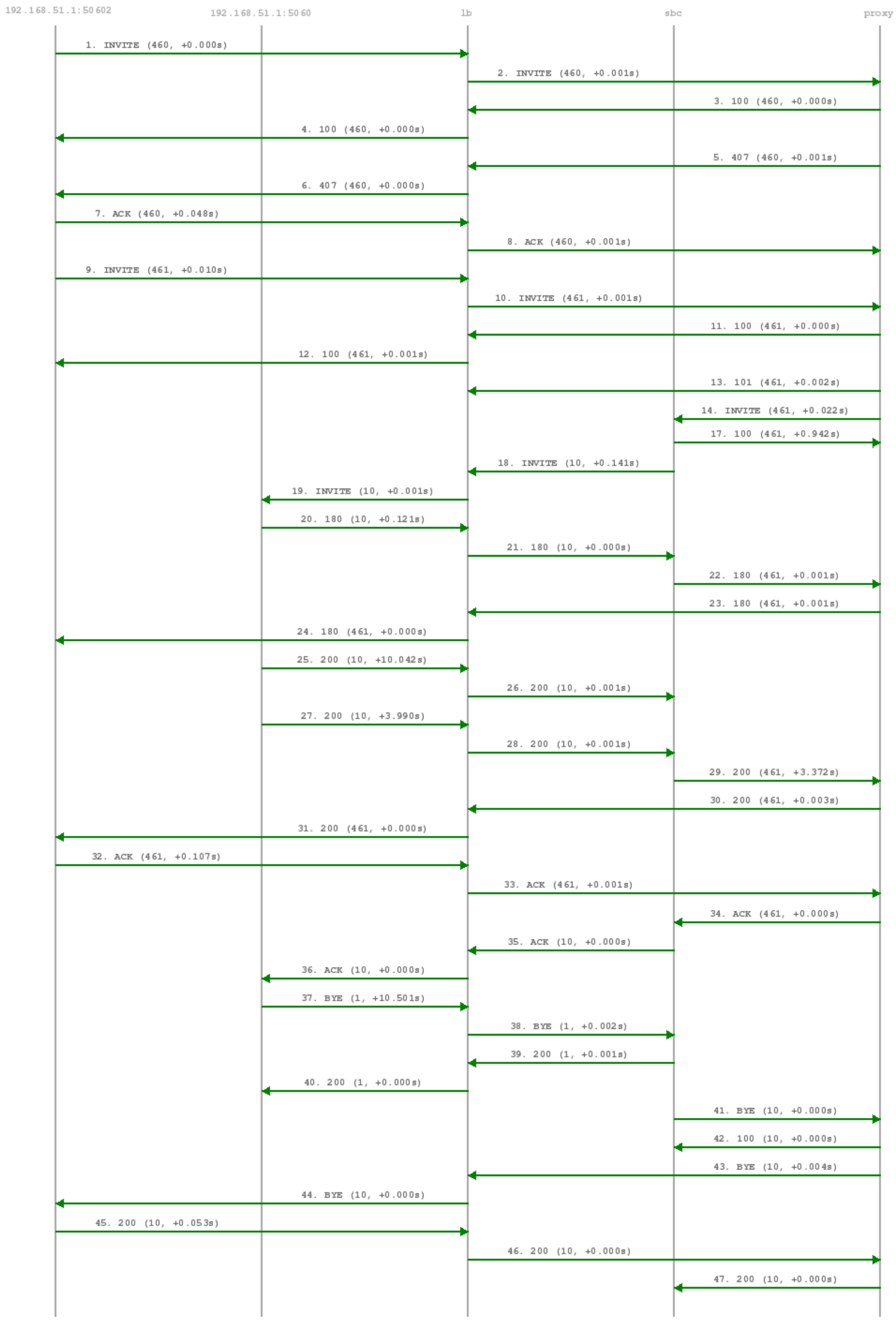


Figure 70: Basic Call Call-Flow

The calling party sends an INVITE (e.g. `sip:someuser@example.org`) via the SIP load-balancer to the SIP proxy. The proxy replies with an authorization challenge in the 407 response, and the calling party sends the INVITE again with authentication credentials. The SIP proxy checks if the called party is a local user. If it is, and if there is a registered contact found for this user, then (after various feature-related tasks for both the caller and the callee) the Request-URI is replaced by the URI of the registered contact (e.g. `sip:me@1.2.3.4:1234;transport=UDP`). If it's not a local user but a numeric user, a proper PSTN gateway is being selected by the SIP proxy, and the Request-URI is rewritten accordingly (e.g. `sip:+431234567890@2.3.4.5:5060`).

Once the proxy has finished working through the call features of both parties involved and has selected the final destination for the call, and - optionally - has invoked the Media Relay for this call, the INVITE is sent to the SIP B2BUA. The B2BUA creates a new INVITE message from scratch (using a new Call-ID and a new From-Tag), copies only various and explicitly allowed SIP headers from the old message to the new one, filters out unwanted media capabilities from the SDP body (e.g. to force audio calls to use G.711 as a codec) and then sends the new message via the SIP load-balancer to the called party.

SIP replies from the called party are passed through the elements back to the calling party (replacing various fields on the B2BUA to match the first call leg again). If a reply with an SDP body is received by the SIP proxy (e.g. a 183 or a 200), the Media Relay is invoked again to prepare the ports for the media stream.

Once the 200 is routed from the called party to the calling party, the media stream is fully negotiated, and the endpoints can start sending traffic to each outer (either end-to-end or via the Media Relay). Upon reception of the 200, the SIP proxy writes a start record for the accounting process. The 200 is also acknowledged with an ACK message from the calling party to the called party, according to the SIP 3-way handshake.

Either of the parties can tear down the media session at any time by sending a BYE, which is passed through to the other party. Once the BYE reaches the SIP proxy, it instructs the Media Relay to close the media ports, and it writes a stop record for accounting purposes. Both the start- and the stop-records are picked up by the *mediator* service in a regular interval and are converted into a Call Detail Record (CDR), which will be rated by the *rate-o-mat* process and can be billed to the calling party.

A.4 Session Keep-Alive

The SIP B2BUA acts as refresher for the Session-Timer mechanism as defined in RFC 4028. If the endpoints indicate support for the UPDATE method during call-setup, then the SIP B2BUA will use an UPDATE message if enabled per peer, domain or subscriber via Provisioning to check if the endpoints are still alive and responsive. Both endpoints can renegotiate the timer within a configurable range. All values can be tuned using the Admin Panel or the APIs using Peer-, Domain- and Subscriber-Preferences.

Tip

Keep in mind that the values being used in the signaling are always half the value being configured. So if you want to send a keep-alive every 300 seconds, you need to provision *sst_expires* to 600.

If one of the endpoints doesn't respond to the keep-alive messages or answers with `481 Call/Transaction Does Not Exist`, then the call is torn down on both sides. This mechanism prevents excessive over-billing of calls if one of the endpoints is not reachable anymore or "forgets" about the call. The BYE message sent by the B2BUA triggers a stop-record for accounting and also closes the media ports on the Media Relay to stop the call.

Beside the Session-Timer mechanism to prevent calls from being lost or kept open, there is a **maximum call length** of 21600 seconds per default defined in the B2BUA. This is a security/anti-fraud mechanism to prevent overly long calls causing excessive costs.

A.5 Voicebox Calls

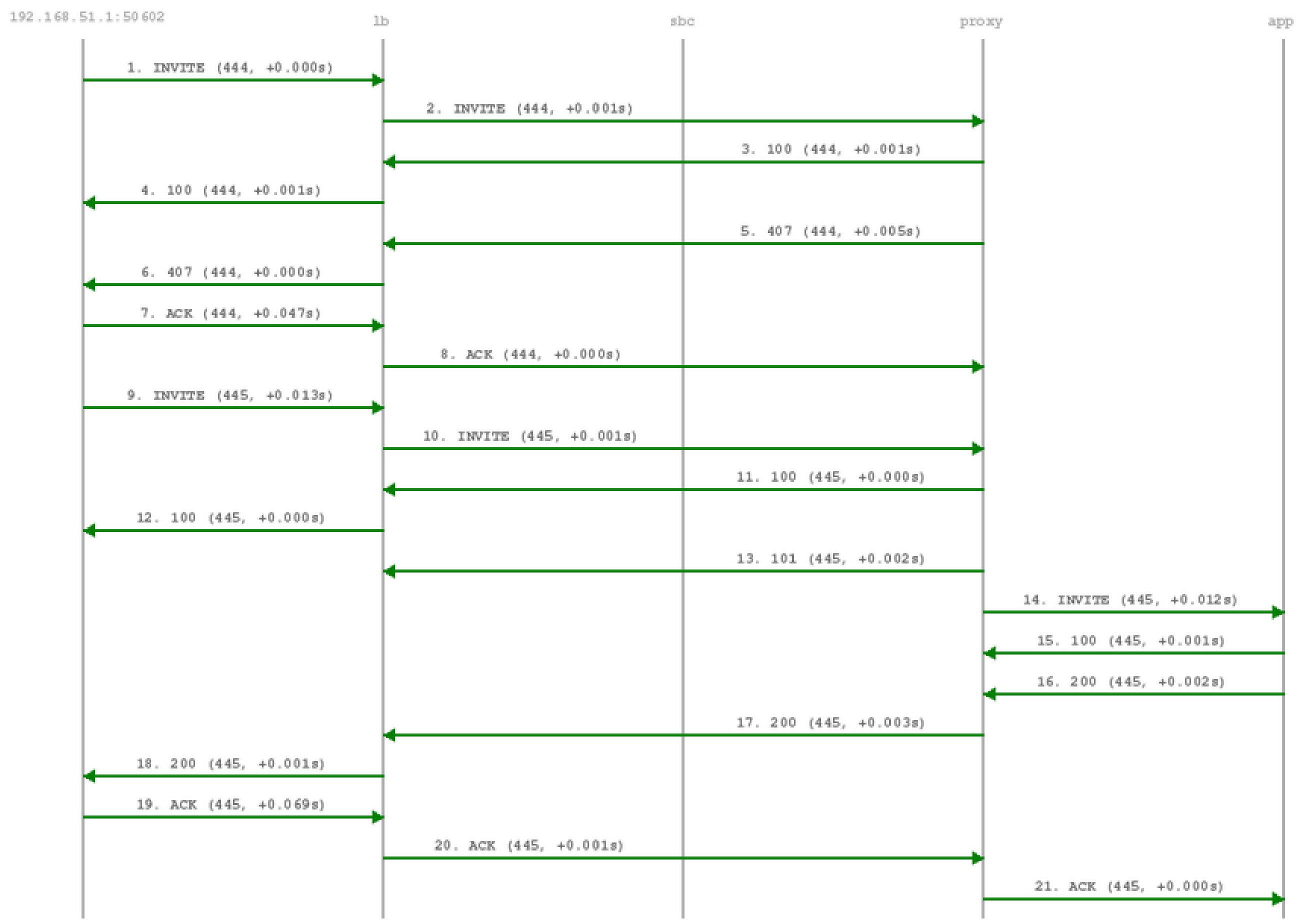


Figure 71: Voicebox Call-Flow

Calls to the Voicebox (both for callers leaving a voicemail message and for voicebox owners managing it via the IVR menu) are passed directly from the SIP proxy to the App-Server without a B2BUA. The App-Server maintains its own timers, so there is no risk of over-billing or overly long calls.

In such a case where an endpoint talks via the Media Relay to a system-internal endpoint, the Media Relay bridges the media streams between the public in the system-internal network.

In case of an endpoint leaving a new message on the voicebox, the Message-Waiting-Indication (MWI) mechanism triggers the sending of a unsolicited NOTIFY message, passing the number of new messages in the body. As soon as the voicebox owner dials into his voicebox (e.g. by calling `sip:voicebox@example.org` from his SIP account), another NOTIFY message is sent to his devices, resetting the number of new messages.

Important

The sip:provider CE does not require your device to subscribe to the MWI service by sending a SUBSCRIBE (it would rather reject it). On the other hand, the endpoints need to accept unsolicited NOTIFY messages (that is, a NOTIFY without a valid subscription), otherwise the MWI service will not work with these endpoints.

B NGCP configs overview

B.1 config.yml Overview

`/etc/ngcp-config/config.yml` is the main configuration YAML file used by Sipwise NGCP. After every changes it need to run the command `ngcpcfg apply` *`my commit message`* to apply changes (followed by `ngcpcfg push` in the PRO version to apply changes to sp2). The following is a brief description of the main variables contained into `/etc/ngcp-config/config.yml` file.

B.1.1 apps

This section contains parameters for the additional applications that may be activated on sip:provider CE.

```
apps:
  malicious_call: no
```

- malicious_call: If set to *yes*, the Malicious Call Identification (MCID) application will be enabled.

B.1.2 asterisk

The following is the asterisk section:

```
asterisk:
  log:
    facility: local6
  rtp:
    maxport: 20000
    minport: 10000
  sip:
    bindport: 5070
    dtmfmode: rfc2833
  voicemail:
    enable: 'no'
    fromstring: 'Voicemail server'
    greeting:
      busy_custom_greeting: '/home/user/file_no_extension'
      busy_overwrite_default: 'no'
      busy_overwrite_subscriber: 'no'
      unavail_custom_greeting: '/home/user/file_no_extension'
      unavail_overwrite_default: 'no'
      unavail_overwrite_subscriber: 'no'
    mailbody: 'You have received a new message from ${VM_CALLERID} in voicebox ${VM_MAILBOX ↩
        } on ${VM_DATE}.'
    mailsubject: '[Voicebox] New message ${VM_MSGNUM} in voicebox ${VM_MAILBOX}'
    max_msg_length: 180
```

```
    maxgreet: 60
    maxmsg: 30
    maxsilence: 0
    min_msg_length: 3
    normalize_match: '^00|\+([1-9][0-9]+)$'
    normalize_replace: '$1'
    serveremail: voicebox@sip.sipwise.com
```

- log.facility: rsyslog facility for asterisk log, defined in /etc/asterisk/logger.conf.
- rtp.maxport: RTP maximum port used by asterisk.
- rtp.minport: RTP minimun port used by asterisk.
- sip.bindport: SIP asterisk internal bindport.
- voicemail.greetings.*: set the audio file path for voicemail custom unavailable/busy greetings
- voicemail.mailbody: Mail body for incoming voicemail.
- voicemail.mailsubject: Mail subject for incoming voicemail.
- voicemail.max_msg_length: Sets the maximum length of a voicemail message, in seconds.
- voicemail.maxgreet: Sets the maximum length of voicemail greetings, in seconds.
- voicemail.maxmsg: Sets the maximum number of messages that may be kept in any voicemail folder.
- voicemail.min_msg_length: Sets the minimun length of a voicemail message, in seconds.
- voicemail.maxsilence: Maxsilence defines how long Asterisk will wait for a contiguous period of silence before terminating an incoming call to voice mail. The default value is 0, which means the silence detector is disabled and the wait time is infinite.
- voicemail.serveremail: Provides the email address from which voicemail notifications should be sent.
- voicemail.normalize_match: Regular expression to match the From number for calls to voicebox.
- voicemail.normalize_replace: Replacement string to return, in order to match an existing voicebox.

B.1.3 autoprov

The following is the autoprovisioning section:

```
autoprov:
  hardphone:
    skip_vendor_redirect: 'no'
  server:
    bootstrap_port: 1445
    ca_certfile: '/etc/ngcp-config/ssl/client-auth-ca.crt'
    host: localhost
    port: 1444
```

```
    server_certfile: '/etc/ngcp-config/ssl/myserver.crt'
    server_keyfile: '/etc/ngcp-config/ssl/myserver.key'
    ssl_enabled: 'yes'
  softphone:
    config_lockdown: 0
    webauth: 0
```

- autoprov.skip_vendor_redirect: Skip phone vendor redirection to the vendor provisioning web site.

B.1.4 backuptools

The following is the backup tools section:

```
backuptools:
  cdrexport_backup:
    enable: 'no'
  etc_backup:
    enable: 'no'
  mail:
    address: noc@company.org
    error_subject: '[ngcp-backup] Problems detected during daily backup'
    log_subject: '[ngcp-backup] Daily backup report'
    send_errors: 'no'
    send_log: 'no'
  mysql_backup:
    enable: 'no'
    exclude_dbs: 'syslog sipstats information_schema'
  rotate_days: 7
  storage_dir: '/var/backup/ngcp_backup'
  temp_backup_dir: '/tmp/ngcp_backup'
```

- backuptools.cdrexport_backup.enable: Enable backup of cdrexport (.csv) directory.
- backuptools.etc_backup.enable: Enable backup of /etc/* directory.
- backuptools.mail.address: Destination email address for backup emails.
- backuptools.mail.error_subject: Subject for error emails.
- backuptools.mail.log_subjetc: Subject for daily backup report.
- backuptools.mail.send_error: Send daily backup error report.
- backuptools.mail.send_log: Send daily backup log report.
- backuptools.mysql_backup.enable: Enable daily mysql backup.
- backuptools.mysql_backup.exclude_dbs: exclude mysql databases from backup.

- backuptools.rotate_days: Number of days backup files should be kept. All files older than specified number of days are deleted from the storage directory.
- backuptools.storage_dir: Storage directory of backups.
- backuptools.storage_group: Name of the group that backup files should be owned by.
- backuptools.storage_user: Name of the user that backup files should be owned by.
- backuptools.temp_backup_dir: Temporary storage directory of backups.

B.1.5 cdrexport

The following is the cdr export section:

```
cdrexport:
  daily_folder: 'yes'
  export_failed: 'no'
  export_incoming: 'no'
  exportpath: '/home/jail/home/cdrexport'
  full_names: 'yes'
  monthly_folder: 'yes'
```

- cdrexport.daily_folder: Set *yes* if you want to create a daily folder for CDRs under the configured path.
- cdrexport.export_failed: Export CDR for failed calls.
- cdrexport.export_incoming: Export CDR for incoming calls.
- cdrexport.exportpath: The path to store CDRs in .csv format.
- cdrexport.full_names: Use full namen for CDRs instead of short ones.
- cdrexport.monthly_folder: Set *yes* if you want to create a monthly folder (ex. 201301 for January 2013) for CDRs under configured path.

B.1.6 checktools

The following is the check tools section:

```
checktools:
  active_check_enable: '1'
  asr_ner_statistics: '1'
  collcheck:
    cpuidle: '0.1'
    dfused: '0.9'
    eximmaxqueue: '15'
    kamminshmem: '1048576'
    lbminshmem: '1048576'
```

```
    loadlong: '2'
    loadmedium: '2'
    loadshort: '3'
    maxage: 30
    memused: 0.98
    siptimeout: '15'
    sslcert_timetoexpiry: '30'
    sslcert_whitelist: []
    swapfree: 0.02
  exim_check_enable: '1'
  force: '0'
  kamailio_check_concurrent_calls_enable: '1'
  kamailio_check_dialog_active_enable: '1'
  kamailio_check_dialog_early_enable: '1'
  kamailio_check_dialog_incoming_enable: '1'
  kamailio_check_dialog_local_enable: '1'
  kamailio_check_dialog_outgoing_enable: '1'
  kamailio_check_dialog_relay_enable: '1'
  kamailio_check_shmem_enable: '1'
  kamailio_check_usrloc_regdevices_enable: '1'
  kamailio_check_usrloc_regusers_enable: '1'
  monitor_peering_groups: '1'
  mpt_check_enable: '0'
  mysql_check_enable: '1'
  mysql_check_replication: '1'
  mysql_replicate_check_interval: '3600'
  mysql_replicate_check_tables:
  - accounting
  - billing
  - carrier
  - kamailio
  - ngcp
  - provisioning
  - prosody
  - rtcengine
  - stats
  mysql_replicate_ignore_tables:
  - accounting.acc_backup
  - accounting.acc_trash
  - kamailio.acc_backup
  - kamailio.acc_trash
  - ngcp.pt_checksums_sp1
  - ngcp.pt_checksums_sp2
  - ngcp.pt_checksums
  oss_check_provisioned_subscribers_enable: '1'
  sip_check_enable: '1'
  sipstats_check_num_packets: '1'
  sipstats_check_num_packets_perday: '1'
```

```
sipstats_check_partition_size: '1'
snmpd:
  communities:
    public:
    - localhost
  trap_communities:
    public:
    - localhost
```

- checktools.collcheck.cpuidle: Sets the minimum value for CPU usage (0.1 means 10%).
- checktools.collcheck.dfused: Sets the maximun value for DISK usage (0.9 means 90%).
- checktools.collcheck.loadlong/loadlong/loadshort: Max values for load (long, short, medium term).
- checktools.collcheck.maxage: Max age in seconds.
- checktools.collcheck.memused: Sets the maximun value for MEM usage (0.7 means 70%).
- checktools.collcheck.siptimeout: Max timeout for sip options.
- checktools.collcheck.swapfree: Sets the minimun value for SWAP free (0.5 means 50%).
- checktools.exim_check_enable: Exim queue check plugin for *ngcp-witnessd*.
- checktools.active_check_enable: Active node check plugin for *ngcp-witnessd*.
- checktools.asr_ner_statistics: enable/Disable ASR/NER statistics.
- checktools.force: Perform checks even if not active from ngcp-check-active command.
- checktools.kamailio_check_*: Enable/Disable SNMP collective check pluglin for Kamailio.
- checktools.mpt_check_enable: MPT raid SNMP check plugin.
- checktools.mysql_check_enable: Enable/disable MySQL check SNMP plugin.
- checktools.mysql_check_replication: Enable/disable MySQL replication check.
- checktools.mysql_replicate_check_interval: MySQL replication check interval in seconds.
- checktools.mysql_replicate_check_tables: List of tables that need to be checked for replication issues.
- checktools.mysql_replicate_ignore_tables: List of tables that need to be ignored during replication check.
- checktools.oss_check_provisioned_subscribers_enable: OSS provisioned subscribers count plugin.
- checktools.sip_check_enable/sipstats_check_*: Enable/Disable SIP check plugins.
- checktools.snmpd.communities.*: Sets the SNMP community and sources. Entries (i.e. the *sources*) under a community (like `public` in the example) are in a list format, each line starting with "-" and followed by the source address.
- checktools.snmpd.trap_communities.*: Sets the SNMP TRAP community and destination for traps sent by NGCP. Format is the same as for `checktools.snmpd.communities`.

B.1.7 cleanuptools

The following is the cleanup tools section:

```
cleanuptools:
  acc_cleanup_days: 90
  archive_targetdir: '/var/backups/cdr'
  binlog_days: 15
  cdr_archive_months: 2
  cdr_backup_months: 2
  cdr_backup_retro: 3
  compress: gzip
  delete_old_cdr_files:
    enabled: 'no'
    max_age_days: 30
    paths:
      -
        max_age_days: ~
        path: '/home/jail/home/*/20[0-9][0-9][0-9][0-9]/[0-9][0-9]'
        remove_empty_directories: 'yes'
        wildcard: 'yes'
      -
        max_age_days: ~
        path: '/home/jail/home/cdrexport/resellers/*/20[0-9][0-9][0-9][0-9]/[0-9][0-9]'
        remove_empty_directories: 'yes'
        wildcard: 'yes'
      -
        max_age_days: ~
        path: '/home/jail/home/cdrexport/system/20[0-9][0-9][0-9][0-9]/[0-9][0-9]'
        remove_empty_directories: 'yes'
        wildcard: 'yes'
  sql_batch: 10000
  trash_cleanup_days: 30
```

- cleanuptools.acc_cleanup_days: CDR records in `acc` table in `kamailio` database will be deleted after this time
- cleanuptools.binlog_days: Time after MySQL binlogs will be deleted.
- cleanuptools.cdr_archive_months: How many months worth of records to keep in monthly CDR backup tables, instead of dumping them into archive files and dropping them from database.
- cleanuptools.cdr_backup_months: How many months worth of records to keep in the current *cdr* table, instead of moving them into the monthly CDR backup tables.
- cleanuptools.cdr_backup_retro: How many months to process for backups, going backwards in time and skipping `cdr_backup_months` months first, and store them in backup tables. Any older record will be left untouched.
- cleanuptools.delete_old_cdr_files:

  - enabled: Enable (yes) or disable (no) exported CDR cleanup.
  - max_age_days: Gives the expiration time of the exported CDR files in days. There is a general value which may be overridden by a local value provided at a specific path. The local value is valid for the particular path only.
  - paths: an array of path definitions
    - path: a path where CDR files are to be found and deleted; this may contain wildcard characters
    - wildcard: Enable (yes) or disable (no) using wildcards in the path
    - remove_empty_directories: Enable (yes) or disable (no) removing empty directories if those are found in the given path
    - max_age_days: the local expiration time value for files in the particular path

- cleanuptools.sql_batch: How many records to process within a single SQL statement.
- cleanuptools.trash_cleanup_days: Time after CDRs from acc_trash and acc_backup tables in kamailio database will be deleted.

For the description of *cleanuptools* please visit Cleanuptools Description Section 15.4 section of the handbook.

B.1.8 cluster_sets

The following is the cluster sets section:

```
cluster_sets:
  default:
    dispatcher_id: 50
  default_set: default
  type: central
```

- cluster_sets.<label>: an arbitrary label of the cluster set; in the above example we have default
- cluster_sets.<label>.dispatcher_id: a unique, numeric value that identifies a particular cluster set
- cluster_sets.default_set: selects the default cluster set
- cluster_sets.type: the type of cluster set; can be central or distributed

B.1.9 database

The following is the database section:

```
database:
  bufferpoolsize: 24768M
```

- database.bufferpoolsize: Innodb_buffer_pool_size value in /etc/mysql/my.cnf

B.1.10 faxserver

The following is the fax server section:

```
faxserver:
  enable: yes
  fail_attempts: '3'
  fail_retry_secs: '60'
  mail_from: 'Sipwise NGCP FaxServer <voipfax@ngcp.sipwise.local>'
```

- faxserver.enable: *yes*/*no* to enable or disable ngcp-faxserver on the platform respectively.
- faxserver.fail_attempts: Amount of attempts to send a fax after which it is marked as *failed*.
- faxserver.fail_retry_secs: Amount of seconds to wait between "fail_attemts".
- faxserver.mail_from: Sets the e-mail From Header for incoming fax.

B.1.11 general

The following is the general section:

```
general:
  adminmail: adjust@example.org
  companyname: sipwise
  lang: en
  maintenance: no
  production: yes
  timezone: localtime
```

- general.adminmail: Email address used by monit to send notifications to.
- general.companyname: Label used in SNMPd configuration.
- general.lang: Sets sounds language (e.g: *de* for German)
- general.production: Label to hint self-check scripts about installation mode.
- general.maintenance: maintenance mode necessary for safe upgrades.
- general.timezone: sip:provider CE Timezone

B.1.12 heartbeat

The following is the heartbeat section:

```
heartbeat:
  hb_watchdog:
    action_max: 5
    enable: 'yes'
    interval: 10
    transition_max: 10
  pingnodes:
    - 10.60.1.1
    - 192.168.3.4
```

- heartbeat.hb_watchdog.enable: Enable heartbeat watchdog in order to prevent and fix split brain scenario.
- heartbeat.hb_watchdog.action_max: Max errors before taking any action.
- heartbeat.hb_watchdog.interval: Interval in secs for the check.
- heartbeat.hb_watchdog.transition_max: Max checks in transition state.
- heartbeat.pingnodes: List of pingnodes for heartbeat. Minimun 2 entries, otherwise by default NGCP will set the default gateway and DNS servers as pingnodes.

B.1.13 intercept

The following is the legal intercept section:

```
intercept:
  captagent:
    port: 18090
    schema: http
  enabled: 'no'
```

- intercept.captagent.enable: Enable captagent for Lawful Interception (addiotional NGCP module).

B.1.14 kamailio

The following is the kamailio section:

```
kamailio:
  lb:
    cfgt: no
    debug:
      enable: no
      modules:
      - level: '1'
        name: core
      - level: '3'
```

```
        name: xlog
    debug_level: '1'
    external_sbc: []
    extra_sockets: ~
    max_forwards: '70'
    mem_log: '1'
    mem_summary: '12'
    nattest_exception_ips:
    - 1.2.3.4
    - 5.6.7.8
    pkg_mem: '16'
    port: '5060'
    remove_isup_body_from_replies: no
    security:
      dos_ban_enable: yes
      dos_ban_time: '300'
      dos_reqs_density_per_unit: '50'
      dos_sampling_time_unit: '5'
      dos_whitelisted_ips: []
      dos_whitelisted_subnets: []
      failed_auth_attempts: '3'
      failed_auth_ban_enable: yes
      failed_auth_ban_time: '3600'
      topoh:
        enable: no
        mask_callid: no
        mask_ip: 127.0.0.8
    shm_mem: '64'
    skip_contact_alias_for_ua_when_tcp:
      enable: no
      user_agent_patterns: []
    start: yes
    strict_routing_safe: no
    syslog_options: yes
    tcp_children: 1
    tcp_max_connections: '2048'
    tls:
      enable: no
      port: '5061'
      sslcertfile: /etc/ngcp-config/ssl/myserver.crt
      sslcertkeyfile: /etc/ngcp-config/ssl/myserver.key
    udp_children: 1
    use_dns_cache: on
  proxy:
    allow_info_method: no
    allow_msg_method: no
    allow_peer_relay: no
    allow_refer_method: no
```

```
  always_anonymize_from_user: no
  authenticate_bye: no
  cf_depth_limit: '10'
  cfgt: no
  check_prev_forwarder_as_upn: no
  children: 1
  debug:
    enable: no
    modules:
    - level: '1'
      name: core
    - level: '3'
      name: xlog
  debug_level: '1'
  default_expires: '3600'
  default_expires_range: '30'
  dlg_timeout: '43200'
  early_rejects:
    block_admin:
      announce_code: '403'
      announce_reason: Blocked by Admin
    block_callee:
      announce_code: '403'
      announce_reason: Blocked by Callee
    block_caller:
      announce_code: '403'
      announce_reason: Blocked by Caller
    block_contract:
      announce_code: '403'
      announce_reason: Blocked by Contract
    block_in:
      announce_code: '403'
      announce_reason: Block in
    block_out:
      announce_code: '403'
      announce_reason: Blocked out
    block_override_pin_wrong:
      announce_code: '403'
      announce_reason: Incorrect Override PIN
    callee_busy:
      announce_code: '486'
      announce_reason: Busy Here
    callee_offline:
      announce_code: '480'
      announce_reason: Offline
    callee_tmp_unavailable:
      announce_code: '480'
      announce_reason: Temporarily Unavailable
```

```
    callee_tmp_unavailable_gp:
      announce_code: '480'
      announce_reason: Unavailable
    callee_tmp_unavailable_tm:
      announce_code: '408'
      announce_reason: Request Timeout
    callee_unknown:
      announce_code: '404'
      announce_reason: Not Found
    cf_loop:
      announce_code: '480'
      announce_reason: Unavailable
    emergency_invalid:
      announce_code: '404'
      announce_reason: Emergency code not available in this region
    emergency_unsupported:
      announce_code: '403'
      announce_reason: Emergency Calls Not Supported
    invalid_speeddial:
      announce_code: '484'
      announce_reason: Speed-Dial slot empty
    locked_in:
      announce_code: '403'
      announce_reason: Callee locked
    locked_out:
      announce_code: '403'
      announce_reason: Caller locked
    max_calls_in:
      announce_code: '486'
      announce_reason: Busy
    max_calls_out:
      announce_code: '403'
      announce_reason: Maximum parallel calls exceeded
    no_credit:
      announce_code: '402'
      announce_reason: Insufficient Credit
    peering_unavailable:
      announce_code: '503'
      announce_reason: PSTN Termination Currently Unavailable
    reject_vsc:
      announce_code: '403'
      announce_reason: VSC Forbidden
    relaying_denied:
      announce_code: '403'
      announce_reason: Relaying Denied
    unauth_caller_ip:
      announce_code: '403'
      announce_reason: Unauthorized IP detected
```

```
  emergency_priorization:
    enabled: no
    register_fake_200: yes
    register_fake_expires: '3600'
    reject_code: '503'
    reject_reason: Temporary Unavailable
    retry_after: '3600'
  enum_suffix: e164.arpa.
  expires_range: '30'
  filter_100rel_from_supported: no
  filter_failover_response: 408|500|503
  foreign_domain_via_peer: no
  fritzbox:
    enable: no
    prefixes:
    - 0$avp(caller_ac)
    - $avp(caller_cc)$avp(caller_ac)
    - \+$avp(caller_cc)$avp(caller_ac)
    - 00$avp(caller_cc)$avp(caller_ac)
    special_numbers:
    - '112'
    - '110'
    - 118[0-9]{2}
  ignore_auth_realm: no
  ignore_subscriber_allowed_clis: no
  keep_original_to: no
  latency_limit_action: '100'
  latency_limit_db: '500'
  latency_log_level: '1'
  latency_runtime_action: 1000
  lnp:
    api:
      add_caller_cc_to_lnp_dst: no
      invalid_lnp_routing_codes:
      - ^EE00
      - ^DD00
      keepalive_interval: '3'
      lnp_request_blacklist: []
      lnp_request_whitelist: []
      port: '8991'
      reply_error_on_lnp_failure: no
      request_timeout: '1000'
      server: localhost
    enabled: no
    skip_callee_lnp_lookup_from_any_peer: no
    type: api
  lookup_peer_destination_domain_for_pbx: no
  loop_detection:
```

```
      enable: no
      expire: '1'
      max: '5'
    max_expires: '43200'
    max_gw_lcr: '128'
    max_registrations_per_subscriber: '5'
    mem_log: '1'
    mem_summary: '12'
    min_expires: '60'
    nathelper:
      sipping_from: sip:pinger@sipwise.local
    nathelper_dbro: no
    natping_interval: '30'
    natping_processes: 1
    nonce_expire: '300'
    pbx:
      hunt_display_fallback_format: '[H %s]'
      hunt_display_fallback_indicator: $var(cloud_pbx_hg_ext)
      hunt_display_format: '[H %s]'
      hunt_display_indicator: $var(cloud_pbx_hg_displayname)
      hunt_display_maxlength: 8
      ignore_cf_when_hunting: no
    peer_probe:
      available_treshold: '1'
      enable: yes
      from_uri_domain: probe.ngcp.local
      from_uri_user: ping
      interval: '10'
      method: OPTIONS
      reply_codes: class=2;class=3;code=403;code=404;code=405
      timeout: '5'
      unavailable_treshold: '1'
    perform_peer_failover_on_tm_timeout: yes
    perform_peer_lcr: no
    pkg_mem: '32'
    port: '5062'
    presence:
      enable: yes
      max_expires: '3600'
      reginfo_domain: example.org
    proxy_lookup: no
    push:
      apns_alert: New call
      apns_sound: incoming_call.xaf
    report_mos: yes
    set_ruri_to_peer_auth_realm: no
    shm_mem: '125'
    start: yes
```

```
    store_recentcalls: no
    syslog_options: yes
    tcp_children: 1
    tm:
      fr_inv_timer: '180000'
      fr_timer: '9000'
    treat_600_as_busy: yes
    use_enum: no
    usrloc_dbmode: '1'
    voicebox_first_caller_cli: yes
```

- kamailio.lb.cfgt: Enable/disable unit test config file execution tracing.
- kamailio.lb.debug.enable: Enable per-module debug options.
- kamailio.lb.debug.modules: List of modules to be traced with respective debug level.
- kamailio.lb.debug_level: Default debug level for `kamailio-lb`.
- kamailio.lb.external_sbc: SIP URI of external SBC used in the Via Route option of peering server.
- kamailio.lb.extra_sockets: Add here extra sockets for Load Balancer.
- kamailio.lb.max_forwards: Set the value for the Max Forwards SIP header for outgoing messages.
- kamailio.lb.mem_log: Specifies on which log level the memory statistics will be logged.
- kamailio.lb.mem_summary: Parameter to control printing of memory debugging information on exit or SIGUSR1 to log.
- kamailio.lb.nattest_exception_ips: List of IPs that don't need the NAT test.
- kamailio.lb.shm_mem: Shared memory used by Kamailio Load Balancer.
- kamailio.lb.pkg_mem: PKG memory used by Kamailio Load Balancer.
- kamailio.lb.port: Default listen port.
- kamailio.lb.remove_isup_body_from_replies: Enable/disable stripping of ISUP part from the message body.
- kamailio.lb.security.dos_ban_enable: Enable/Disable DoS Ban.
- kamailio.lb.security.dos_ban_time: Sets the ban time.
- kamailio.lb.security.dos_reqs_density_per_unit: Sets the requests density per unit (if we receive more then * lb.dos_reqs_density_per_u within dos_sampling_time_unit the user will be banned).
- kamailio.lb.security.dos_sampling_time_unit: Sets the DoS unit time.
- kamailio.lb.security.dos_whitelisted_ips: Write here the whitelisted IPs.
- kamailio.lb.security.dos_whitelisted_subnets: Write here the whitelisted IP subnets.
- kamailio.lb.security.failed_auth_attempts: Sets how many authentication attempts allowed before ban.

- kamailio.lb.security.failed_auth_ban_enable: Enable/Disable authentication ban.
- kamailio.lb.security.failed_auth_ban_time: Sets how long a user/IP has be banned.
- kamailio.lb.topoh.enable: Enable topology hiding module (see the Topology Hiding Section 16.6 subchapter for a detailed description).
- kamailio.lb.topoh.mask_callid: if set to `yes`, the SIP Call-ID header will also be encoded.
- kamailio.lb.topoh.mask_ip: an IP address that will be used to create valid SIP URIs, after encoding the real/original header content.
- kamailio.lb.start: Enable/disable kamailio-lb service.
- kamailio.lb.strict_routing_safe: Enable strict routing handle feature.
- kamailio.lb.syslog_options: Enable/disable logging of SIP OPTIONS messages to `kamailio-options-lb.log`.
- kamailio.lb.tcp_children: Number of TCP worker processes.
- kamailio.lb.tcp_max_connections: Maximum number of open TCP connections.
- kamailio.lb.tls.enable: Enable TLS socket.
- kamailio.lb.tls.port: Set TLS listening port.
- kamailio.lb.tls.sslcertificate: Path for the SSL certificate.
- kamailio.lb.tls.sslcertkeyfile: Path for the SSL key file.
- kamailio.lb.udp_children: Number of UDP worker processes.
- kamailio.lb.use_dns_cache: Enable/disable use of internal DNS cache.
- kamailio.proxy.allow_info_method: Allow INFO method.
- kamailio.proxy.allow_msg_method: Allow MESSAGE method.
- kamailio.proxy.allow_peer_relay: Allow peer relay. Call coming from a peer that doesn't match a local subscriber will try to go out again, matching the peering rules.
- kamailio.proxy.allow_refer_method: Allow REFER method. Enable it with caution.
- kamailio.proxy.always_anonymize_from_user: Enable anonymization of full From URI (as opposed to just From Display-name part by default), has same effect as enabling the preference `anonymize_from_user` for all peers.
- kamailio.proxy.authenticate_bye: Enable BYE authentication.
- kamailio.proxy.cf_depth_limit: CF loop detector. How many CF loops are allowed before drop the call.
- kamailio.proxy.cfgt: Enable/disable unit test config file execution tracing.
- kamailio.proxy.check_prev_forwarder_as_upn: Enable/disable validation of the forwarder's number taken from the `Diversion` or `History-Info` header.
- kamailio.proxy.children: Number of UDP worker processes.

- kamailio.proxy.debug.enable: Enable per-module debug options.
- kamailio.proxy.debug.modules: List of modules to be traced with respective debug level.
- kamailio.proxy.debug_level: Default debug level for `kamailio-proxy`.
- kamailio.proxy.default_expires: Default expires value in seconds for a new registration (for REGISTER messages that contains neither Expires HFs nor expires contact parameters).
- kamailio.proxy.default_expires_range: This parameter specifies that the expiry used for the registration should be randomly chosen within `default_expires_range` seconds of the `default_expires` parameter.
- kamailio.proxy.dlg_timeout: Dialog timeout in seconds (by default 43200 sec - 12 hours).
- kamailio.proxy.early_rejects: Customize here the response codes and sound prompts for various reject scenarios. See the subchapter Configuring Early Reject Sound Sets Section 7.13.1 for a detailed description.
- kamailio.proxy.emergency_prioritization.enabled: Enable an emergency mode support.
- kamailio.proxy.emergency_prioritization.register_fake_200: When enabled, generates a fake 200 response to REGISTER from non-prioritized subscriber in emergency mode.
- kamailio.proxy.emergency_prioritization.register_fake_expires: Expires value for the fake 200 response to REGISTER.
- kamailio.proxy.emergency_prioritization.reject_code: Reject code for the non-emergency request.
- kamailio.proxy.emergency_prioritization.reject_reason: Reject reason for the non-emergency request.
- kamailio.proxy.emergency_prioritization.retry_after: Retry-After value when rejecting the non-emergency request.

Tip

In order to learn about details of *emergency priorization* function of NGCP please refer to Section 7.6 part of the handbook.

- kamailio.proxy.enum_suffix: Sets ENUM suffix - don't forget *.* (dot).
- kamailio.proxy.expires_range: Set randomization of expires for REGISTER messages (similar to default_expires_range but applies to recieved expires value).
- kamailio.proxy.filter_100rel_from_supported: Enable filtering of *100rel* from Supported header, to disable PRACK.
- kamailio.proxy.filter_failover_response: Specify the list of SIP responses that trigger a failover on the next available peering server.
- kamailio.proxy.foreign_domain_via_peer: Enable/disable of routing of calls to foreign SIP URI via peering servers.
- kamailio.proxy.fritzbox.enable: Enable detection for Fritzbox special numbers. Ex. Fritzbox add some prefix to emergency numbers.
- kamailio.proxy.fritzbox.prefixes: Fritybox prefixes to check. Ex. *0$avp(caller_ac)*
- kamailio.proxy.fritzbox.special_numbers: Specifies Fritzbox special number patterns. They will be checked with the prefixes defined. Ex. *112*, so the performed check will be *sip:0$avp(caller_ac)112@* if prefix is *0$avp(caller_ac)*

- kamailio.proxy.ignore_auth_realm: Ignore SIP authentication realm.
- kamailio.proxy.ignore_subscriber_allowed_clis: Set to *yes* to ignore the subscriber's `allowed_clis` preference so that the User-Provided CLI is only checked against customer's `allowed_clis` preference.
- kamailio.proxy.latency_limit_action: Limit of runtime in ms for config actions. If a config action executed by cfg interpreter takes longer than this value, a message is printed in the logs.
- kamailio.proxy.latency_limit_db: Limit of runtime in ms for DB queries. If a DB operation takes longer than this value, a warning is printed in the logs.
- kamailio.proxy.latency_log_level: Log level to print the messages related to latency. Defaut is 1 (INFO).
- kamailio.proxy.latency_runtime_action: Limit of runtime in ms for SIP message processing cycle. If the SIP message processing takes longer than this value, a warning is printed in the logs.
- kamailio.proxy.keep_original_to: Not used now.
- kamailio.proxy.lnp.api.add_caller_cc_to_lnp_dst: Enable/disable adding of caller country code to LNP routing number of the result (*no* by default, LNP result in E.164 format is assumed).
- kamailio.proxy.lnp.api.invalid_lnp_routing_codes [only for `api` type]: number matching pattern for routing numbers that represent invalid call destinations; an announcement is played in that case and the call is dropped.
- kamailio.proxy.lnp.api.keepalive_interval: Not used now.
- kamailio.proxy.lnp.api.lnp_request_whitelist [only for `api` type]: list of matching patterns of called numbers for which LNP lookup must be done.
- kamailio.proxy.lnp.api.lnp_request_blacklist [only for `api` type]: list of matching patterns of called numbers for which LNP lookup must not be done.
- kamailio.proxy.lnp.api.port: Not used now.
- kamailio.proxy.lnp.api.reply_error_on_lnp_failure: Specifies whether platform should drop the call in case of LNP API server failure or continue routing the call to the original callee without LNP.
- kamailio.proxy.lnp.api.request_timeout [only for `api` type]: timeout in milliseconds while Proxy waits for the response of an LNP query from *Sipwise LNP daemon*.
- kamailio.proxy.lnp.api.server: Not used now.
- kamailio.proxy.lnp.enabled: Enable/disable LNP (local number portability) lookup during call setup.
- kamailio.proxy.lnp.skip_callee_lnp_lookup_from_any_peer: if set to *yes*, the destination LNP lookup is skipped (has same effect as enabling preference skip_callee_lnp_lookup_from_any_peer for all peers).
- kamailio.proxy.lnp.type: method of LNP lookup; valid values are: `local` (local LNP database) and `api` (LNP lookup through external gateways). *PLEASE NOTE:* the `api` type of LNP lookup is only available for NGCP PRO / CARRIER installations.
- kamailio.proxy.lookup_peer_destination_domain_for_pbx: one of [yes, no, peer_host_name] - Sets the content of destination_domain CDR field for calls between CloudPBX subscribers. In case of *no* this field contains name of CloudPBX domain; *yes*: peer destination domain; *peer_host_name*: human-readable name of the peering server.

- kamailio.proxy.loop_detection.enable: Enable the SIP loop detection based on the combination of SIP-URI, To and From header URIs.
- kamailio.proxy.loop_detection.expire: Sampling interval in seconds for the incoming INVITE requests (by default 1 sec).
- kamailio.proxy.loop_detection.max: Maximum allowed number of SIP requests with the same SIP-URI, To and From header URIs within sampling interval. Requests in excess of this limit will be rejected with 482 Loop Detected response.
- kamailio.proxy.max_expires: Sets the maximum expires in seconds for registration.
- kamailio.proxy.max_gw_lcr: Defines the maximum number of gateways in lcr_gw table
- kamailio.proxy.max_registrations_per_subscriber: Sets the maximum registration per subscribers.
- kamailio.proxy.mem_log: Specifies on which log level the memory statistics will be logged.
- kamailio.proxy.mem_summary: Parameter to control printing of memory debugging information on exit or SIGUSR1 to log.
- kamailio.proxy.min_expires: Sets the minimum expires in seconds for registration.
- kamailio.proxy.nathelper.sipping_from: Set the From header in OPTIONS NAT ping.
- kamailio.proxy.nathelper_dbro: Default is "no". This will be "yes" on CARRIER in order to activate the use of a read-only connection using LOCAL_URL
- kamailio.proxy.natping_interval: Sets the NAT ping interval in seconds.
- kamailio.proxy.natping_processes: Set the number of NAT ping worker processes.
- kamailio.proxy.nonce_expire: Nonce expire time in seconds.
- kamailio.proxy.pbx.hunt_display_fallback_format: Default is *[H %s]*. Sets the format of the hunt group indicator that is sent as initial part of the From Display Name when subscriber is called as a member of PBX hunt group if the preferred format defined by the `hunt_display_format` and `hunt_display_indicator` can not be used (as in the case of not provisioned subscriber settings). The *%s* part is replaced with the value of the `hunt_display_fallback_indicator` variable.
- kamailio.proxy.pbx.hunt_display_fallback_indicator: The internal kamailio variable that sets the number or extension of the hunt group. Default is `$var(cloud_pbx_hg_ext)` which is populated during call routing with the extension of the hunt group.
- kamailio.proxy.pbx.hunt_display_format: Default is *[H %s]*. Sets the format of hunt group indicator that is sent as initial part of the From Display Name when subscriber is called as a member of PBX hunt group. This is the preferred (default) indicator format with Display Name, where the *%s* part is replaced with the value of the `hunt_display_indicator` variable.
- kamailio.proxy.pbx.hunt_display_indicator: The internal kamailio variable that contains the preferred identifier of the hunt group. Default is `$var(cloud_pbx_hg_displayname)` which is populated during call routing with the provisioned Display Name of the hunt group.
- kamailio.proxy.pbx.hunt_display_maxlength: Default is *8*. Sets the maximum length of the variable used as the part of hunt group indicator in Display Name. The characters beyond this limit are truncated in order for hunt group indicator and calling party information to fit on display of most phones.
- kamailio.proxy.pbx.ignore_cf_when_hunting: Default is *no*. Whether to disregard all individual call forwards (CFU, CFB, CFT and CFNA) of PBX extensions when they are called via hunt groups. Note that call forwards configured to local services such as Voicebox or Conference are always skipped from group hunting.

- kamailio.proxy.peer_probe.enable: Enable the peer probing, must be also checked per individual peer in the panel/API.
- kamailio.proxy.peer_probe.interval: Peer probe interval in seconds.
- kamailio.proxy.peer_probe.timeout: Peer probe response wait timeout in seconds.
- kamailio.proxy.peer_probe.reply_codes: Defines the response codes that are considered successful response to the configured probe request, e.g. `class=2;class=3;code=403;code=404;code=405`, with class defining a code range.
- kamailio.proxy.peer_probe.unavailable_treshold: Defines after how many failed probes a peer is considered unavailable.
- kamailio.proxy.peer_probe.available_treshold: Defines after how many successful probes a peer is considered available.
- kamailio.proxy.peer_probe.from_uri_user: From-userpart for the probe requests.
- kamailio.proxy.peer_probe.from_uri_domain From-hostpart for the probe requests.
- kamailio.proxy.peer_probe.method: [OPTIONS|INFO] - Request method for probe request.

> **Tip**
>
> You can find more information about peer probing configuration in Section 7.10.2 of the handbook.

- kamailio.proxy.perform_peer_failover_on_tm_timeout: Specifies the failover behavior when maximum ring timeout (fr_inv_timer) has been reached. In case it is set to *yes*: failover to the next peer if any; in case of *no* stop trying other peers.
- kamailio.proxy.perform_peer_lcr: Enable/Disable Least Cost Routing based on peering fees.
- kamailio.proxy.pkg_mem: PKG memory used by Kamailio Proxy.
- kamailio.proxy.shm_mem: Shared memory used by Kamailio Proxy.
- kamailio.proxy.port: SIP listening port.
- kamailio.proxy.presence.enable: Enable/disable presence feature
- kamailio.proxy.presence.max_expires: Sets the maximum expires value for PUBLISH/SUBSCRIBE message. Defines expiration of the presentity record.
- kamailio.proxy.presence.reginfo_domain: Set FQDN of the NGCP domain used in callback for mobile push.
- kamailio.proxy.push.apns_alert: Set the content of *alert* field towards APNS.
- kamailio.proxy.push.apns_sound: Set the content of *sound* field towards APNS.
- kamailio.proxy.report_mos: Enable MOS reporting in the log file.
- kamailio.proxy.set_ruri_to_peer_auth_realm: Set R-URI using peer auth realm.
- kamailio.proxy.start: Enable/disable kamailio-proxy service.
- kamailio.proxy.store_recentcalls: Store recent calls to redis (used by Malicious Call Identification application).
- kamailio.proxy.syslog_options: Enable/disable logging of SIP OPTIONS messages to `kamailio-options-proxy.log`.
- kamailio.proxy.tcp_children: Number of TCP worker processes.

- kamailio.proxy.tm.fr_inv_timer: Set INVITE transaction timeout if no final reply for an INVITE arrives after a provisional message was received (ringing timeout).
- kamailio.proxy.tm.fr_timer: Set INVITE transaction timeout if the destination is not responding with provisional response message.
- kamailio.proxy.treat_600_as_busy: Enable the 6xx response handling according to RFC3261. When enabled, the 6xx response should stop the serial forking. Also, CFB will be triggered or busy prompt played as in case of 486 Busy response.
- kamailio.proxy.use_enum: Enable/Disable ENUM feature.
- kamailio.proxy.usrloc_dbmode: Set the mode of database usage for persistent contact storage.
- kamailio.proxy.voicebox_first_caller_cli: When enabled the previous forwarder's CLI will be used as caller CLI in case of chained Call Forwards.

B.1.15 mediator

The following is the mediator section:

```
mediator:
  interval: 10
```

- mediator.interval: Running interval of mediator.

B.1.16 modules

The following is the modules section:

```
modules:
  - enable: no
    name: dummy
    options: numdummies=2
```

- modules: list of configs needed for load kernel modules on boot.
- enable: Enable/disable loading of the specific module (yes/no)
- name: kernel module name
- options: kernel module options if needed

B.1.17 nginx

The following is the nginx section:

```
nginx:
  status_port: 8081
  xcap_port: 1080
```

- nginx.status_port: Status port used by nginx server
- nginx.xcap_port: XCAP port used by nginx server

B.1.18 ntp

The following is the ntp server section:

```
ntp:
  servers:
    - 0.debian.pool.ntp.org
    - 1.debian.pool.ntp.org
    - 2.debian.pool.ntp.org
    - 3.debian.pool.ntp.org
```

- ntp.servers: Define your NTP server list.

B.1.19 ossbss

The following is the ossbss section:

```
ossbss:
  apache:
    port: 2443
    proxyluport: 1080
    restapi:
      sslcertfile: '/etc/ngcp-panel/api_ssl/api_ca.crt'
      sslcertkeyfile: '/etc/ngcp-panel/api_ssl/api_ca.key'
    serveradmin: support@sipwise.com
    servername: "\"myserver\""
    ssl_enable: 'yes'
    sslcertfile: '/etc/ngcp-config/ssl/myserver.crt'
    sslcertkeyfile: '/etc/ngcp-config/ssl/myserver.key'
  frontend: 'no'
  htpasswd:
    -
      pass: '{SHA}w4zj3mxbmynIQ1jsUEjSkN2z2pk='
      user: ngcpsoap
  logging:
    apache:
      acc:
```

```
        facility: daemon
        identity: oss
        level: info
      err:
        facility: local7
        level: info
    ossbss:
      facility: local0
      identity: provisioning
      level: DEBUG
    web:
      facility: local0
      level: DEBUG
  provisioning:
    allow_ip_as_domain: 1
    allow_numeric_usernames: 0
    auto_allow_cli: 1
    carrier:
      account_distribution_function: roundrobin
      prov_distribution_function: roundrobin
    credit_warnings:
      -
        domain: example.com
        recipients:
          - nobody@example.com
        threshold: 1000
    faxpw_min_char: 0
    log_passwords: 0
    no_logline_truncate: 0
    pw_min_char: 6
    routing:
      ac_regex: '[1-9]\d{0,4}'
      cc_regex: '[1-9]\d{0,3}'
      sn_regex: '[1-9]\d+'
    tmpdir: '/tmp'
```


- ossbss.frontend: Enable disable SOAP interface. Set value to *fcgi* to enable old SOAP interface.
- ossbss.htpasswd: Sets the username and SHA hashed password for SOAP access. You can generate the password using the following command: htpasswd -nbs myuser mypassword.
- ossbss.provisioning.allow_ip_as_domain: Allow or not allow IP address as SIP domain (0 is not allowed).
- ossbss.provisioning.allow_numeric_usernames: Allow or not allow numeric SIP username (0 is not allowed).
- ossbss.provisioning.faxpw_min_char: Minimum number of characters for fax passwords.
- ossbss.provisioning.pw_min_char: Minimum number of characters for sip passwords.

- ossbss.provisioning.log_password: Enable logging of passwords.
- ossbss.provisioning.routing: Regexp for allowed AC (Area Code), CC (Country Code) and SN (Subscriber Number).

B.1.20 pbx (only with additional cloud PBX module installed)

The following is the PBX section:

```
pbx:
  bindport: 5085
  enable: 'no'
  highport: 55000
  lowport: 50001
  media_processor_threads: 10
  session_processor_threads: 10
  xmlrpcport: 8095
```

- pbx.enable: Enable Cloud PBX module.

B.1.21 prosody

The following is the prosody section:

```
prosody:
  ctrl_port: 5582
  log_level: info
```

- prosody.ctrl_port: XMPP server control port.
- prosody.log_level: Prosody loglevel.

B.1.22 pushd

The following is the pushd section:

```
pushd:
  apns:
    enable: yes
    endpoint: api.push.apple.com
    endpoint_port: 0
    extra_instances:
    - certificate: '/etc/ngcp-config/ssl/PushCallkitCert.pem'
      enable: yes
      key: '/etc/ngcp-config/ssl/PushCallkitKey.pem'
      type: callkit
    http2_jwt:
```

```
    ec_key: '/etc/ngcp-config/ssl/AuthKey_ABCDE12345.pem'
    ec_key_id: 'ABCDE12345'
    enable: yes
    issuer: 'VWXYZ67890'
    tls_certificate: ''
    tls_key: ''
    topic: 'com.example.appID'
  legacy:
    certificate: '/etc/ngcp-config/ssl/PushChatCert.pem'
    feedback_endpoint: feedback.push.apple.com
    feedback_interval: '3600'
    key: '/etc/ngcp-config/ssl/PushChatKey.pem'
  socket_timeout: 0
domains:
- apns:
    endpoint: api.push.apple.com
    extra_instances:
    - certificate: '/etc/ngcp-config/ssl/PushCallkitCert-example.com.pem'
      enable: no
      key: '/etc/ngcp-config/ssl/PushCallkitKey-example.com.pem''
      type: callkit
    http2_jwt:
      ec_key: '/etc/ngcp-config/ssl/AuthKey_54321EDCBA.pem'
      ec_key_id: '54321EDCBA'
      issuer: '09876ZYXWV'
      tls_certificate: ''
      tls_key: ''
      topic: 'com.example.otherAppID'
    legacy:
      certificate: '/etc/ngcp-config/ssl/PushChatCert-example.com.pem'
      feedback_endpoint: feedback.push.apple.com
      key: '/etc/ngcp-config/ssl/PushChatKey-example.com.pem'
  domain: example.com
  enable: yes
  gcm:
    key: 'google_api_key_for_example.com_here'
enable: yes
gcm:
  enable: yes
  key: 'google_api_key_here'
  priority:
    call: high
    groupchat: normal
    invite: normal
    message: normal
muc:
  exclude: []
  force_persistent: 'true'
```

```
    owner_on_join: 'true'
  one_device_per_subscriber: no
  port: 45060
  processes: 4
  ssl: yes
  sslcertfile: /etc/ngcp-config/ssl/CAsigned.crt
  sslcertkeyfile: /etc/ngcp-config/ssl/CAsigned.key
  unique_device_ids: no
```

- pushd.enable: Enable/Disable the Push Notification feature.
- pushd.apns.enable: Enable/Disable Apple push notification.
- pushd.apns.endpoint: API endpoint hostname or address. Should be one of *api.push.apple.com* or *api.development.push.apple.com* for the newer HTTP2/JWT based protocol, or one of *gateway.push.apple.com* or *gateway.sandbox.push.apple.com* for the legacy protocol.
- pushd.apns.endpoint_port: API endpoint port. Normally 443 or alternatively 2197 for the newer HTTP2/JWT based protocol, or 2195 for the legacy protocol.
- pushd.apns.legacy: Contains all options specific to the legacy APNS protocol. Ignored when HTTP2/JWT is in use.
- pushd.apns.legacy.certificate: Specify the Apple certificate for push notification https requests from the NGCP to an endpoint.
- pushd.apns.legacy.key: Specify the Apple key for push notification https requests from the NGCP to an endpoint.
- pushd.apns.legacy.feedback_endpoint: Hostname or address of the APNS feedback service. Normally one of *feedback.push.apple.com* or *feedback.sandbox.push.apple.com*.
- pushd.apns.legacy.feedback_interval: How often to poll the feedback service, in seconds.
- pushd.apns.extra_instances: If the iOS app supports Callkit push notifications, they can be enabled here and the required separate certificate and key can be specified. Ignored if HTTP2/JWT is enabled.
- pushd.http2_jwt: Contains all options specific to the newer HTTP2/JWT based APNS API protocol.
- pushd.http2_jwt.ec_key: Name of file that contains the elliptic-curve (EC) cryptographic key provided by Apple, in PEM format.
- pushd.http2_jwt.ec_key_id: 10-digit identification string of the EC key in use.
- pushd.http2_jwt.enable: Master switch for the HTTP2/JWT based protocol. Disables the legacy protocol when enabled.
- pushd.http2_jwt.issuer: Issuer string for the JWT token. Normally the 10-digit team ID string for which the EC key was issued.
- pushd.http2_jwt.tls_certificate: Optional client certificate to use for the TLS connection.
- pushd.http2_jwt.tls_key: Optional private key for the client certificate to use for the TLS connection.
- pushd.http2_jwt.topic: Topic string for the JWT token. Normally the bundle ID for the iOS app.
- pushd.gcm.enable: Enable/Disable Google push notification.
- pushd.gcm.key: Specify the Google key for push notification https requests from the NGCP to an endpoint.

- pushd.domains: Supports a separate set of push configurations (API keys, certificates, etc) for all subscribers of the given domain.
- pushd.muc.exclude: list of MUC room jids excluded from sending push notifications.
- pushd.muc.force_persistent: Enable/Disable MUC rooms to be persistent. Needed for NGCP app to work with other clients.
- pushd.muc.owner_on_join: Enable/Disable all MUC participants to be owners of the MUC room. Needed for NGCP app to work with other clients.
- pushd.ssl: The security protocol the NGCP uses for https requests from the app in the push notification process.
- pushd.sslcertfile: The trusted certificate file purchased from a CA
- pushd.sslcertkeyfile: The key file that purchased from a CA
- pushd.unique_device_ids: Allows a subscriber to register the app and have the push notification enabled on more than one mobile device.

B.1.23 qos

The following is the QOS section:

```
qos:
  tos_rtp: 184
  tos_sip: 184
```

- qos.tos_rtp: TOS value for RTP traffic.
- qos.tos_sip: TOS value for SIP traffic.

B.1.24 rate-o-mat

The following is the rate-o-mat section:

```
rateomat:
  enable: 'yes'
  loopinterval: 10
  splitpeakparts: 0
```

- rateomat.enable: Enable/Disable Rate-o-mat
- rateomat.loopinterval: How long we shall sleep before looking for unrated CDRs again.
- rateomat.splitpeakparts: Whether we should split CDRs on peaktime borders.

B.1.25 redis

The following is the redis section:

```
redis:
  database_amount: 16
  port: 6379
  syslog_ident: redis
```

- redis.database_amout: Set the number of databases in redis. The default database is DB 0.
- redis.port: Accept connections on the specified port, default is 6379
- redis.syslog_ident: Specify the syslog identity.

B.1.26 reminder

The following is the reminder section:

```
reminder:
  retries: 2
  retry_time: 60
  sip_fromdomain: voicebox.sipwise.local
  sip_fromuser: reminder
  wait_time: 30
  weekdays: '2, 3, 4, 5, 6, 7'
```

- reminder.retries: How many times the reminder feature have to try to call you.
- reminder.retry_time: Seconds between retries.
- reminder.wait_time: Seconds to wait for an answer.

B.1.27 rsyslog

The following is the rsyslog section:

```
rsyslog:
  elasticsearch:
    action:
      resumeretrycount: '-1'
    bulkmode: 'on'
    dynSearchIndex: 'on'
    enable: 'yes'
    queue:
      dequeuebatchsize: 300
      size: 5000
```

```
      type: linkedlist
  external_address:
  external_log: 0
  external_loglevel: warning
  external_port: 514
  external_proto: udp
  ngcp_logs_preserve_days: 93
```

- rsyslog.elasticsearch.enable: Enable/Disable Elasticsearch web interface
- rsyslog.external_address: Set the remote rsyslog server.
- rsyslog.ngcp_logs_preserve_days: Specify how many days to preserve old rotated log files in /var/log/ngcp/old path.

B.1.28 rtpproxy

The following is the rtp proxy section:

```
rtpproxy:
  allow_userspace_only: yes
  enabled: yes
  firewall_iptables_chain: ''
  log_level: '6'
  maxport: '40000'
  minport: '30000'
  prefer_bind_on_internal: no
  recording:
    enabled: no
    mp3_bitrate: '48000'
    nfs_host: 192.168.1.1
    nfs_remote_path: /var/recordings
    output_dir: /var/lib/rtpengine-recording
    output_format: wav
    output_mixed: yes
    output_single: yes
    resample: no
    resample_to: '16000'
    spool_dir: /var/spool/rtpengine
  rtp_timeout: '60'
  rtp_timeout_onhold: '3600'
```

- rtpproxy.allow_userspace_only: Enable/Disable the user space failover for rtpengine (*yes* means enable). By default rtpengine works in kernel space.
- rtpproxy.firewall_iptables_chain: If set, rtpengine will create an iptables rule for each individual media port opened in this chain.
- rtpproxy.log_level: Verbosity of log messages. The default *6* logs everything except debug messages. Increase to 7 to log everything, or decrease to make logging more quiet.

- rtpproxy.maxport: Maximum port used by rtpengine for RTP traffic.
- rtpproxy.minport: Minimum port used by rtpengine for RTP traffic.
- rtpproxy.recording.enabled: Enable support for call recording.
- rtpproxy.recording.mp3_bitrate: If saving audio as MP3, bitrate of the output file.
- rtpproxy.recording.nfs_host: Mount an NFS share from this host for storage.
- rtpproxy.recording.nfs_remote_path: Remote path of the NFS share to mount.
- rtpproxy.recording.output_dir: Local mount point for the NFS share.
- rtpproxy.recording.output_format: Either *wav* for PCM output or *mp3*.
- rtpproxy.recording.output_mixed: Create output audio files with all contributing audio streams mixed together.
- rtpproxy.recording.output_single: Create separate audio files for each contributing audio stream.
- rtpproxy.recording.resample: Resample all audio to a fixed bitrate (*yes* or *no*).
- rtpproxy.recording.resample_to: If resampling is enabled, resample to this sample rate.
- rtpproxy.recording.spool_dir: Local directory for temporary metadata file storage.
- rtpproxy.rtp_timeout: Consider a call dead if no RTP is received for this long (60 seconds).
- rtpproxy.rtp_timeout_onhold: Maximum limit in seconds for an onhold (1h).

B.1.29 security

The following is the security section. Usage of the firewall subsection is described in Section 16.2:

```
security:
  firewall:
    enable: no
    logging:
      days_kept: '7'
      enable: yes
      file: /var/log/firewall.log
      tag: NGCPFW
    nat_rules4: ~
    nat_rules6: ~
    policies:
      forward: DROP
      input: DROP
      output: ACCEPT
    rules4: ~
    rules6: ~
```

- security.firewall.enable: Enable/disable iptables configuration and rule generation for IPv4 and IPv6 (default: `no`)

- security.firewall.logging.days_kept: Number of days logfiles are kept on the system before being deleted (log files are rotated daily, default: `7`)
- security.firewall.logging.enable: Enables/disables logging of all packets dropped by the NGCP firewall (default: `yes`)
- security.firewall.logging.file: File firewall log messages go to (default: `/var/log/firewall.log`)
- security.firewall.logging.tag: String prepended to all log messages (internally `DROP` is added to any tag indicating the action triggering the message, default: `NGCPFW`)
- security.firewall.nat_rules4: Optional list of IPv4 firewall rules added to table `nat` using iptables-persistent syntax (default: `undef`)
- security.firewall.nat_rules6: Optional list of IPv6 firewall rules added to table `nat` using iptables-persistent syntax (default: `undef`)
- security.firewall.policies.forward: Default policy for iptables `FORWARD` chain (default: `DROP`)
- security.firewall.policies.input: Default policy for iptables `INPUT` chain (default: `DROP`)
- security.firewall.policies.output: Default policy for iptables `OUTPUT` chain (default: `ACCEPT`)
- security.firewall.rules4: Optional list of IPv4 firewall rules added to table `filter` using iptables-persistent syntax (default: `undef`)
- security.firewall.rules6: Optional list of IPv6 firewall rules added to table `filter` using iptables-persistent syntax (default: `undef`)

B.1.30 sems

The following is the SEMS section:

```
sems:
  bindport: 5080
  conference:
    enable: 'yes'
    max_participants: 10
  debug: 'no'
  highport: 50000
  lowport: 40001
  media_processor_threads: 10
  prepaid:
    enable: 'yes'
  sbc:
    calltimer_enable: 'yes'
    calltimer_max: 3600
    outbound_timeout: 6000
    sdp_filter:
      codecs: PCMA,PCMU,telephone-event
      enable: 'yes'
      mode: whitelist
```

```
    session_timer:
      enable: 'yes'
      max_timer: 7200
      min_timer: 90
      session_expires: 300
  session_processor_threads: 10
  vsc:
    block_override_code: 80
    cfb_code: 90
    cfna_code: 93
    cft_code: 92
    cfu_code: 72
    clir_code: 31
    directed_pickup_code: 99
    enable: 'yes'
    park_code: 97
    reminder_code: 55
    speedial_code: 50
    unpark_code: 98
    voicemail_number: 2000
  xmlrpcport: 8090
```

- sems.conference.enable: Enable/Disable conference feature.
- sems.conference.max_participants: Sets the number of concurrent participant.
- sems.highport: Maximum ports used by sems for RTP traffic.
- sems.debug: Enable/Disable debug mode.
- sems.lowport: Minimum ports used by sems for RTP traffic.
- sems.prepaid.enable: Enable/Disable prepaid feature.
- sems.sbc.calltimer_max: Set the default maximum call duration (used if otherwise is not defined by preference).
- sems.sbc.outbound_timeout: Set INVITE transaction timeout if the destination is not responding with provisional response message.
- sems.sbc.session_timer.enable: Enable/Disable session timers (deprecated, use the web interface configuration).
- sems.vsc.*: Define here the VSC codes.

B.1.31 sms

This section provides configuration of **S**hort **M**essage **S**ervice on the NGCP. Description of the SMS module is provided earlier in this handbook here Section 7.24.

In the below example you can see the default values of the configuration parameters.

```
sms:
  core:
    admin_port: '13000'
    smsbox_port: '13001'
  enable: no
  loglevel: '0'
  sendsms:
    max_parts_per_message: '5'
    port: '13002'
  smsc:
    dest_addr_npi: '1'
    dest_addr_ton: '1'
    enquire_link_interval: '58'
    host: 1.2.3.4
    id: default_smsc
    max_pending_submits: '10'
    no_dlr: yes
    password: password
    port: '2775'
    source_addr_npi: '1'
    source_addr_ton: '1'
    system_type: ''
    throughput: '5'
    transceiver_mode: '1'
    username: username
```

- sms.core.admin_port: Port number of admin interface of SMS core module (running on LB nodes).
- sms.core.smsbox_port: Port number used for internal communication between *bearerbox* module on LB nodes and *smsbox* module on PRX nodes. This is a listening port of the *bearerbox* module (running on LB nodes).
- sms.enable: Set to `yes` if you want to enable SMS module.
- sms.loglevel: Log level of SMS module; the default *0* will result in writing only the most important information into the log file.
- sms.sendsms.max_parts_per_message: If the SM needs to be sent as concatenated SM, this parameter sets the max. number of parts for a single (logical) message.
- sms.sendsms.port: Port number of *smsbox* module (running on PRX nodes).
- sms.smsc. : Parameters of the connection to an SMSC
  - dest_addr_npi: Telephony numbering plan indicator for the SM destination, as defined by standards (e.g. *1* stands for E.164)
  - dest_addr_ton: Type of number for the SM destination, as defined by standards (e.g. *1* stands for "international" format)
  - enquire_link_interval: Interval of SMSC link status check in seconds
  - host: IP address of the SMSC
  - id: An arbitrary string for identification of the SMSC; may be used in log files and for routing SMs.

- max_pending_submits: The maximum number of outstanding (i.e. not acknowledged) SMPP operations between the NGCP and SMSC. As a guideline it is recommended that no more than 10 (default) SMPP messages are outstanding at any time.
- no_dlr: Do not request delivery report; when sending an SM and this parameter is set to `yes`, NGCP will not request DR for the message(s). May be required for some particular SMSCs, in order to avoid "Incorrect status report request parameter usage" error messages from the SMSC.
- password: This is the password used for authentication on the SMSC.
- port: Port number of the SMSC where NGCP will connect to.
- source_addr_npi: Telephony numbering plan indicator for the SM source, as defined by standards (e.g. *1* stands for E.164)
- source_addr_ton: Type of number for the SM source, as defined by standards (e.g. *1* stands for "international" format)
- system_type: Defines the SMSC client category in which NGCP belongs to; defaults to "VMA" (Voice Mail Alert) when no value is given. (No need to set any value)
- throughput: The max. number of messages per second that NGCP will send towards the SMSC. (Value type: float)
- transceiver_mode: If set to `1` (yes / true), NGCP will attempt to use a TRANSCEIVER mode connection to the SMSC. It uses the standard transmit port of the SMSC for receiving SMs too.
- username: This is the username used for authentication on the SMSC.

B.1.32 snmpagent

The following is the SNMP Agent section:

```
snmpagent:
  daemonize: '1'
  debug: '0'
  update_interval: '30'
```

- daemonize: Enable/Disable ngcp-snmp-agent daemonization.
- debug: Enable/Disable debug output.
- update_interval: Sets the interval in seconds used to update the fetched data.

B.1.33 sshd

The following is the sshd section:

```
sshd:
  listen_addresses:
    - 0.0.0.0
```

- sshd: specify interface where SSHD should run on. By default sshd listens on all IPs found in network.yml with type *ssh_ext*. Unfortunately sshd can be limited to IPs only and not to interfaces. The current option makes it possible to specify allowed IPs (or all IPs with 0.0.0.0).

B.1.34 sudo

The following is in the sudo section:

```
sudo:
  logging: no
  max_log_sessions: 0
```

- logging: enable/disable the I/O logging feature of sudo. See man page of *sudoreplay(8)*.
- max_log_sessions: when I/O logging is enabled, specifies how many log sessions per individual user sudo should keep before it starts overwriting old ones. The default *0* means no limit.

B.1.35 www_admin

The following is the WEB Admin interface (www_admin) section:

```
www_admin:
  ac_dial_prefix: 0
  apache:
    autoprov_port: 1444
  billing_features: 1
  callingcard_features: 0
  callthru_features: 0
  cc_dial_prefix: 00
  conference_features: 1
  contactmail: adjust@example.org
  dashboard:
    enabled: 1
  default_admin_settings:
    call_data: 0
    is_active: 1
    is_master: 0
    read_only: 0
    show_passwords: 1
  domain:
    preference_features: 1
    rewrite_features: 1
    vsc_features: 0
  fastcgi_workers: 2
  fax_features: 1
  fees_csv:
    element_order:
      - source
      - destination
      - direction
      - zone
```

```
    - zone_detail
    - onpeak_init_rate
    - onpeak_init_interval
    - onpeak_follow_rate
    - onpeak_follow_interval
    - offpeak_init_rate
    - offpeak_init_interval
    - offpeak_follow_rate
    - offpeak_follow_interval
    - use_free_time
http_admin:
  autoprov_port: 1444
  port: 1443
  serveradmin: support@sipwise.com
  servername: "\"myserver\""
  ssl_enable: 'yes'
  sslcertfile: '/etc/ngcp-config/ssl/myserver.crt'
  sslcertkeyfile: '/etc/ngcp-config/ssl/myserver.key'
http_csc:
  autoprov_bootstrap_port: 1445
  autoprov_port: 1444
  port: 443
  serveradmin: support@sipwise.com
  servername: "\"myserver\""
  ssl_enable: 'yes'
  sslcertfile: '/etc/ngcp-config/ssl/myserver.crt'
  sslcertkeyfile: '/etc/ngcp-config/ssl/myserver.key'
logging:
  apache:
    acc:
      facility: daemon
      identity: oss
      level: info
    err:
      facility: local7
      level: info
peer:
  preference_features: 1
peering_features: 1
security:
  password_allow_recovery: 0
  password_max_length: 40
  password_min_length: 6
  password_musthave_digit: 0
  password_musthave_lowercase: 1
  password_musthave_specialchar: 0
  password_musthave_uppercase: 0
  password_sip_autogenerate: 0
```

```
    password_sip_expose_subadmin: 1
    password_web_autogenerate: 0
    password_web_expose_subadmin: 1
  speed_dial_vsc_presets:
    vsc:
      - '*0'
      - '*1'
      - '*2'
      - '*3'
      - '*4'
      - '*5'
      - '*6'
      - '*7'
      - '*8'
      - '*9'
  subscriber:
    auto_allow_cli: 0
    extension_features: 0
  voicemail_features: 1
```

- www_admin.http_admin.*: Define the Administration interface and certificates.
- www_admin.http_csc.*: Define the Customers interface and certificates.
- www_admin.contactmail: Email to show in the GUI's Error page.

B.2 constants.yml Overview

`/etc/ngcp-config/constants.yml` is one of the main configuration files that contains important (static) configuration parameters, like NGCP system-user data.

> **Caution**
> NGCP platform administrator should not change content of `constants.yml` file unless absolutely necessary. Please contact Sipwise Support before changing any of the parameters within the `constants.yml` file!

B.3 network.yml Overview

`/etc/ngcp-config/network.yml` is one of the main configuration files that contains network-related configuration parameters, like IP addresses and roles of the node(s) in sip:provider CE system.

The next example shows a part of the `network.yml` configuration file. Explanation of all the configuration parameters is provided in Network Configuration Section 12 section of the handbook.

Sample host configuration for sip:provider CE

```
self:
  dbnode: '1'
  eth0:
    ip: 10.0.2.15
    netmask: 255.255.255.0
    type:
      - web_ext
      - web_int
      - ssh_ext
  eth1:
    ip: 10.15.20.143
    netmask: 255.255.255.0
    type:
      - ssh_ext
      - web_ext
      - web_int
      - sip_ext
      - rtp_ext
      - mon_ext
  interfaces:
    - lo
    - eth0
    - eth1
  lo:
    cluster_sets:
      - default
    ip: 127.0.0.1
    netmask: 255.255.255.0
    shared_ip: []
    shared_v6ip: []
    type:
      - sip_int
      - ha_int
      - aux_ext
      - ssh_ext
      - api_int
    v6ip: '::1'
  role:
    - proxy
    - lb
    - mgmt
    - rtp
    - db
  status: 'online'
```

C NGCP Internals

This chapter documents internals of the sip:provider CE that should not be usually needed, but might be helpful to understand the overall system.

C.1 Pending reboot marker

The sip:provider CE has the ability to mark a pending reboot for any server, using the file */var/run/reboot-required*. As soon as the file exists, several components will report about a pending reboot to the end-user. The following components report about a pending reboot right now: *ngcp-status*, *ngcpcfg status*, *motd*, *ngcp-upgrade*. Also, ngcp-upgrade will NOT allow proceeding with an upgrade if it notices a pending reboot. It might affect *rtpengine* dkms module building if there is a pending reboot requested by a newly installed kernel, etc.

C.2 Redis id constants

The list of current sip:provider CE Redis DB IDs:

Service	Redis DB N:	central (role *db*)	local	Release	Ticket	Description
sems	redis_db:	-	0	mr3.7.1+	-	HA switchover
rtpengine	redis_db:	-	1	mr3.7.1+	-	HA switchover
proxy	redis_db:	2	-	mr3.7.1+	-	Counter of hunting groups
proxy	redis_db:	3	-	mr3.7.1+	-	Concurrent dialog counters
proxy	redis_db:	-	4	mr3.7.1+	-	List of keys of the central counters
prosody	redis_db:	5	-	mr3.7.1+	-	XMPP cluster
sems PBX	redis_db:	-	6	mr3.7.1+	-	HA switchover
sems	redis_db:	7	-	mr4.1.1+	MT#12707	Sems malicious_call app
captagent	redis_db:	-	8	mr4.1.1+	MT#15427	Captagent internal data
monitoring	redis_db:	9	-	mr4.3+	MT#31	Old SNMP agent monitoring data (unused)
proxy	redis_db:	10	-	mr4.3+	MT#16079	SIP Loop detection

C.2.1 InfluxDB monitoring keys

The *InfluxDB ngcp* monitoring database contains time series of several monitoring sources. The following are some of the current measurements:

node	Cluster node information.
memory	System memory information.
proc_count	Process counts.
monit	Monit supervised processes information.
mail	MTA information.
mysql	MySQL database information.
kamailio	Kamailio statistics information.
sip	SIP statistics information.

The *node* measurement contains the following fields:

active	Cluster node HA state (boolean: 1/0).
hb_proc_state	Cluster node heartbeat process state (boolean: stopped/running).
hb_host_state	Cluster node host state (boolean: up/down).
hb_node_state	Cluster node HA state (ngcp-check-active -p).

The *monit* measurement contains the following fields:

name	The process name.
proc_status	The process status.
monit_status	The monit status.
pid	The process ID.
ppid	The process parent ID.
children	The number of children.
uptime	The process uptime.
cpu_percent	The CPU usage in percent for this process.
cpu_percent_total	The CPU usage in percent for the process group.
memory	The memory in bytes for this process.
memory_total	The memory in bytes for the process group.
memory_percent	The memory in percent for this process.
memory_percent_total	The memory in percent for the process group.
data_collected	The timestamp when the data was collected.

The *mysql* measurement contains the following fields:

last_io_error	Last IO error description.
last_sql_error	Last SQL error description.
queries_per_second_average	Average of queries per second.

replication_discrepancies	Number of replication discrepancies.

C.3 Enum preferences

All tables are in database "provisioning".

So called "enum preferences" allow a fixed set of possible values, an enumeration, for preferences. Following the differences between other preferences are described.

Setting the attribute "data_type" of table "voip_preferences" to "enum" marks a preferences as an enum. The list of possible options is stored in table "voip_preferences_enum".

voip_preferences_enum is:

```
id
    boring pkey

preference_id
    Reference to table voip_preferences.

label
    A label to be displayed in frontends.

value
    Value that will be written to voip_[usr|dom|peer]_preferences.value
    if it is NOT NULL. Will not be written if it IS NULL. This can be
    used to implement a "default value" for a preference that is visible
    in frontends as such (will be listed first if nothing is actually
    selected), but will not be written to
    voip_[usr|dom|peer]_preferences.value. Usually forcing a domain or peer
    default. Should also be named clearly (eg. __"use domain default"__).
    (Note: Therefore will also not be written to any kamailio table.)

usr_pref
dom_pref
peer_pref
    Flag if this is to be used for [usr|dom|peer] preferences.

default_val
    Flag indicating if this should be used as a default value when
    creating new entities or introducing new enum preferences (both done
    via triggers). (Note: For this to work, value must also be set.)
```

Relevant triggers:

```
enum_update
    Propagates changes of voip_preferences_enum.value to
    voip_[usr|dom|peer]_preferences.value

enum_set_default
    Will create entries for default values when adding a new enum
    preference. The default value is the tuple from voip_preferences_enum
    WHERE default_val=1 AND value NOT NULL.

trigger voip_dom_crepl_trig
trigger voip_phost_crepl_trig
trigger voip_sub_crepl_trig
    These three triggers will set possible default values (same condition
    as for enum_set_default) when creating new subscribers/domains/peers.
```

Find a usage example in a section in *db-schema/db_scripts/diff/9086.up*.