The sip:provider PRO Handbook mr4.5.3

Sipwise GmbH

<support@sipwise.com>

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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 PRO.

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 PRO?

The sip:provider PRO 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 PRO comes pre-installed on two servers. Apart from your product specific configuration, there is no initial configuration or installation to be done to get started.

1.3 The Advantages of the sip:provider PRO

Opposed to other free VoIP software, the sip:provider PRO 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.

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 PRO. 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 PRO for?

The sip:provider PRO is specifically tailored to companies who want to provide 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. The sip:provider PRO is already deployed all around the world by all kinds of VoIP operators, using it as Class5 soft-switch, as Class4 termination platform or even as Session Border Controller with all kinds of access networks, like Cable, DSL, WiFi and Mobile networks.

1.5 Getting Help

1.5.1 Phone Support

Depending on your support contract, you are eligible to contact our Support Team by phone either in business hours or around the clock. Business hours refer to the UTC+1 time zone (Europe/Vienna). Please check your support contract to find out the type of support you've purchased.

Before calling our Support Team, please also open a ticket in our Ticket System and provide as much detail as you can for us to understand the problems, fix them and investigate the cause. Please provide the number of your newly created ticket when asked by our support personnel on the phone.

You can find phone numbers, Ticket System URL, and account information in your support contract. Please make this information available to the persons in your company maintaining the sip:provider PRO.

1.5.2 Ticket System

Depending on your support contract, you can create either a limited or an unlimited amount of support tickets on our Web-based Ticket System. Please provide as much information as possible when opening a ticket, especially the following:

- **WHAT** is affected (e.g. the whole system is unreachable, or customers can't register or place calls)
- **WHO** is affected (e.g. all customers, only parts of it, and **WHICH** parts - only customers in a particular domain or customers with specific devices, etc.)
- **WHEN** did the problem occur (time frames, or after the firmware of specific devices types have been updated, etc.)

Our Support Team will ask further questions via the Ticket System along the way of troubleshooting your issue. Please provide the information as soon as possible to solve your issue promptly.

2 Architecture

The sip:provider PRO platform consists of two identical appliances working in active/standby mode. The components of a node are outlined in the following figure:

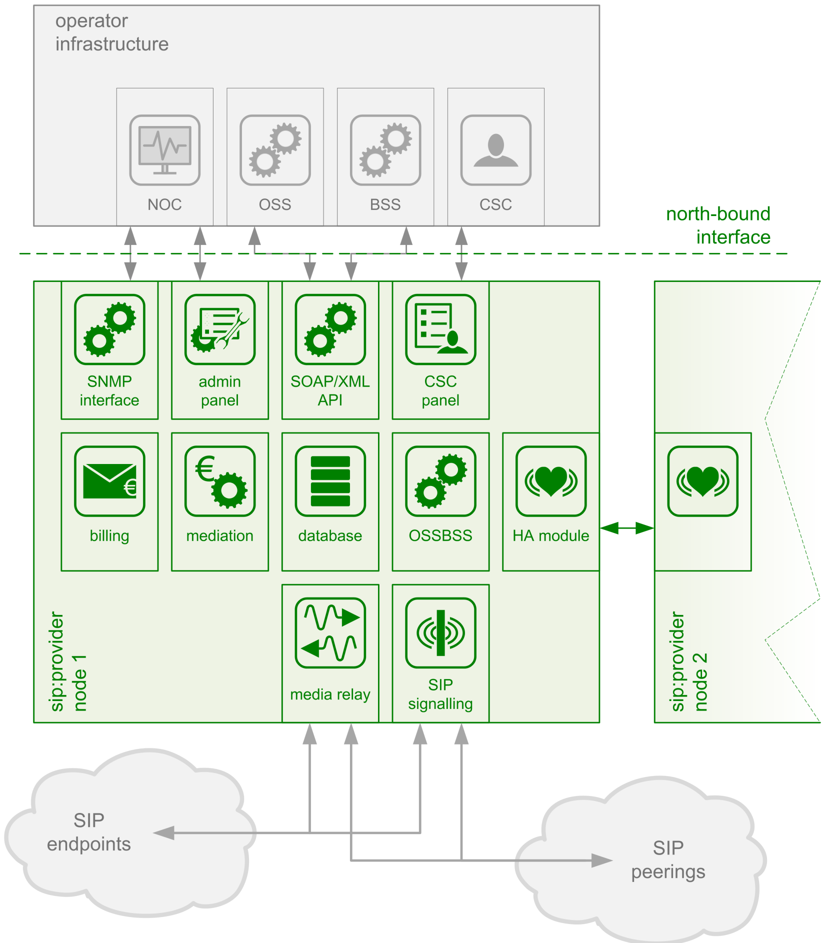


Figure 1: Architecture Overview

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

- SIP Signaling and Media Relay
- Provisioning
- Mediation and Billing
- Monitoring and Alerting
- High Availability and Fail-Over

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 PRO are shown in the following figure:

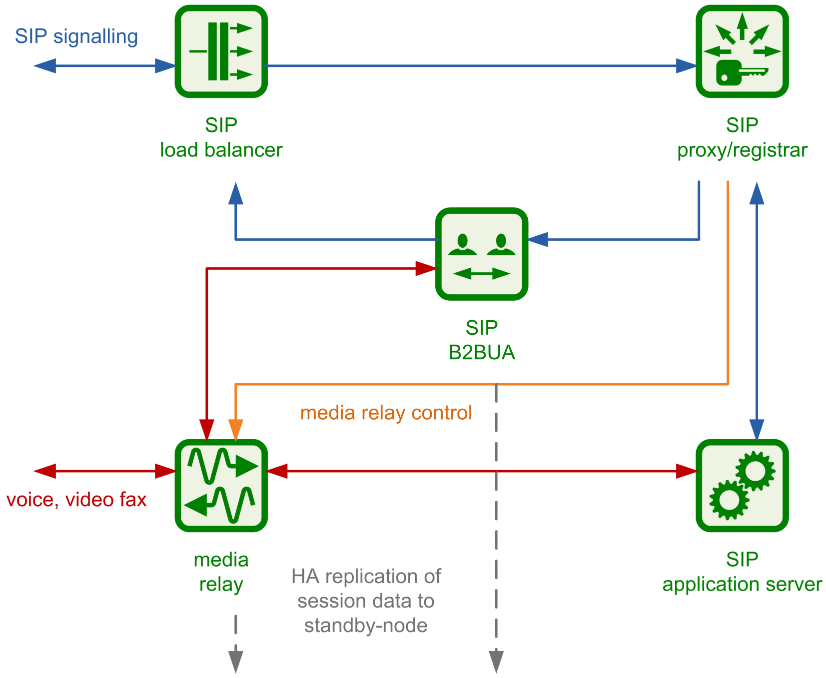


Figure 2: SIP and Media Relay Components

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 PRO, 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 when scaling out a sip:provider PRO 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 `monit start lb`, `monit stop lb` and `monit restart lb`. Its status can be queried by executing `monit summary | grep 'lb'`. 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 PRO. 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 PRO, 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 `monit start proxy`, `monit stop proxy` and `monit restart proxy`. Its status can be queried by executing `monit summary | grep 'proxy'`. 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 PRO) from the second call-leg (sip:provider PRO to the called party).

The software part used for this element is a commercial version of SEMS, with the main difference to the open-source version that it includes a replication module to share its call states with the stand-by node.

This element is typically optional in SIP systems, but it is always used for SIP calls (INVITE) that don't have the sip:provider PRO 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, Prepaid accounting 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 PRO) 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 PRO. 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 `monit start sbc`, `monit stop sbc` and `monit restart sbc`. Its status can be queried by executing `monit summary | grep 'sbc'`

2.1.4 SIP App-Server

The SIP App-Server is an Asterisk instance used for voice applications like Voicemail and Reminder Calls. It is also used in the software-based Faxserver solution to transcode SIP and RTP into the IAX protocol and vice versa, in order to talk to the Software Fax Modems. 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 PRO 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 `monit start asterisk`, `monit stop asterisk` and `monit restart asterisk`. Its status can be queried by executing `monit summary | grep 'asterisk'`

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 `monit start rtpengine`, `monit stop rtpengine` and `monit restart rtpengine`. Its status can be queried by executing `monit summary | grep 'rtpengine'`

2.2 MySQL Database

The MySQL database consists of a pair of active/standby MySQL servers. They run a MySQL master/master replication with replication integrity checks to ensure data consistency and redundancy.

The MySQL servers on both physical nodes synchronize via the row-based master/master replication. In theory, any of the two servers in the pair can be used to write data to the database, however, in practice the shared IP address is used towards clients accessing the service, hence only the active MySQL server will receive the write requests and replicate them to the standby one.

2.3 Redis Database

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

The active-standby replication ensures that the data is immediately copied from the active node to the standby one. As all sensitive call information is held in the shared storage, sip:provider PRO allows to switch the operational state from active to standby on one physical node and from standby to active on the other node without any call interruptions. Your subscribers will never notice that their calls being established on one physical server, were successfully moved to another one and successfully completed there.

2.4 High Availability and Fail-Over

2.4.1 Overview

The two servers of a complete sip:provider PRO system form a pair, a simple cluster with two nodes. Their names are fixed as `sp1` and `+sp2`, however neither of them is inherently a *first* or a *second*. They're both equal and identical and either can be the active node of the cluster at any time. Only one node is always ever active, the other one is in standby mode and doesn't perform any active functions.

High availability is achieved through constant communication between the two nodes and constant state replication from the active node to the standby one. Whenever the standby node detects that the other node has become unresponsive, has gone offline and has failed in any other way, it will proceed with taking over all resources and becoming the active node, with all operations resuming where the failed node has left off. Through that, the system will remain fully operational and service disruption will be minimal.

When the failed node comes back to life, it will become the new standby node, replicate everything that has changed in the meantime from the new active node, and then the cluster will be back in fully highly available state.

Tip

The login banner at the SSH shell provides information about whether the local system is currently the active one or the standby one. See Section 2.4.3 for other ways to differentiate between the active and the standby node.

2.4.2 Core Concepts and Configuration

The direct Ethernet crosslink between the two nodes provides the main mechanism of HA communication between them. All state replication happens over this link. Additionally, the HA daemon *heartbeat* uses this link to communicate with the other node to see if it's still alive and active. A break in this link will therefore result in a *split brain* scenario, with either node trying to become the active one. This is to be avoided at all costs.

The `config.yml` file allows specification of a list of *ping nodes* under the key `heartbeat.pingnodes`, which are used by *heartbeat* to determine if local network communications are healthy. Both servers will then constantly compare the number of locally reachable ping nodes with each other, and if the standby server is able to reach more of them, then it will become the active one.

The main resource that *heartbeat* manages is the shared service IP address. Each node has its own static IP address configured on its first Ethernet interface (`eth0`), which is done outside of the sip:provider PRO configuration framework (i.e. in the Debian-specific config file `/etc/network/interfaces`). The shared service IP is specified in `network.yml` at the key `hosts.sp1|sp2.eth0.shared_ip`. *Heartbeat* will configure it as a secondary IP address on the first Ethernet interface (`eth0:0`) on the active node and will deconfigure it on the standby node. Thus, all network communications with this IP address will always go only to the currently active node.

2.4.3 Administration

The current status of the local sip:provider PRO node can be determined using the `ngcp-check_active` shell command. This command produces no output, but returns an exit status of *0* for the active node and *1* for the standby node. A more complete shell command to produce visible output could be: `ngcp-check_active && echo active || echo standby`

To force a currently active node into standby mode, use the command `/usr/lib64/heartbeat/hb_standby`. For the opposite effect, use the command `/usr/lib64/heartbeat/hb_takeover`. This will also always affect the state of the other node, as the system automatically makes sure that always only one node is active at a time.

3 Platform Deployment

This chapter will provide the step by step instructions on how to put the sip:provider PRO into operations.

3.1 Hardware Specifications

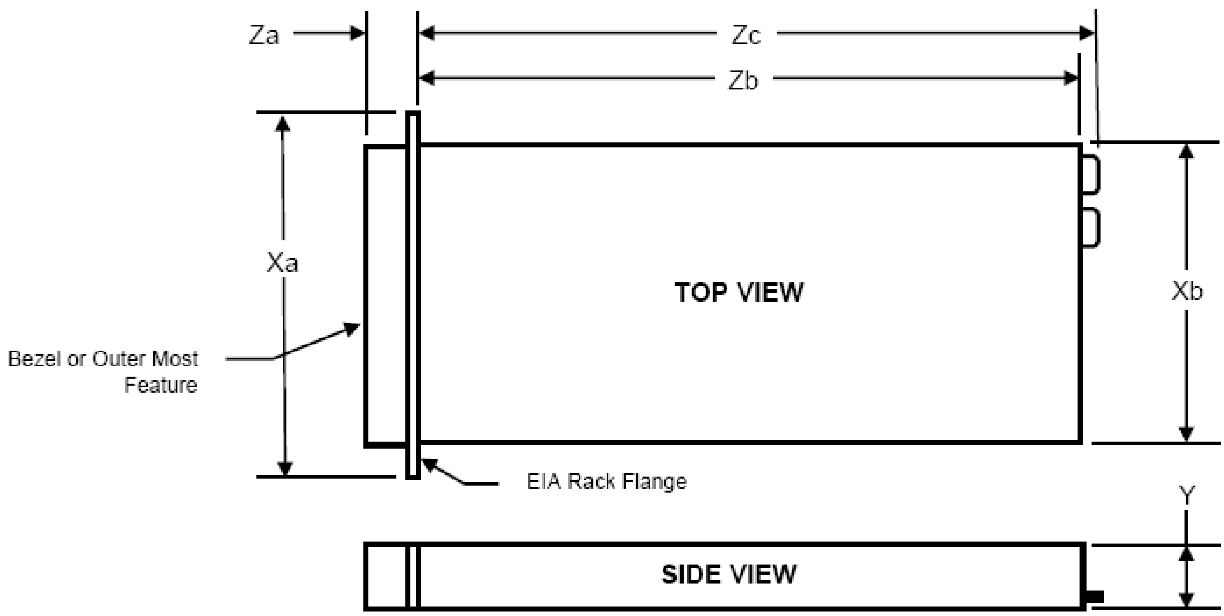
Sipwise provides the sip:provider PRO platform fully pre-installed on two Dell PowerEdge R330 servers. Their most important characteristics are:

- Up to 8 pcs. of 2.5" storage drives (HDD or SSD); shipped with 4 drives installed and configured as RAID10 array
- Gbit Ethernet ports: 2 on-board and 2 additional ports (optional)
- iDRAC module for remote maintenance

Note
Please be aware that prior to Q3 2016 Sipwise used to provide its sip:provider PRO platform on older Dell PowerEdge server models: R310 and R320.

3.1.1 Dimensions and Weight

The hardware **dimensions** are defined in the following figure:



Xa	Xb (Width)	Y (Height)	Za w/ bezel	Za w/o bezel	Zb (Depth)	Zc
482.4mm	434mm	42.8mm	35mm	21mm	610mm	639.5mm

Weight of the server with storage drives and internal components installed: 13.4kg

3.1.2 Front View

The front view of a current sip:provider PRO **Dell R330** server:

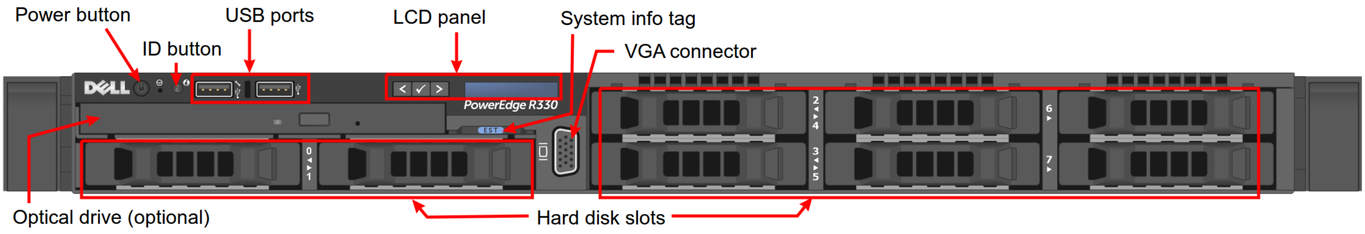


Figure 3: Dell R330 Front View

The front view of a former sip:provider PRO **Dell R310**. . . :

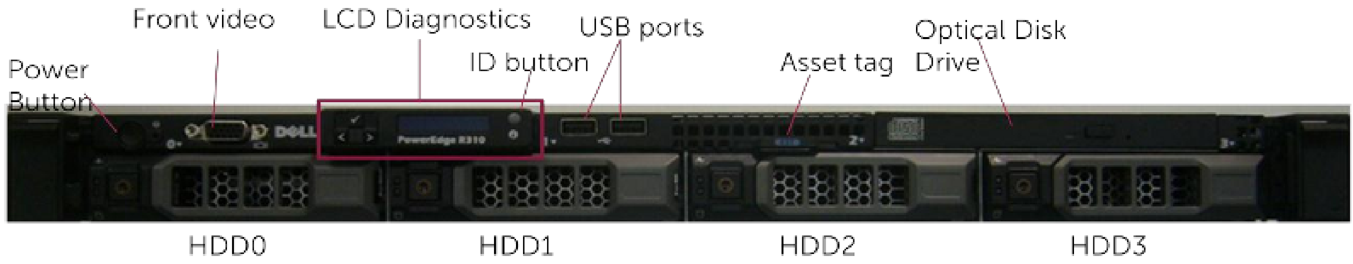


Figure 4: Dell R310 Front View

. . . and **Dell R320** server:

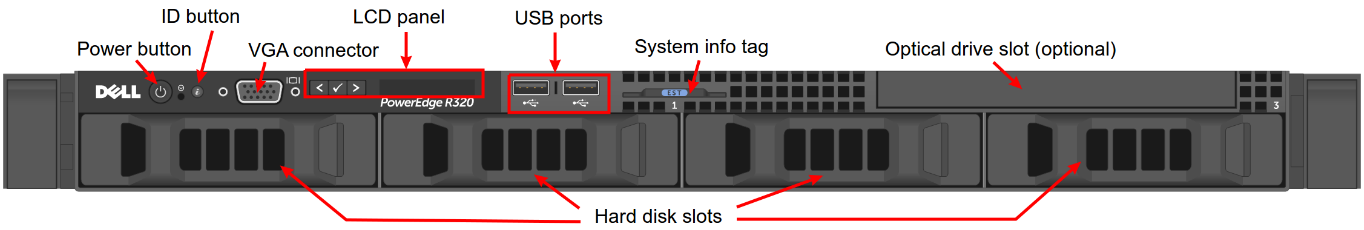


Figure 5: Dell R320 Front View

3.1.3 Rear View

The rear view of a current sip:provider PRO **Dell R330** server:

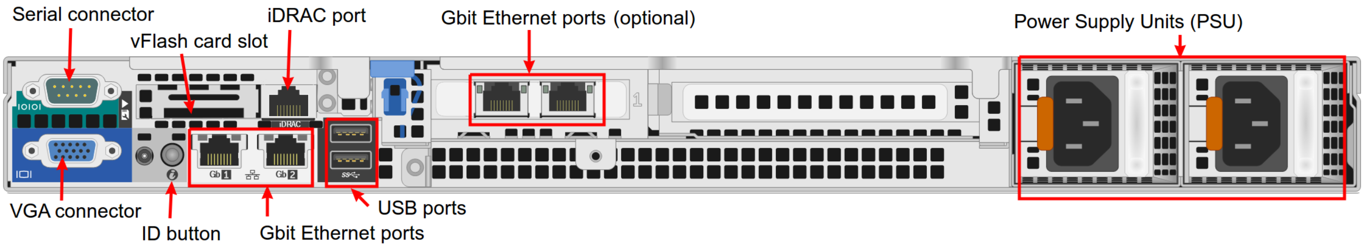


Figure 6: Dell R330 Rear View

The rear view of a former sip:provider PRO **Dell R310**. . . :

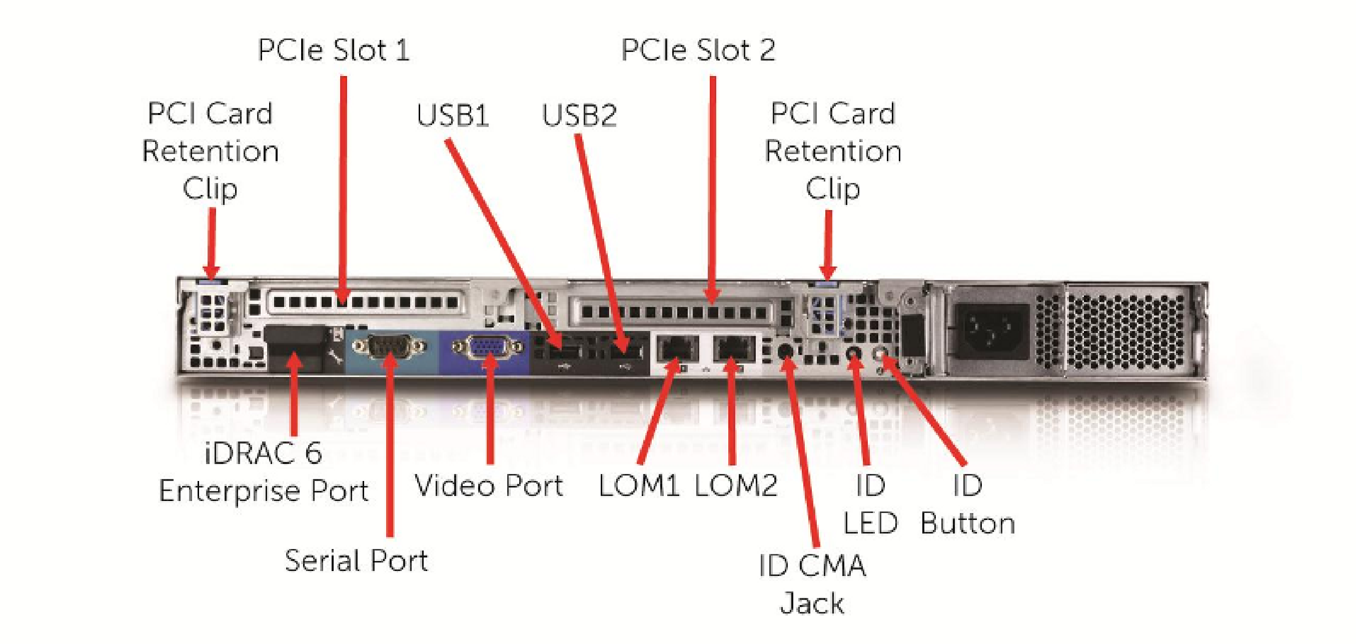


Figure 7: Dell R310 Rear View

. . . and **Dell R320** server:

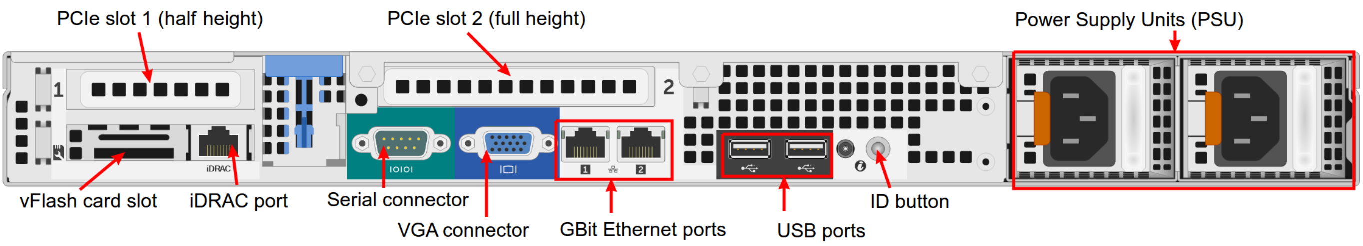


Figure 8: Dell R320 Rear View

3.1.4 Power Supply Units (PSU)

The servers are equipped with 2 redundant, hot-swappable PSUs, which are accessible from the rear side and located on the right of the chassis:

Figure 9: Redundant PSUs

The redundant PSUs include LEDs that indicate the status of the PSU:

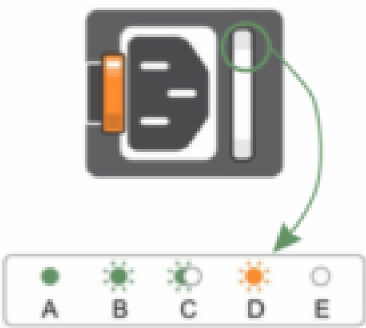


Figure 10: PSU Indicators

A. The indicator is solidly lit green: A valid power source is connected to the PSU and the PSU is operational.

B. The indicator is flashing green: The PSU firmware is being updated.

> **Caution**
> Do not disconnect the power cord or unplug the PSU when updating the firmware. If a firmware update is interrupted, the PSUs will not function. You must roll back the PSU firmware by using Dell Lifecycle Controller. For more information, see Dell Lifecycle Controller User's Guide at Dell.com/idracmanuals.

C. The indicator is flashing green and turns off: When hot-adding a PSU, the PSU handle flashes green five times at 4 Hz rate and turns off. This indicates that there is a PSU mismatch with respect to efficiency, feature set, health status, and supported voltage. Ensure that both the PSUs are the same.

D. The indicator is flashing amber: Indicates a problem with the PSU.

> **Caution**
> When correcting a PSU mismatch, replace only the PSU with the flashing indicator. Swapping the other PSU to make a matched pair can result in an error condition and unexpected system shutdown. To change from a High Output configuration to a Low Output configuration or vice versa, you must turn off the system.

> **Caution**
> AC PSUs support both 220 V and 110 V input voltages with the exception of Titanium PSUs, which support only 220 V. When two identical PSUs receive different input voltages, they can output different wattages, and trigger a mismatch.

> **Caution**
> If two PSUs are used, they must be of the same type and have the same maximum output power.

> **Caution**
> Combining AC and DC PSUs is not supported and triggers a mismatch.

E. The indicator is not lit: Power is not connected.

3.2 Installation Prerequisites

In order to put the sip:provider PRO into operations, you need to rack-mount it into 19" racks.

You will find the following equipment in the box:

- 2 servers

- 2 pairs of rails to rack-mount the servers
- 2 cable management arms

You will additionally need the following parts as they are not part of the distribution:

- 4 power cables

> **Note**
> The exact type required depends on the location of installation, e.g. there are various forms of power outlets in different countries.

- At least 2 CAT5 cables to connect the servers to the access switches for external communication
- 1 CAT5 cable to directly connect the two servers for internal communication

3.3 Rack-Mount Installation

Install the two servers into the rack (either into a single one or into two geographically distributed ones).

The rails shipped with the servers fit into standard 4-Post 19" racks. If they do not fit, please consult your rack vendor to get proper rails.

The following figure shows the mounted rails:

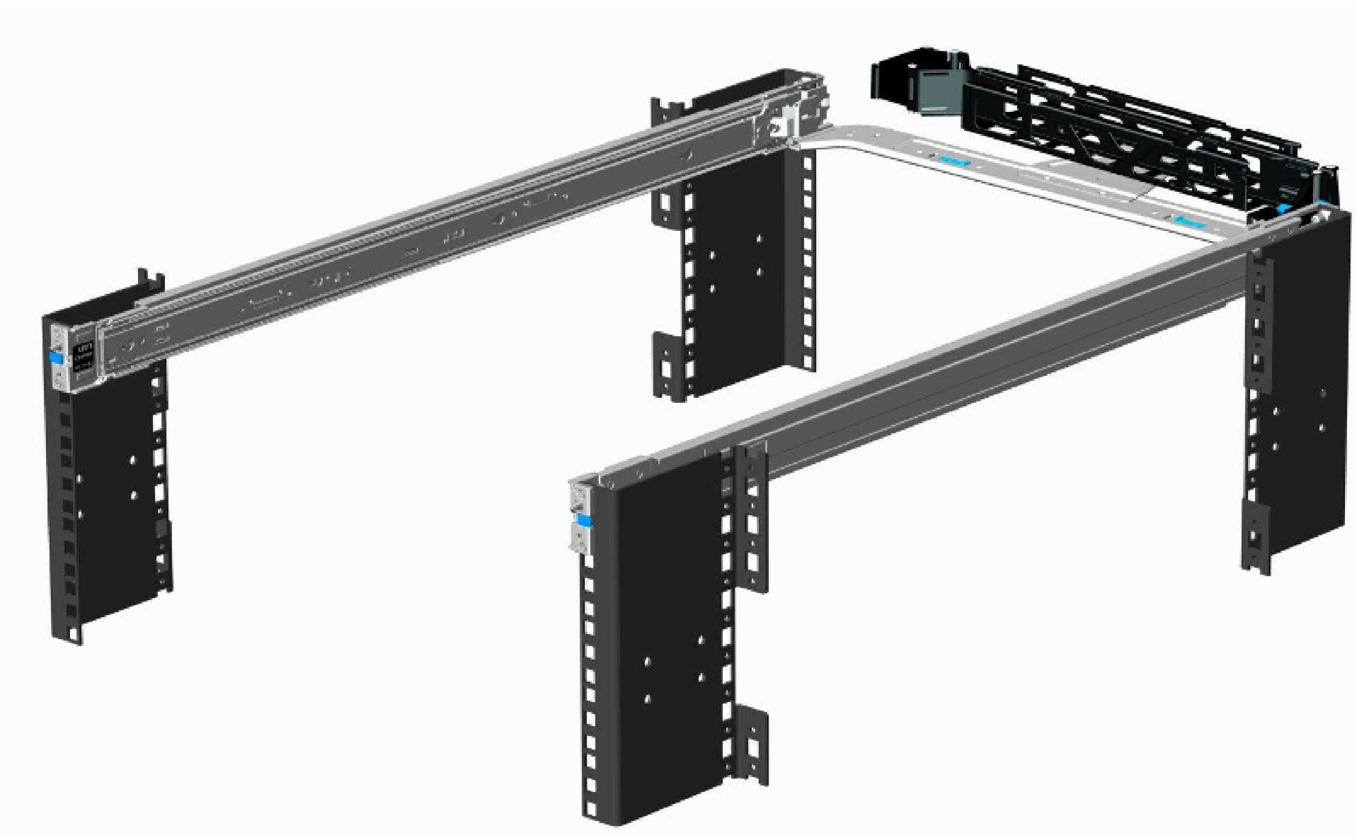


Figure 11: Rack-mounted Rails

3.4 Power Supply Cabling

Each server has two redundant Power Supply Units (PSU). Connect one PSU to your normal power circuit and the other one to an Uninterruptible Power Supply Unit (UPS) to gain the maximum protection against power failures.

The cabling should look like in the following picture to prevent accidental power cuts:

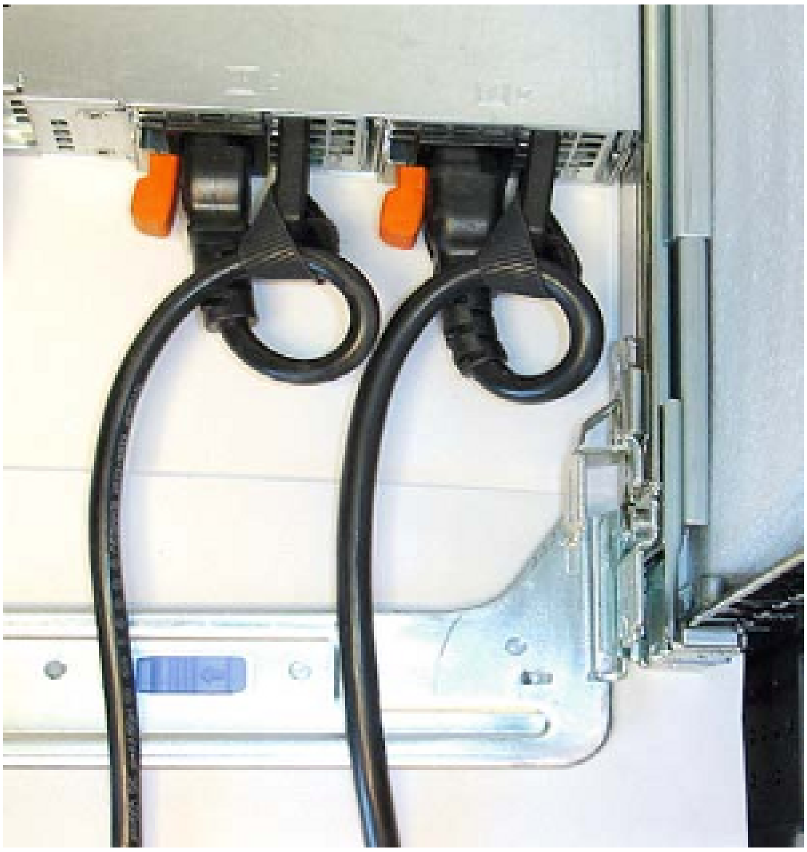


Figure 12: Proper PSU Cabling

3.5 Network Cabling

Internal Communication
The *high availability (HA)* feature of sip:provider PRO requires that a direct Ethernet connection between the servers is established. One of the network interfaces must be dedicated to this functionality.

External Communication
Remaining network interfaces may be used to make the servers publicly available for communication services (SIP, messaging, etc.) and also for their management and maintenance.

3.5.1 Internal Communication

Patch a cross-link with a straight CAT5 cable between the two servers by connecting the cable to the network interface assigned to the HA component by Sipwise. The direct cross cable is applied for maximum availability because this connection is used by the servers to communicate with each other internally.

Important

We strongly suggest against using a switch in between the servers for this internal interface. Using a switch is acceptable only if there is no another way to connect the two ports (e.g. if you configure a geographically distributed installation).

Note

In case you are using a switch for cross-link make sure to enable *portfast* mode on Cisco switches. The thing is that STP puts the port into learning mode for 90 seconds, after it comes up for the first time. During this learning phase, the link is technically up, but no traffic passes through, so `heartbeat` service will detect the other node as dead during boot. The *portfast* mode tells the switch to skip the learning phase and go to forwarding state right away: `spanning-tree portfast [trunk]`.

3.5.2 External Communication

For both servers, depending on the network configuration, connect one or more straight CAT5 cables to the ports on the servers network cards and plug them into the corresponding switch ports. Information about proper ports of the servers to be used for this purpose are provided by Sipwise.

4 VoIP Service Configuration Scenario

To be able to configure your first test clients, you will need a Customer, a SIP domain and some subscribers in this domain. Throughout this steps, let's assume you're running the NGCP on the IP address *1.2.3.4*, and you want this IP to be used as SIP domain. This means that your subscribers will have an URI like *user1@1.2.3.4*.

Tip

You can of course set up a DNS name for your IP address (e.g. letting sip.yourdomain.com point to 1.2.3.4) and use this DNS name throughout the next steps, but we'll keep it simple and stick directly with the IP as a SIP domain for now.

Warning

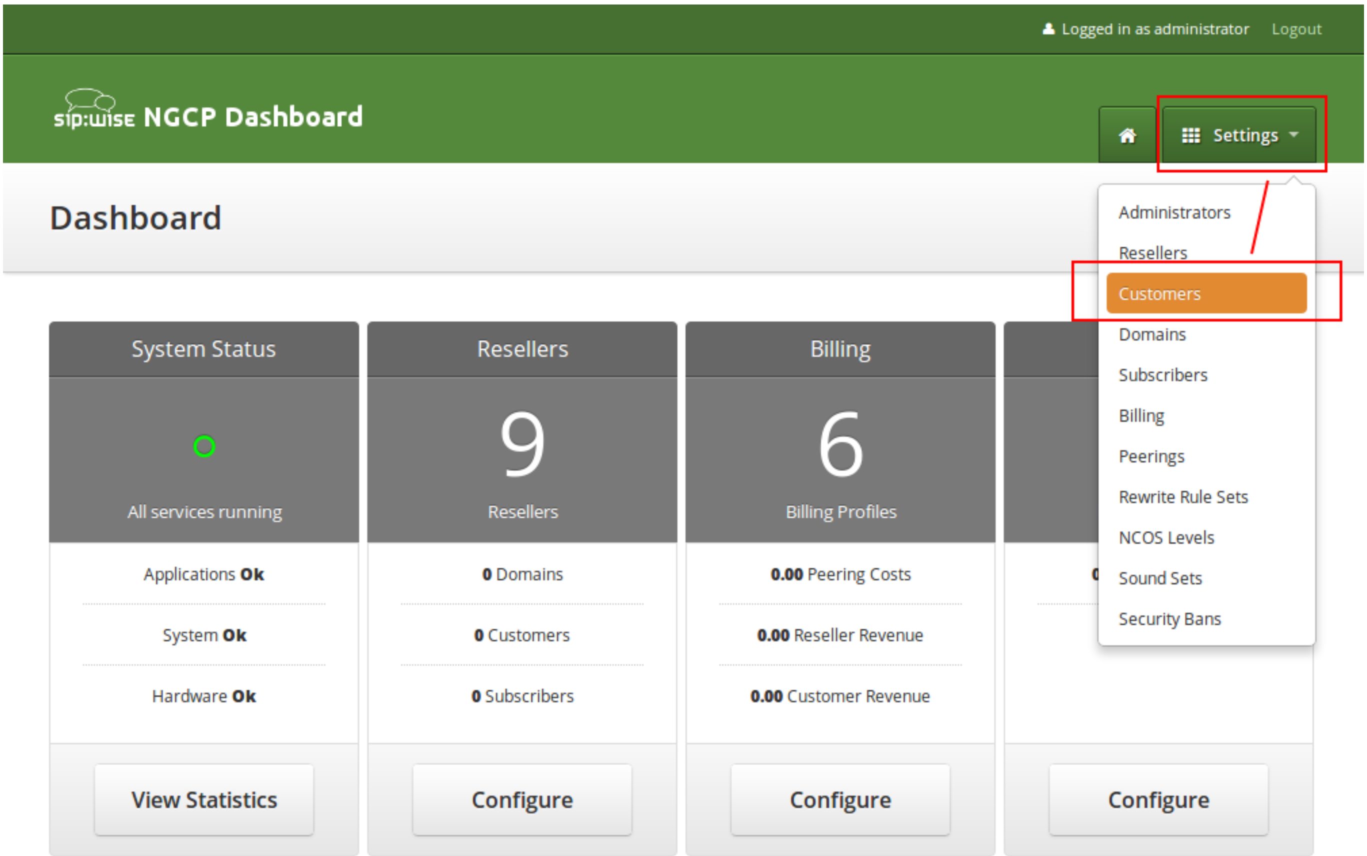
Once you started adding subscribers to a SIP domain, and later decide to change the domain, e.g. from 1.2.3.4 to sip.yourdomain.com, you'll need to recreate all your subscribers in this new domain. It's currently not possible to easily change the domain part of a subscriber.

Go to the *Administrative Web Panel* (*Admin Panel*) running on *https://<ip>:1443/login/admin* and follow the steps below. The default user on the system is *administrator* with the password *administrator*, if you haven't changed it already.

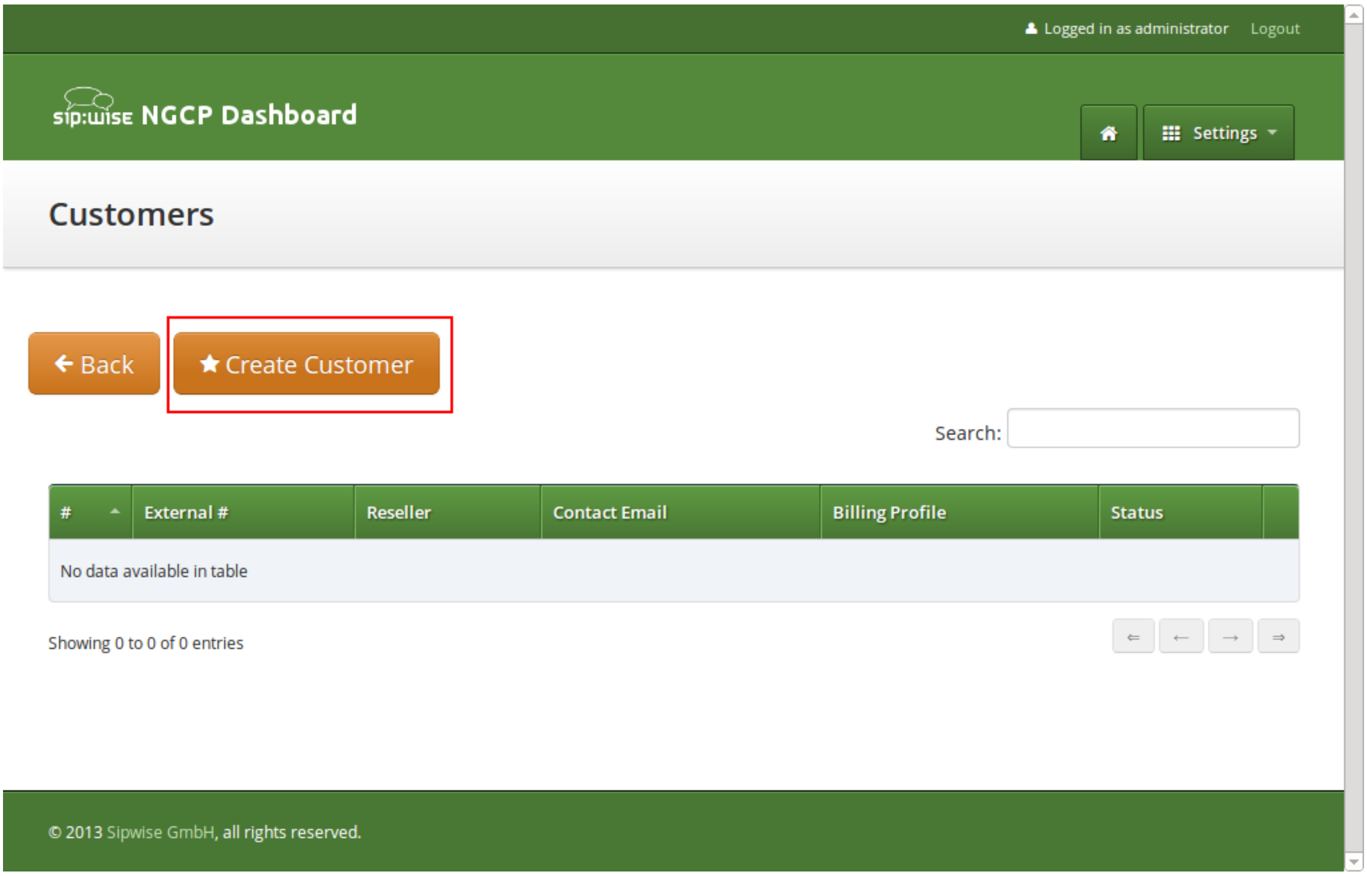
4.1 Creating a Customer

A Customer is a special type of contract on the system acting as billing container for SIP subscribers. You can create as many SIP subscribers within a Customer as you want.

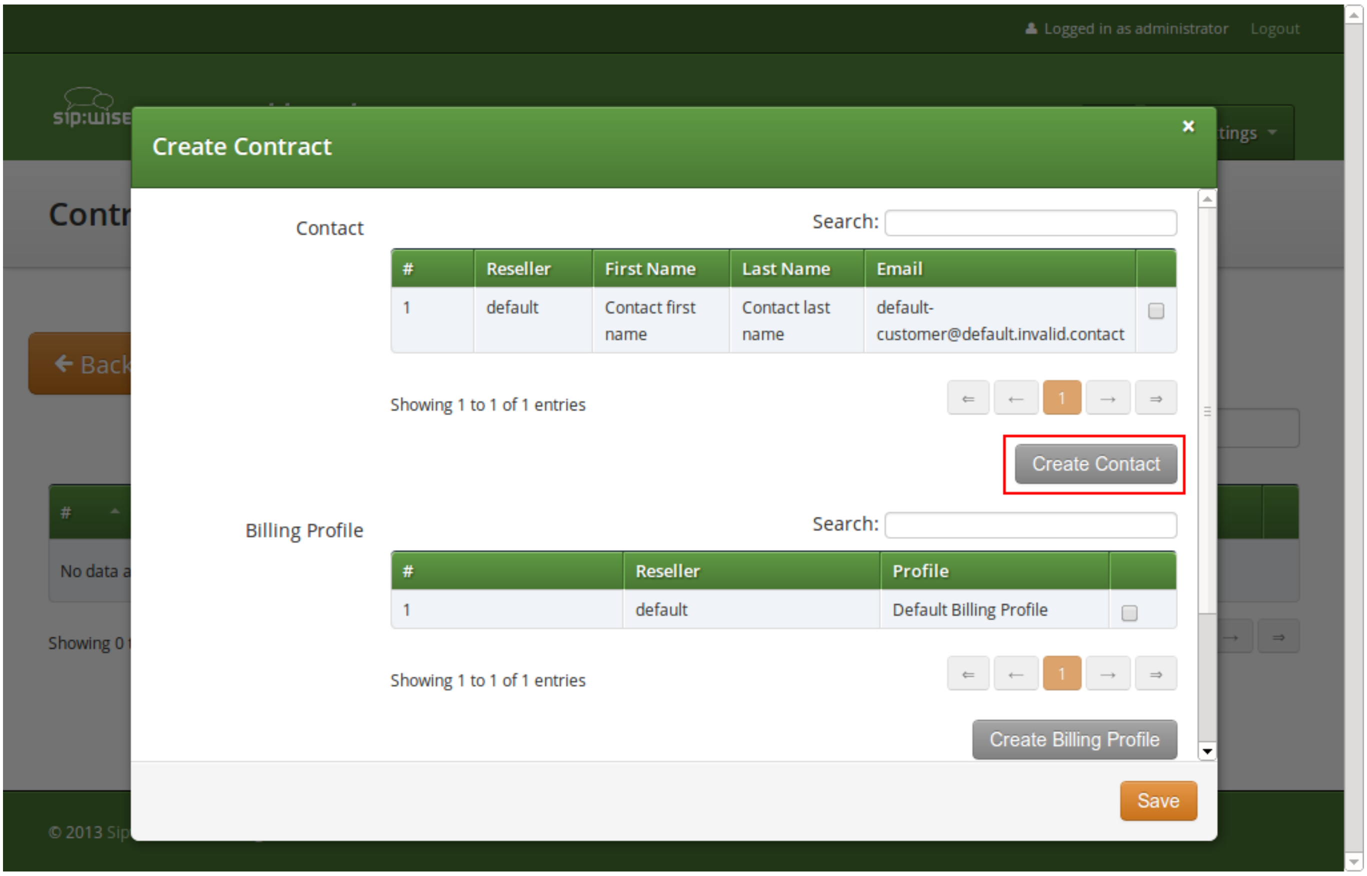
To create a Customer, got to *Settings→Customers*.



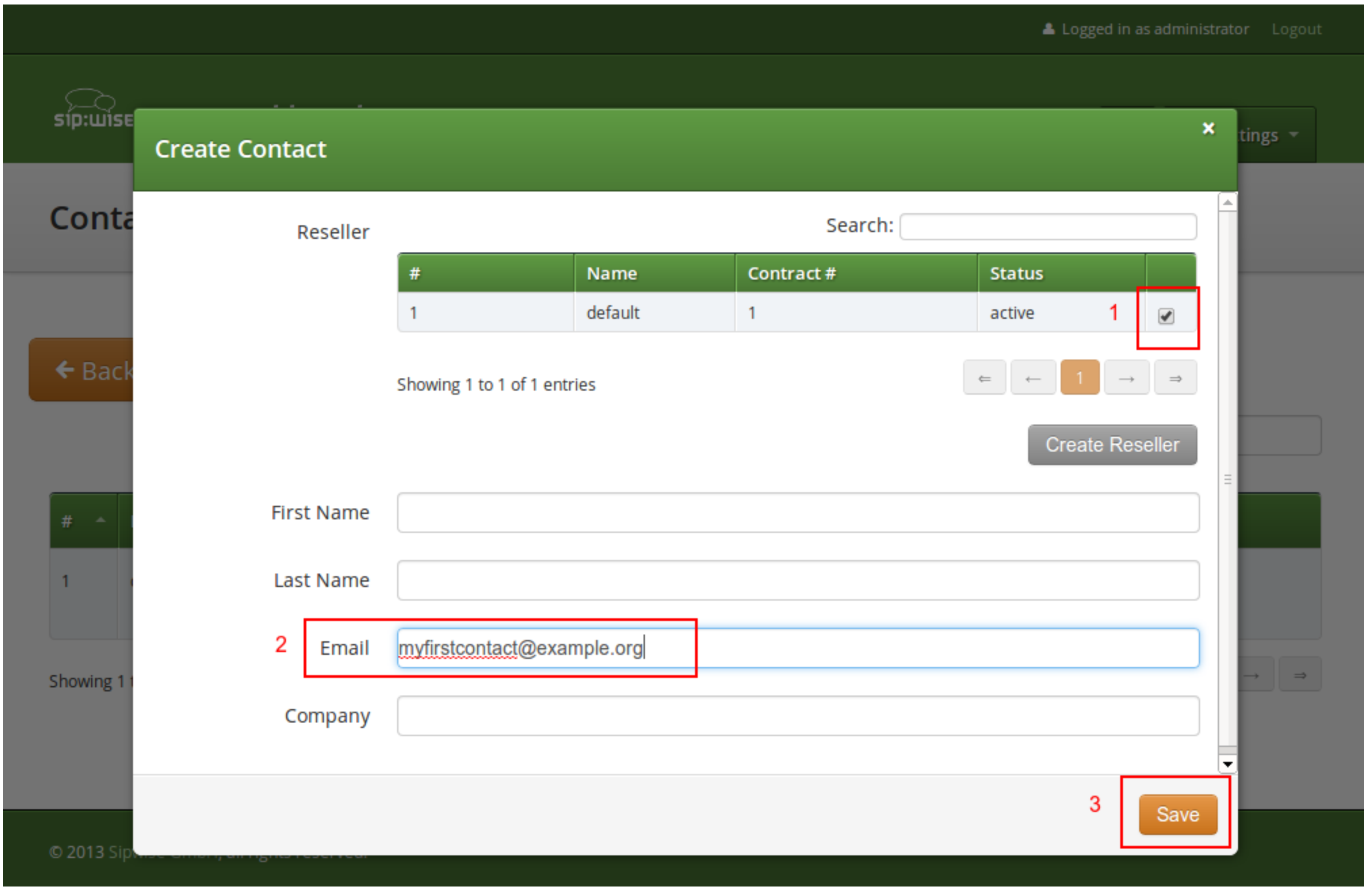
Click on *Create Customer*.



Each *Customer* needs a *Contact*. We can either reuse the default one, but for a clean setup, we create a new *Contact* for each *Customer* to be able to identify the *Customer*. Click on *Create Contact* to create a new *Contact*.

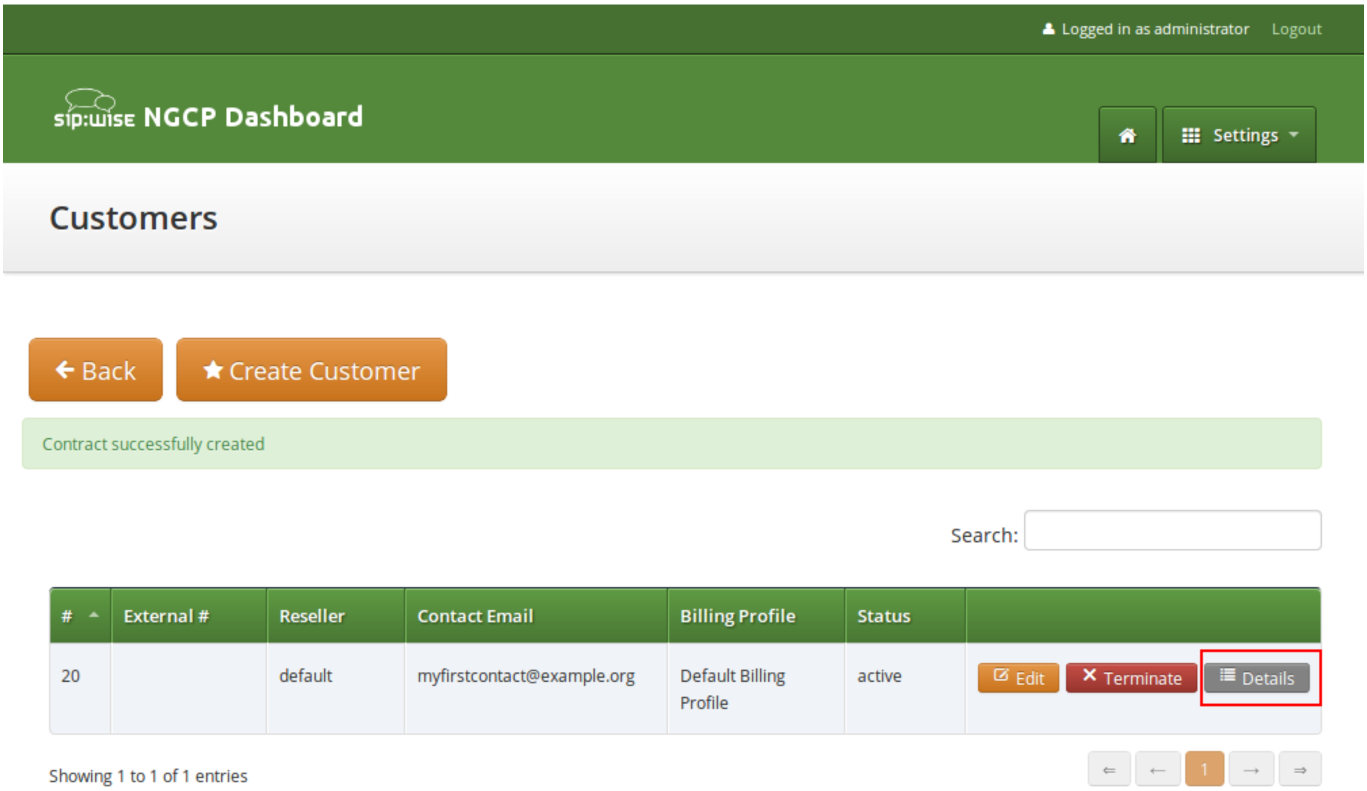


We assign the Contact to the default *Reseller*. You can create a new one if you want, but for a simple setup the default *Reseller* is sufficient. Select the *Reseller* and enter the contact details (at least an *Email* is required), then press *Save*.



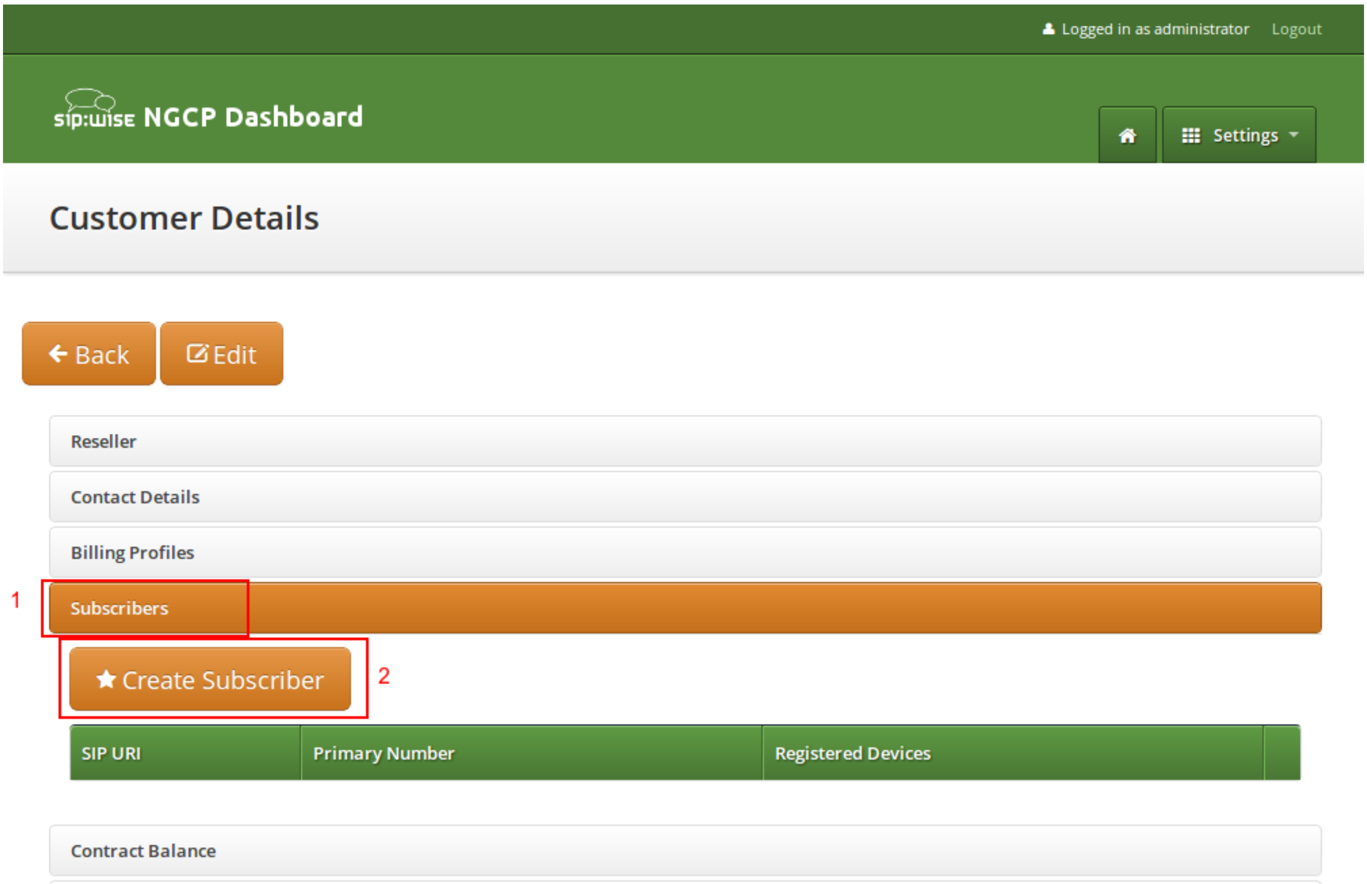
You will be redirected back to the *Customer* form. The newly created *Contact* is selected by default now, so you only have to select a *Billing Profile*. Again you can create a new one on the fly, but we will go with the default profile for now. Select it and press *Save*.

You will now see your first *Customer* in the list. Hover over the customer and click *Details* to view the details.

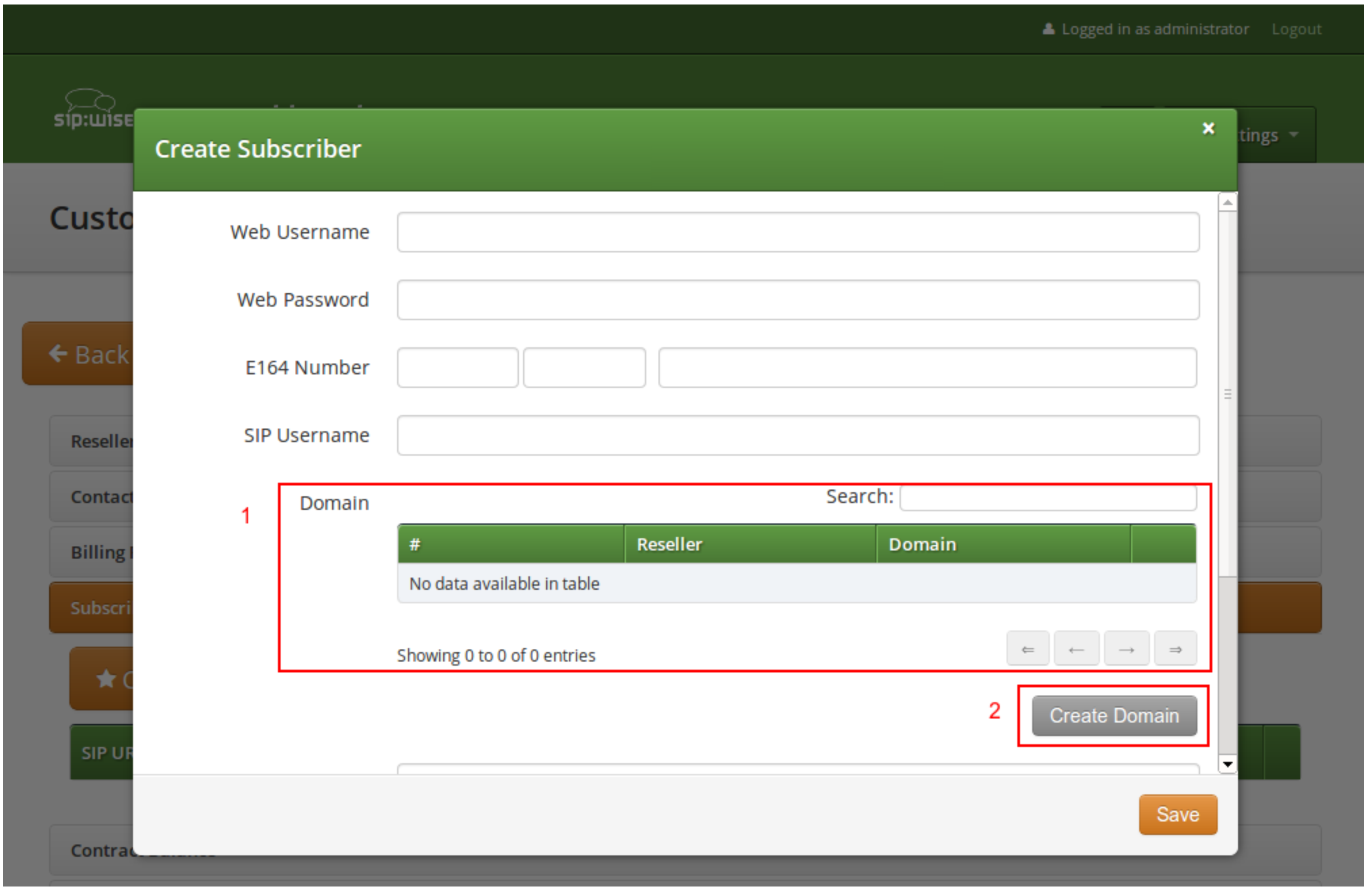


4.2 Creating a Subscriber

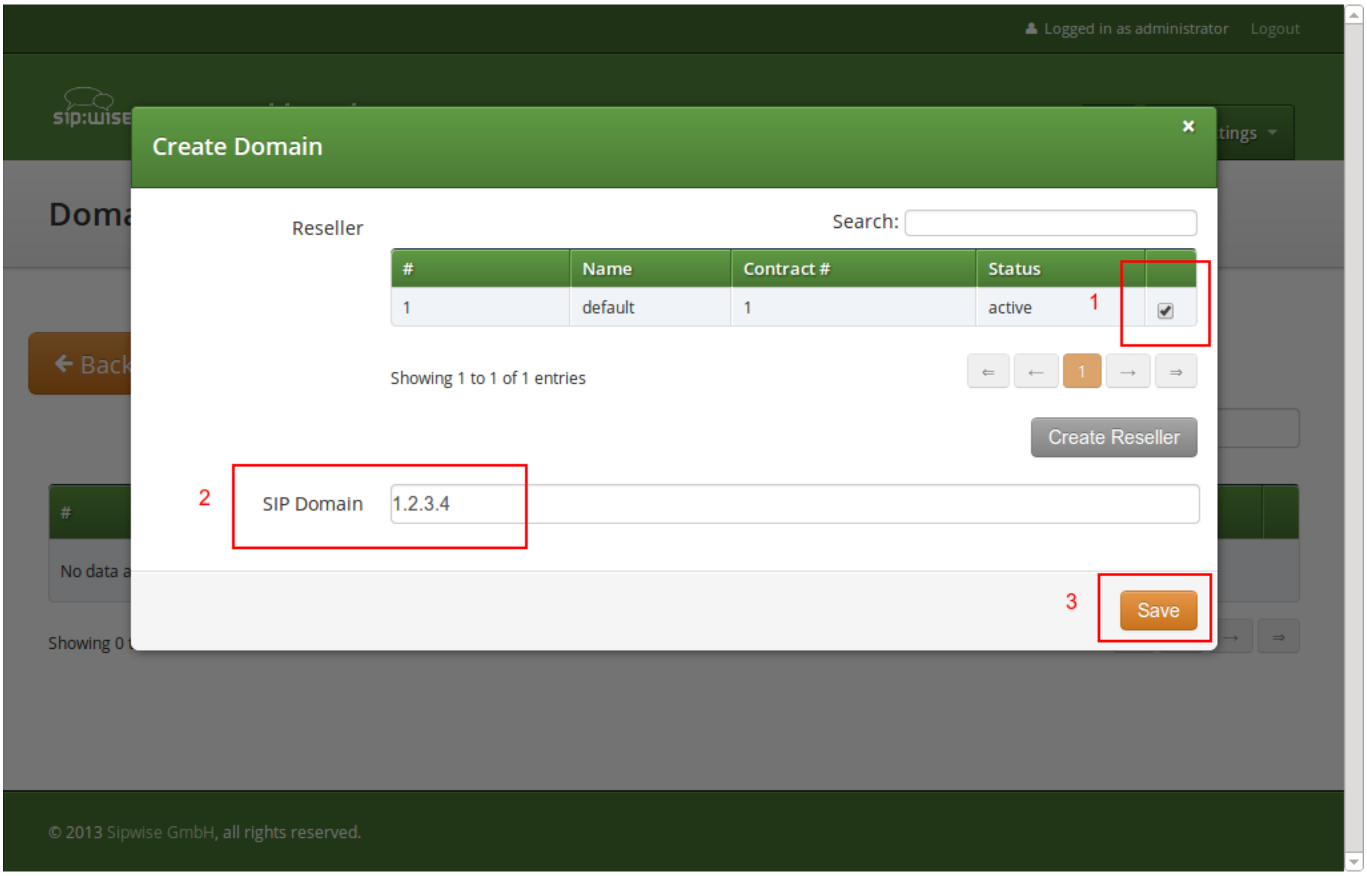
In your *Customer* details view, click on the *Subscribers* row, then click the *Create Subscriber*.



As you can see, we don't have any *SIP Domains* yet, so click on *Create Domain* to create one.



Select the *Reseller* (make sure to use the same reseller where your *Customer* is created in) and enter your domain name, then press *Save*.



Your *Domain* will be preselected now, so fill out the rest of the form:

- **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.
- **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 a made-up 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 phantasy number *+43 99 1001*.

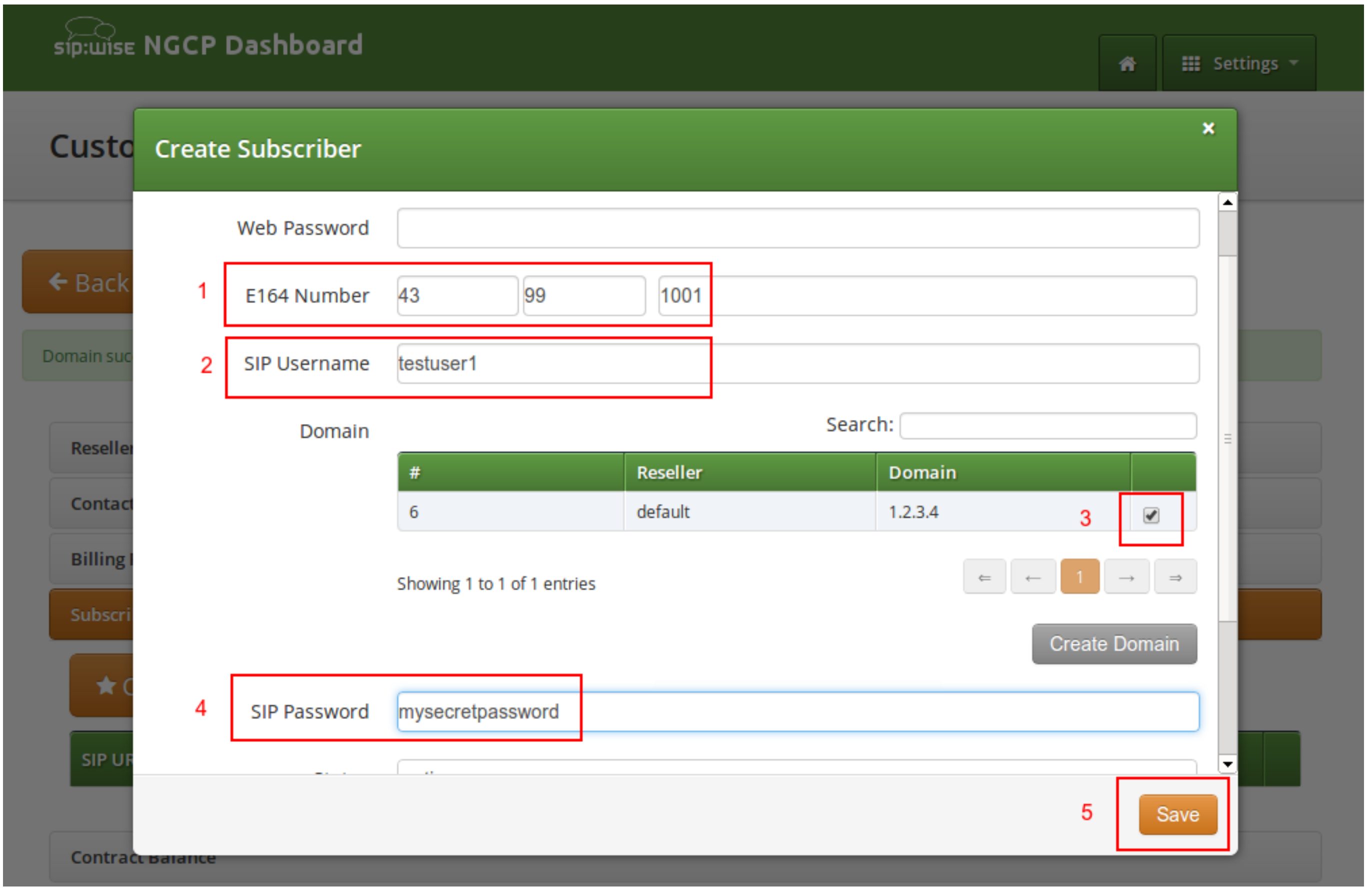
Tip

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 4.6.

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).

- **SIP Username**: The user part of the SIP URI for your subscriber.
- **SIP Domain**: The domain 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 *active*.
- **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*.



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.

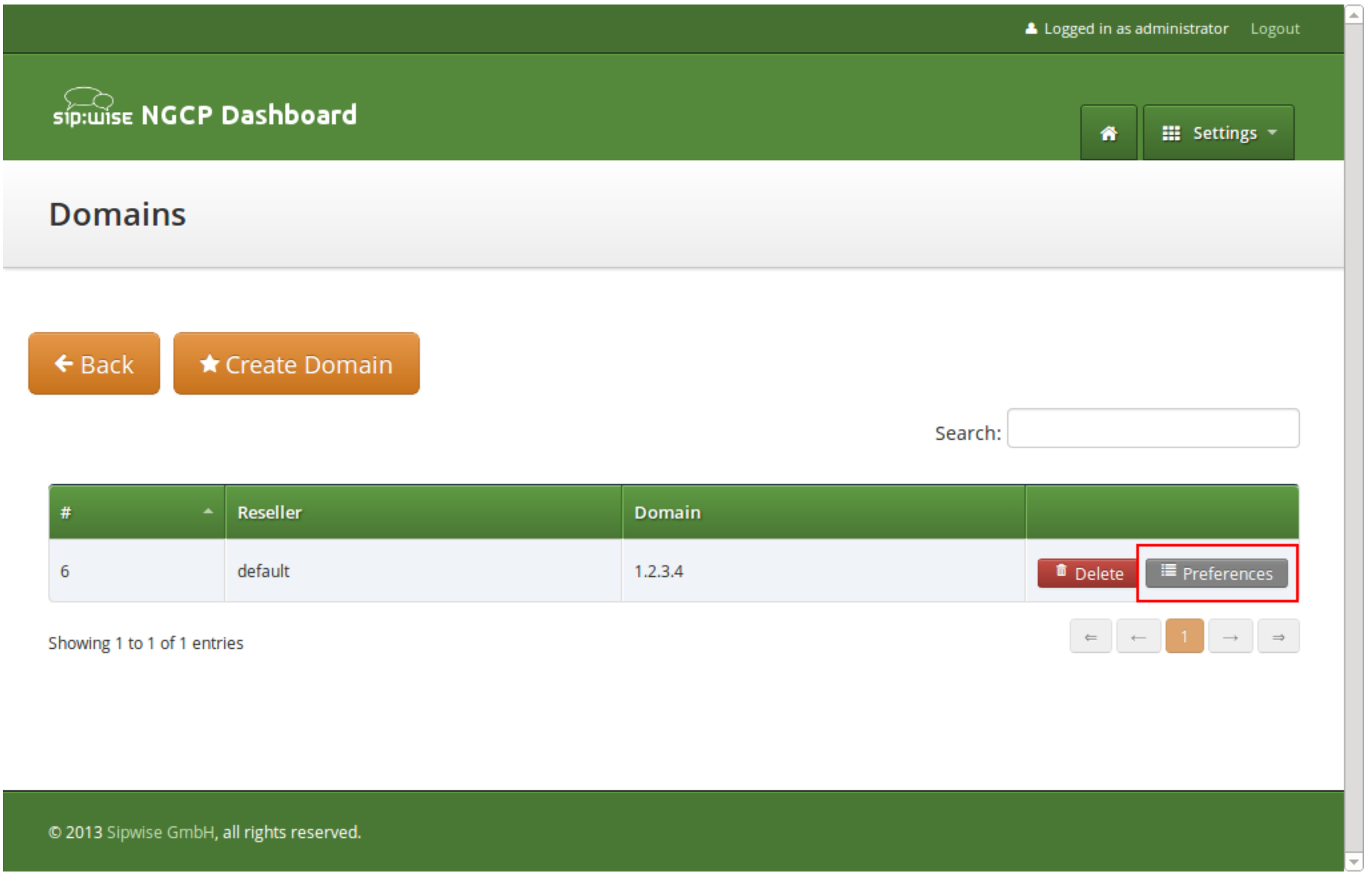
Tip

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.

4.3 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.



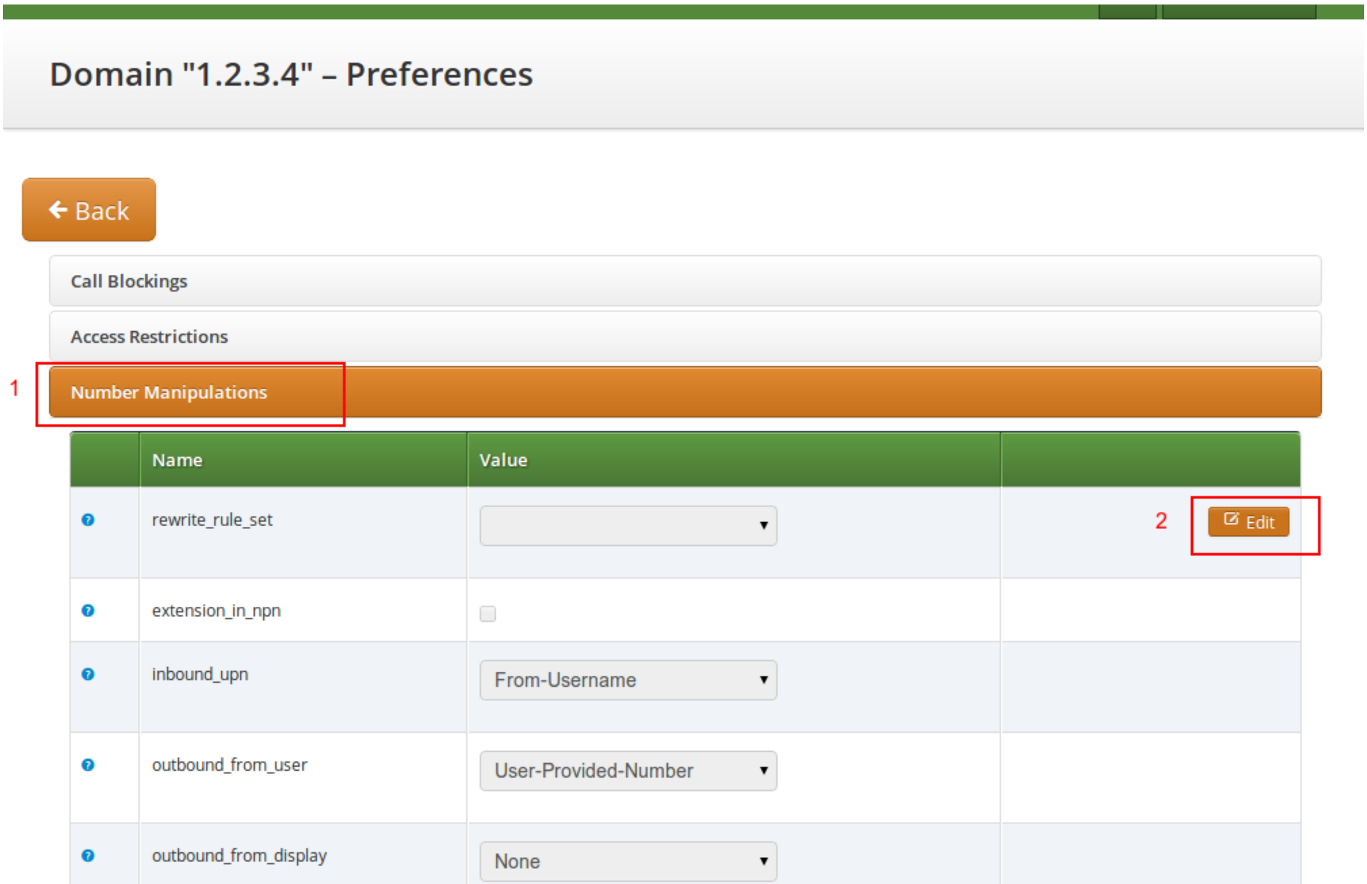
The most important settings are in the *Number Manipulations* group.

Here you can configure the following:

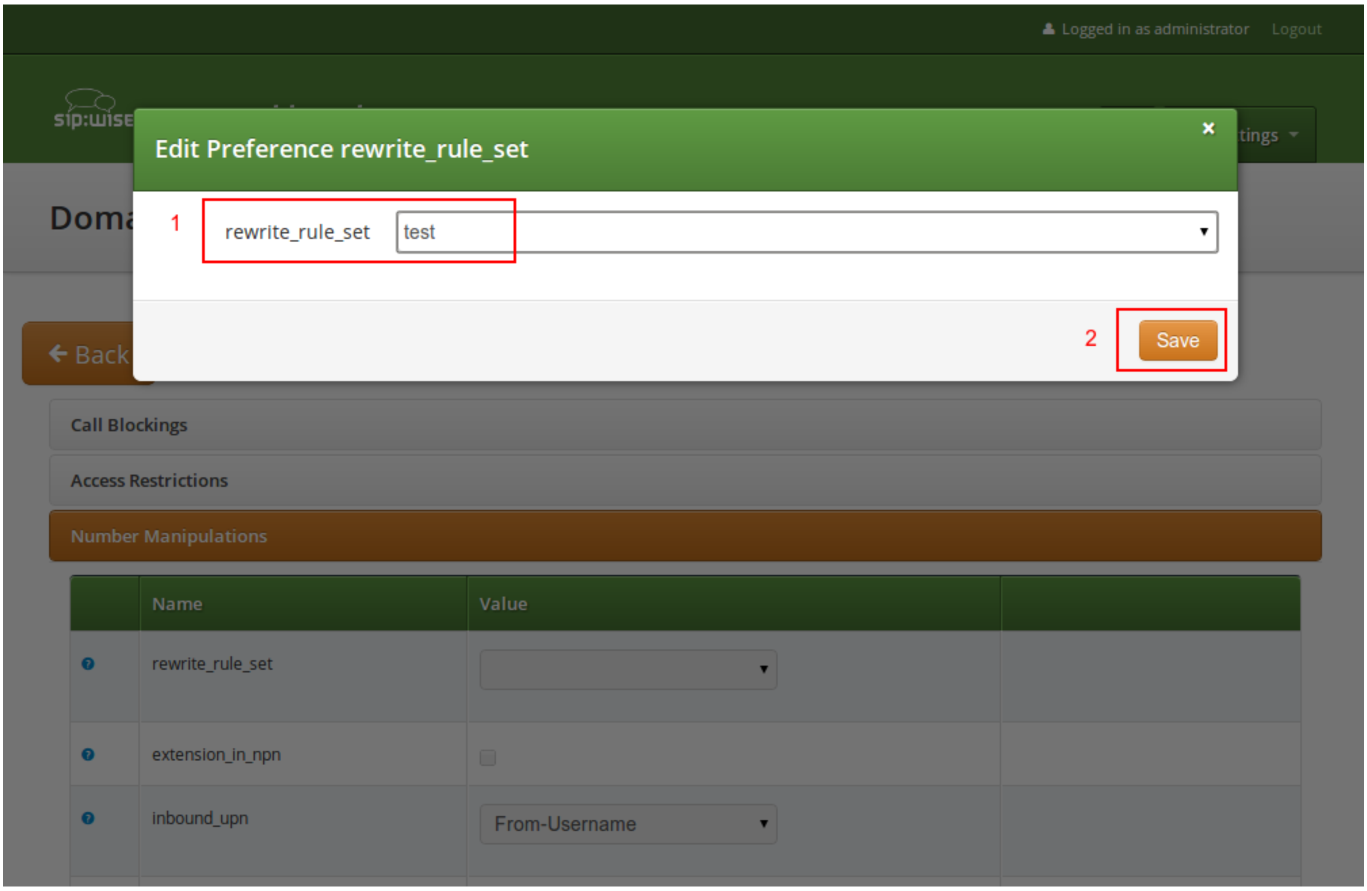
- for incoming calls - which SIP message headers to take numbers from
- for outgoing calls - where in the SIP messages to put certain numbers to

- 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 4.6, 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.

4.4 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 *oss-bss→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 and aliases if *oss-bss→provisioning→auto_sync_cli* is set to **yes** in */etc/ngcp-config/config.yml*.

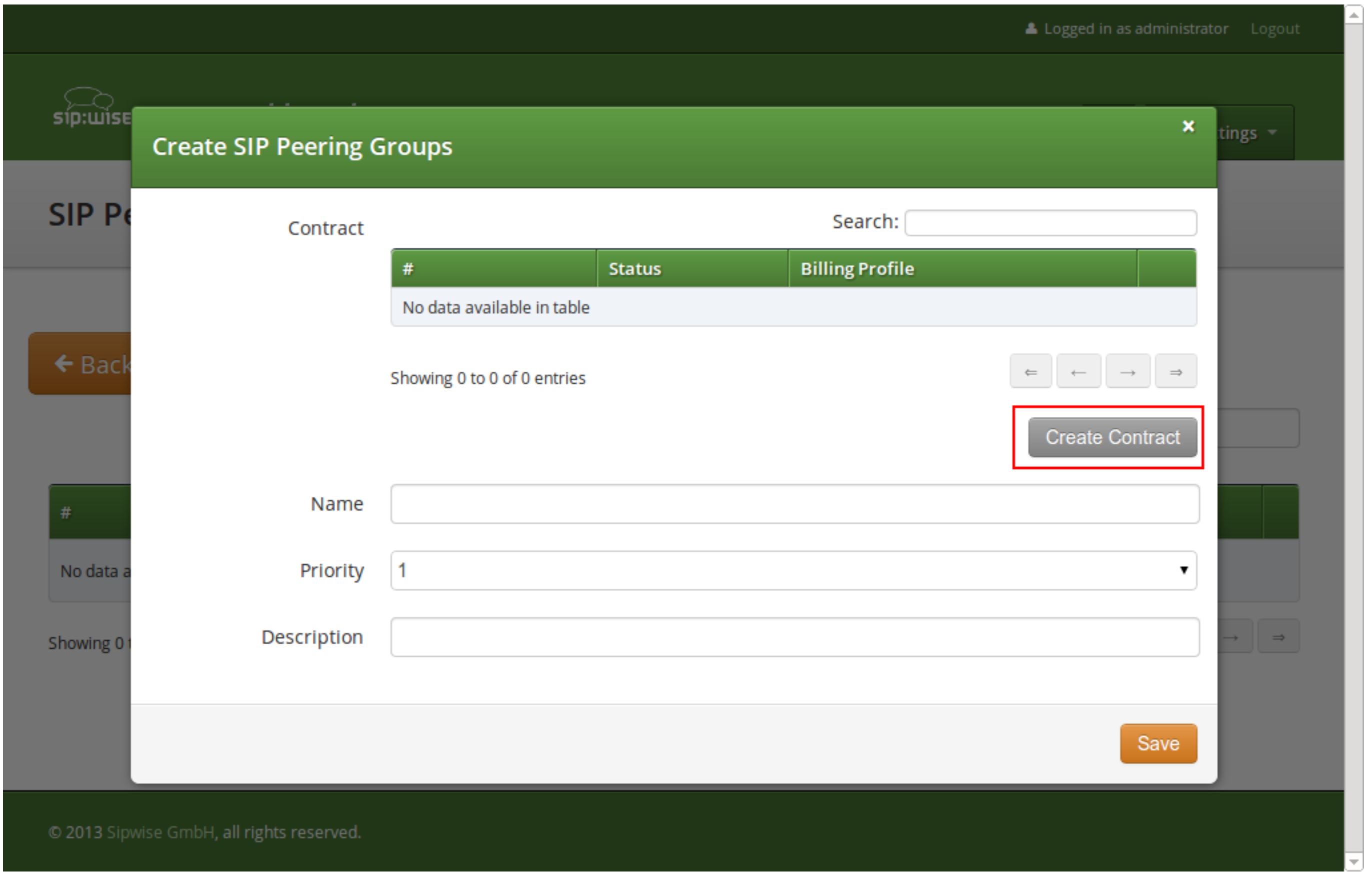
4.5 Creating Peerings

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.

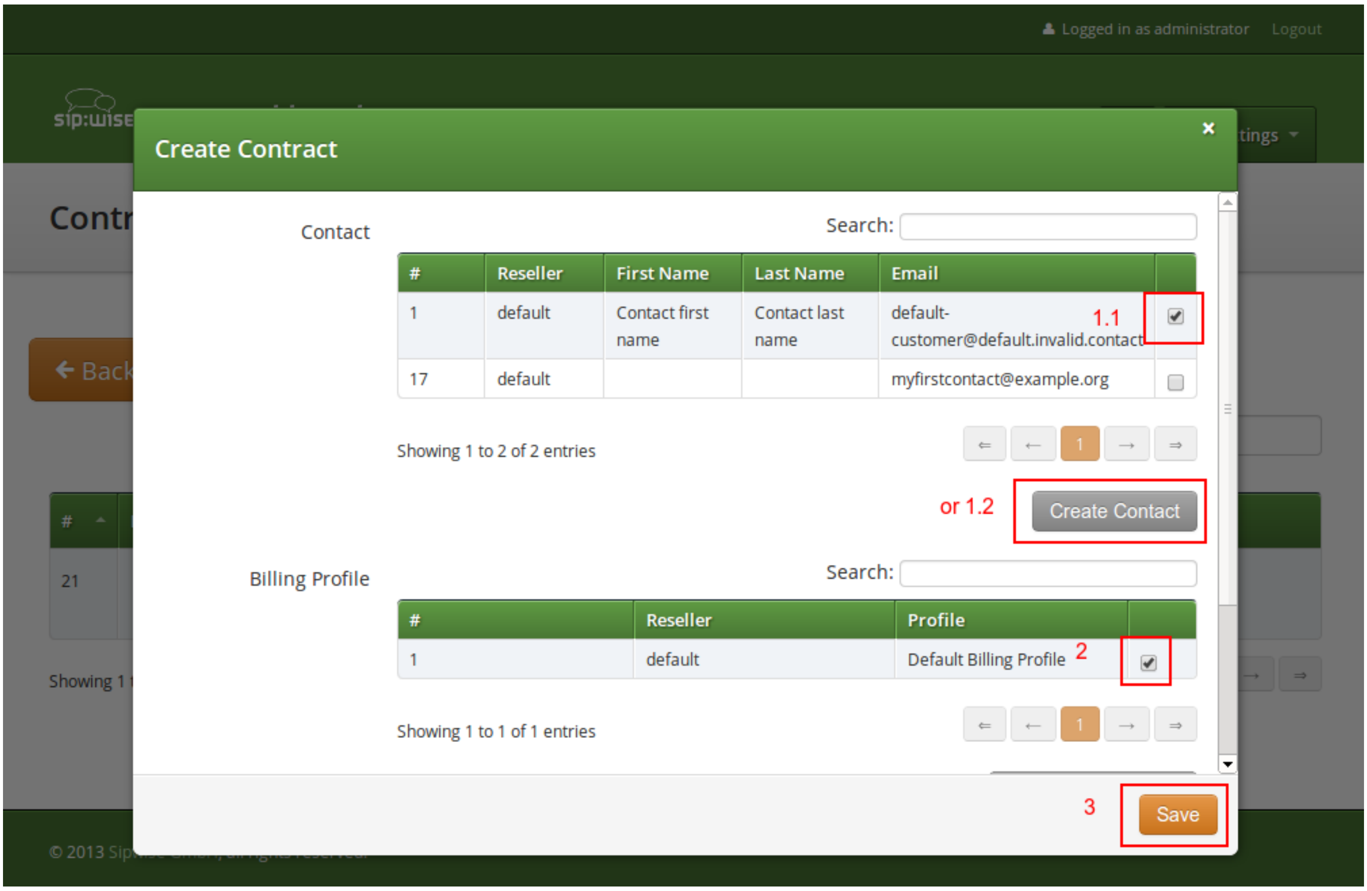
4.5.1 Creating Peering Groups

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.

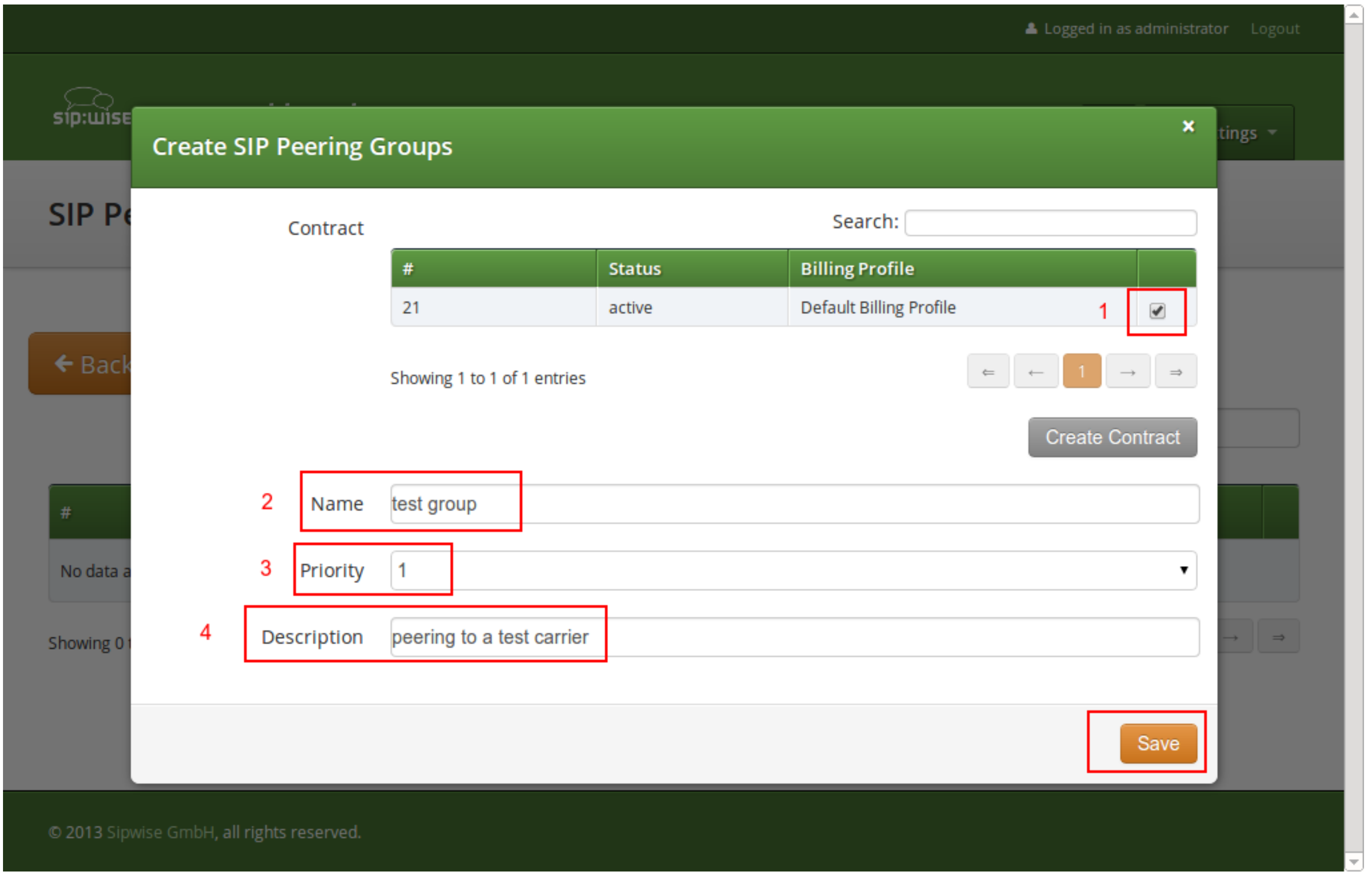


Click on *Create Contract* create a *Contact*, then select a *Billing Profile*.



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`



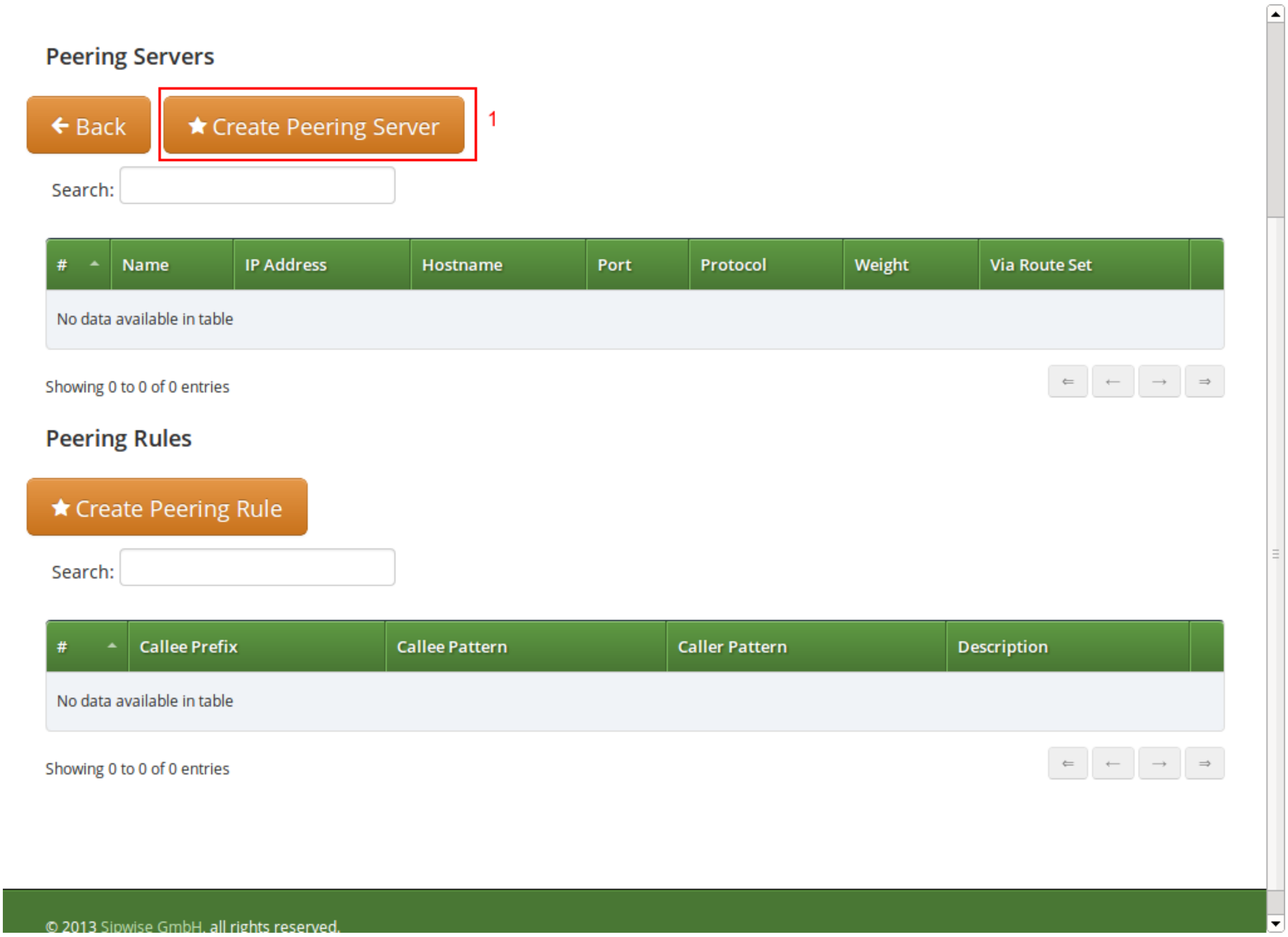
The *Priority* option defines which *Peering Group* to favor if two peering groups have peering rules matching an outbound call. *Peering Rules* are described below.

Then click *Save* to create the group.

4.5.2 Creating Peering Servers

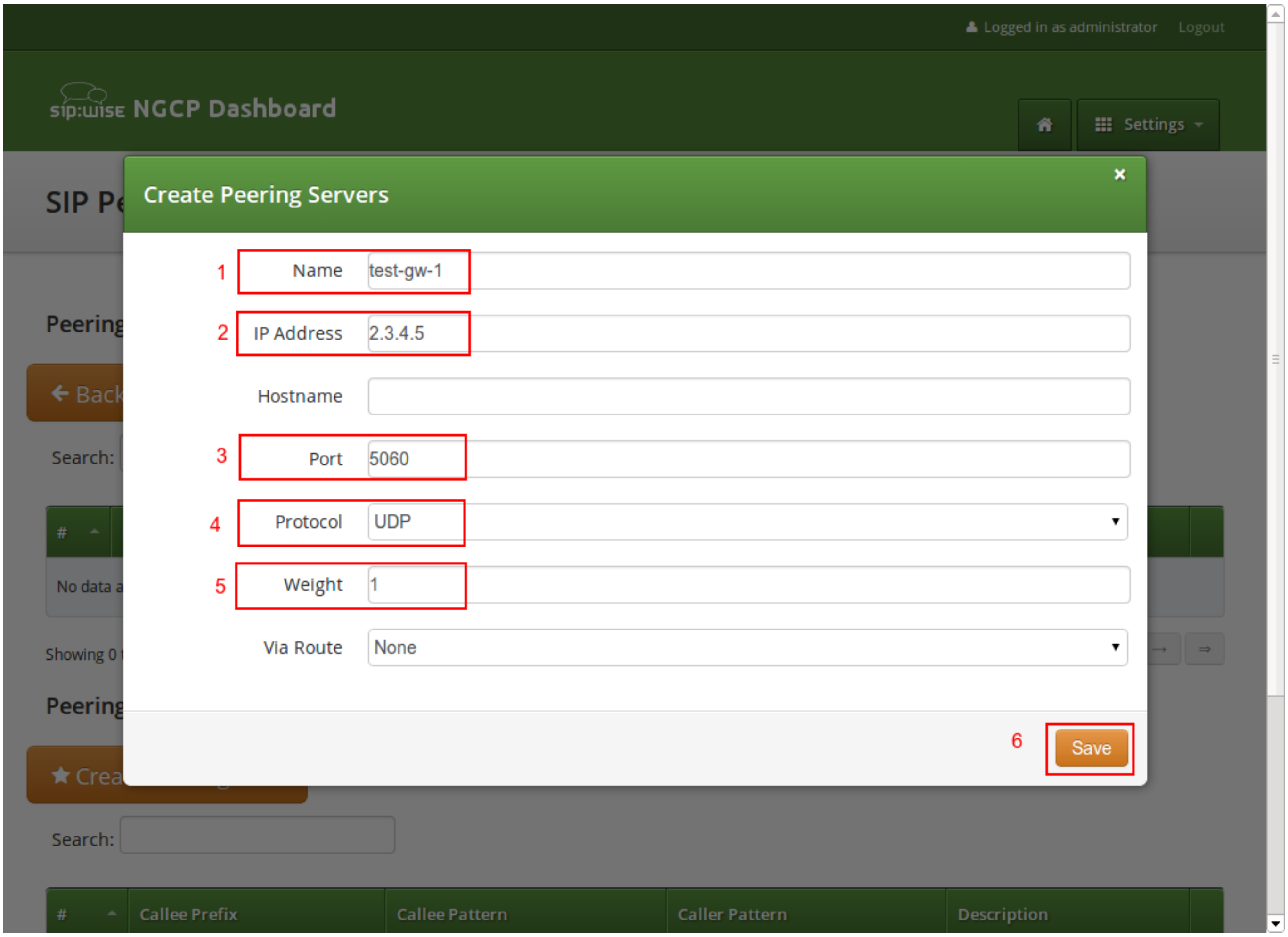
In the group created before, you need to add peering servers to route calls to and receive calls from. To do so, click on *Details* on the row of your new group in your peering group list.

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



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`



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:

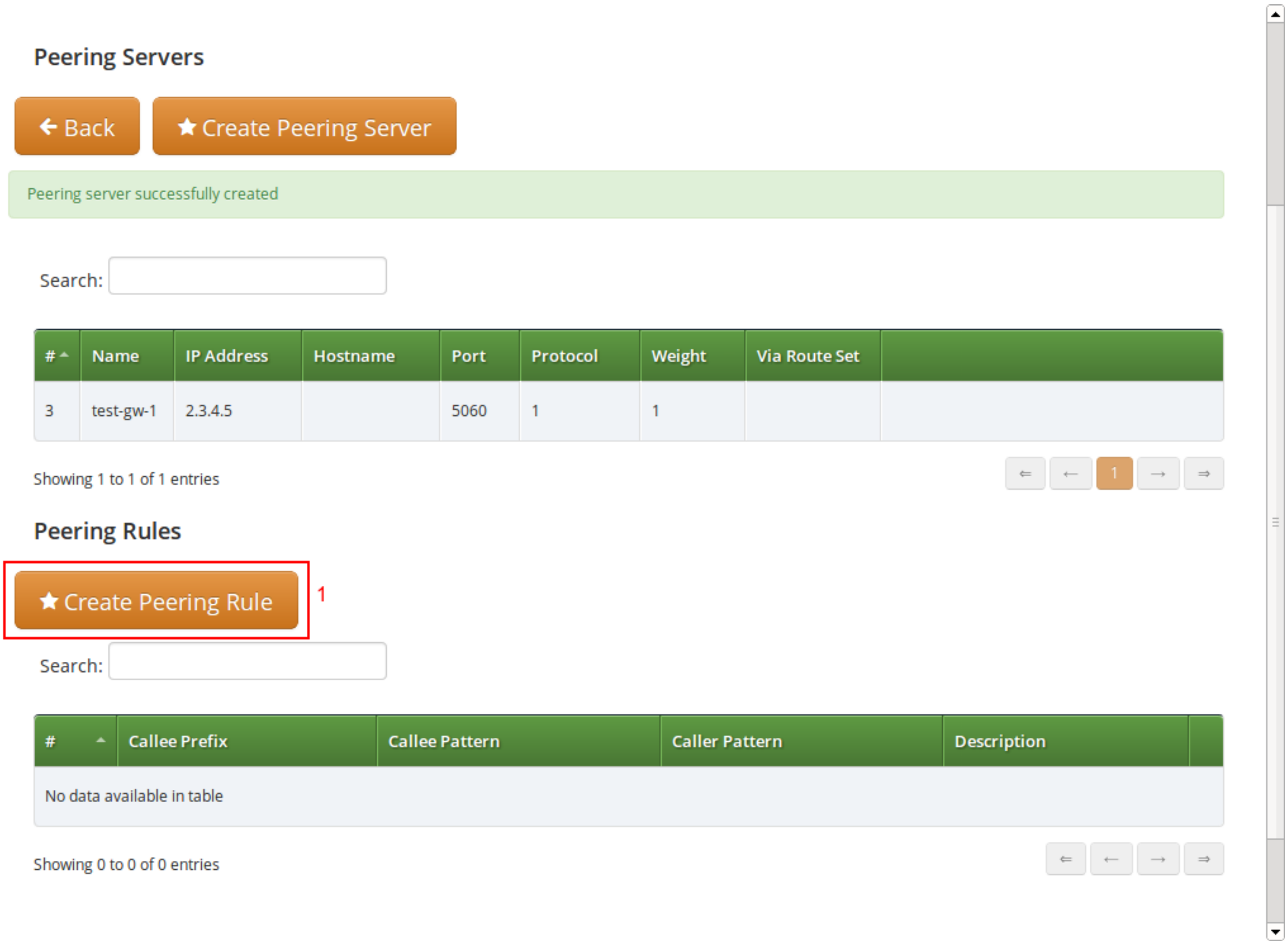
```
kamailio:
  lb:
    external_sbc:
      - sip:192.168.0.1:5060
```

```
    - 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.

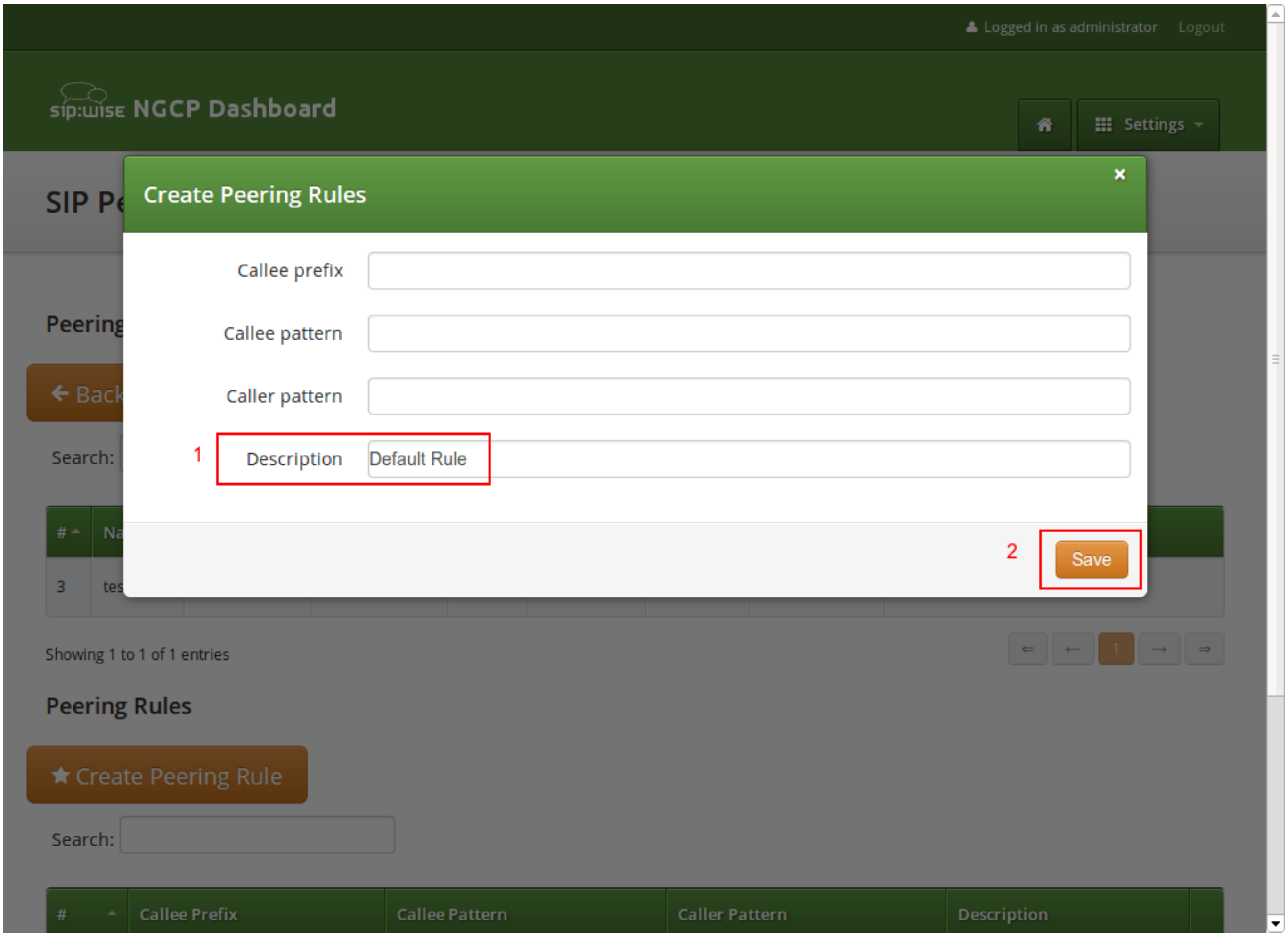
> **Important**
> To be able to send outbound calls towards the servers in the *Peering Group*, you also need to define *Peering Rules*. They specify which source and destination numbers are going to be terminated over this group. To create a rule, click the *Create Peering Rule* button.



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

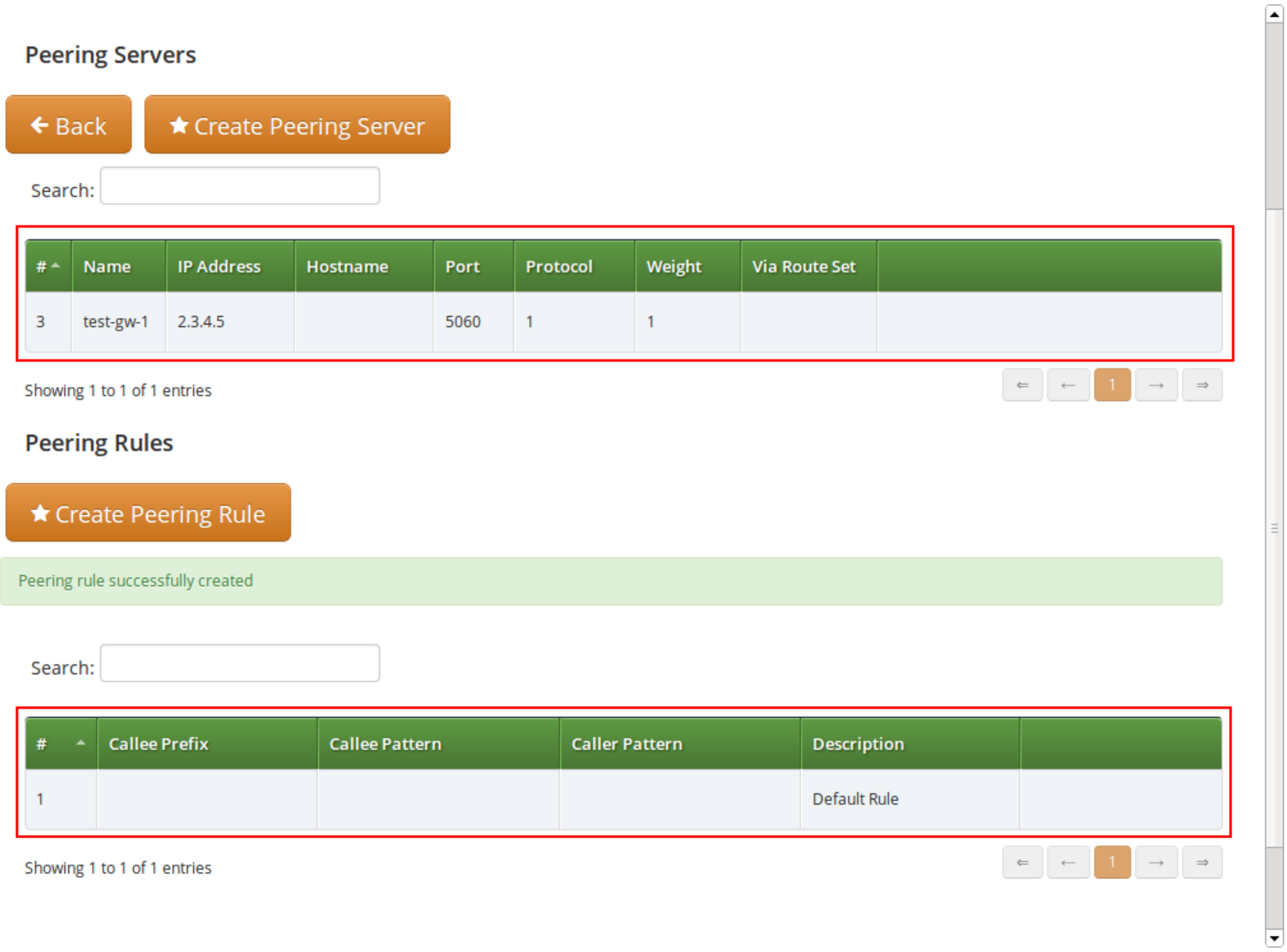
- **Description:** `Default Rule`



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.



4.5.2.1 Peering Group Selection Procedure

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

1. All peering groups are selected as candidates that meet the following criteria:
   - Callee's username matches *callee prefix*
   - Callee's URI matches *callee pattern*
   - Caller's URI matches *caller pattern*

   of the outbound peering rule.
2. From the peering group candidates those are considered for further selection where the longest match with *callee prefix* occurs.
3. ***Priority*** of the peering group will decide the order in which peering groups will be tried for routing the outbound call.

> **Important**
>
> A lower priority value means higher effective priority.

The valid range of values is: from "0" for highest priority to "255" for lowest priority.

4. All peering servers in the group with the highest priority (i.e. with the lowest priority value) are tried one by one. The ***weight*** of the peering servers in the selected peering group will influence the order in which the servers will be tried for routing the outbound call. The weight of a server is an integer value in the range of "1" to "254".

> **Important**
> A server with higher weight value does *not* always take precedence over a server with lower weight, although the former one has a higher chance to be the first. The weight of a peering server just defines the probability that it will get a call 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:

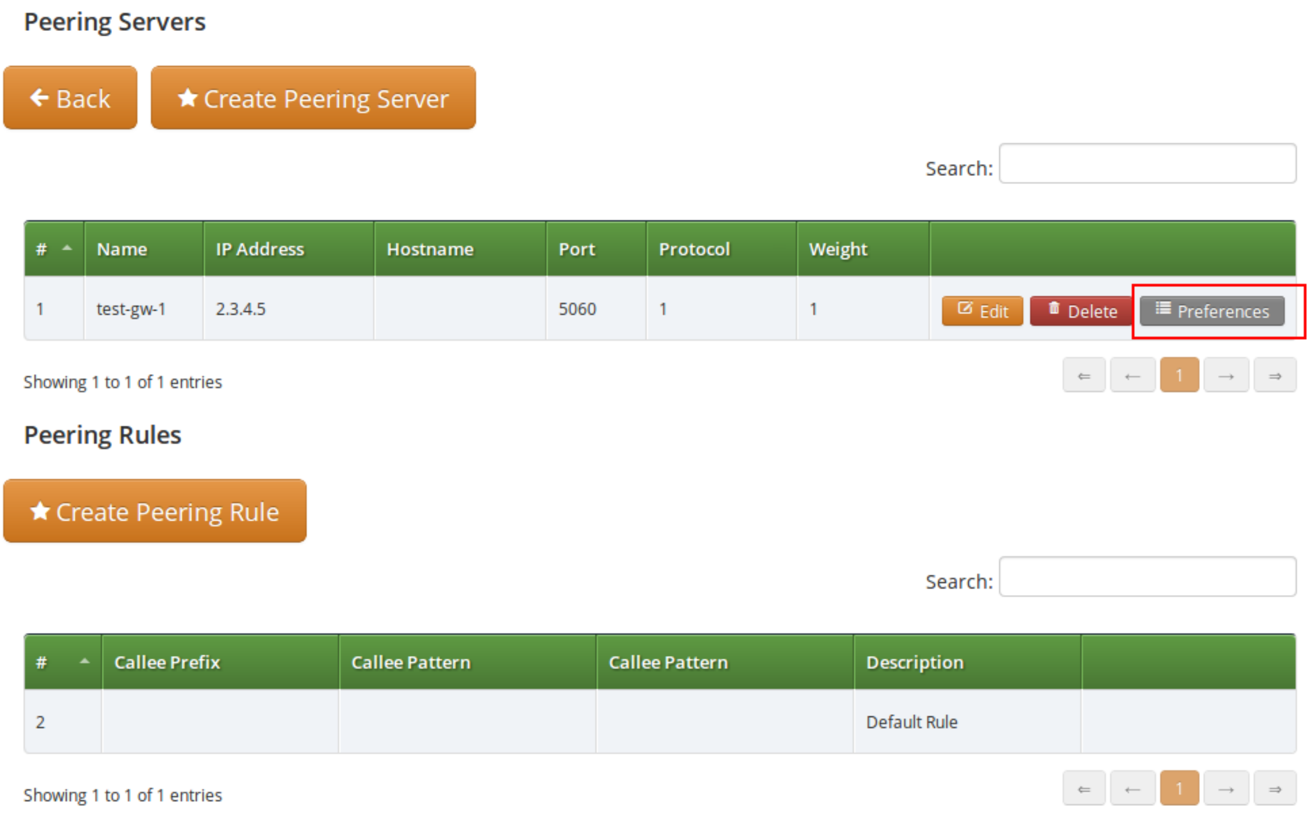
```
# lcr_weight_test.php 1 2

 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.

4.5.3 Authenticating and Registering against Peering Servers

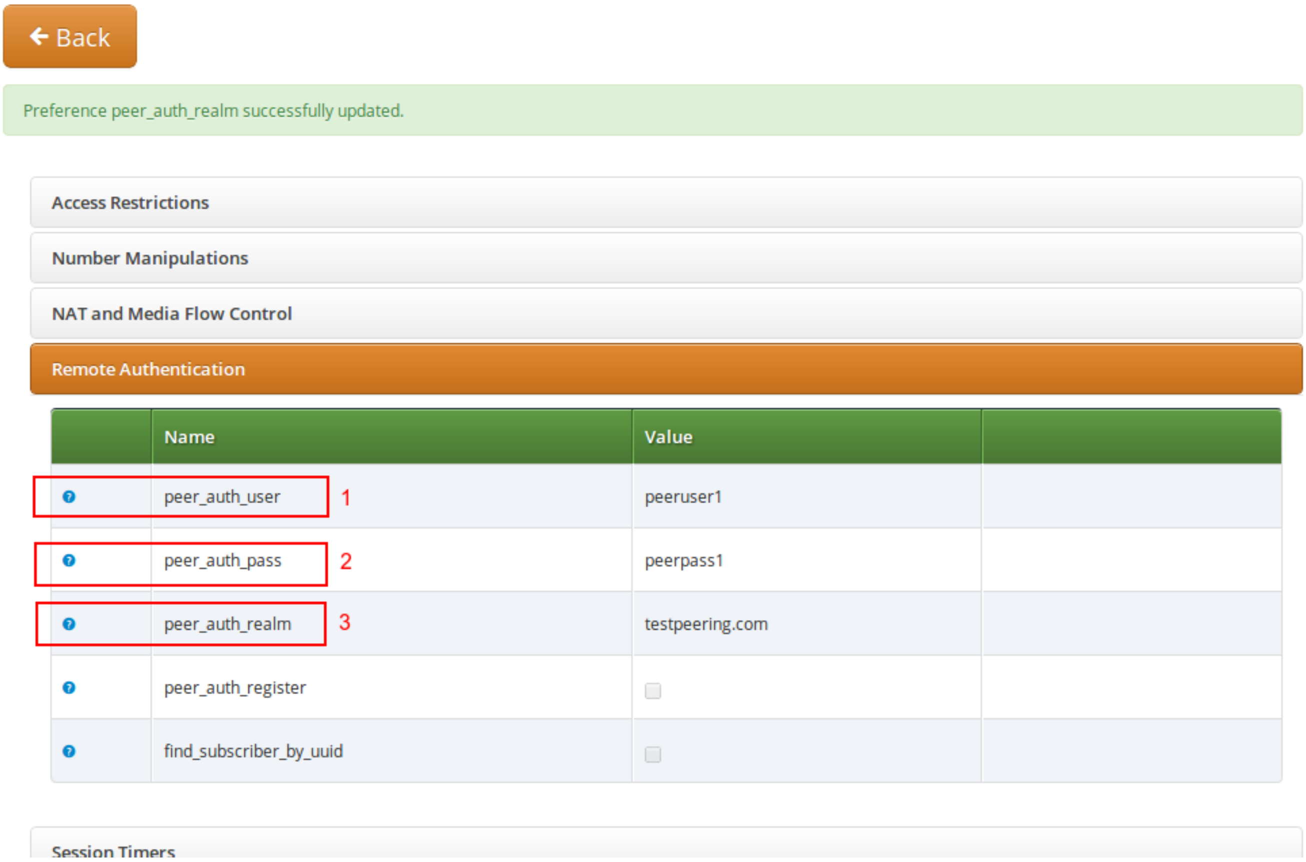
4.5.3.1 Proxy-Authentication for outbound calls

If a peering server requires the sip:provider PRO 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.



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*.

4.5.3.2 Registering at a Peering Server

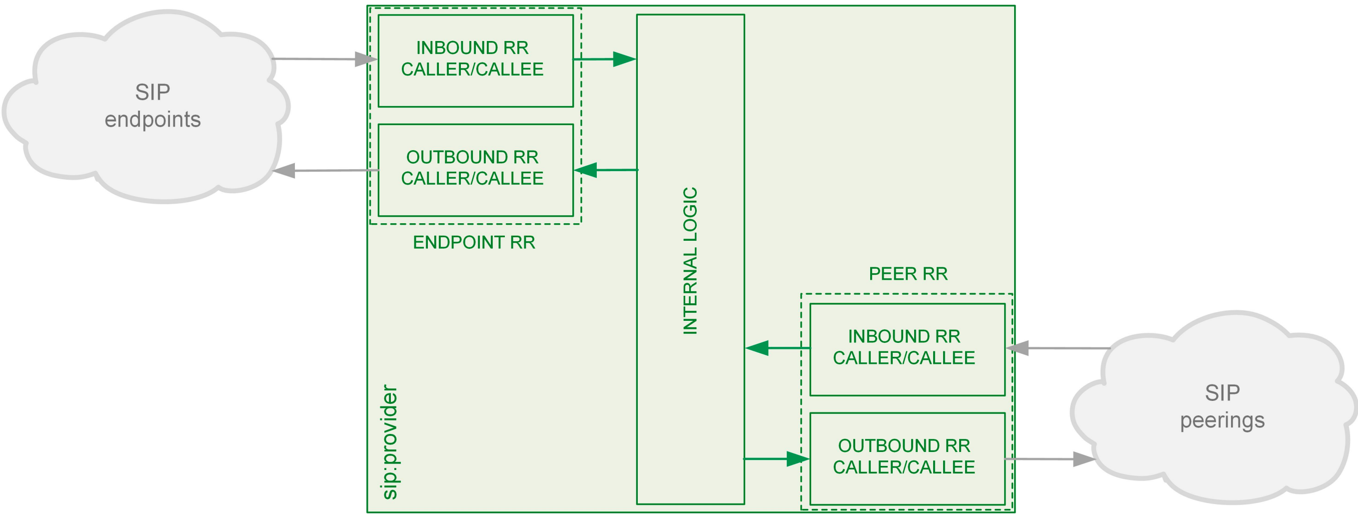
Unfortunately, the credentials configured above are not yet automatically used to register the sip:provider PRO at your peer hosts. There is however an easy manual way to do so, until this is addressed.

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 running conference calls.

4.6 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:

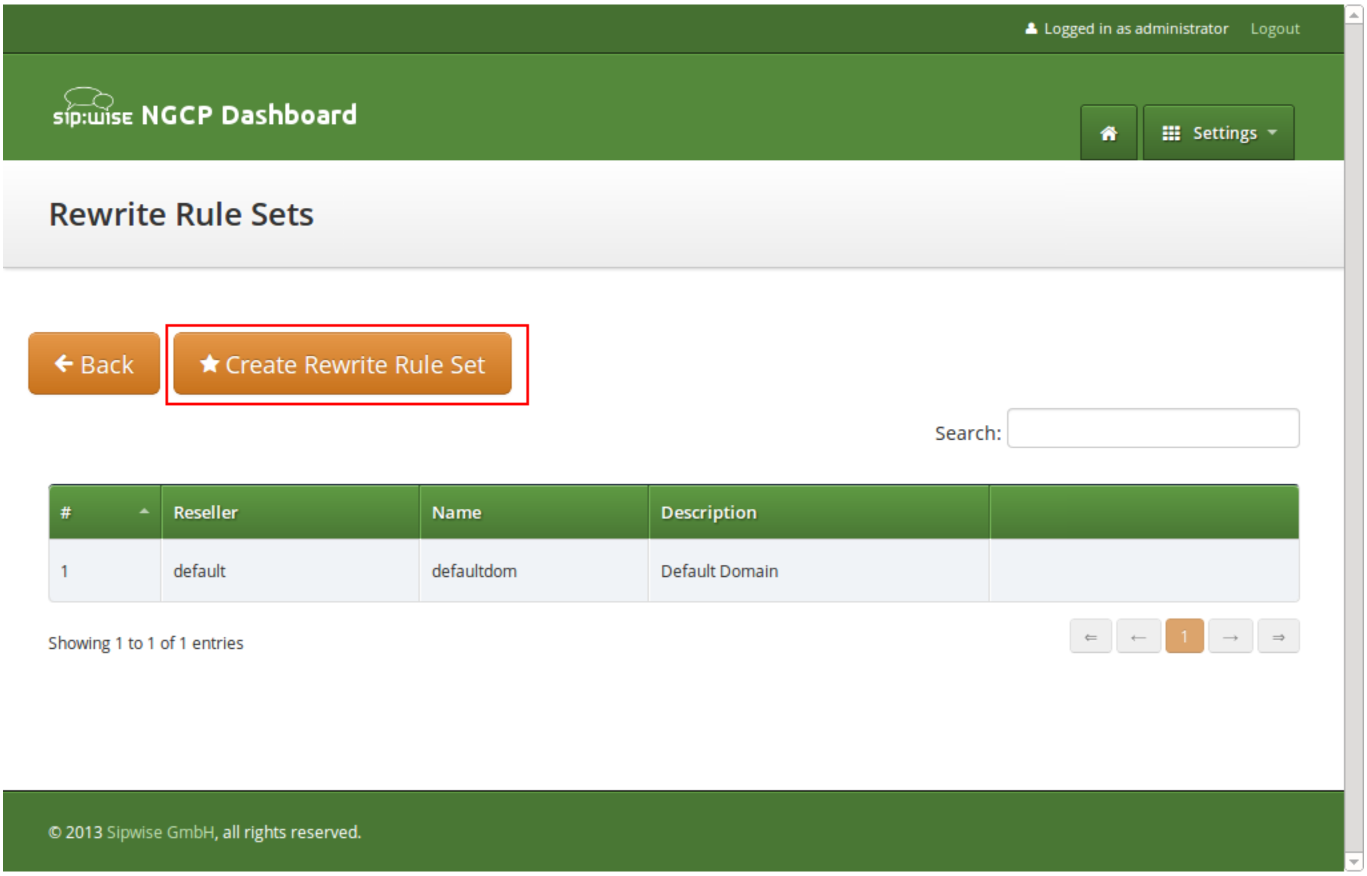
- Call from local subscriber A to local subscriber B: Inbound RR from local Domain/Subscriber A and Outbound Rewrite Rules from local Domain/Subscriber B.
- Call from local subscriber A to the peer: Inbound RR from local Domain/Subscriber A and Outbound Rewrite Rules from the peer.
- Call from peer to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

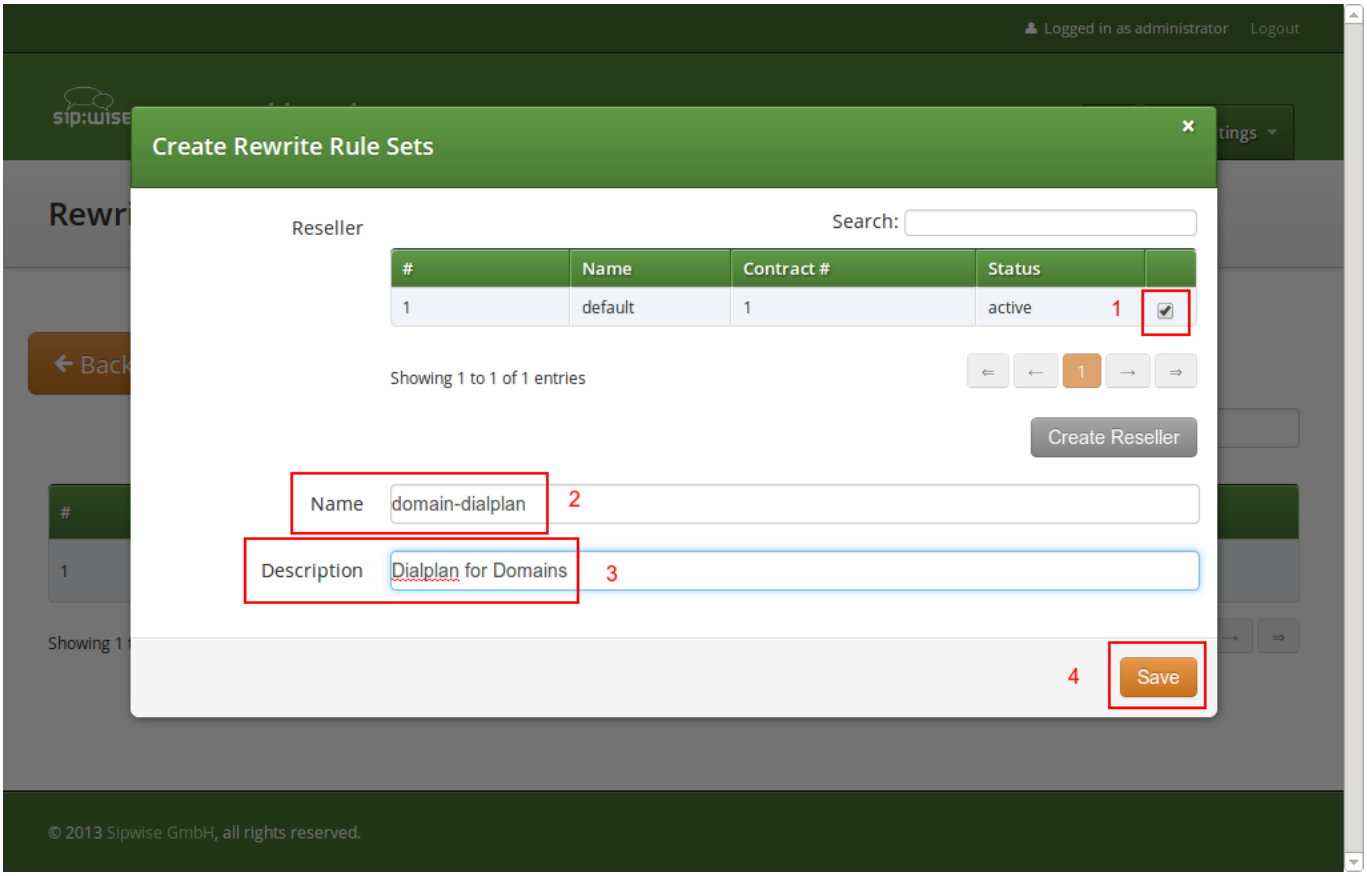
You can use several special variables in the Rewrite Rules, below you can find a list of them. Some examples of how to use them are also provided in the following sections:

- `${caller_cc}` : This is the value taken from the subscriber's preference CC value under Number Manipulation
- `${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
- `${caller_emergency_suffix}` : This is the value taken from the subscriber's preference emergency_suffix value under Number Manipulation
- `${caller_cloud_pbx_base_cli}` :This is the value taken from the *Primary Number* field from section *Details → Master Data* of the *Pilot Subscriber* for a particular PBX customer.

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.

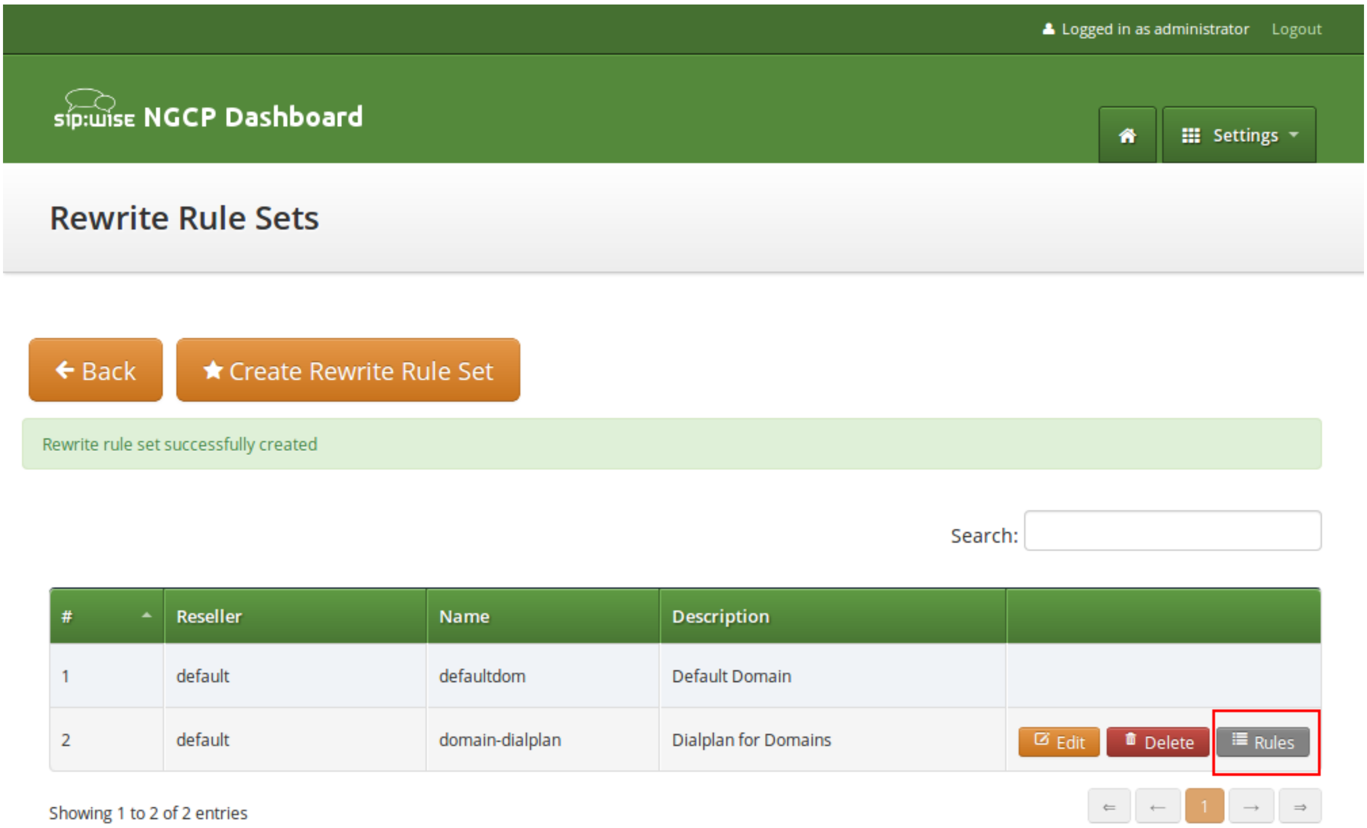


Click *Create Rewrite Rule Set* and fill in the form accordingly.



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.

4.6.1 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`

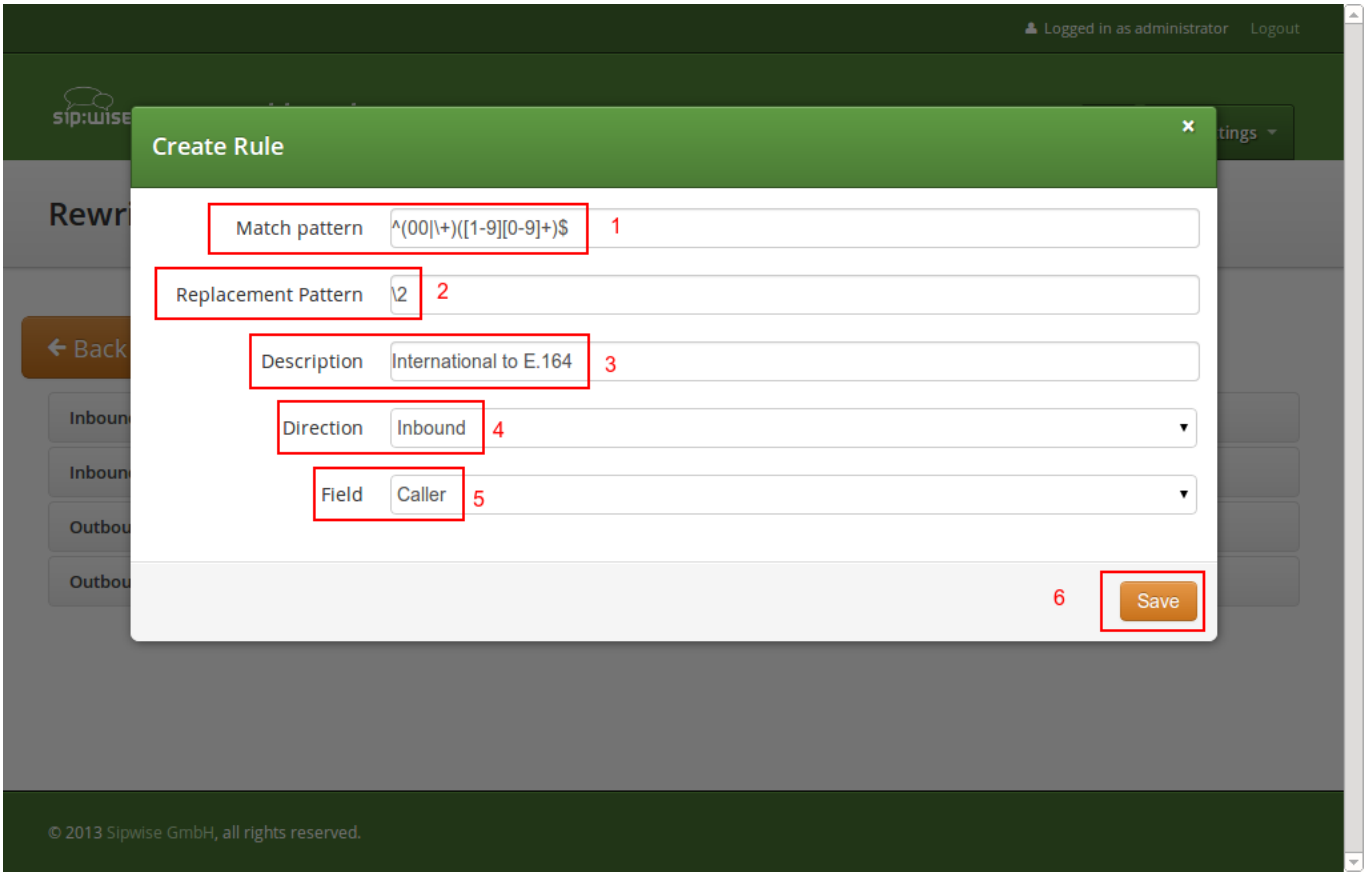
- Field: `Caller`

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`



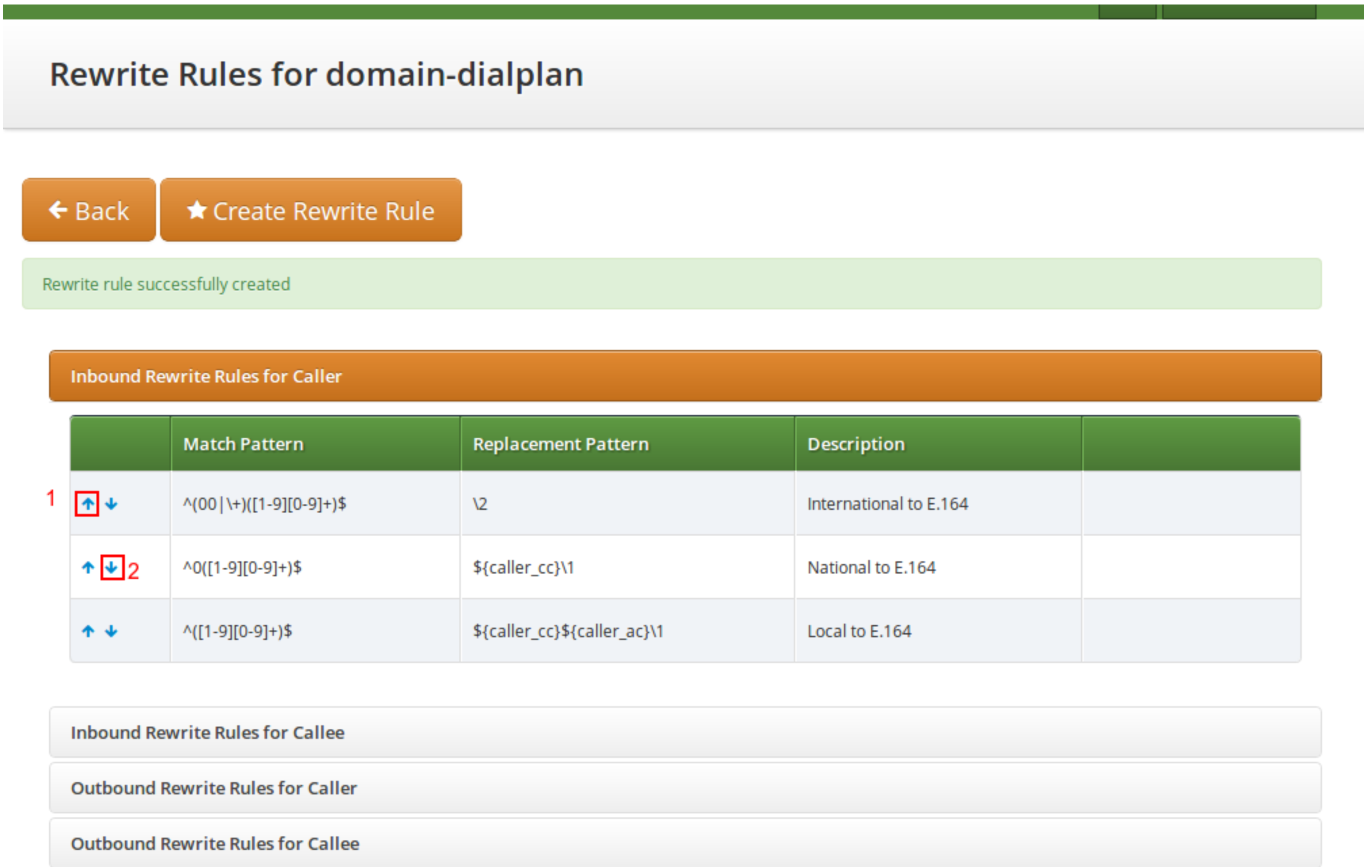
Normalization for national and local calls is possible with special variables `${caller_cc}` and `${caller_ac}` that can be used in Replacement Pattern and are substituted by the country and area code accordingly during the call routing.

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.



4.6.2 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.

4.6.3 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).

4.6.4 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`

4.6.5 Emergency Number Handling

Configuring Emergency Numbers is also done via Rewrite Rules.

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.

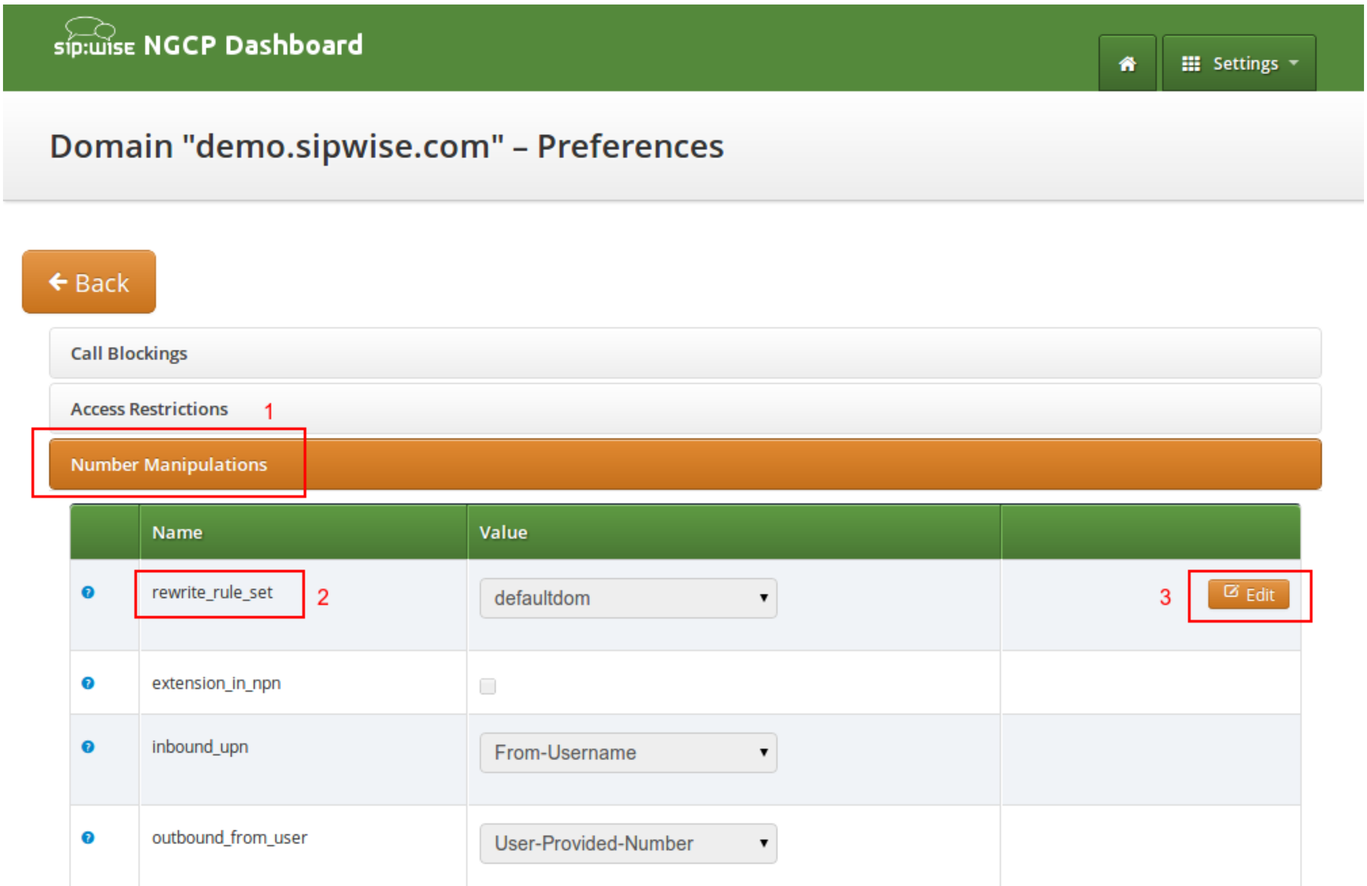
In order to normalize the emergency number to a valid format accepted by the peer, you need to assign an *Outbound Rewrite Rule For Callee*, which strips off the emergency_ prefix. You can also use the variables ${caller_emergency_cli}, ${caller_emergency_prefix} and ${caller_emergency_suffix} as well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

4.6.6 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.

4.6.7 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 4.6 (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.

5 Features

The sip:provider PRO 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.

5.1 Managing System Administrators

The sip:provider PRO 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.

5.1.1 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*.

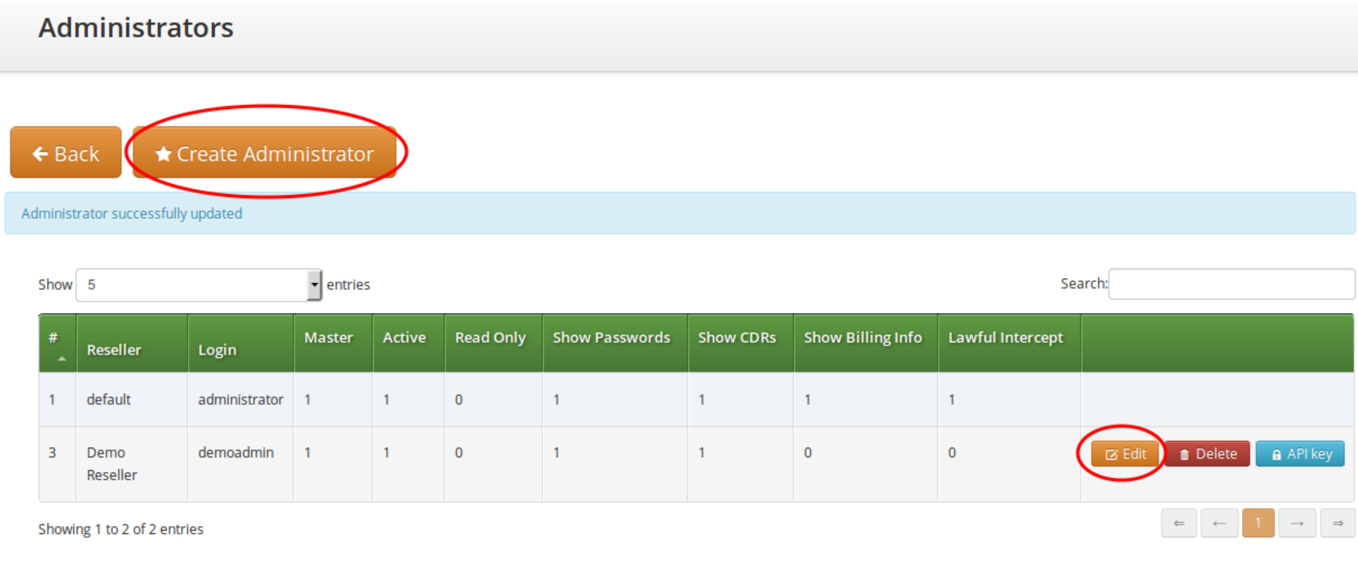


Figure 13: List of System Administrators

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:

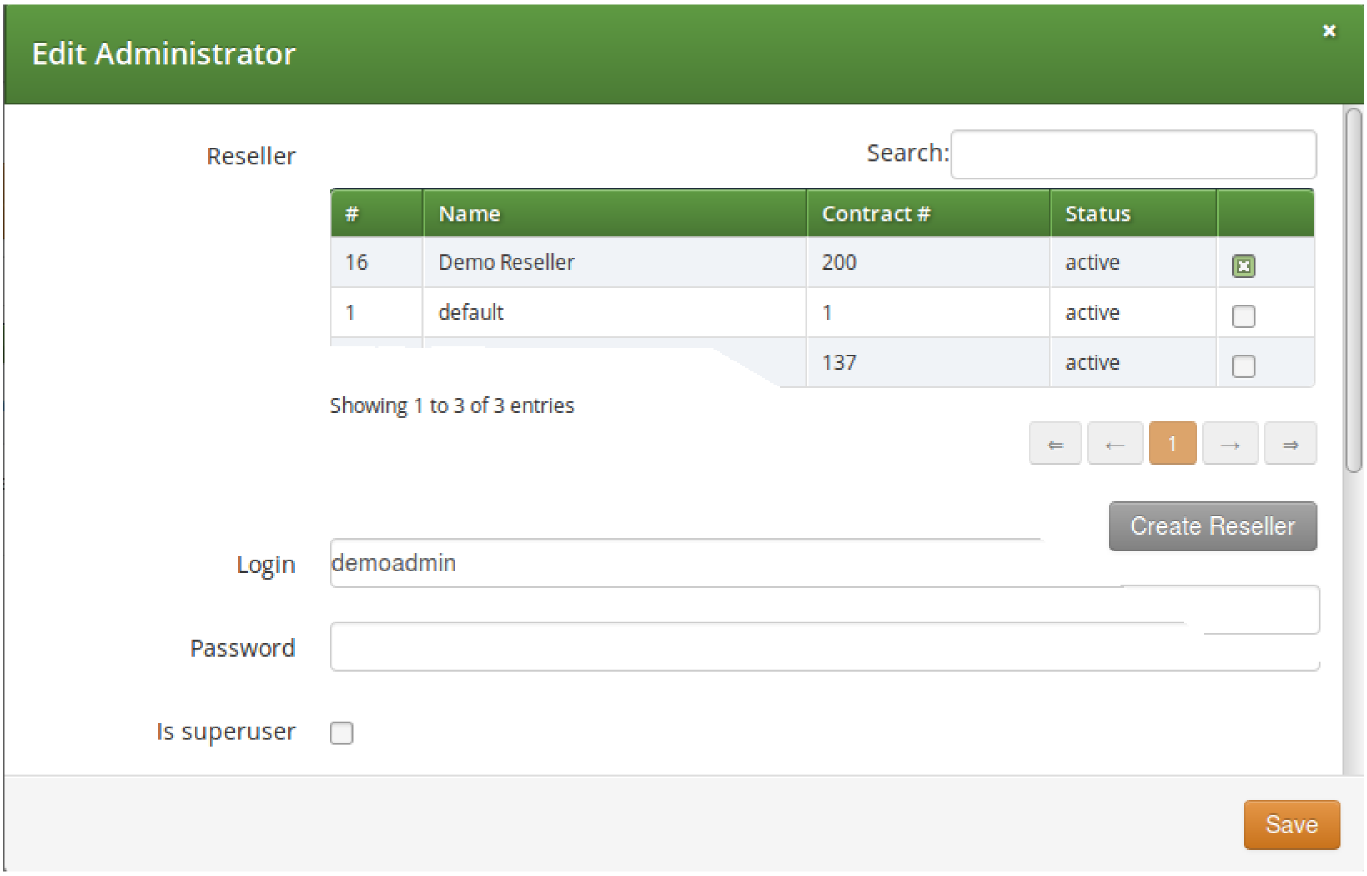


Figure 14: Generic System Administrator Attributes

- *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.

5.1.2 Access Rights of Administrators

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

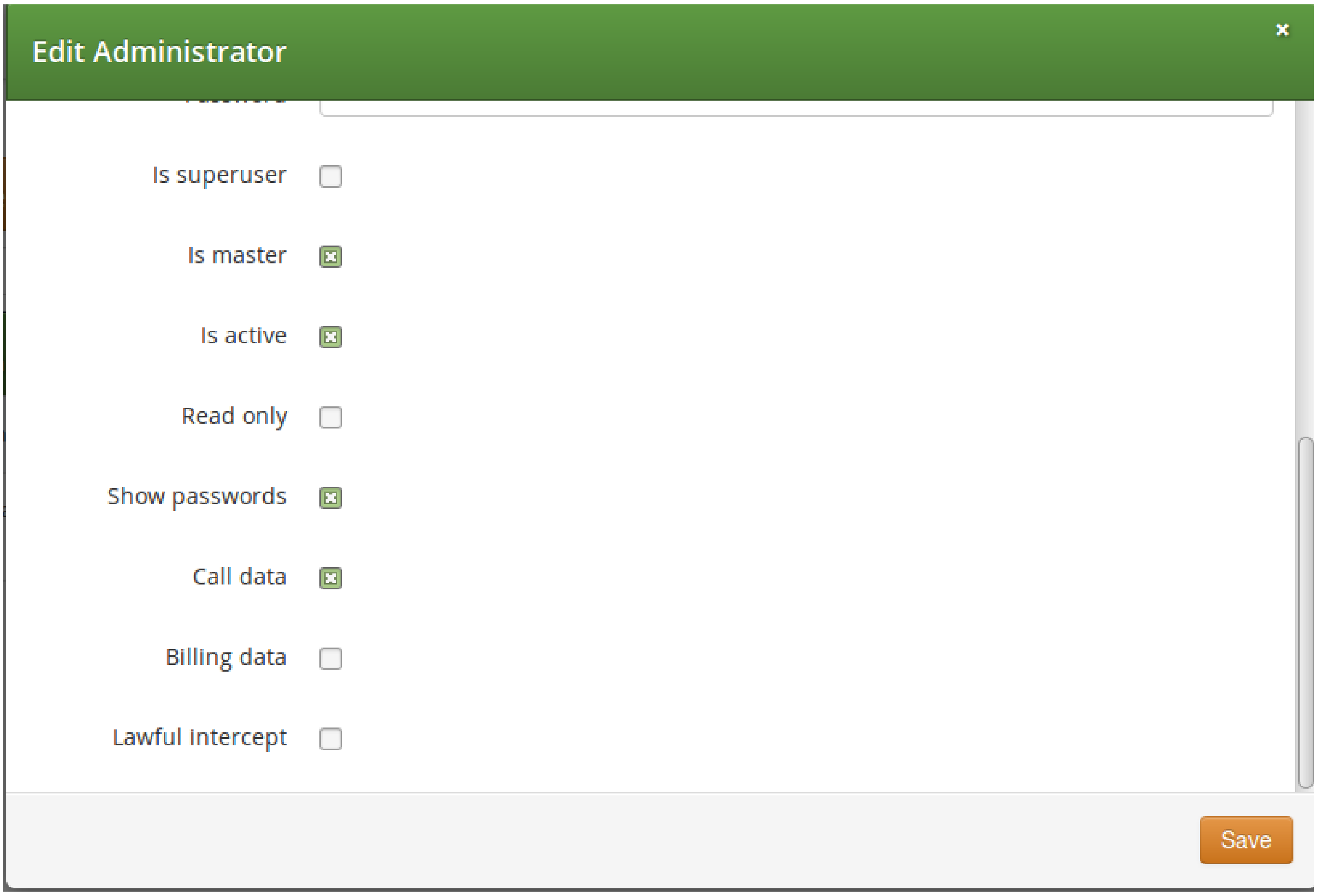


Figure 15: Access Rights of System Administrators

Table 1: 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 1: (continued)

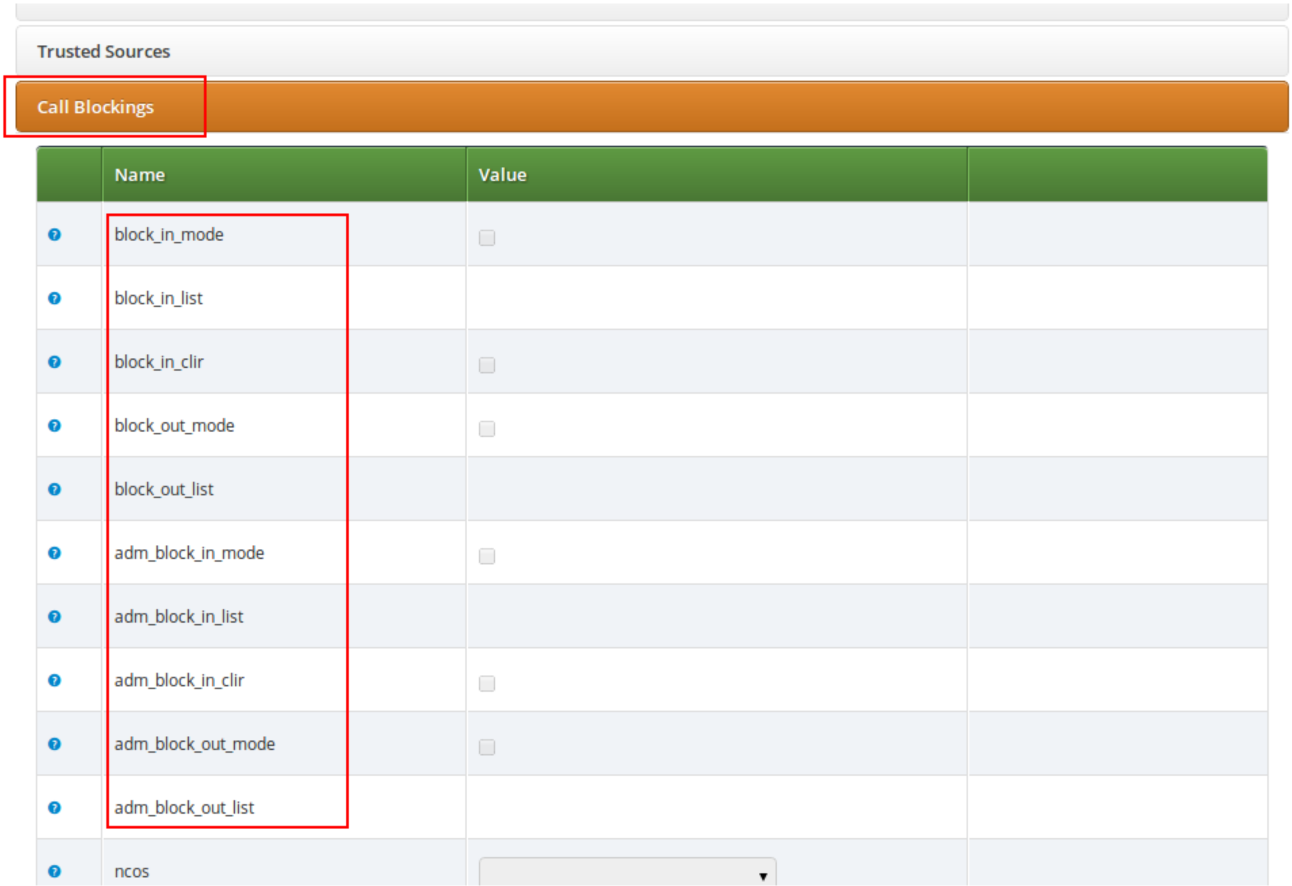
<table>
<tr><th>Label in admin list</th><th>Access Right</th><th>Description</th></tr>
<tr><td>Read Only</td><td><code>Read only</code></td><td>The user will only be able to list various data but is not allowed to modify anything.
• For the web interface this means that Create… and Edit buttons will be hidden or disabled.
• For the REST API this means that only <code>GET, HEAD, OPTIONS</code> HTTP request methods are accepted, and NGCP will reject those targeting data modification: <code>PUT, PATCH, POST, DELETE</code>.</td></tr>
<tr><td>Show Passwords</td><td><code>Show passwords</code></td><td>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.</td></tr>
<tr><td>Show CDRs</td><td><code>Call data</code></td><td>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</td></tr>
<tr><td>Show Billing Info</td><td><code>Billing data</code></td><td>Some REST API resources that are related to billing are disabled: HTTP requests on <code>/api/vouchers, /api/topupcash</code> and <code>/api/topupvoucher</code> resources are rejected.</td></tr>
<tr><td>Lawful Intercept</td><td><code>Lawful intercept</code></td><td>If the privilege is selected then the REST API for interceptions (that is: <code>/api/interceptions</code>) is enabled; if the privilege is not selected then the interceptions API is disabled.
Note
This means that besides enabling LI in <code>config.yml</code> 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.</td></tr>
</table>

5.2 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 PRO 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.

5.2.1 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.

5.2.1.1 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.

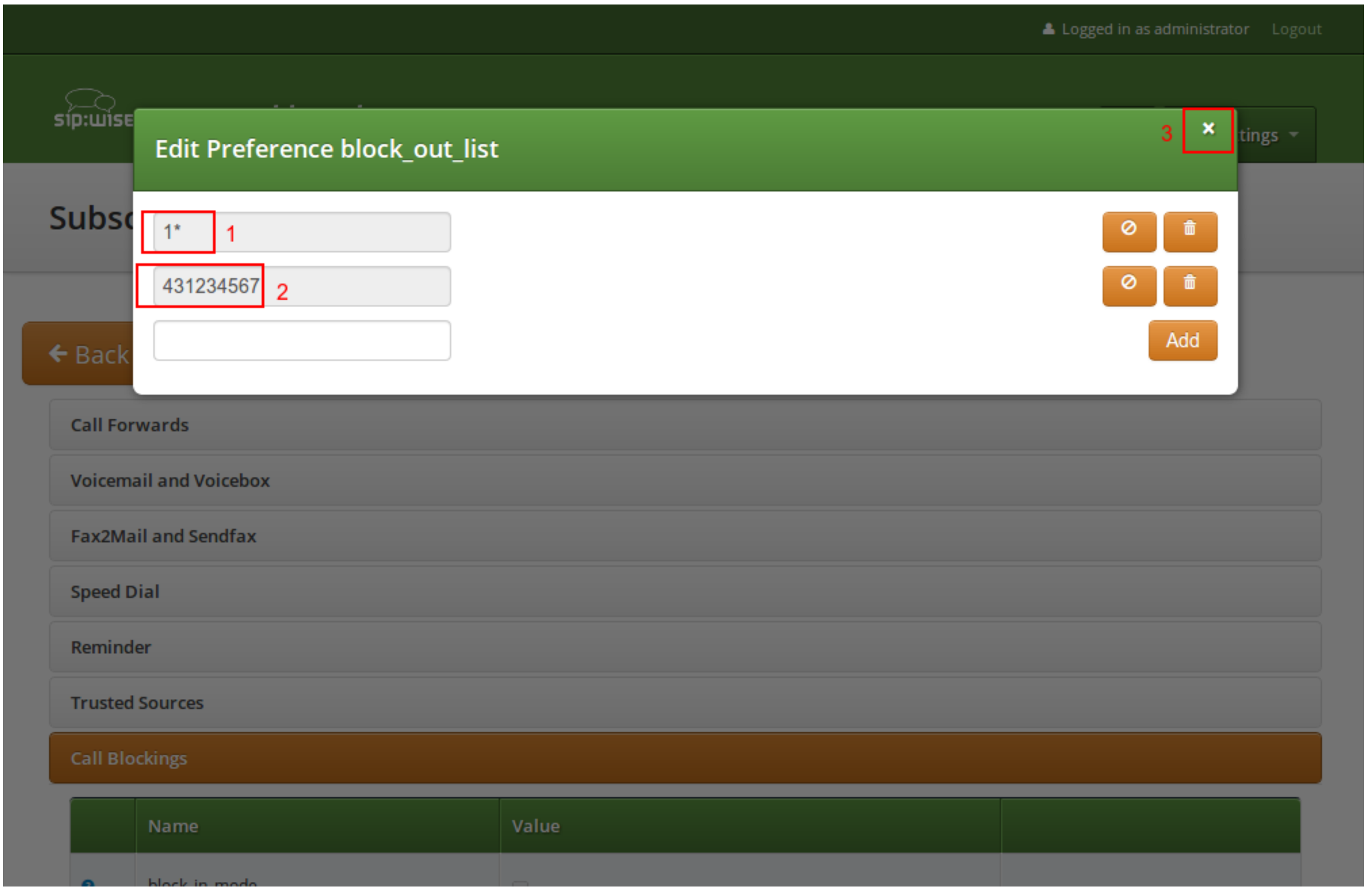
- The *blacklist* mode (option is not checked tells the system to **allow anything except the entries in the list**. Use this mode if you just want to block certain numbers and allow all the rest.
- The *whitelist* mode indicates to **reject anything except the entries in the list**. Use this mode if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

5.2.1.2 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.

5.2.1.3 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.

5.2.2 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.

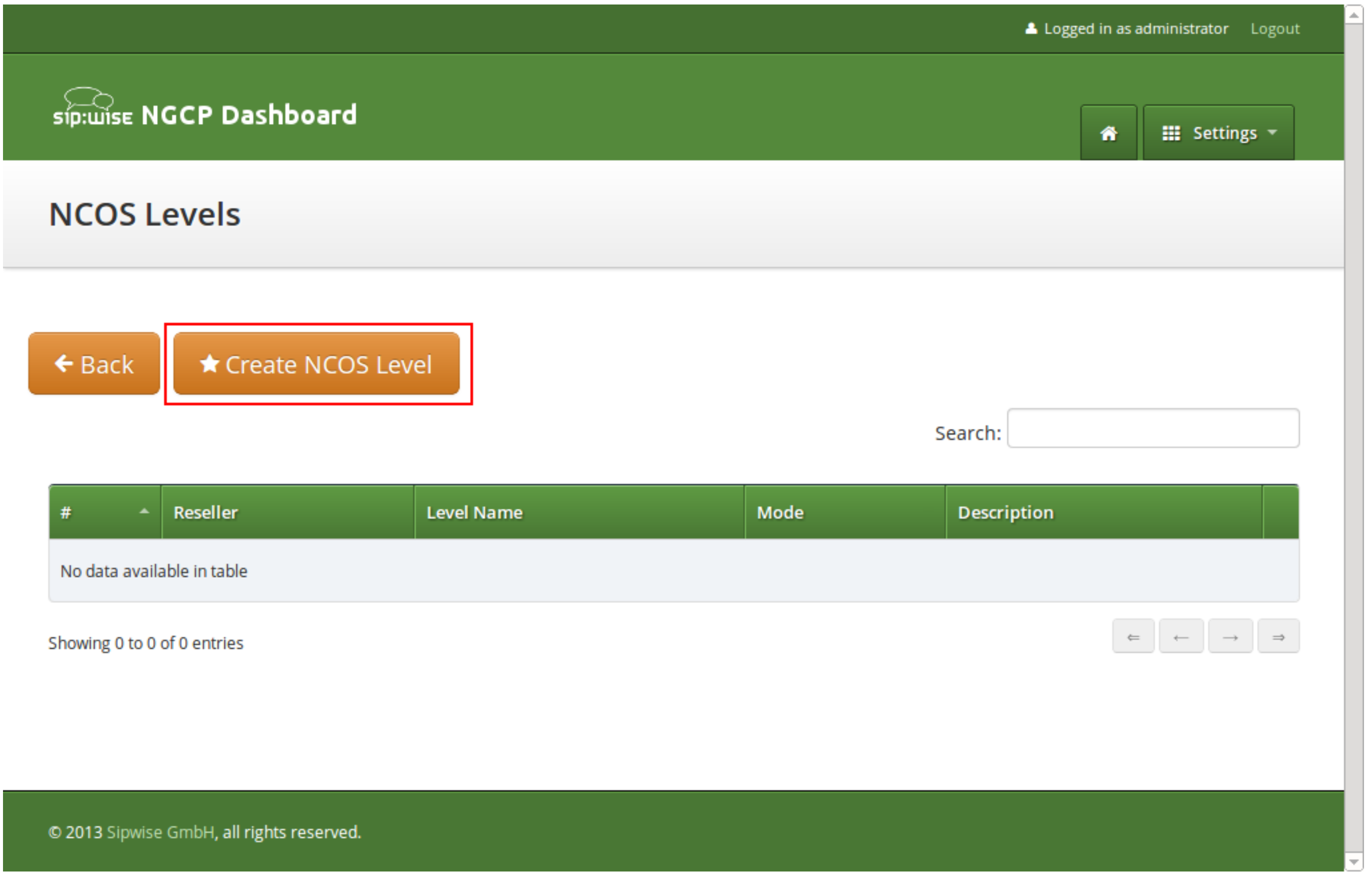
If case of a conflict, when the Block Lists feature allows a number and NCOS Levels rejects the same number or vice versa, the number will be rejected.

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

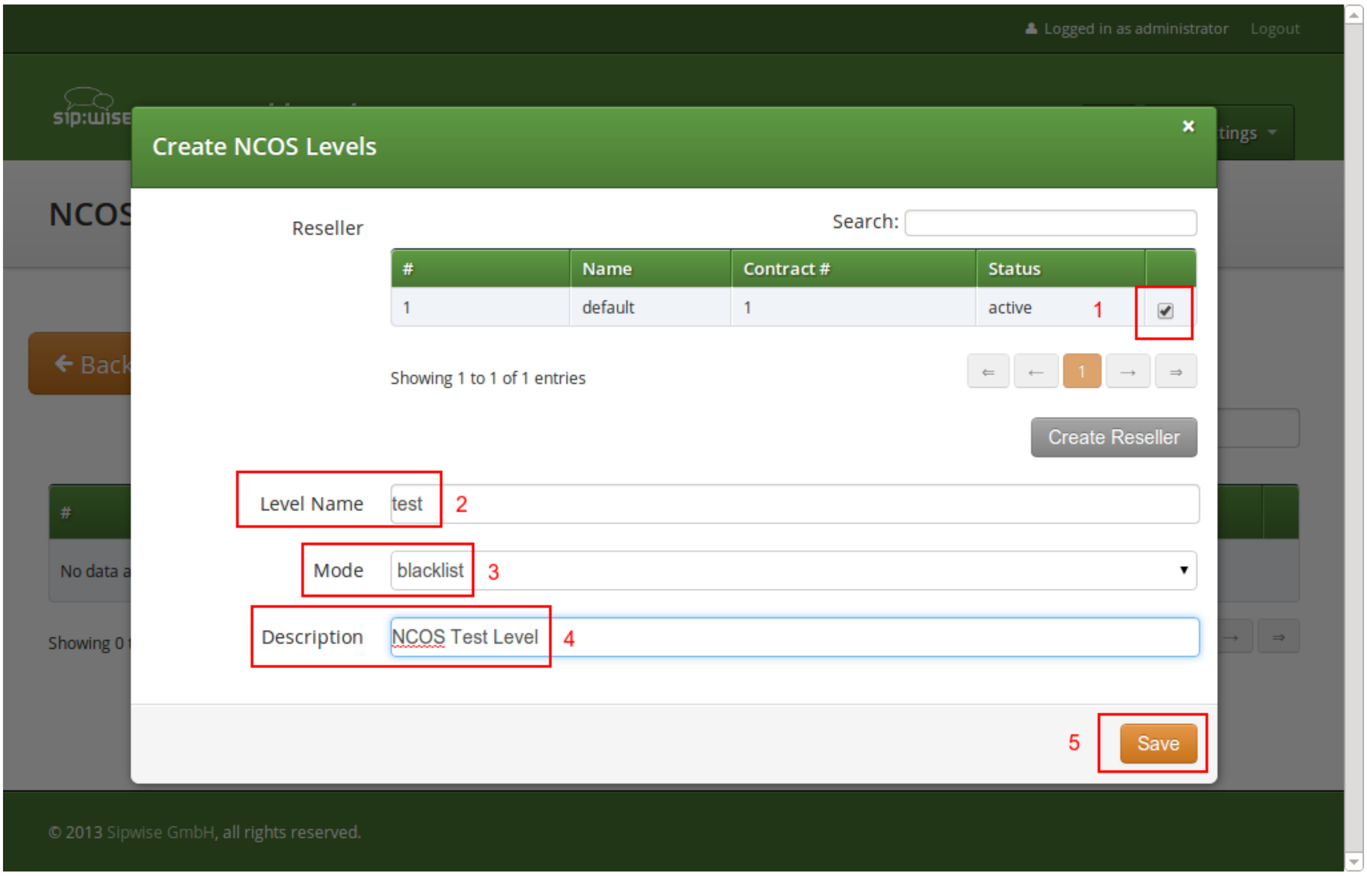
- The *blacklist* mode indicates to **allow everything except the entries in this level**. This mode is used if you want to just block certain destinations and allow all the rest.
- The *whitelist* mode indicates to **reject anything except the entries in this level**. This is used if you want to enforce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

5.2.2.1 Creating NCOS Levels

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

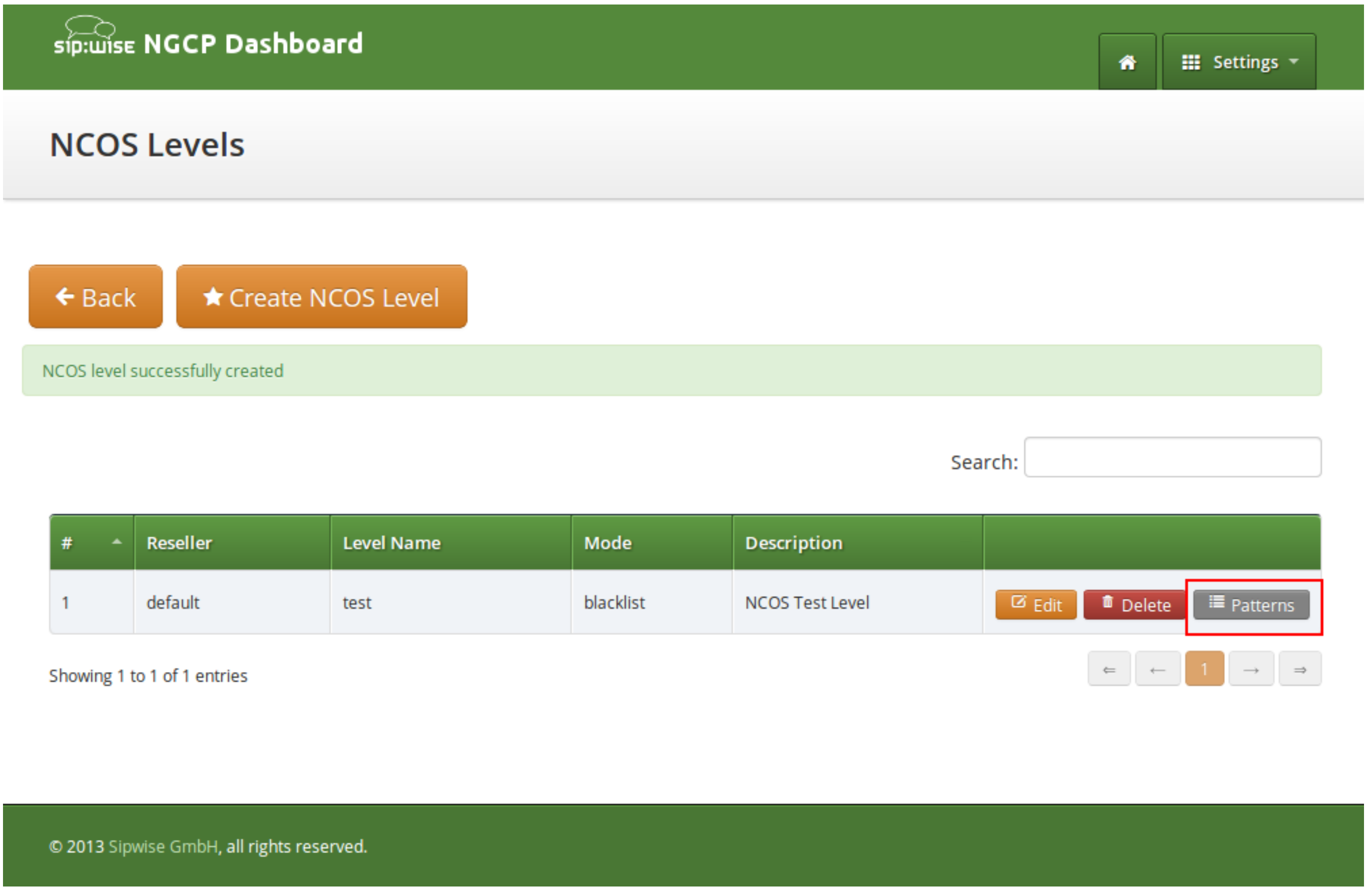


Select a reseller, enter a name, select the mode and add a description, then click the *Save* button.

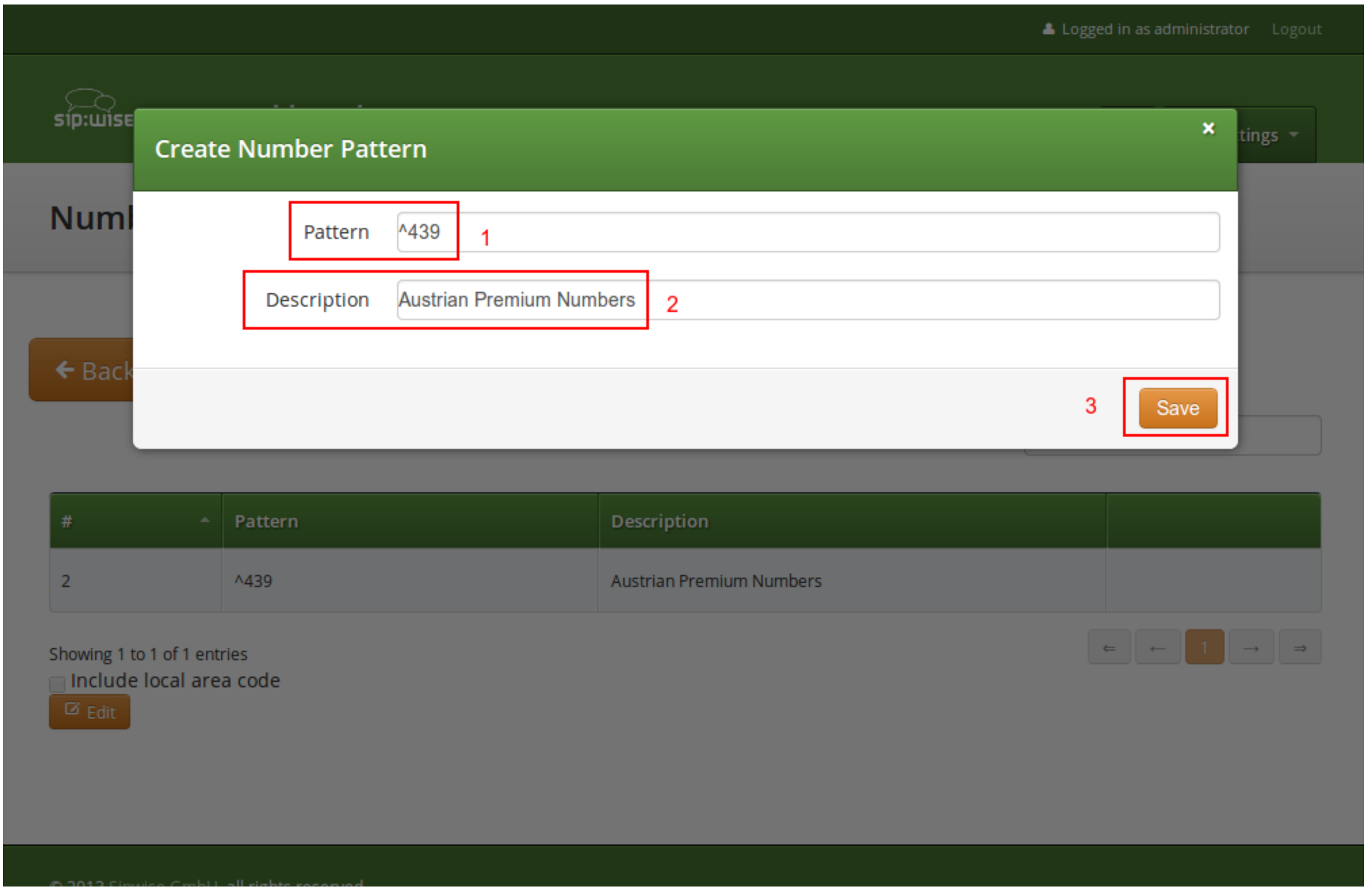


5.2.2.2 Creating Rules per NCOS Level

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



In the *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.

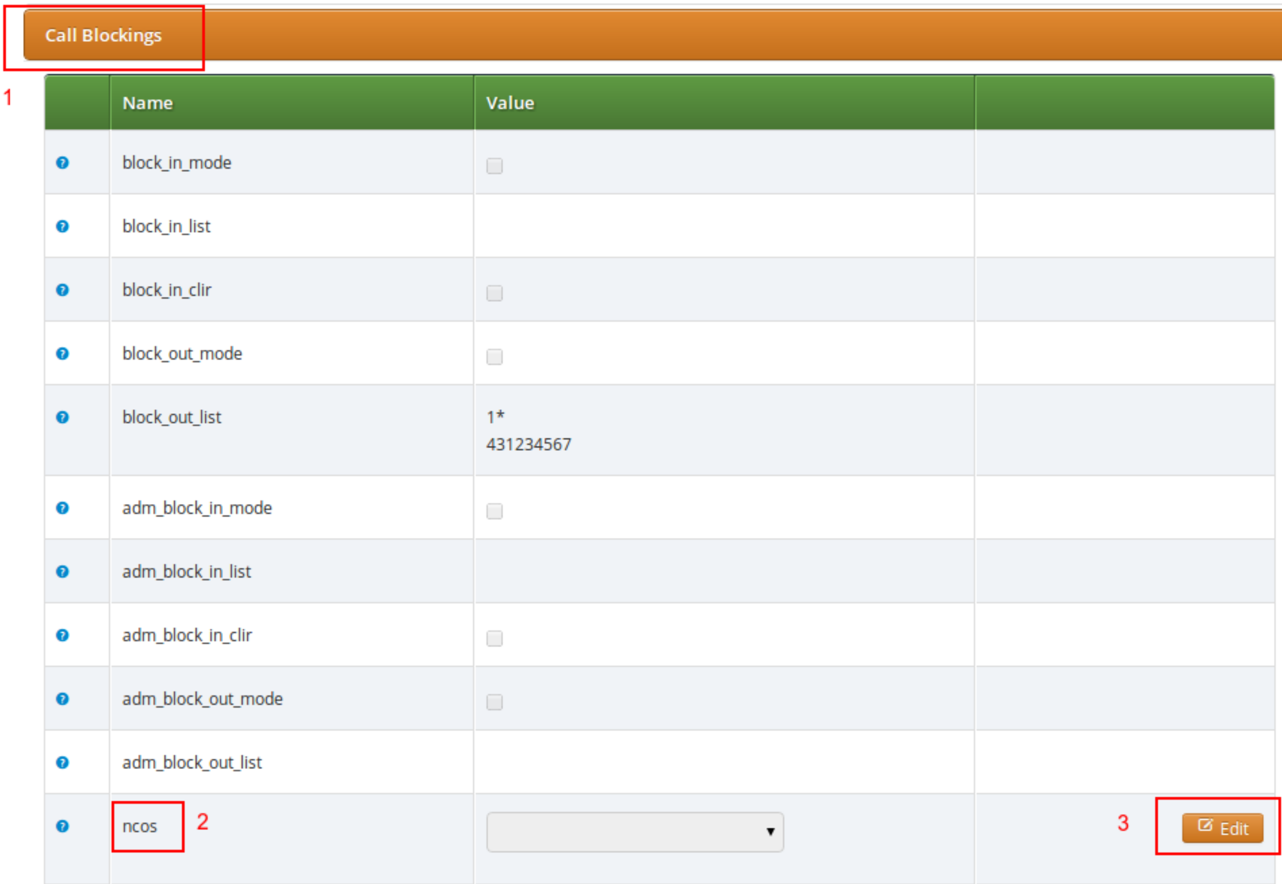


In this example, we block (since the mode of the level is *blacklist*) all numbers starting with `439`. Click the *Save* button to save the entry in the level.

The option *include local area code in list* for a blacklist means that calls within the area code of the subscribers are denied, and for whitelist that they are allowed, respectively. For example if a subscriber has country-code 43 and area-code 1, then selecting this checkbox would result in an implicit entry `^431`.

5.2.2.3 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.

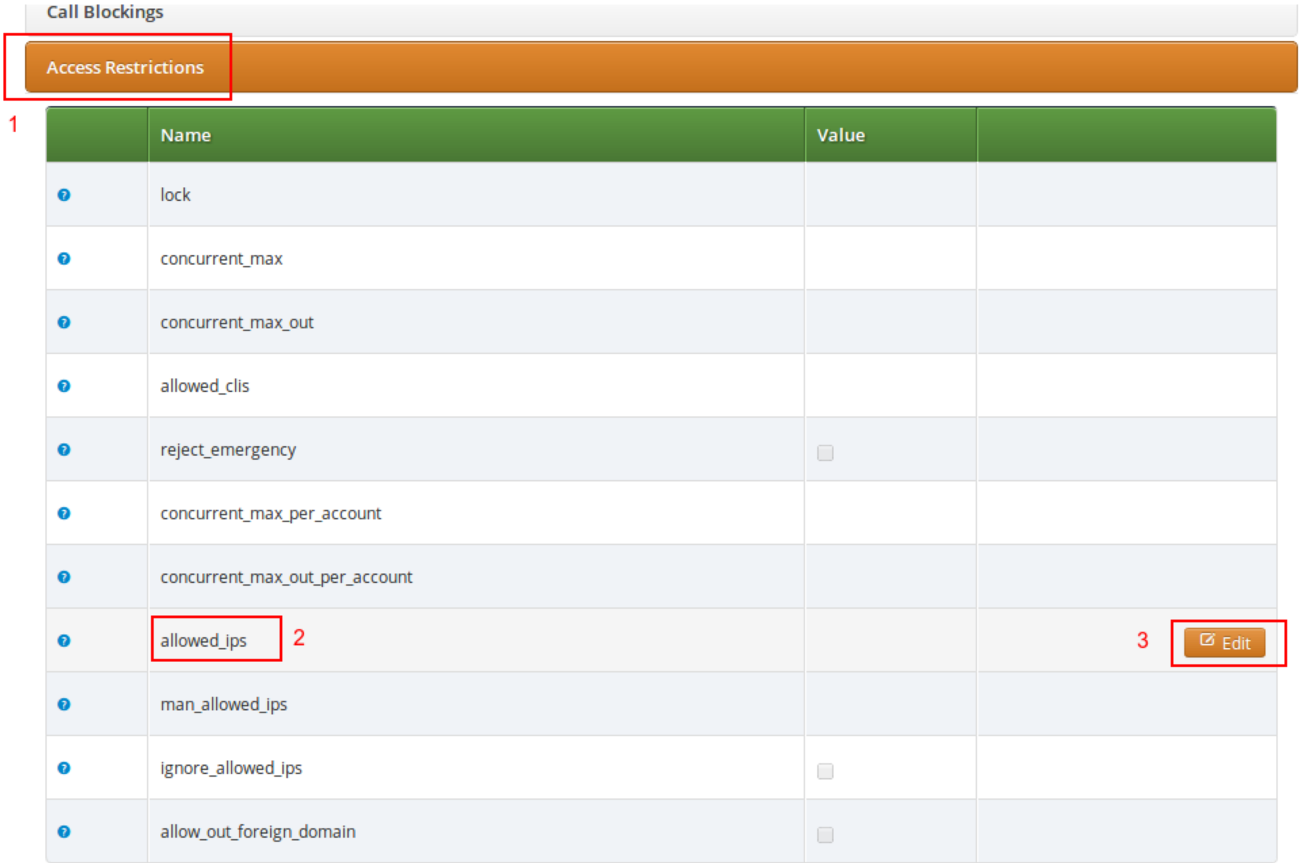
5.2.2.4 Assigning NCOS Level for Forwarded Calls to Subscribers/Domains

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 PRO 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.

5.2.3 IP Address Restriction

The sip:provider PRO 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 PRO 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.

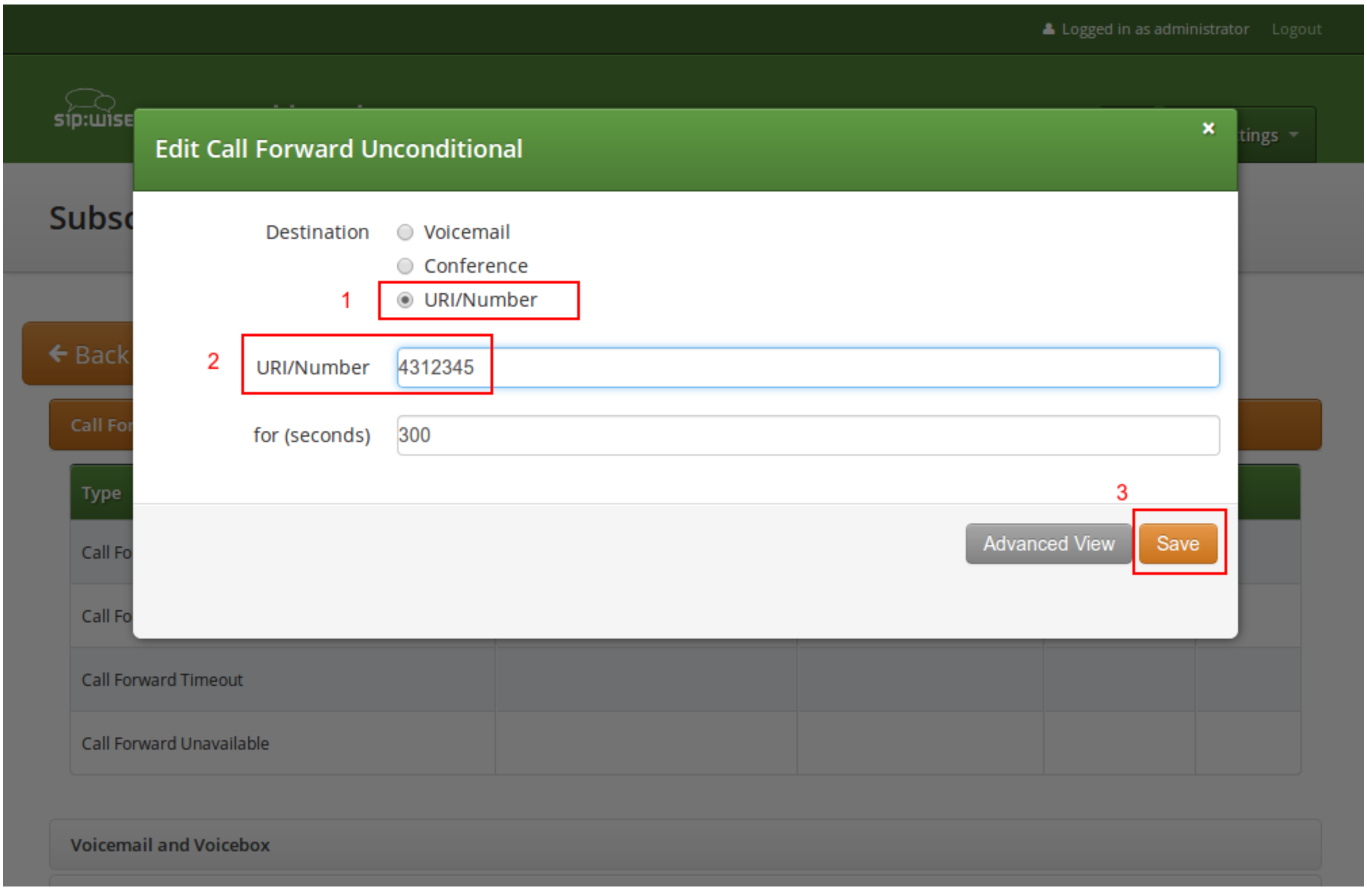
5.3 Call Forwarding and Call Hunting

The sip:provider PRO 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).

5.3.1 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.

5.3.2 Advanced Call Hunting

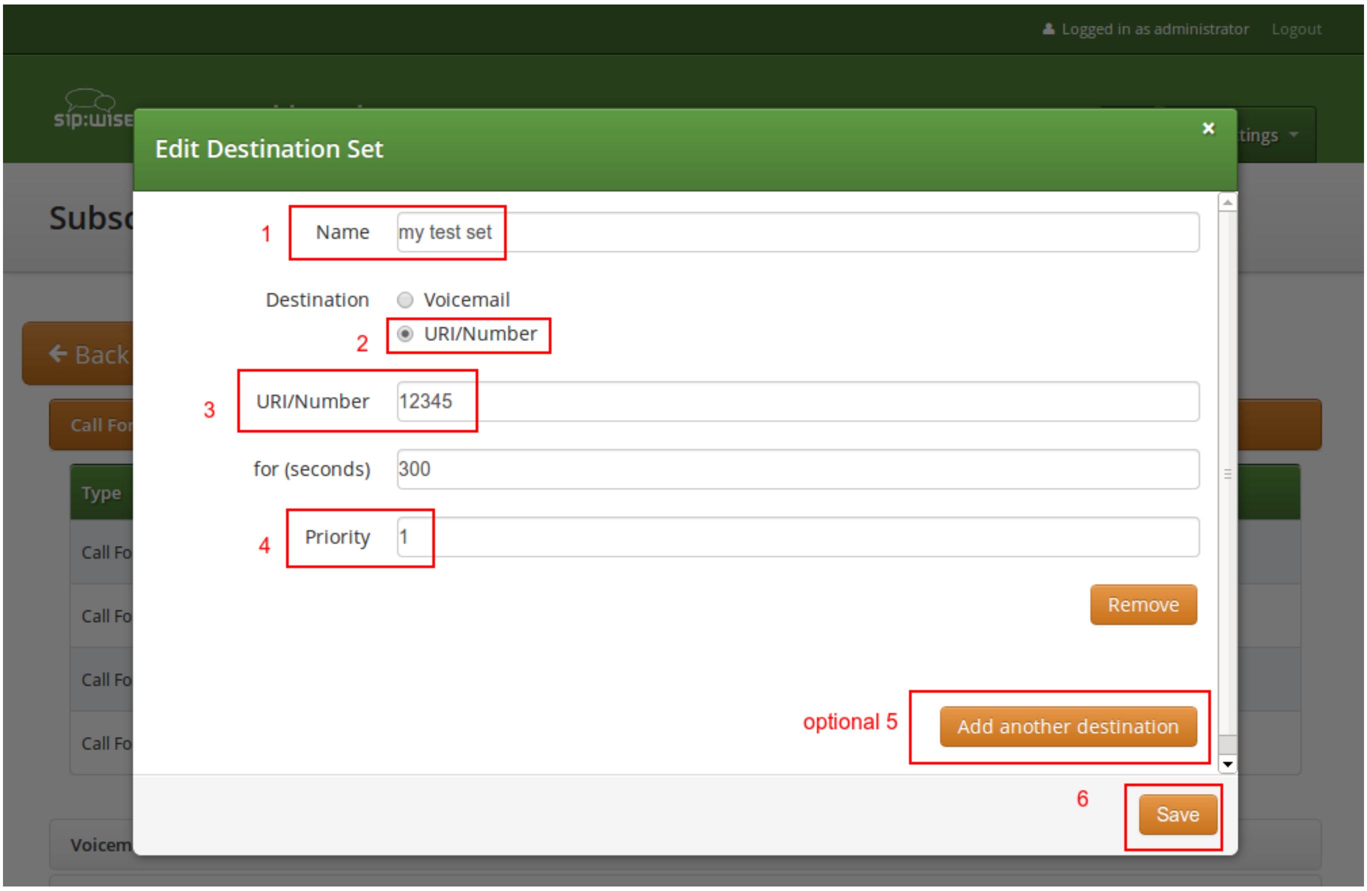
If you want multiple destinations to be executed one after the other, you need to change into the *Advanced View* when editing your call forward. There, you can select multiple *Destination Set*/*Time Set* pairs to be executed.

A *Destination Set* is a list of destinations to be executed one after another.

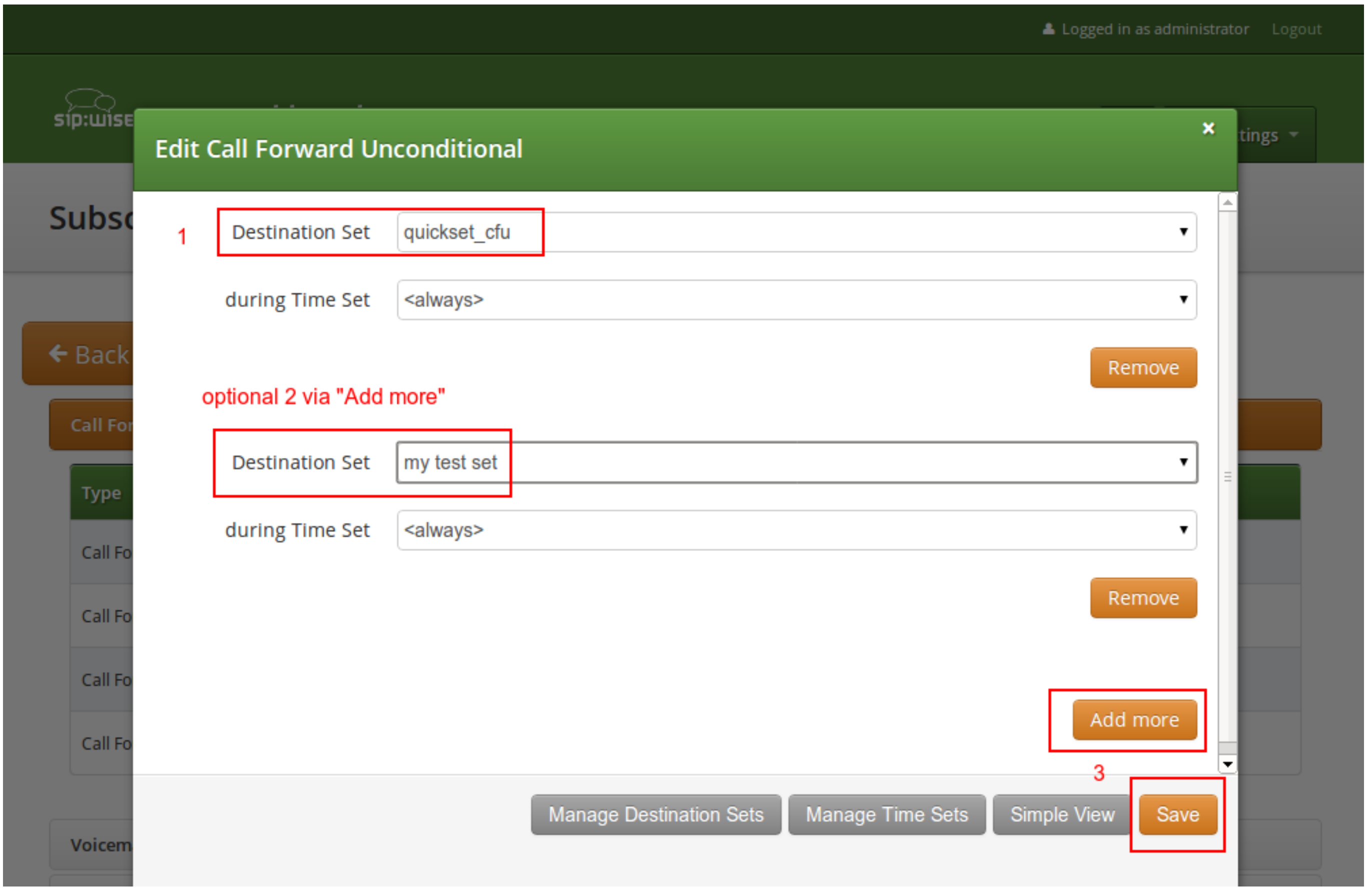
A *Time Set* is a time definition when to execute this *Destination Set*.

5.3.2.1 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.



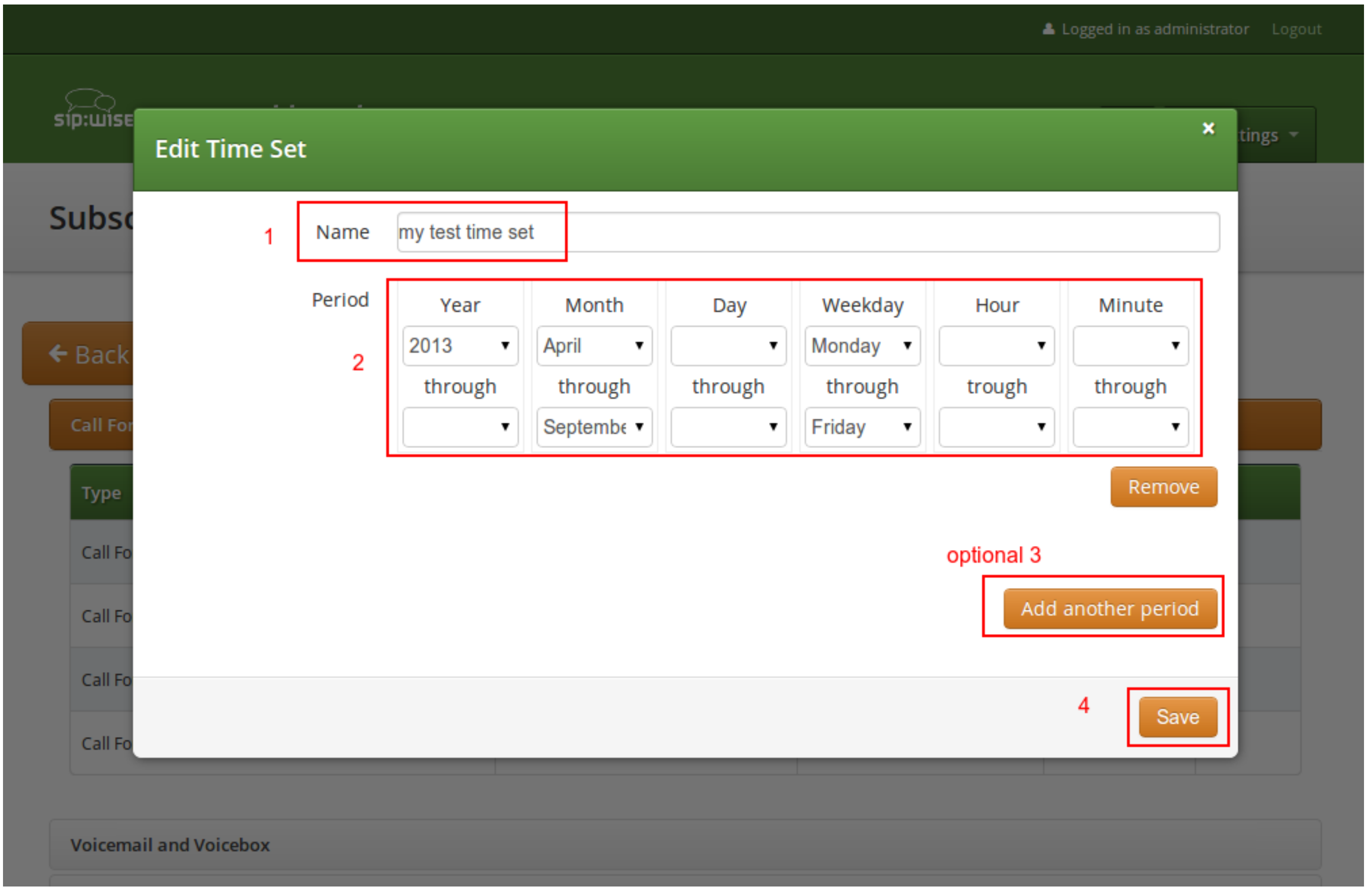
When you close the *Destination Set* Overview, you can now assign your new set in addition or instead of the *quickset_cfu* set.



Press *Save* to store your settings.

5.3.2.2 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.

5.4 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.

5.4.1 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.

5.4.1.1 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 PRO. 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.

5.4.1.2 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.

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.

5.4.1.3 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*.

5.4.1.4 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 solely to find the LNP prefix of the calling peer, which is then stored as *source_lnp_prefix* in the CDR. 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.

5.4.1.5 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.

5.4.1.6 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 2: 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*)
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)

5.4.2 External LNP via LNP API

External LNP relies on the *Sipwise LNP Daemon (lnpd)* which kamailio-proxy is talking to via a defined JSONRPC protocol. The proxy sends the A and B number to *lnpd*, which in the current release translates it to a SIP Message sent to an external server (typically a Squire SIP-to-INAP gateway. This external gateway is performing an SS7 INAP request to fetch the LNP result, which is passed back as a binary blob in a 3xx response to the *lnpd*. The *lnpd* extracts the TCAP body of the response and returns the information back to the proxy.

5.4.2.1 Enabling LNP lookup via API

In order to activate LNP lookup via API during call routing, the feature must be activated in `/etc/ngcp-config/config.yml`. Set these parameters:

- `kamailio→proxy→lnp→enabled` : *yes*
- `kamailio→proxy→lnp→type` : *api*
- `lnpd→enabled` : *yes*

There is a possibility to explicitly allow (whitelist) or deny (blacklist) certain number ranges for which an LNP lookup may be done. The relevant configuration parameters are at `kamailio→proxy→lnp→lnp_request_whitelist` and `kamailio→proxy→lnp→lnp_request_blacklist`. For each entry in the list a POSIX regex expression may be used, see the following example:

```
lnp:
  lnp_request_whitelist:
    - '^9'
    - '^800'
  lnp_request_blacklist:
    - '^1'
    - '^900'
    - '^110'
    - '^112'
```

Interpretation of the above lists (that are based on numbers represented in national format):

- whitelist: *do* LNP lookup for any called number that starts with *9* or *800*
- blacklist: *do not* perform LNP lookup for any called number that starts with *1*, *900*, *110* or *112*

Important

If both whitelist and blacklist are defined, the LNP lookup is only performed when the called number matches any of the whitelist patterns and does not match any of the blacklist patterns.

5.4.2.2 The Redundancy Feature

It is possible to set up *LNP daemon* to provide a kind of redundant service to the Proxy. This means the *LNP daemon* will send its LNP query to more LNP serving nodes that are predefined in a list. (See Configuration of LNP daemon Section 5.4.2.3 chapter for details.) The LNP query may happen in 2 ways:

- **round-robin**: *LNP daemon* sends the query to one of the serving nodes then waits for the response for a configurable timeout. If it does not get the response in time, it sends the LNP query to the next serving node.
- **parallel**: *LNP daemon* sends the query to all of the serving nodes then waits for the response, and will accept the first response that it receives.

5.4.2.3 Configuration of Sipwise LNP Daemon

LNP daemon takes its active configuration from `/etc/ngcp-lnpd/config.yml` file. The file is generated automatically — when a new NGCP configuration is applied (`ngcpcfg apply...`) — from the main Sipwise NGCP configuration file: `/etc/ngcp-config/config.yml` and a template: `/etc/ngcp-config/template/etc/ngcp-lnpd/config.yml.tt2`. System administrators are only expected to modify the `lnpd.config` section of main configuration file `/etc/ngcp-config/config.yml`.

A sample *LNP daemon* configuration file (`/etc/ngcp-lnpd/config.yml`) looks like:

```
daemon:
        json-rpc:
                ports:
                        - 54321
                        - 12345
                interfaces:
                        - 127.0.0.1
                        - 192.168.1.90
                        - ::1

        sip:
                port: 5095
                address: 0.0.0.0

        threads: 4
        foreground: false
        pidfile: /tmp/lnpd.pid
        loglevel: 7

instances:
        default:
                module: sigtran
                destination: 192.168.1.99
                from-domain: test.example.com
                headers:
```

```
                    - header: INAP-Service-Key
                      value: 2
            reply:
                    tcap: raw-tcap
    redundant:
            module: sigtran
            destinations:
                    - 192.168.1.99
                    - 192.168.1.95
                    - 192.168.1.90
            mechanism: round-robin
            retry-time: 30
            timeout: 5
            from-domain: test.example.com
            headers:
                    - header: INAP-Service-Key
                      value: 2
            reply:
                    tcap: raw-tcap
    parallel:
            module: sigtran
            destinations:
                    - 192.168.1.99
                    - 192.168.1.95
                    - 192.168.1.90
            mechanism: parallel
            retry-time: 30
            timeout: 10
            from-domain: test.example.com
            headers:
                    - header: INAP-Service-Key
                      value: 2
            reply:
                    tcap: raw-tcap
    mock1:
            module: mock-tcap
            numbers:
                    - number: '4311003'
                      routing-number: '4318881003'
            reply:
                    tcap: raw-tcap
```

The corresponding NGCP main configuration file contains:

```
daemon:
  foreground: 'false'
  json-rpc:
    ports:
```

```
        - '54321'
        - '12345'
    loglevel: '7'
    sip:
      port: '5095'
    threads: '4'
  instances:
  << These are the same entries as in /etc/ngcp-lnpd/config.yml file >>
```

Description of configuration parameters in `/etc/ngcp-config/config.yml` file

- `daemon` section:
  - `foreground`: determines if the LNP daemon runs as foreground or background process
  - `json-rpc.ports`: port numbers where LNP daemon listens for incoming JSONRPC requests from NGCP Proxy
  - `loglevel`: how detailed information LNP daemon writes in its log file
  - `sip.port`: listening port number used for SIP sessions with LNP serving nodes; LNP daemon will listen on first available (shared) IP address that is taken from `/etc/ngcp-config/network.yml` file
  - `threads`: number of threads LNP daemon will use internally; this value determines how many requests the daemon can serve in parallel
- `instances` section: at least one `default` instance must be defined here. Others are also useful for providing redundancy, please check `redundant` and `parallel` entries above.
  - `module`: only `sigtran` is used for normal operations

    > **Important**
    >
    > The module `mock-tcap` is only meant for developers. In this case the LNP daemon does not produce a SIP request that it sends to LNP serving nodes, but instead it uses the `numbers` parameter to match a called number with a routing number. The `numbers` parameter contains a list of number — routing-number pairs and is used as a database for number lookups. Finally LNP daemon returns the routing number as a response on LNP query.

  - `destinations`: list of nodes to which LNP daemon sends the LNP query
  - `mechanism`: either `parallel` or `round-robin`, defining the method of redundant queries
  - `retry-time`: a period of time in seconds while LNP daemon considers an LNP serving node being unreachable after an LNP query timeout
  - `timeout`: the period of time while LNP daemon waits for a response on an LNP query from one of the LNP serving nodes *PLEASE NOTE*: `retry-time` and `timeout` are used with both the parallel and the round-robin redundancy methods
  - `from-domain`: the domain that will be used in SIP *From* header when LNP daemon sends the LNP query
  - `headers`: this is a list of `header` name — `value` pairs; these custom headers will be included in SIP request that LNP daemon sends to an LNP serving node
  - `reply.tcap`: determines the format of reply sent to NGCP Proxy; currently only `raw-tcap` is supported, which means LNP daemon will not decode the TCAP response it gets from an LNP serving node but it forwards the raw TCAP message body

5.5 Header Manipulation

5.5.1 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 and push 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.

5.5.2 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`* and push the changes .

5.5.3 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)

5.6 SIP Trunking with SIPconnect

5.6.1 User provisioning

For the purpose of external SIP-PBX interconnect with sip:provider PRO 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.

5.6.2 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).

5.6.3 Number manipulations

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

5.6.3.1 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 4.6. 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 PRO 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 (see Section 15.1.2) 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.

5.6.3.2 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

5.6.3.3 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.

5.6.3.4 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.

5.6.4 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 PRO will take care of routing the aliases to the AOR with *e164_to_ruri* preference.

5.6.4.1 Trusted sources

If a SIP-PBX can not be configured to perform digest authentication, you can enable IP-based authentication in sip:provider PRO for it. For this go to the subscriber's preferences (*Details→Preferences→Trusted Sources*) and specify the IP address of the SIP-PBX in the *Source IP* field.

When this feature is configured, the sip:provider PRO will authenticate all calls of this user that arrive from the specified IP without sending a challenge.

Important

If the same IP address is mistakenly specified as trusted for different subscribers, the sip:provider PRO will not know which subscriber should be responsible for the call and will randomly select one.

5.7 Trusted Subscribers

In some cases, when you have a device that cannot authenticate itself towards sip:provider PRO, you may need to create Trusted Subscriber. Trusted Subscribers use IP-based authentication and they have a Permanent SIP Registration URI in order to receive messages from sip:provider PRO. In order to create a Trusted Subscriber you just need to create a normal subscriber, then Create a Permanent Registration via (*Subscribers→Details→Registered Devices→Create Permanent Registration*) and also you need to add the devices IP as Trusted Source in your subscriber's preferences (*Details→Preferences→Trusted Sources*). In this way, all messages coming from your device IP will be trusted (and authenticate just via the source IP), on the other side all the SIP messages to your devices will be sent to the SIP URI specified in the Permanent Registration.

5.8 Fax Server

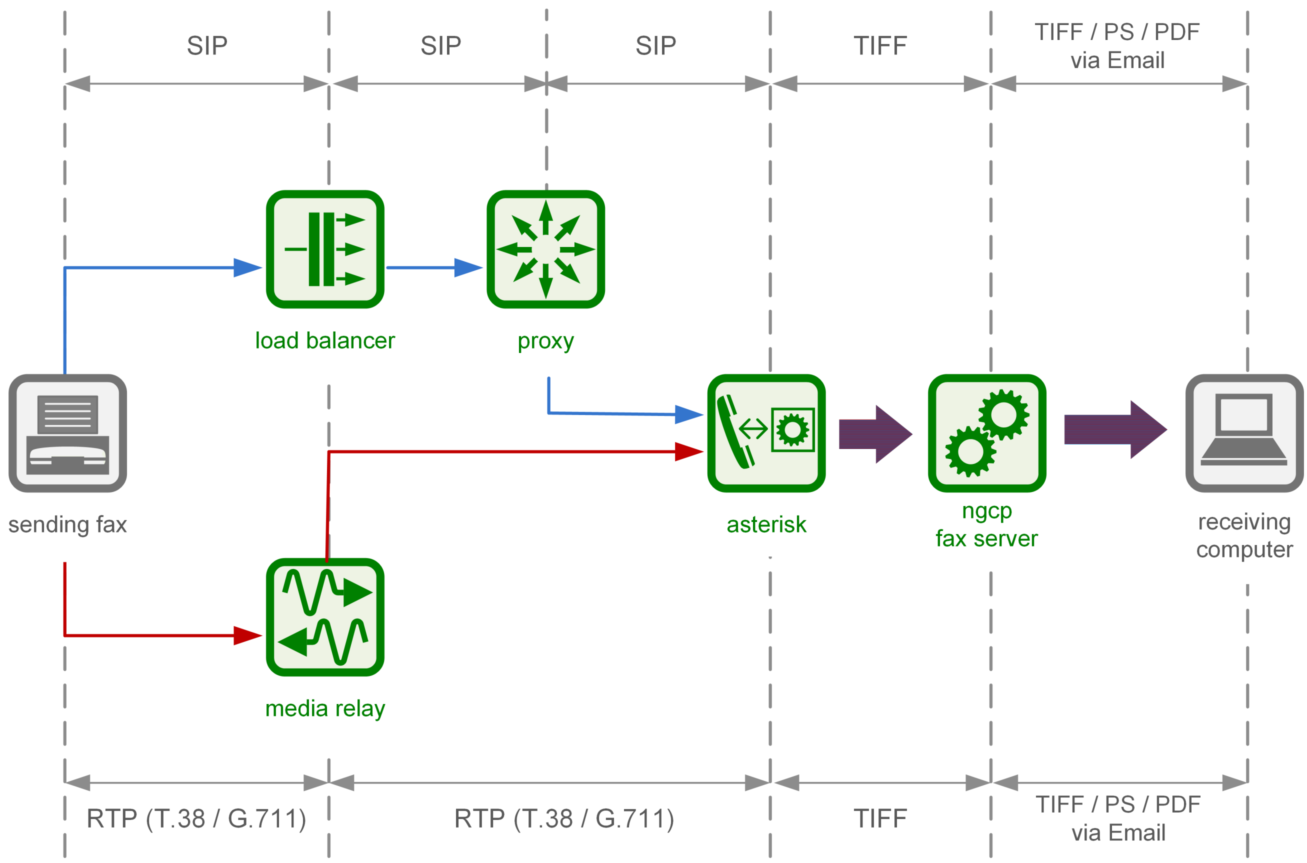
There is a Fax Server included in the sip:provider PRO. The following sections describe its architecture.

The Fax Server is included on the platform and requires no additional hardware. It supports both T38 and G711 codecs and provides a cost-effective paper-free office solution.

For the details of Fax Server configuration options, please see Faxserver Configuration Appendix C chapter in this handbook.

5.8.1 Fax2Mail Architecture

To receive faxes via email, a phone call from a sender is connected to the fax application module (Asterisk + NGCP Fax Server) on the sip:provider PRO. The received fax document is converted to the format the receiver has configured (either PS, PDF or TIFF) via the components outlined in the figure below. The email is delivered to one or more configured addresses.

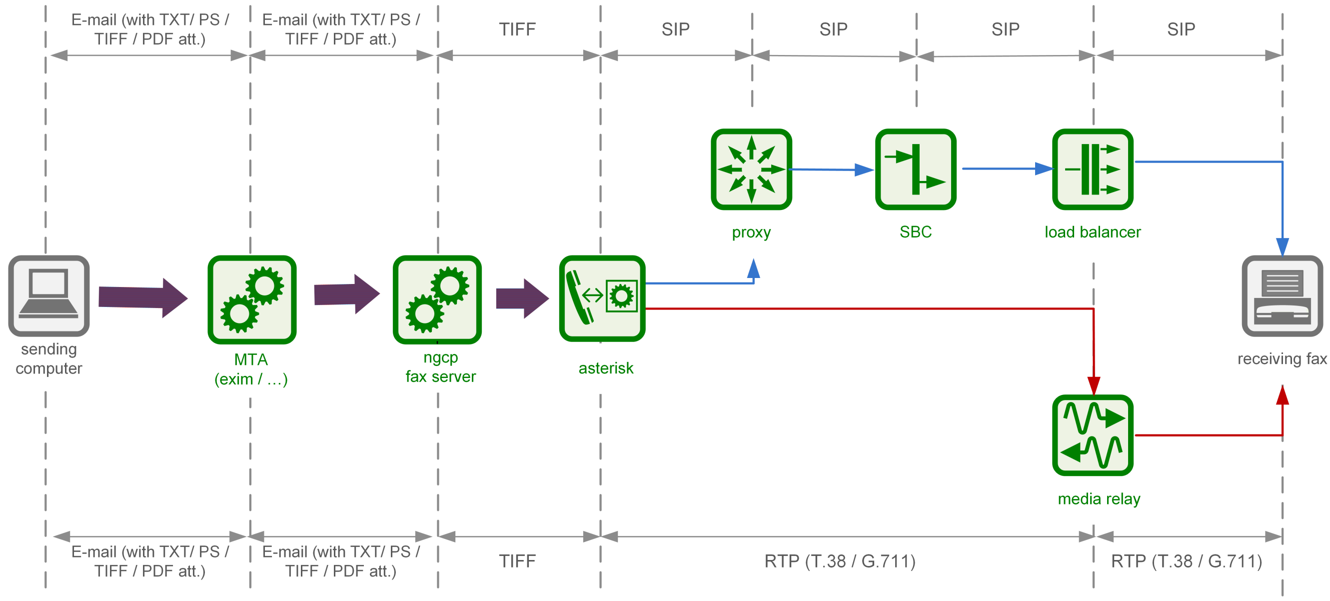


5.8.2 Sendfax and Mail2Fax Architecture

To send faxes via the sip:provider PRO a sender can use any email client or an interface such as Webfax or REST API.

Currently, supported formats are TXT, PS, TIFF and PDF.

The document is sent to the NGCP Fax Server instance on the sip:provider PRO. Once successfully queued by the fax server, it is converted to an internal TIFF format and is sent via the components outlined in the below figure to the specified phone number. Of course, a fax device that can receive the document must be connected on the destination side.



5.9 Voicemail System

5.9.1 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*.

5.9.1.1 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.

5.9.1.2 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 `aster` isk.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.

5.9.2 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

5.9.3 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.

5.9.3.1 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".

5.9.3.2 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."

5.9.3.3 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.

5.9.4 Folders

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

5.9.4.1 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.

5.9.5 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.

5.9.5.1 Listening to New Messages

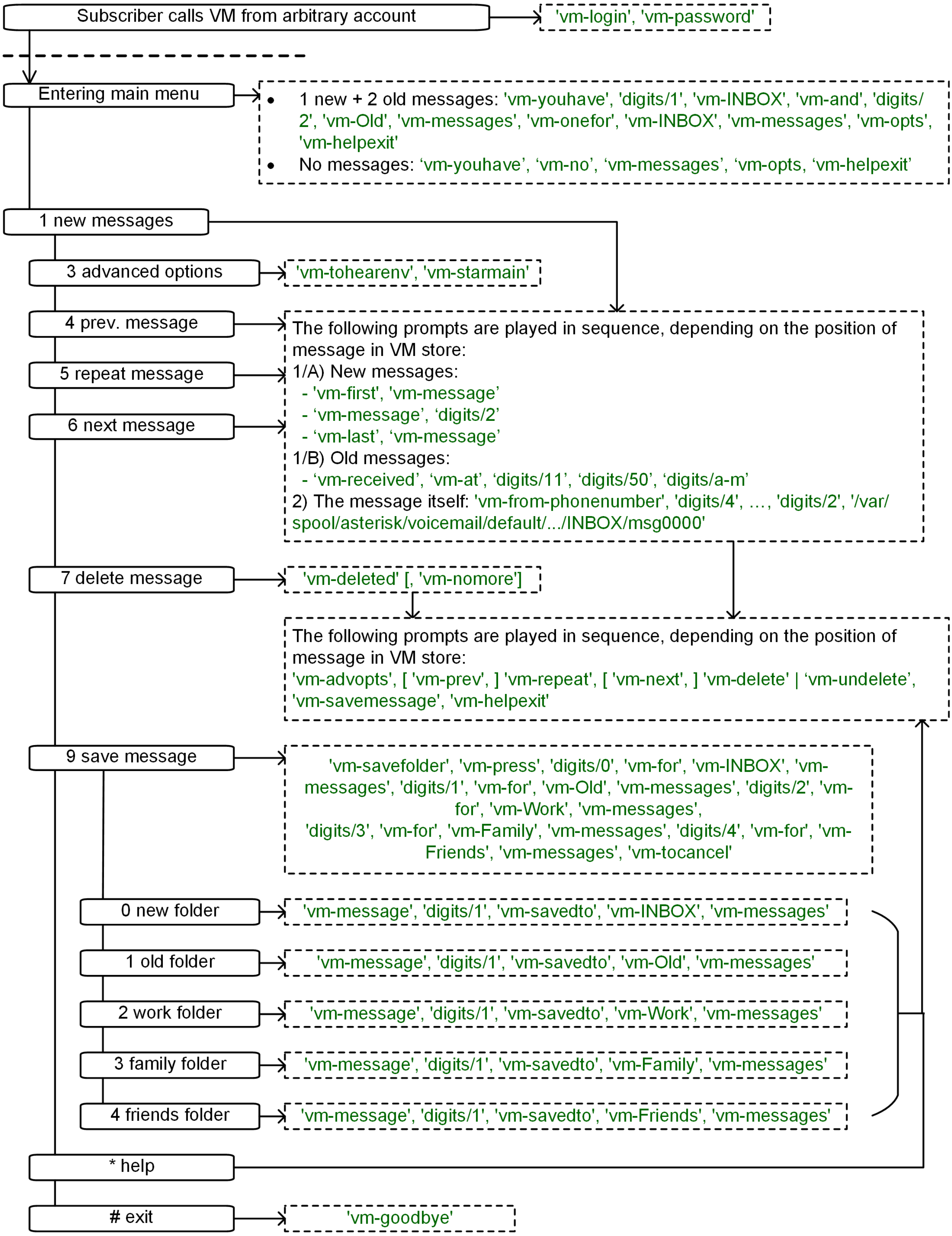


Figure 16: Flowchart of Listening to New Messages

5.9.5.2 Changing Voicemail Folders

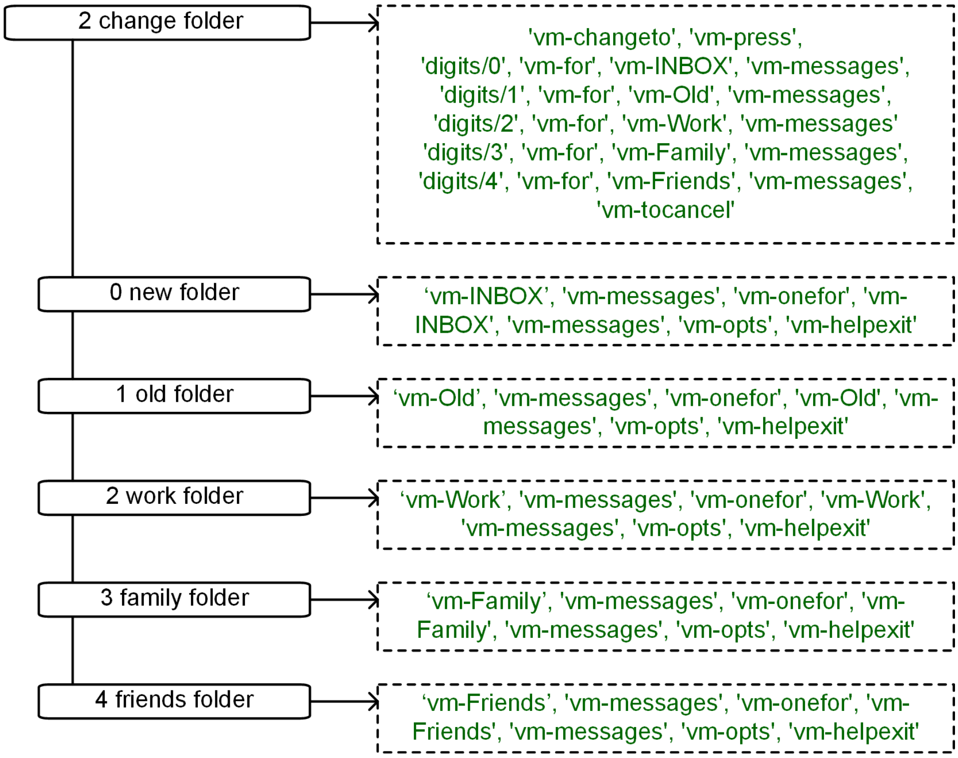


Figure 17: Flowchart of Changing Voicemail Folders

5.9.5.3 Mailbox Options

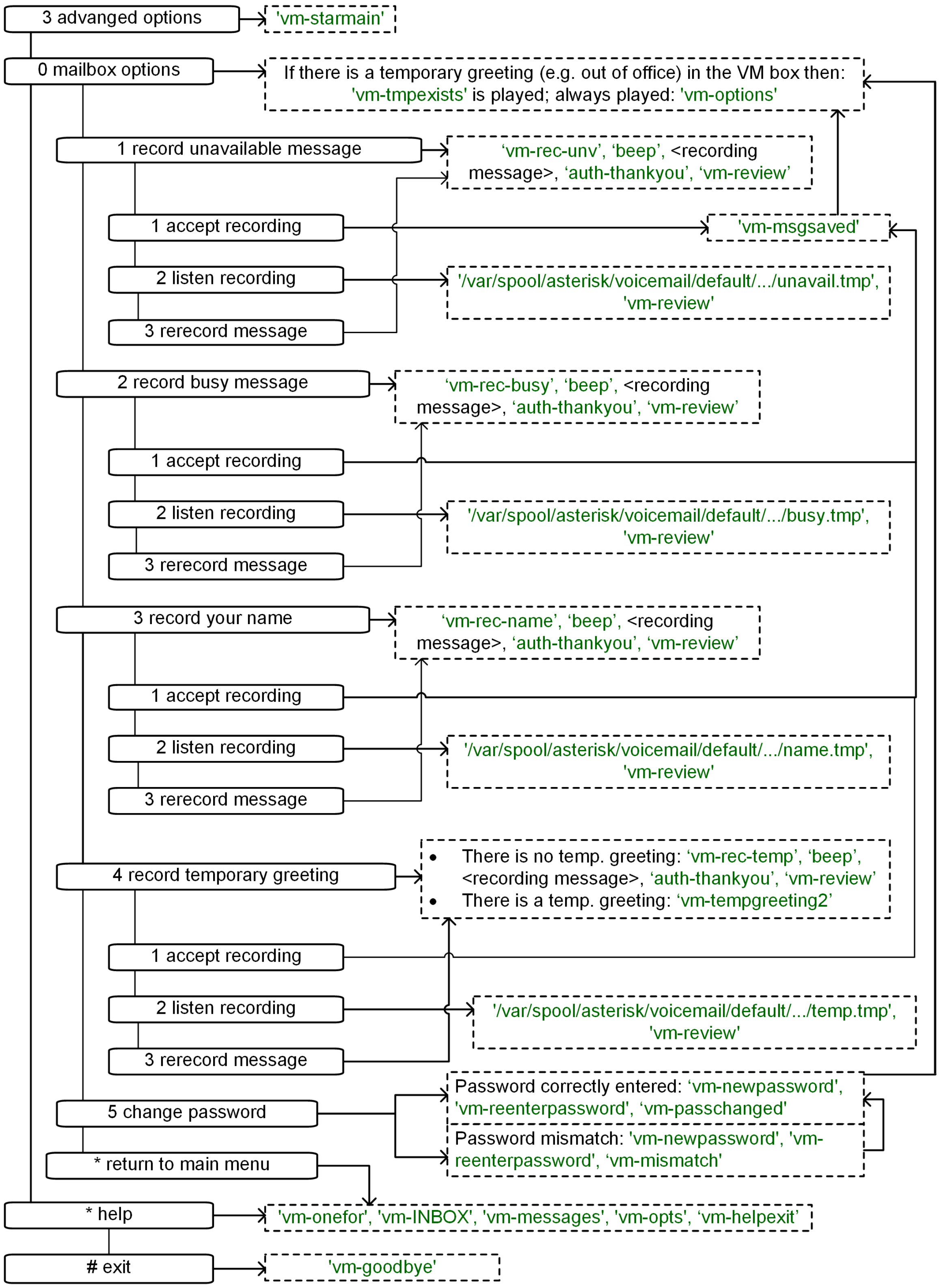


Figure 18: Flowchart of Changing Mailbox Options

5.9.5.4 Leaving a Message

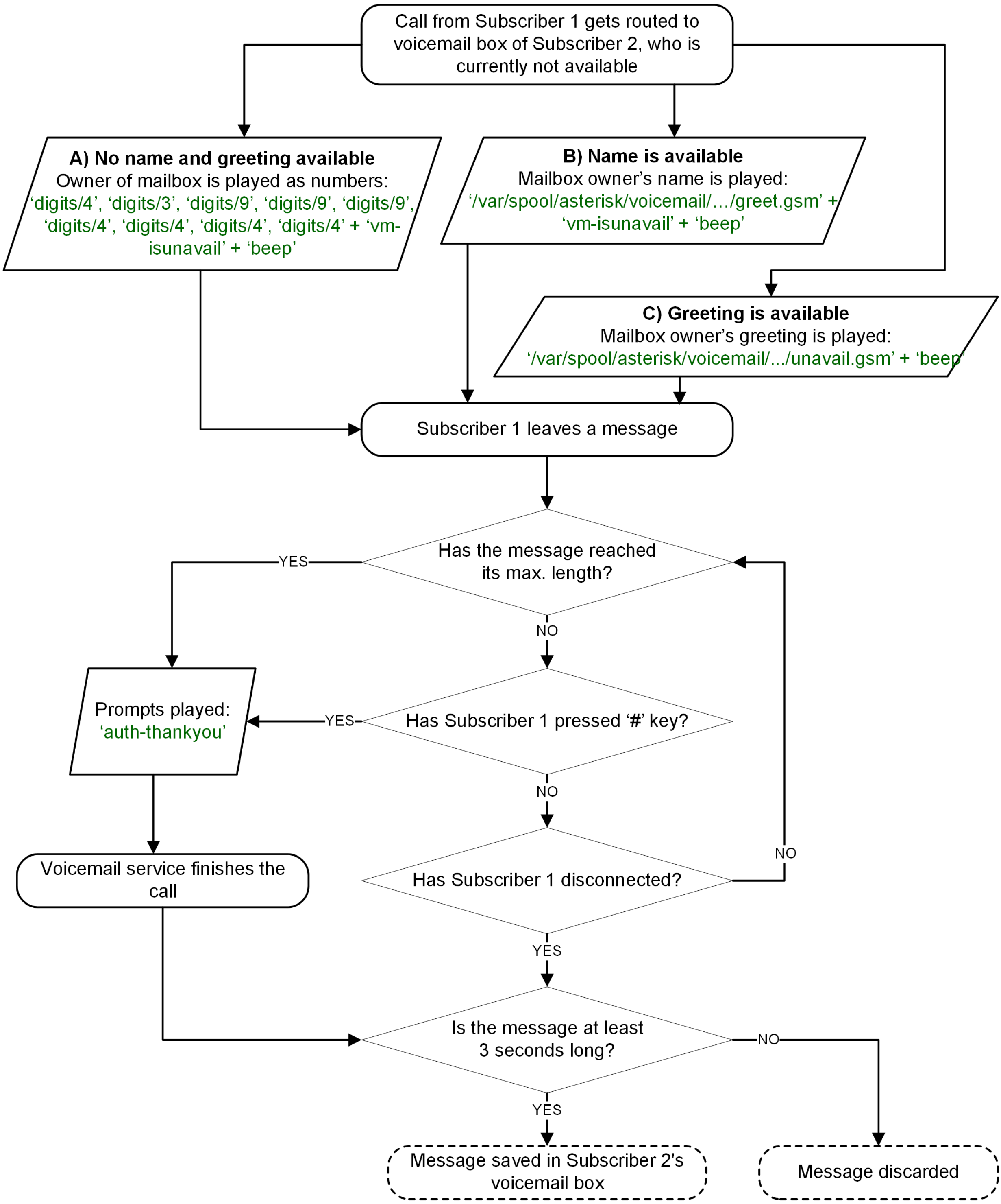
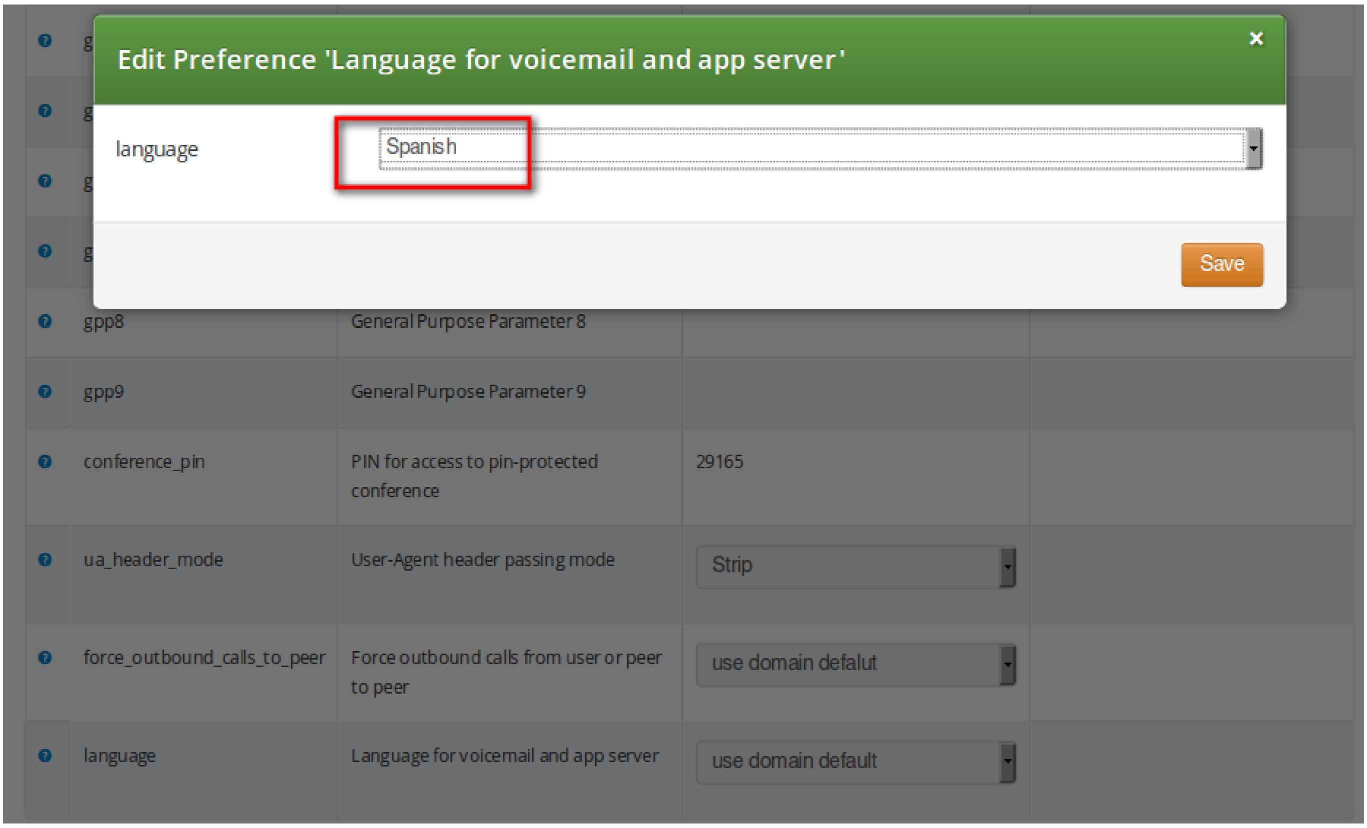


Figure 19: Flowchart of Leaving a Voice Message

5.10 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*.

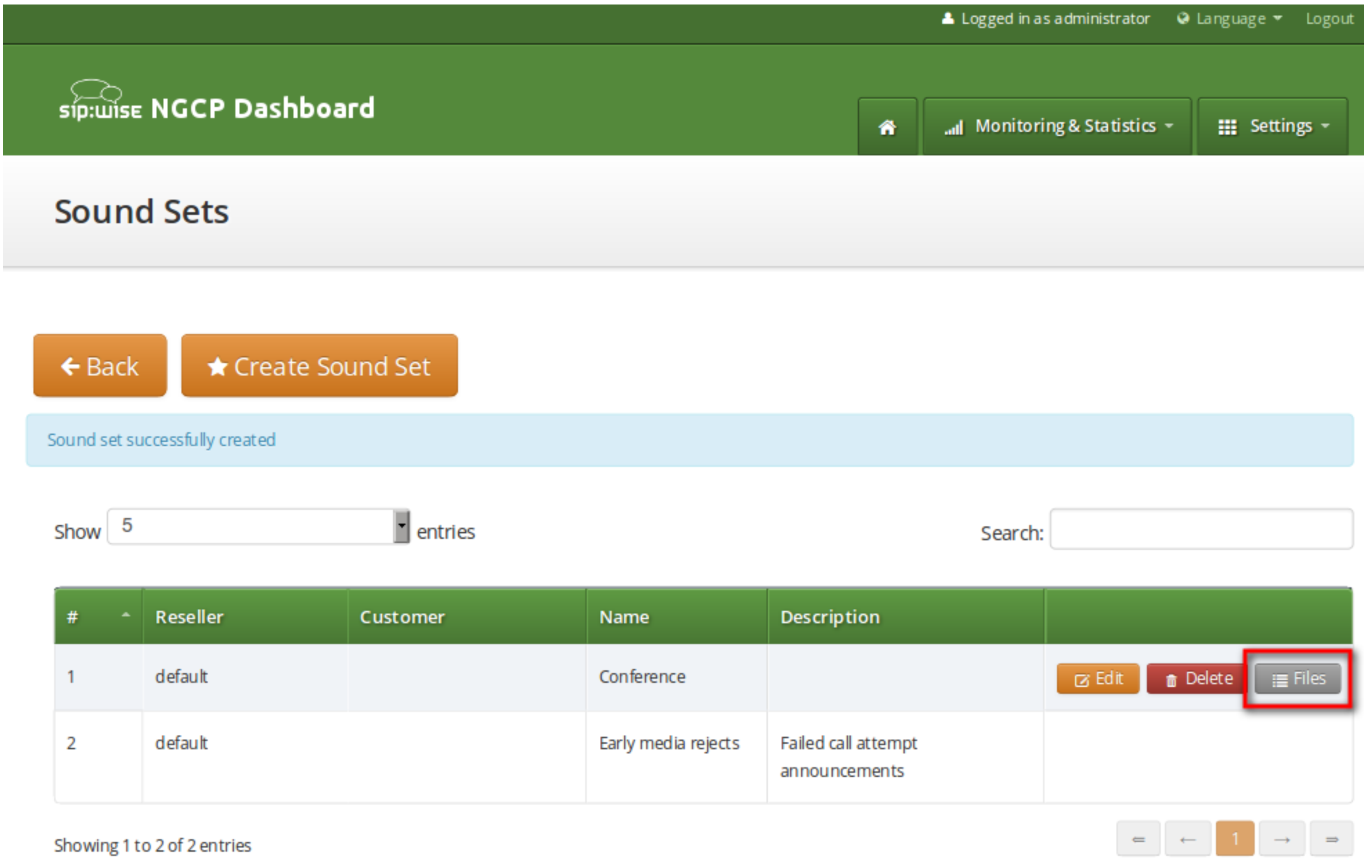


The sip:provider PRO provides the pre-installed prompts for the Voicemail in the English, Spanish, French and Italian languages and the pre-installed prompts for the Vertical Service Codes IVRs in English only.

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.

5.11 Sound Sets

The sip:provider PRO 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.

5.11.1 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 PRO will fallback to sending the error response code back to the user.

Table 3: Early Reject Sound Sets

Handle	Description	Message played
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* subscriber preferences)	Your call is blocked by the number you are trying to reach.
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* 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)	Your call to the number you are trying to reach is not permitted.

Table 3: (continued)

Handle	Description	Message played
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.
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.
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)	The network you are trying to reach is currently busy. Please try again later.
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.

Table 3: (continued)

Handle	Description	Message played
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.
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.
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.
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_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_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.
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.
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.
voicebox_unavailable	Announcement played on call to voicebox if the voicemail server is not configured (system operation is impaired)	The voicemail of the number you are trying to reach is currently not available. Please try again later.
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.
emergency_geo_unavailable	Announcement played when emergency destination is dialed but the destination is not provisioned for the location of the user.	The emergency number you have dialed is not available in your region.

Table 3: (continued)

Handle	Description	Message played
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.

5.12 Conference System

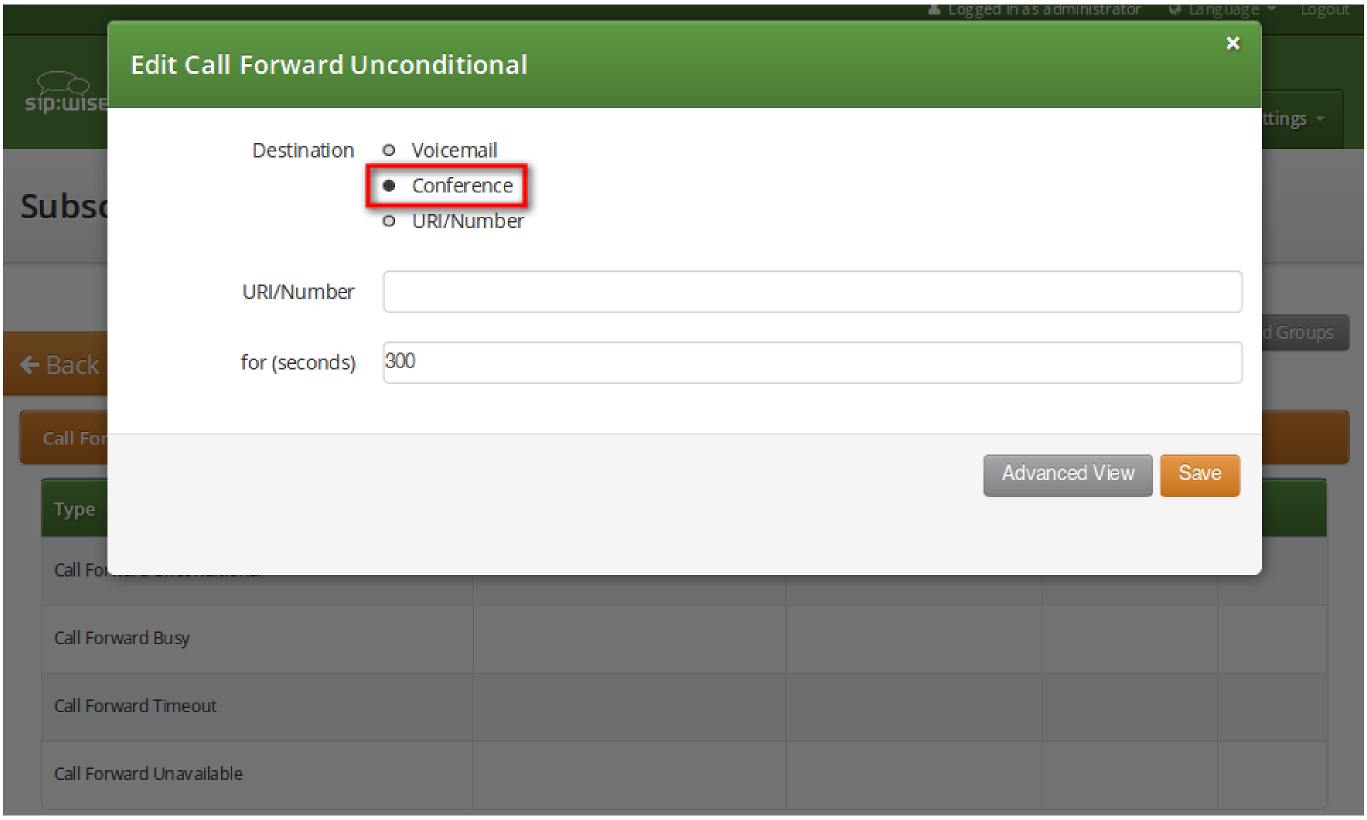
The sip:provider PRO provides the simple pin-protected conferencing service built using the SEMS DSM scripting language. Hence it is open for all kinds of modifications and extensions.

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

5.12.1 Configuring Call Forward to Conference

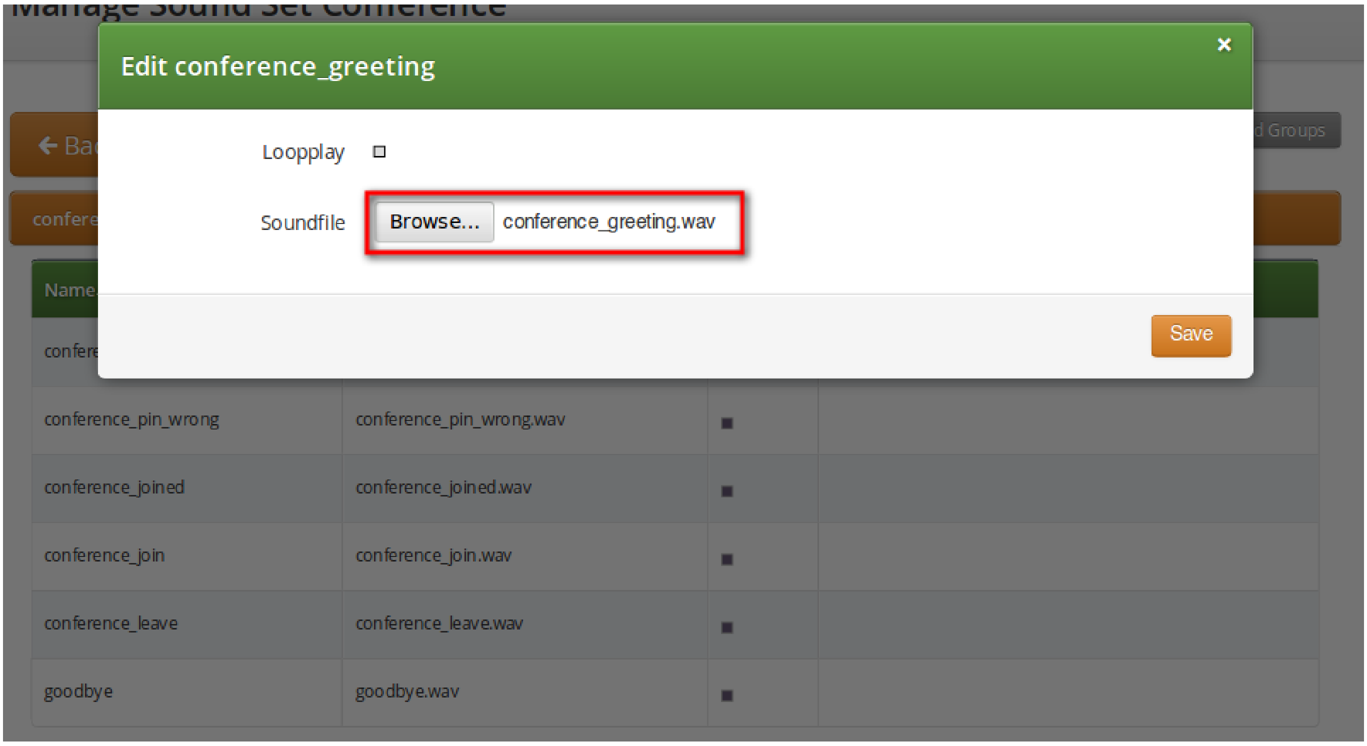
Go to your *Subscriber Preferences* and click *Edit* on the Call Forward Type you want to set (e.g. *Call Forward Unconditional*).



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.

5.12.2 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 4: 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).

5.12.3 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.

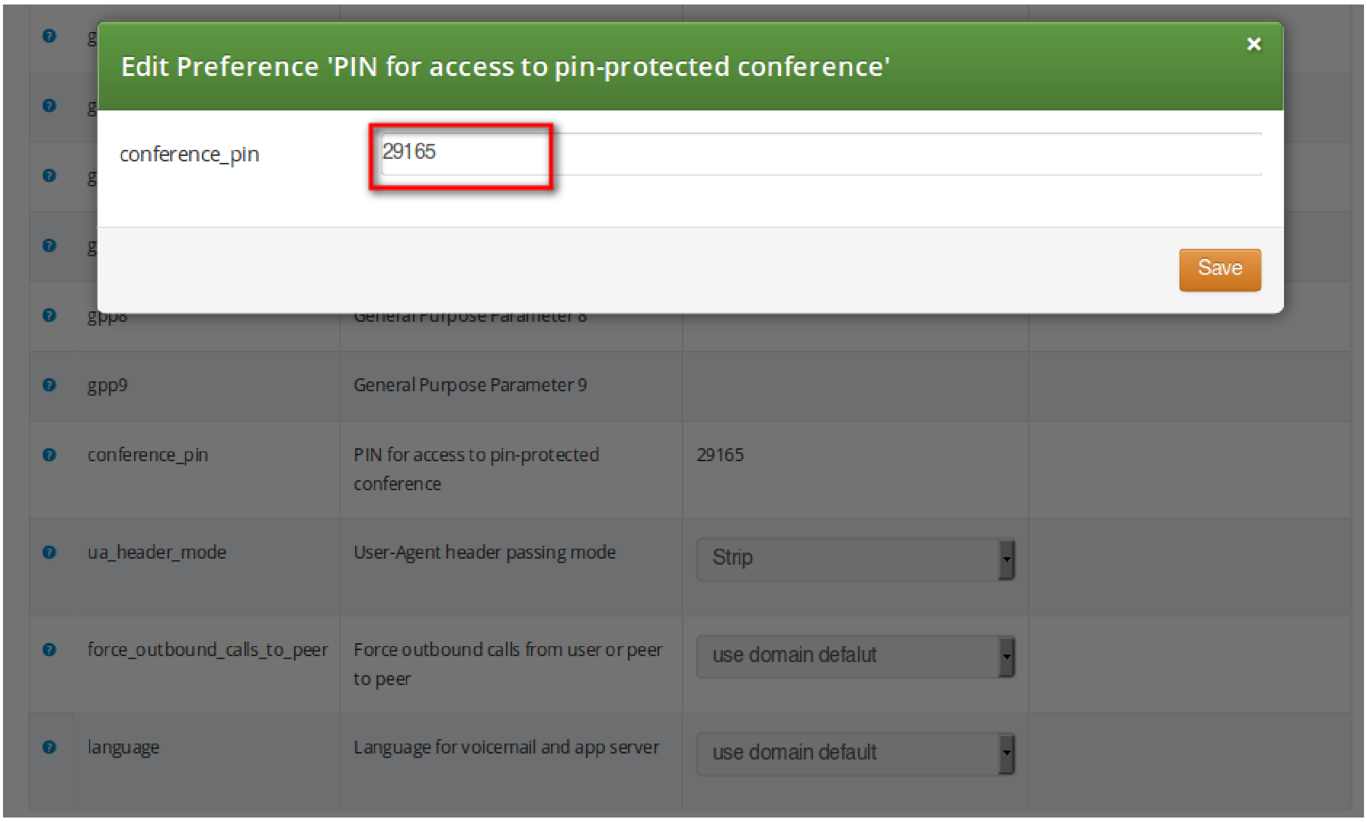


Figure 20: Setting Conference PIN

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.

5.12.4 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.

5.12.4.1 Conference Flowchart with PIN Validation

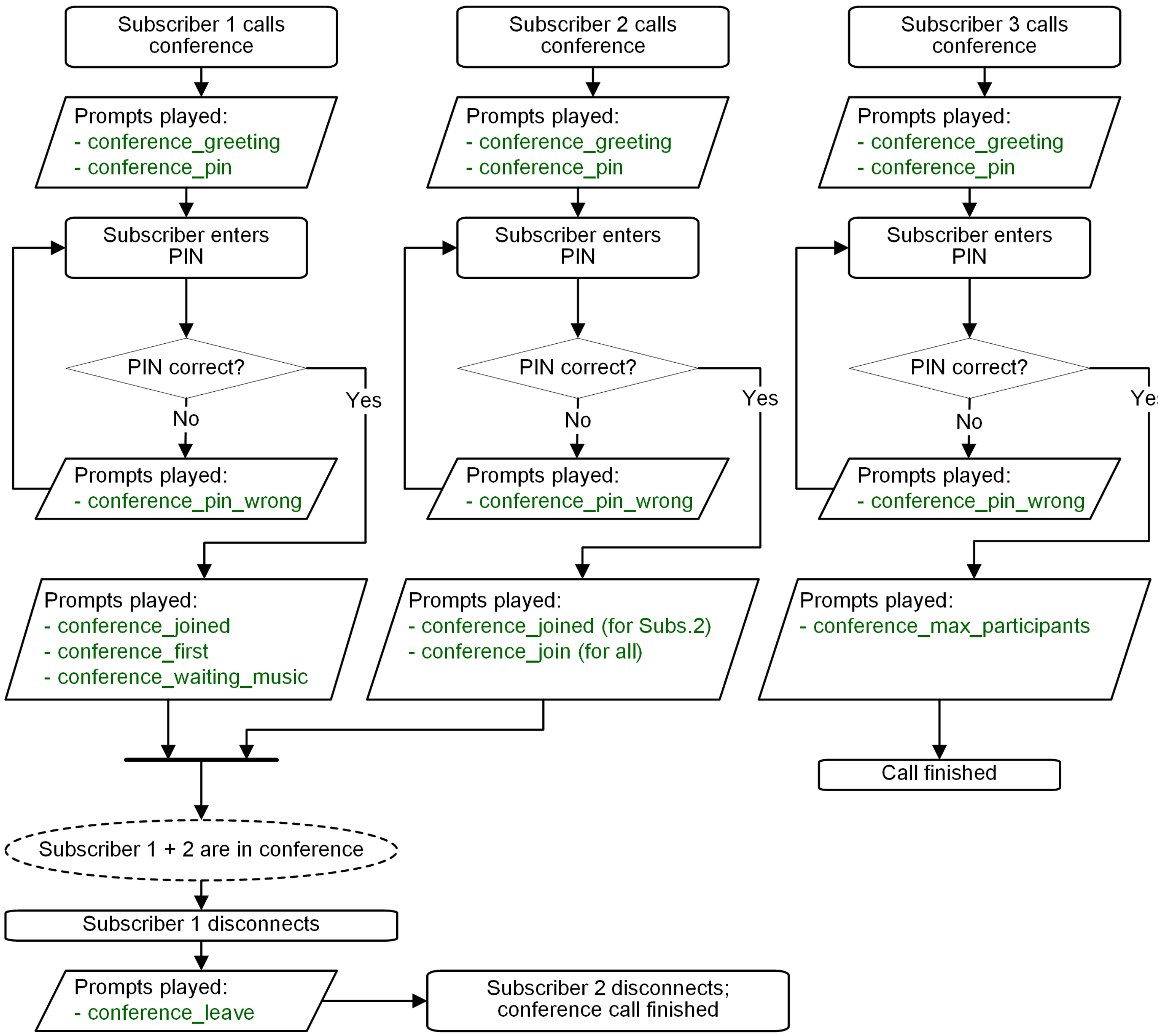


Figure 21: Flowchart of Conference with PIN Validation

5.12.4.2 Conference Flowchart without PIN

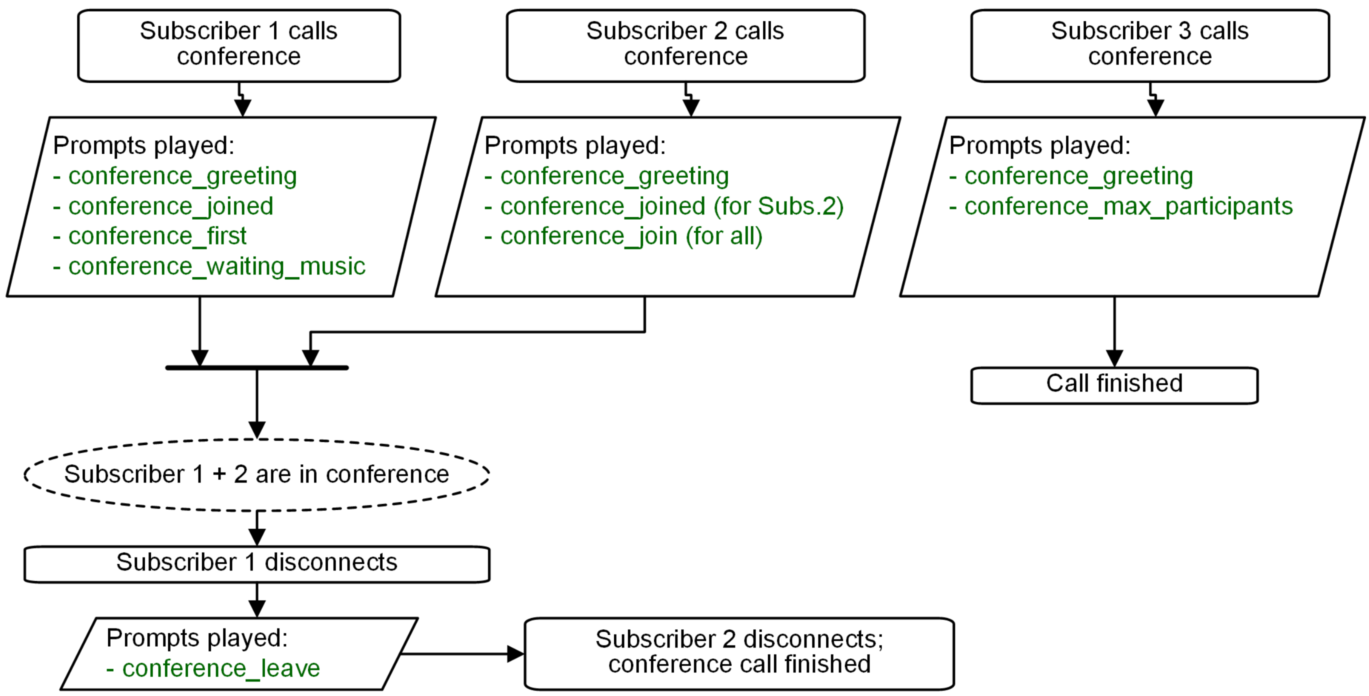


Figure 22: Flowchart of Conference without PIN

5.13 Malicious Call Identification (MCID)

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

5.13.1 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.

5.13.2 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.

5.13.3 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.

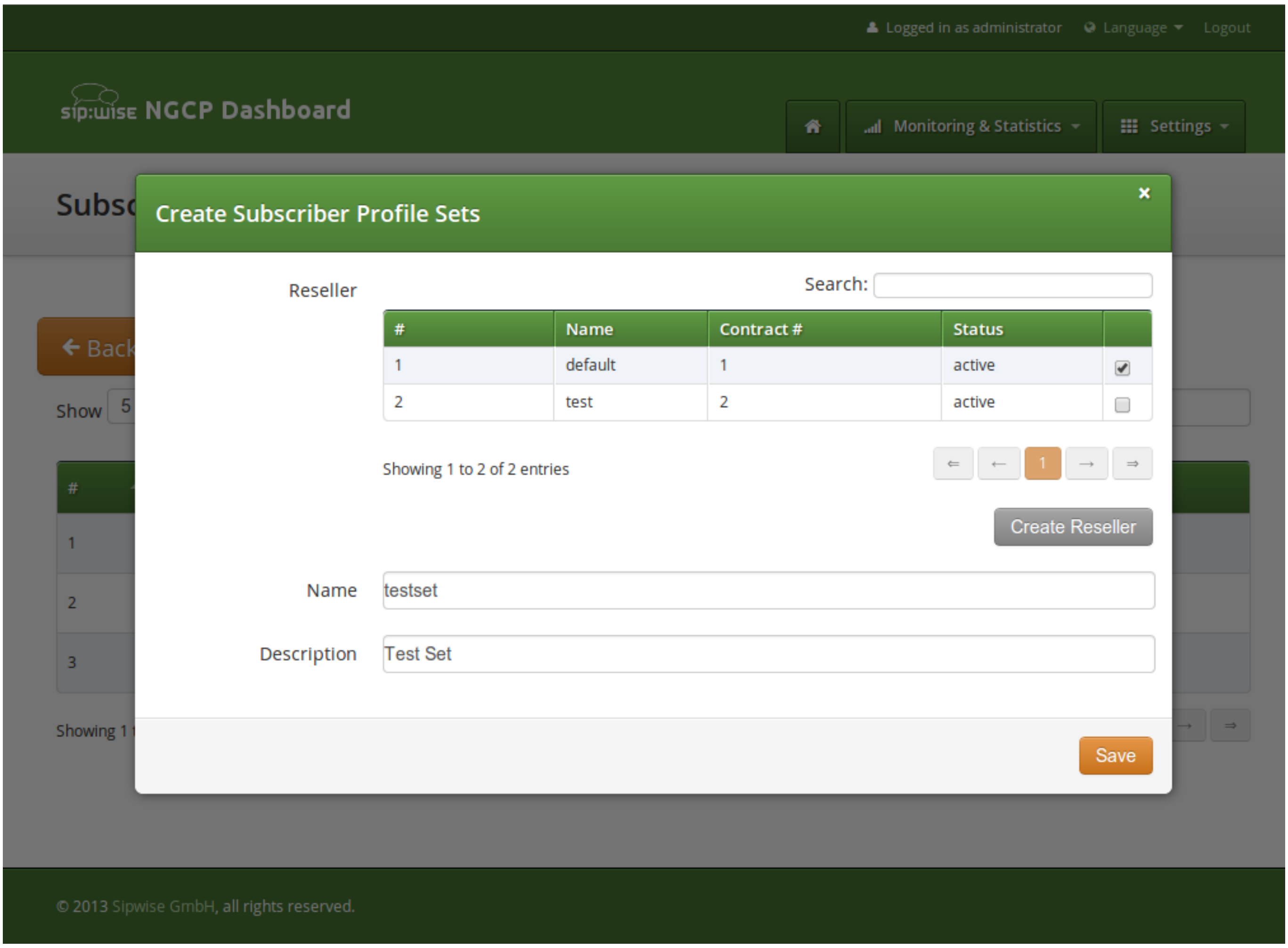
5.14 Subscriber Profiles

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

5.14.1 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.

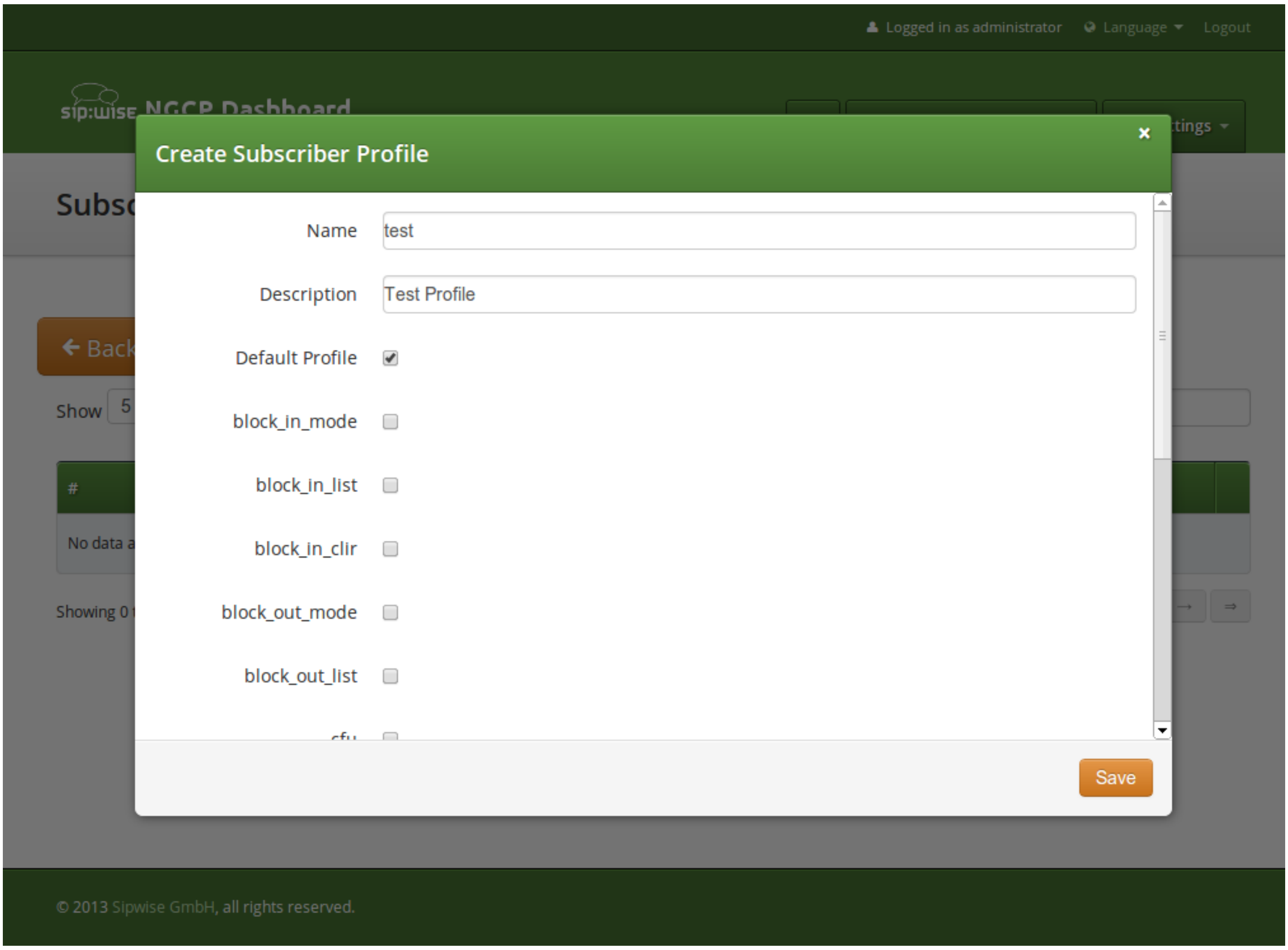
Profile Sets can be defined in *Settings→Subscriber Profiles*. To create a new Profile Set, click *Create Subscriber Profile Set*.



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.

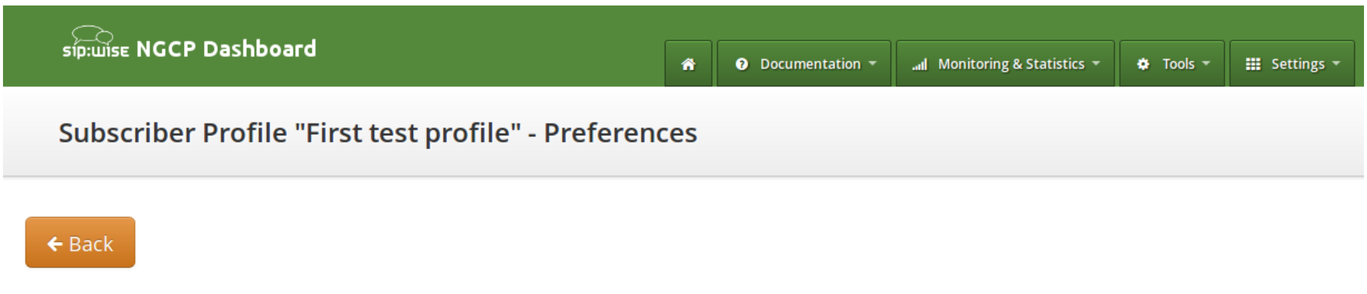
Profiles within a Profile Set can be created by clicking the *Create Subscriber Profile* button.



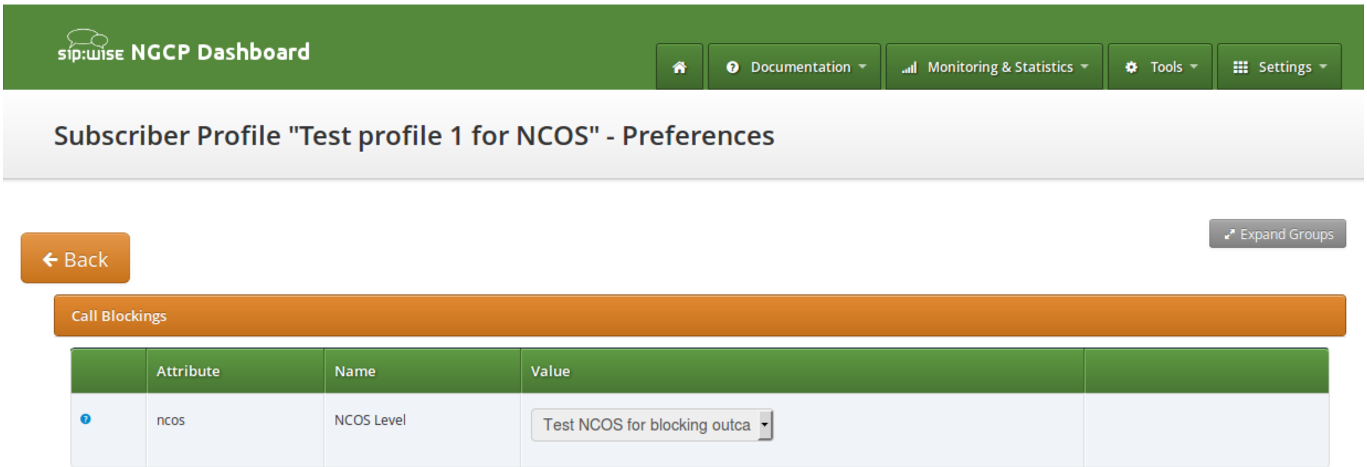
Checking the *Default Profile* option causes this profile to get assigned automatically to all subscribers, who have the profile set assigned. Other options define the user preferences which should be made available to the subscriber.

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:



5.15 SIP Loop Detection

In order to detect a SIP loop (incoming call as a response for a call request) sip:provider PRO 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

5.16 Call-Through Application

Call-through allows telephony client to dial into an IVR system and specify (in two-stage dialing fashion) a new destination number which is then dialed by the sip:provider PRO to connect the client to the destination. As the call-through system needs to be protected from unauthorized use, a list of CLIs which are allowed to use the call-through system is stored in the sip:provider PRO platform.

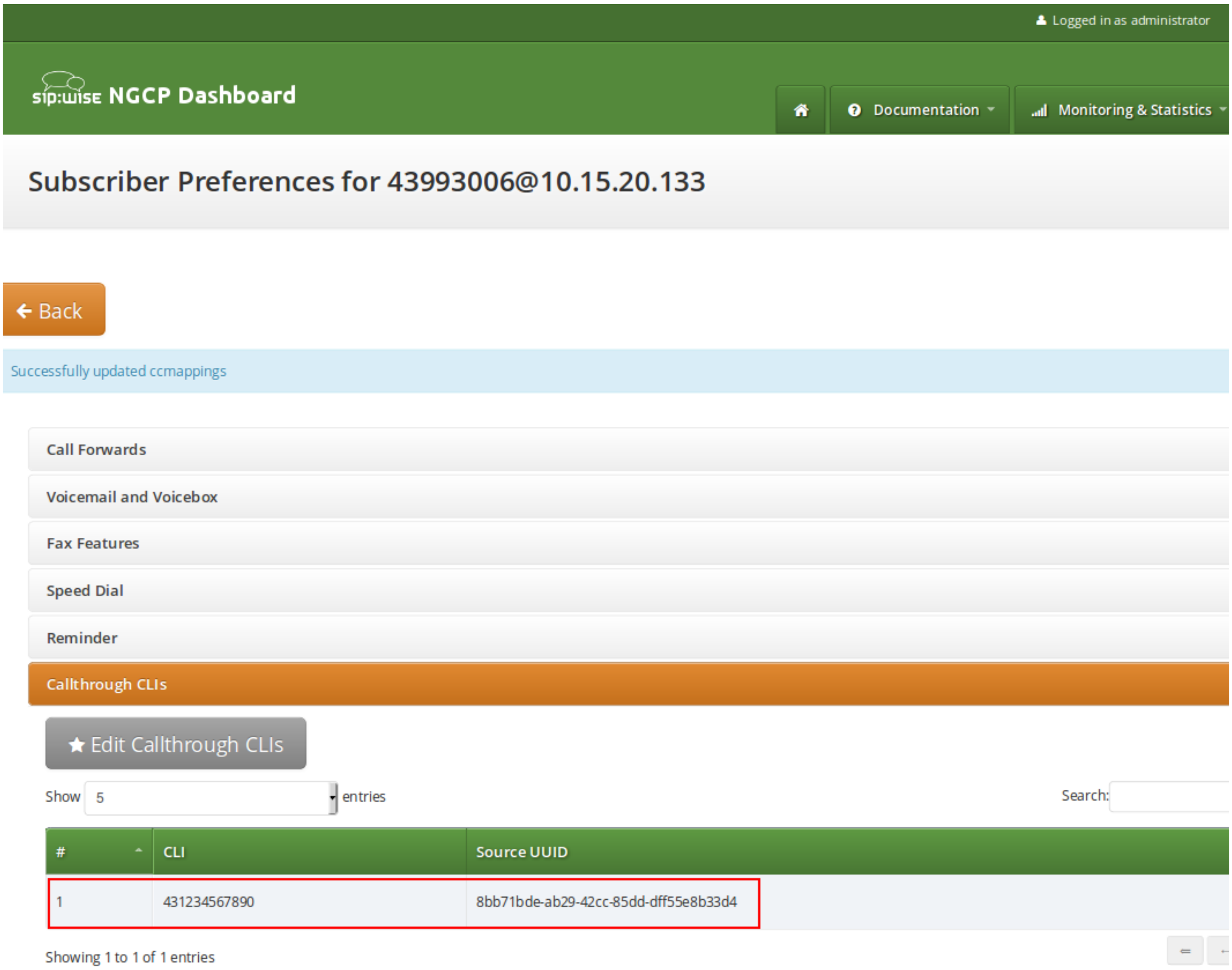
Table 5: Call-Through Mappings

Column	Description
uuid	The internal UUID of the call-through subscriber
auth_key	Authentication key (CLI)
source_uuid	The internal UUID of the subscriber that is authorized for outgoing call leg (same as uuid in call-through scenario)

5.16.1 Administrative Configuration

5.16.1.1 Subscriber provisioning

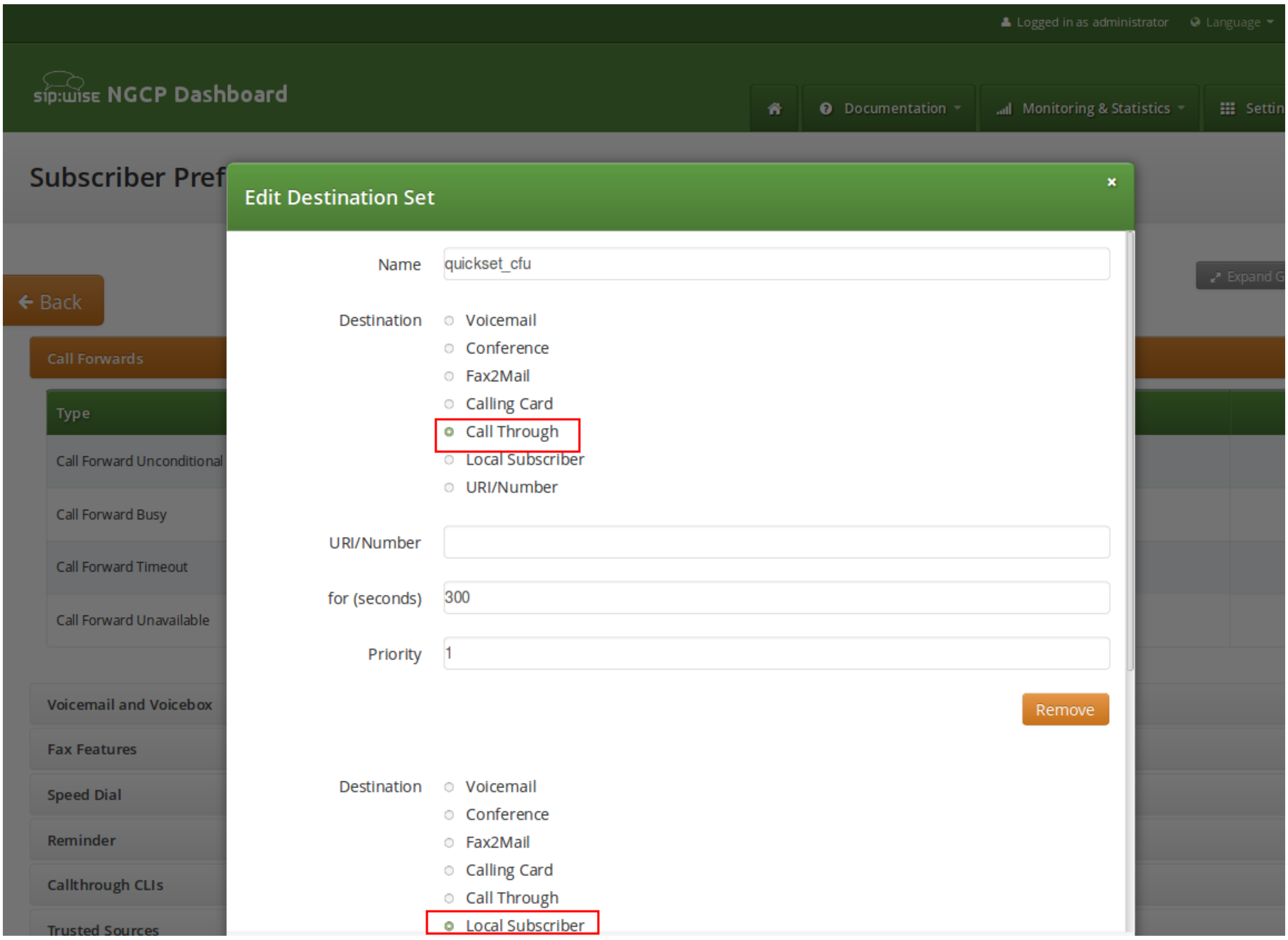
In order to manage the call-through CLIs for subscriber, navigate to *Settings→Subscribers*, search for the subscriber you want to edit, press *Details* and then *Preferences*, scroll down to the *Callthrough CLIs* section and press *Edit Callthrough CLIs* button.



Using the NGCP Panel the user then creates Call Forward to destination *Call Through*.

5.16.1.2 Forward to local user

If the subscriber has a Call Forward to the call-through application but caller's CLI is not in the authorized CLIs list for call-through, sems responds with error back to proxy and proxy advances to the next number in the Call Forward destinations set. User can enter special destination *Local Subscriber* as next target after *Call Through* in the destinations set in order to terminate the call to the subscriber as if the subscriber didn't exist. This way the user may reach the call-through application from his authorized CLI (e.g. mobile number) and all other callers would reach the SIP subscriber's registered phone as usual.

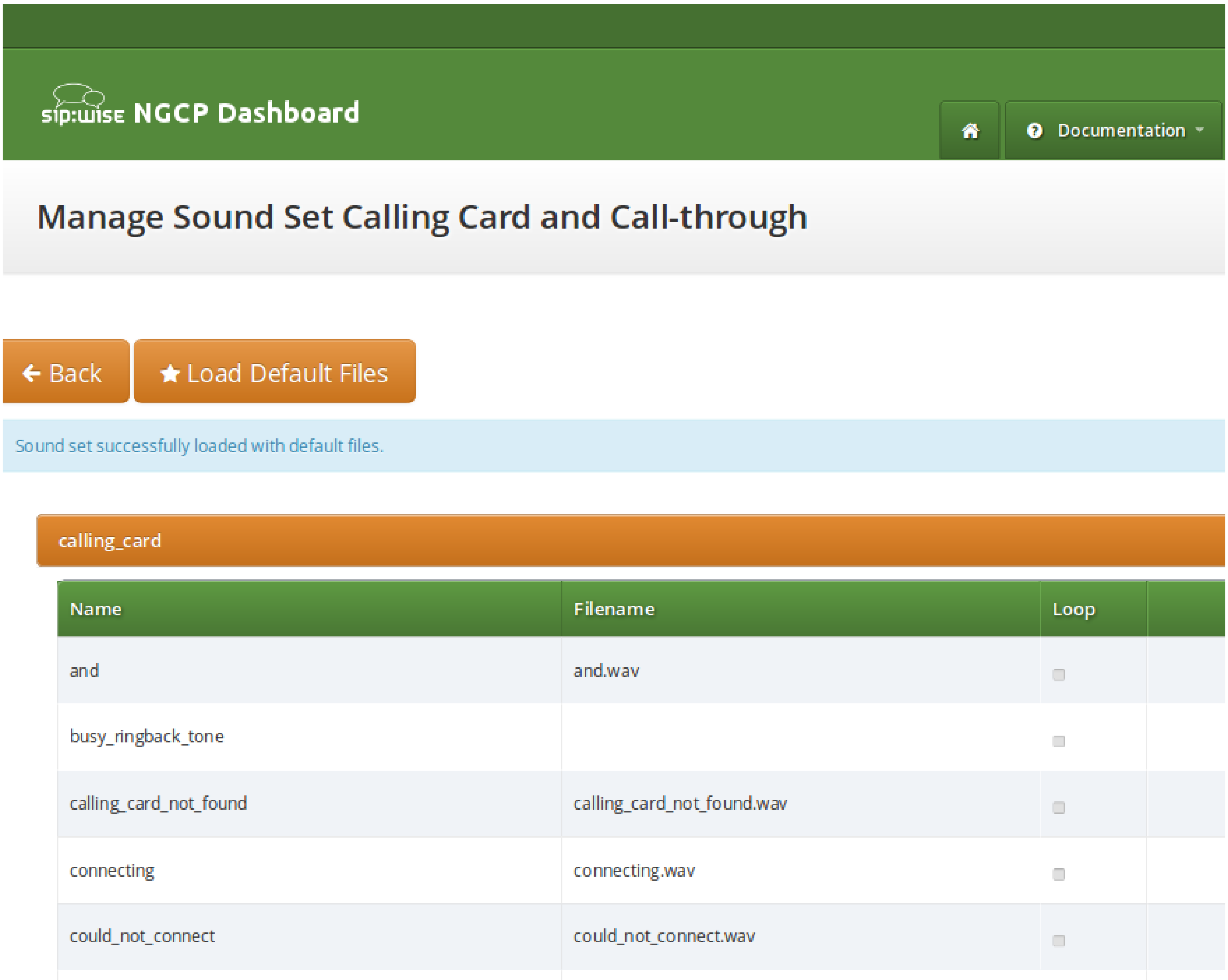


5.16.1.3 Sound Set provisioning

In order for the Callthrough application to work a Sound Set must be created and associated with the Domain or Subscriber.

Sound Sets can be defined in *Settings→Sound Sets*. To create a new Sound Set, click *Create Sound Set*. Then click the *Files* button. Administrator can upload the default sounds in one of supported languages or uploaded by the administrator manually in his language of choice.

There is a preference *sound_set* 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).



Note

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

5.16.2 Call Flow

The call arrives at sems application server with Request-URI user `callthrough`.

5.16.2.1 Internal Header Parameters

The INVITE contains an extra SIP header `P-App-Param` with the following parameters:

Table 6: SIP Header parameters for call-through application

Name	Meaning
uuid	The internal UUID of the call-through subscriber

Table 6: (continued)

Name	Meaning
srcnumber	Caller's CLI for the authentication
outgoing_cli	New CLI to be used by sems application for the outgoing call leg

5.16.2.2 Caller authorization

Caller is authorized using mapping shown in table above: `select source_uuid from provisioning.voip_cc_mapping where uuid=$uuid and auth_key=$srcnumber;`

If the check fails return the configured error response code. Then proceed with the call setup as follows.

5.16.2.3 Outgoing call

Sems requests the user to enter destination and starts digit collection. Digit collection process is terminated after 5 seconds (configurable in sems config file) or by pressing the # key. User can start entering destination while the voice prompt is being played.

Sems sends INVITE to the proxy with Request-URI: `sip:$number@$outboundproxy;sw_domain=$subscriber.domain`

`From:$outgoing_cli`

On receiving the 401 or 407 response from the proxy the application authenticates using the digest credentials retrieved for the call-through subscriber from the `voip_subscribers` table:`select s.username, s.password, d.domain from provisioning.voip_subscribers s, provisioning.voip_domains d where s.uuid=$source_uuid and s.domain_id=d.id;`

If the call setup fails the application plays back the "could_not_connect" sound file. If successful the application acts transparently and does not provide any voice announcements or DTMF detection.

5.16.2.4 CLI configuration

The CLI on the outgoing call from the call-through module is set to the Network-Provided Number (NPN) of the call-through subscriber. There is nothing to configure.

5.17 Calling Card Application

Calling card application uses a similar concept to call-through except that authorization process operates on the PIN code entered by user using DTMF instead of the CLI. The sip:provider PRO maps incoming UUID of the pilot subscriber to the list of PINs for calling card application with their corresponding subscriber UUIDs for outbound call leg using table `provisioning.voip_cc_mapping table {"uuid", "auth_key", "source_uuid"}`

Table 7: Calling Cards

Column	Description
uuid	The internal UUID of the pilot subscriber
auth_key	Authentication key (PIN)
source_uuid	The internal UUID of the subscriber that is authorized for outgoing call leg

5.17.1 Administrative Configuration

5.17.1.1 Subscriber provisioning

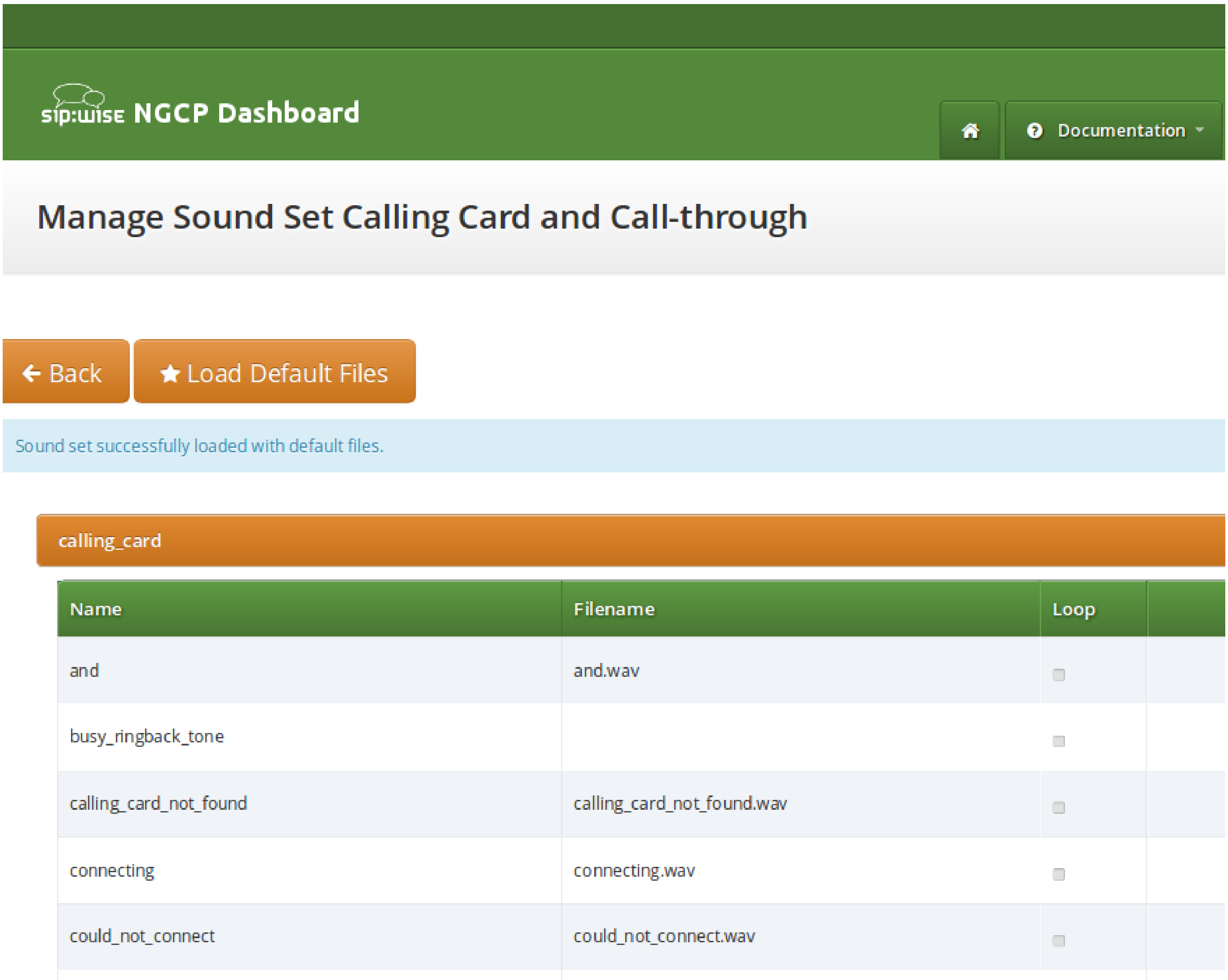
In order to use the calling cards service the user creates a Call Forward to destination *Calling Card* for the designated subscriber that will be used as access number for this service.

5.17.1.2 Sound Set provisioning

In order for the Calling Card application to work a Sound Set must be created and associated with the Domain or Subscriber.

Sound Sets can be defined in *Settings→Sound Sets*. To create a new Sound Set, click *Create Sound Set*. Then click the *Files* button. Administrator can upload the default sounds in one of supported languages or uploaded by the administrator manually in his language of choice.

There is a preference *sound_set* 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).



Note

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

5.17.1.3 CLI configuration

The CLI on the outgoing call from the calling card app can be configured in one of the following ways using subscriber preferences:

1) Show original caller's CLI: the calling card subscriber shall have `allowed_clis:* (any)`. Sems application sends the original caller's CLI in the From header, it is validated by the SIP proxy and sent to outside.

2) Show number of the pilot (calling card) subscriber: the calling card subscriber shall have an empty `allowed_clis` and desired number set as value of `user_cli` preference. The SIP proxy overrides the original caller's CLI in UPN with the value of the `user_cli` preference. The peer must have set `outbound_from_user, outbound_from_display:User-Provided Number (UPN)`.

5.17.2 Call Flow

The call arrives at sems application server with Request-URI user `callingcard`.

5.17.2.1 Internal Header Parameters

The INVITE contains an extra SIP header `P-App-Param` with the following parameters:

Table 8: SIP Header parameters for calling card application

Name	Meaning
uuid	The internal UUID of the pilot subscriber
outgoing_cli	New CLI to be used by sems application for the outgoing call leg

5.17.2.2 Caller authorization

- Sems requests the user to enter PIN and starts digit collection. Digit collection process is terminated after 5 seconds (configurable in sems config file) or by pressing the # key. User can start entering destination while the voice prompt is being played.
- Sems checks that PIN is valid and belongs to the pilot subscriber using mapping as shown in the table. It fetches UUID of the subscriber to be used for outgoing call leg: `select source_uuid from provisioning.voip_cc_mapping where uuid=$uuid and auth_key=$pin;`
- If the check fails sems will request the user to re-enter PIN up to the configured number of times.
- If successful proceed with the call setup making call on behalf of subscriber determined by the `source_uuid` key as follows.

5.17.2.3 Outgoing call

Sems application plays back the available balance of the customer. Sems requests the user to enter destination and starts digit collection. Digit collection process is terminated after 5 seconds (configurable in sems config file) or by pressing the # key. User can start entering destination while the voice prompt is being played.

Sems sends INVITE to the proxy with Request-URI: `sip:$number@$outboundproxy;sw_domain=$subscriber.domain`

`From:$outgoing_cli`

On receiving the 401 or 407 response from the proxy the application authenticates using the digest credentials retrieved for the subscriber for outgoing call leg from the `voip_subscribers` table: `select s.username, s.password, d.domain from provisioning.voip_subscribers s, provisioning.voip_domains d where s.uuid=$source_uuid and s.domain_id=d.id;`

5.17.2.4 Voucher recharge

During the destination collection phase in calling card application user can enter special code *1*<pin># (configurable in sems config file) to transfer balance from other calling card customer to the currently authorized customer. Sems transfers all remaining balance from that customer to the current customer.

5.17.2.5 Billing

The call via calling card application as well as call-through generates three CDRs:

- A to B: The incoming call from any source to the call-through subscriber.
- B to callingcard@app.local or callthrough@app.local: The call forward to the sems application.
- B to C: The outgoing call to the final destination. The three CDRs are handled by the billing process as usual, exported and shown in all call lists. .

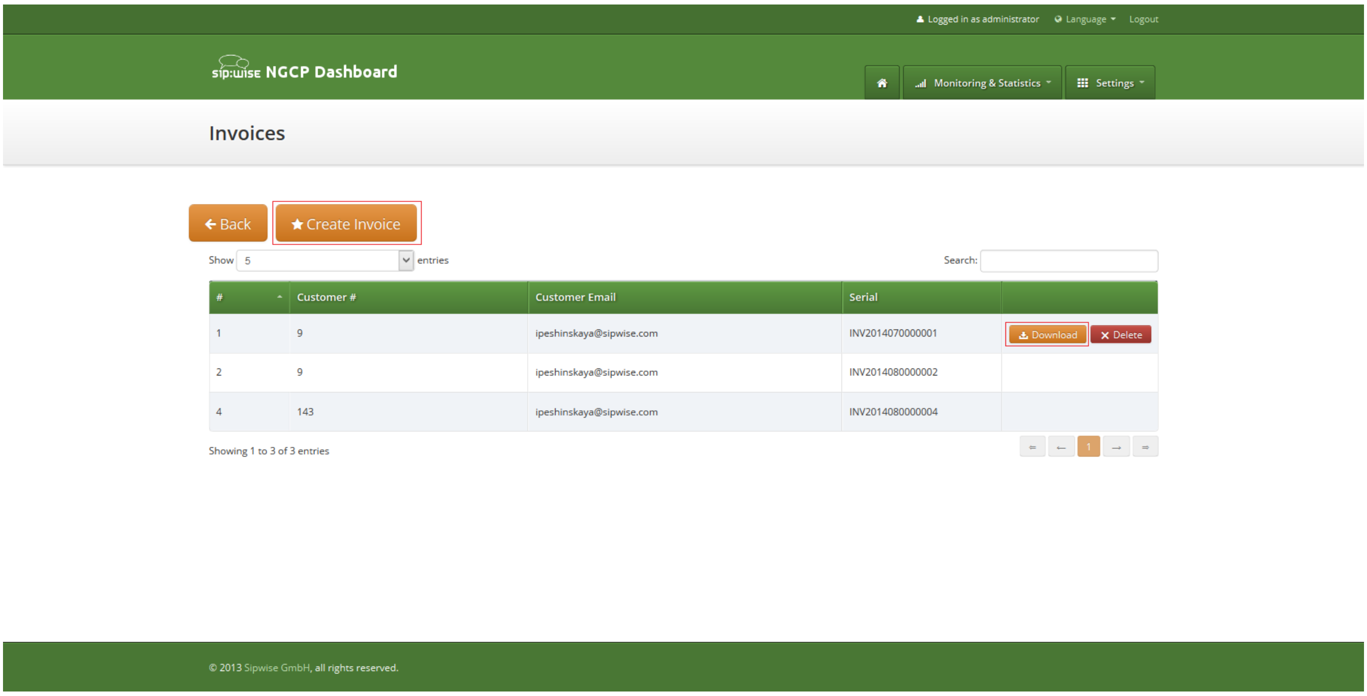
5.18 Invoices and Invoice Templates

Content and vision of the invoices are customizable by invoice templates Section 5.18.2.

> **Note**
> The sip:provider PRO generates invoices in pdf format.

5.18.1 Invoices Management

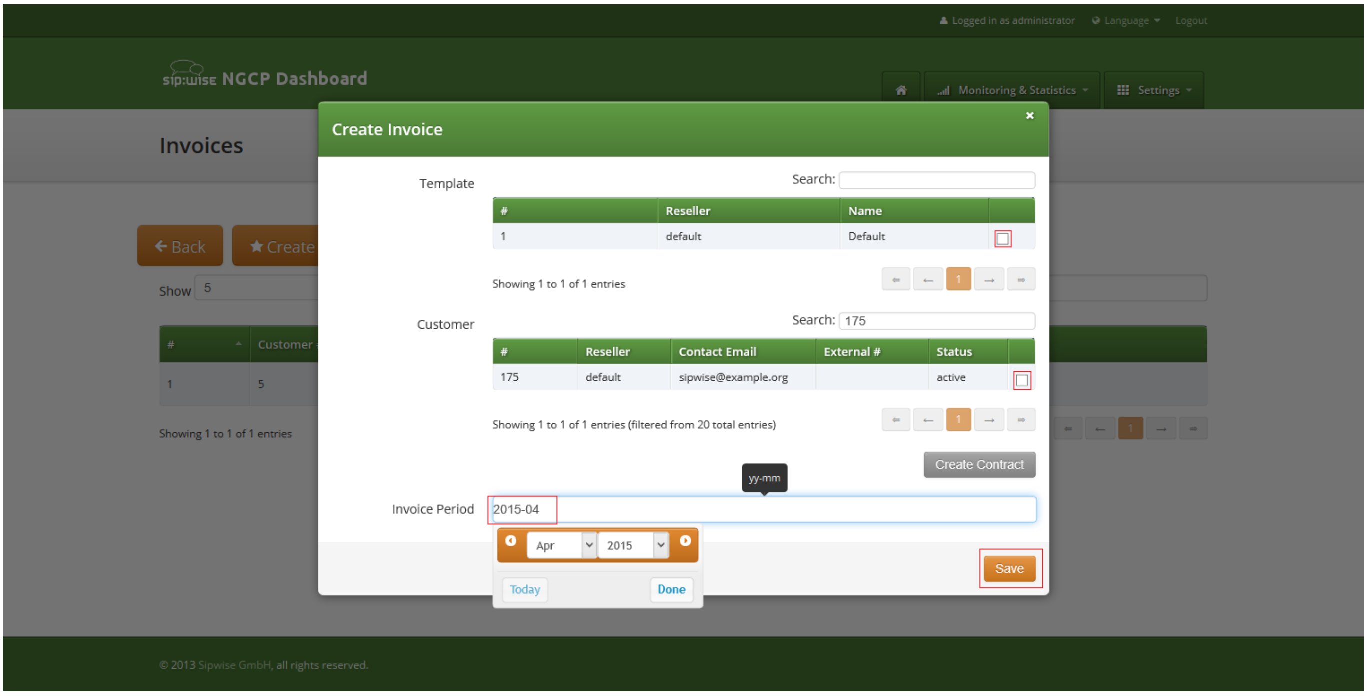
Invoices can be requested for generation, searched, downloaded and deleted in the invoices interface.



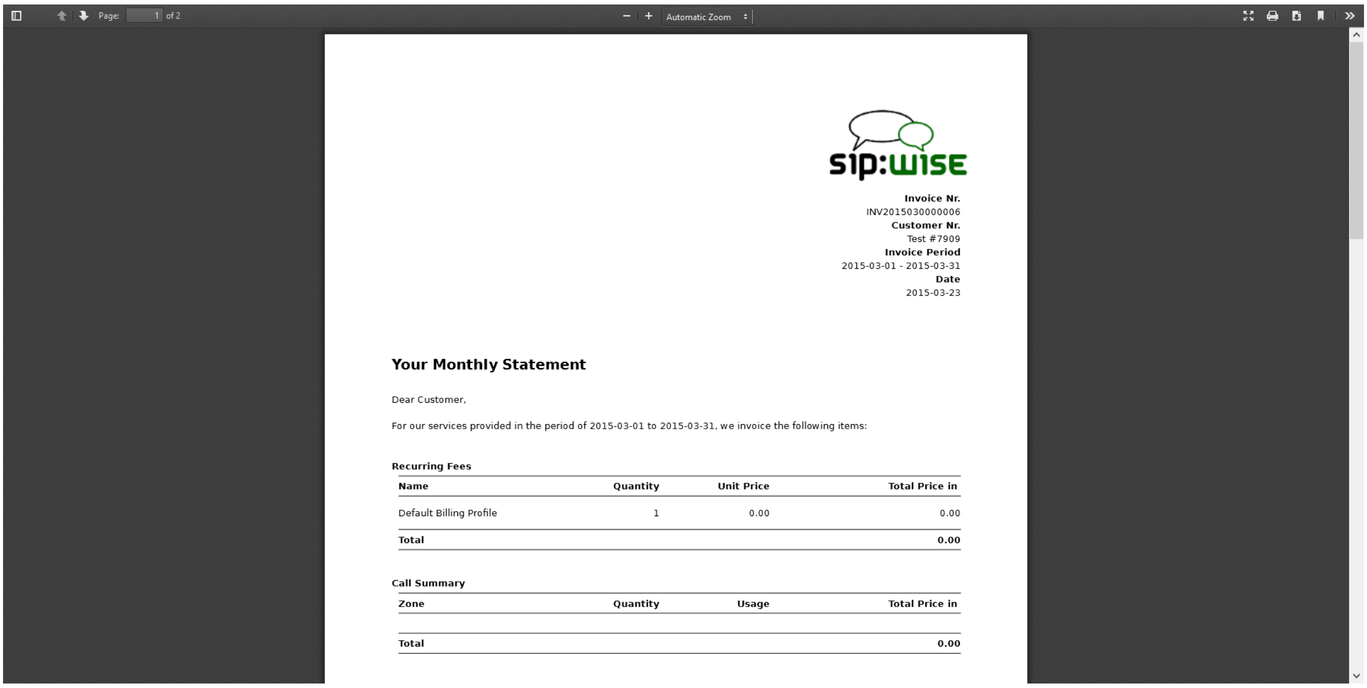
To request invoice generation for the particular customer and period press "Create invoice" button. On the invoice creation form following parameters are available for selection:

- **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.

5.18.2 Invoice Templates

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

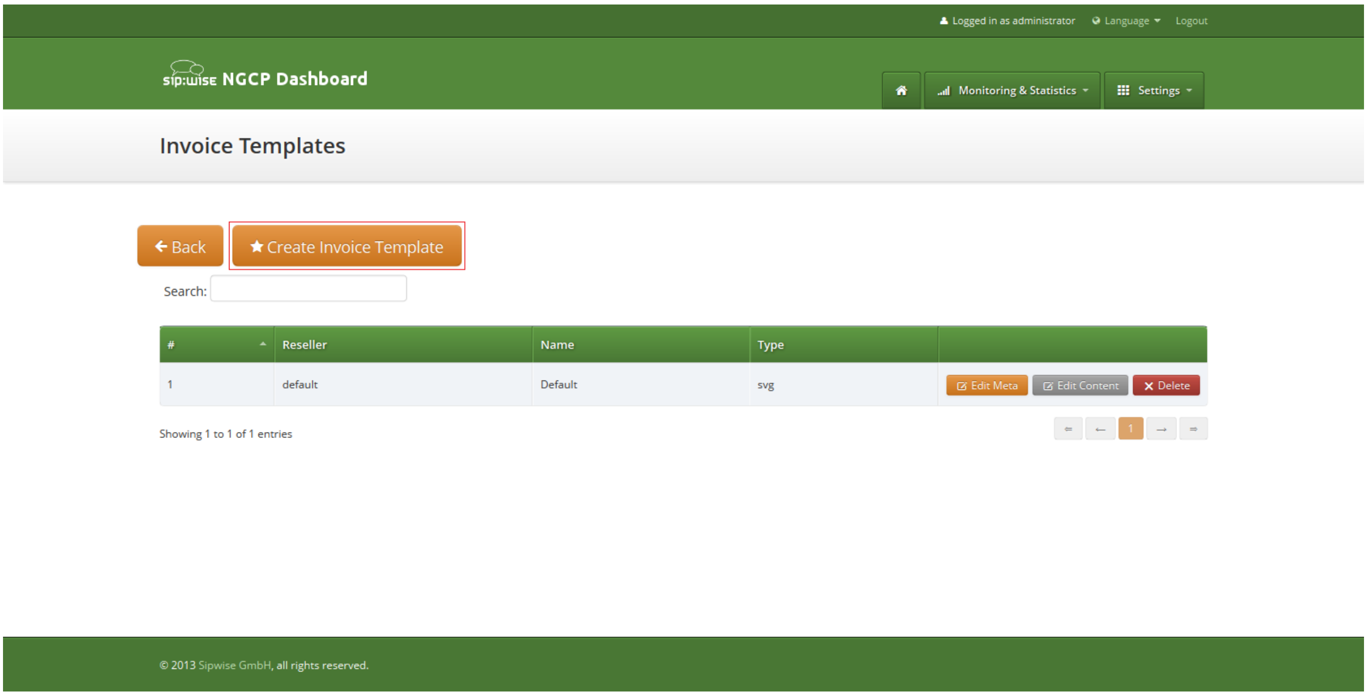
Important

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.

5.18.2.1 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:

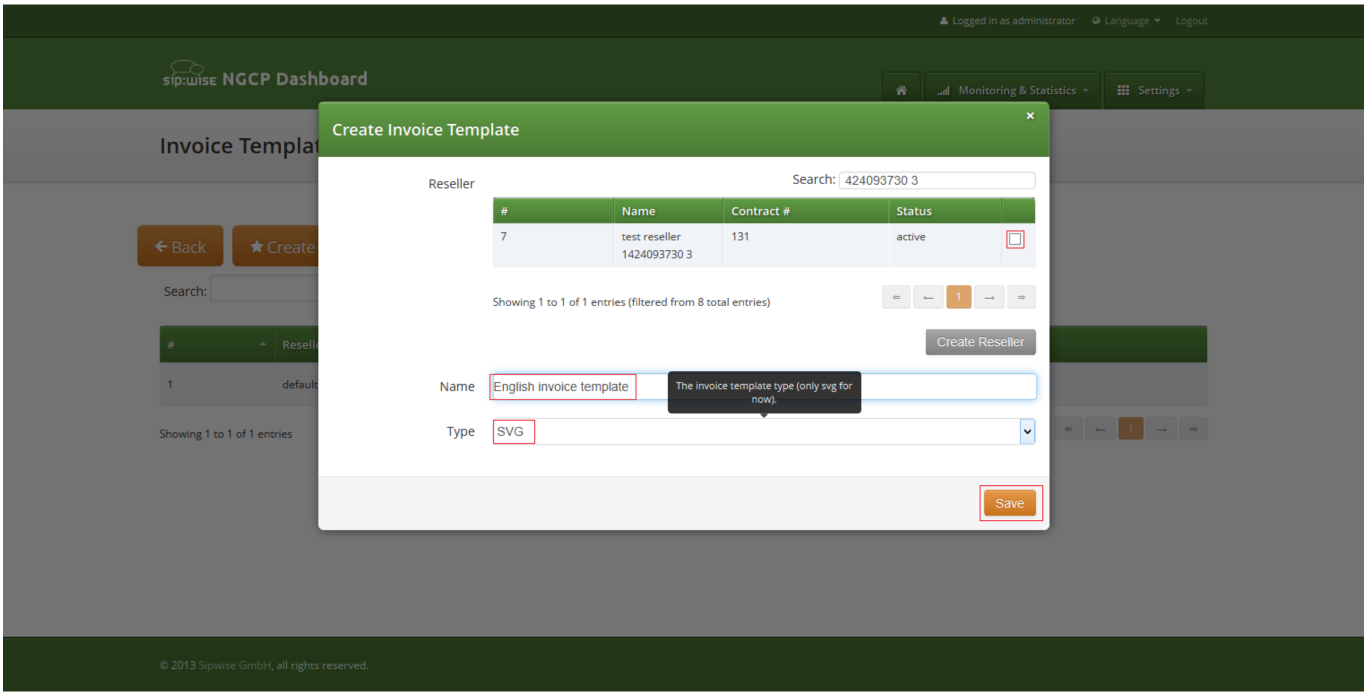
- Register new invoice template meta information.
- Edit content (template itself) of the invoice template.

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 PRO supports only svg format of the invoice templates.

All form fields are mandatory.



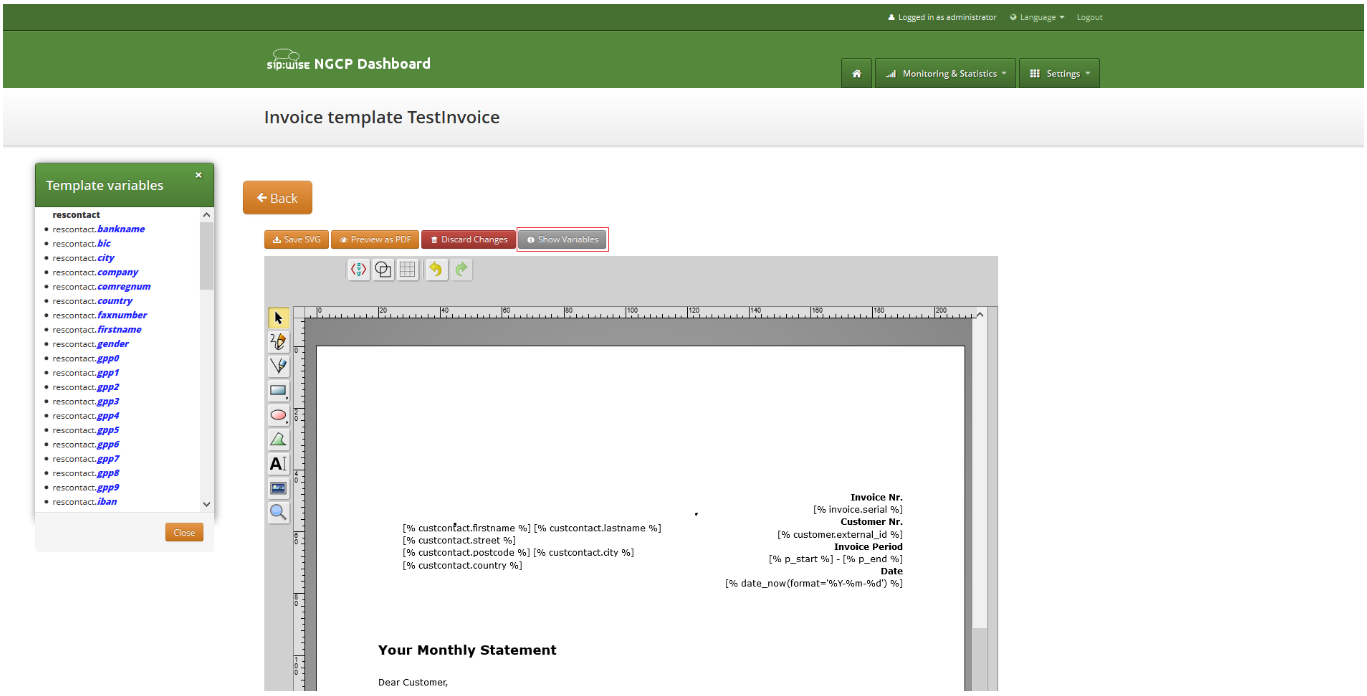
After registering new invoice template you can change invoice template structure in WYSIWYG SVG editor and preview result of the invoice generation based on the template.

5.18.2.2 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 PRO provides embedded WYSIWYG SVG editor svg-edit 2.6 to customize default template. The sip:provider PRO 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 5.18.2.3.

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

Important

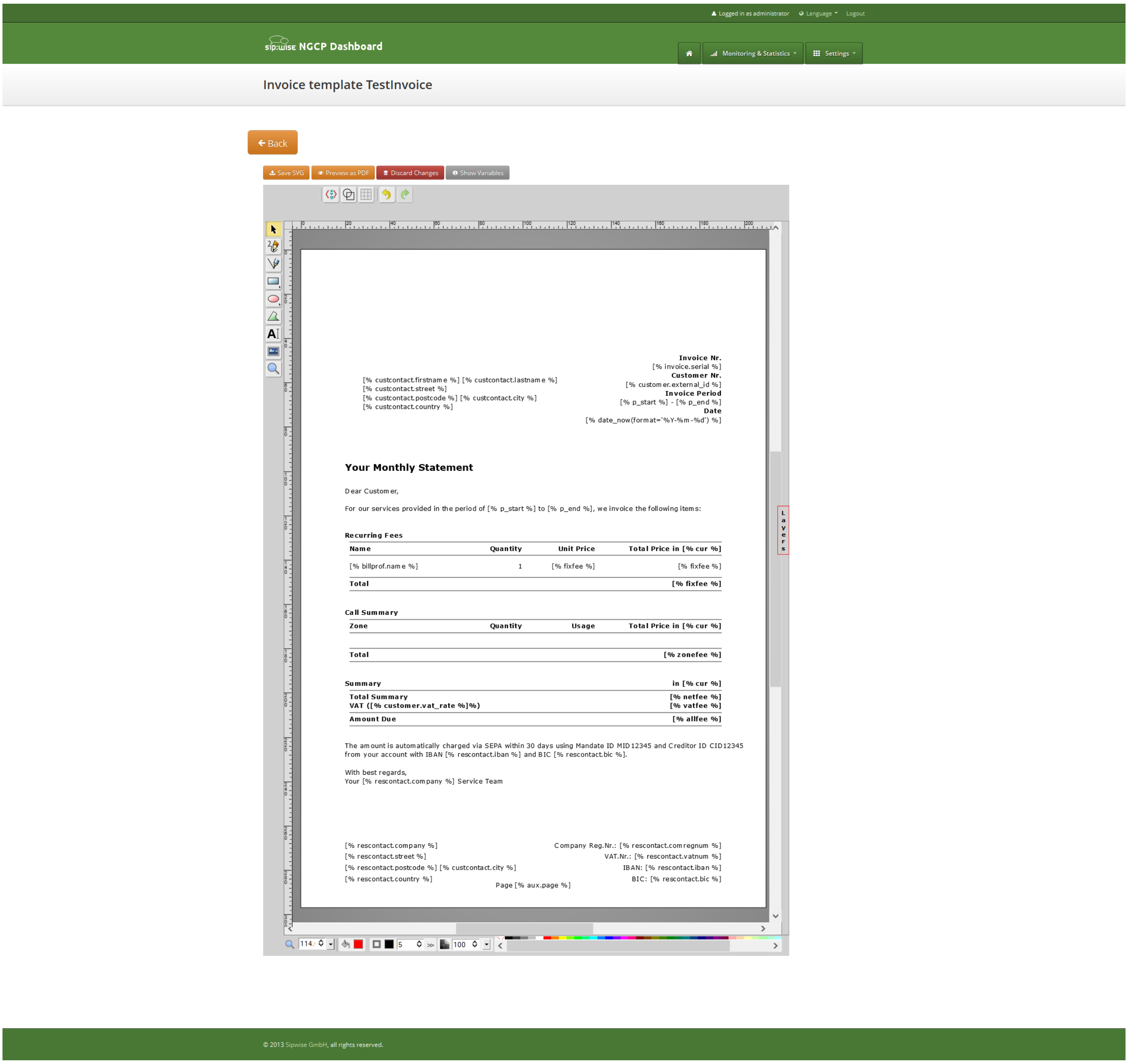
"Discard changes" operation can't be undone.

Layers

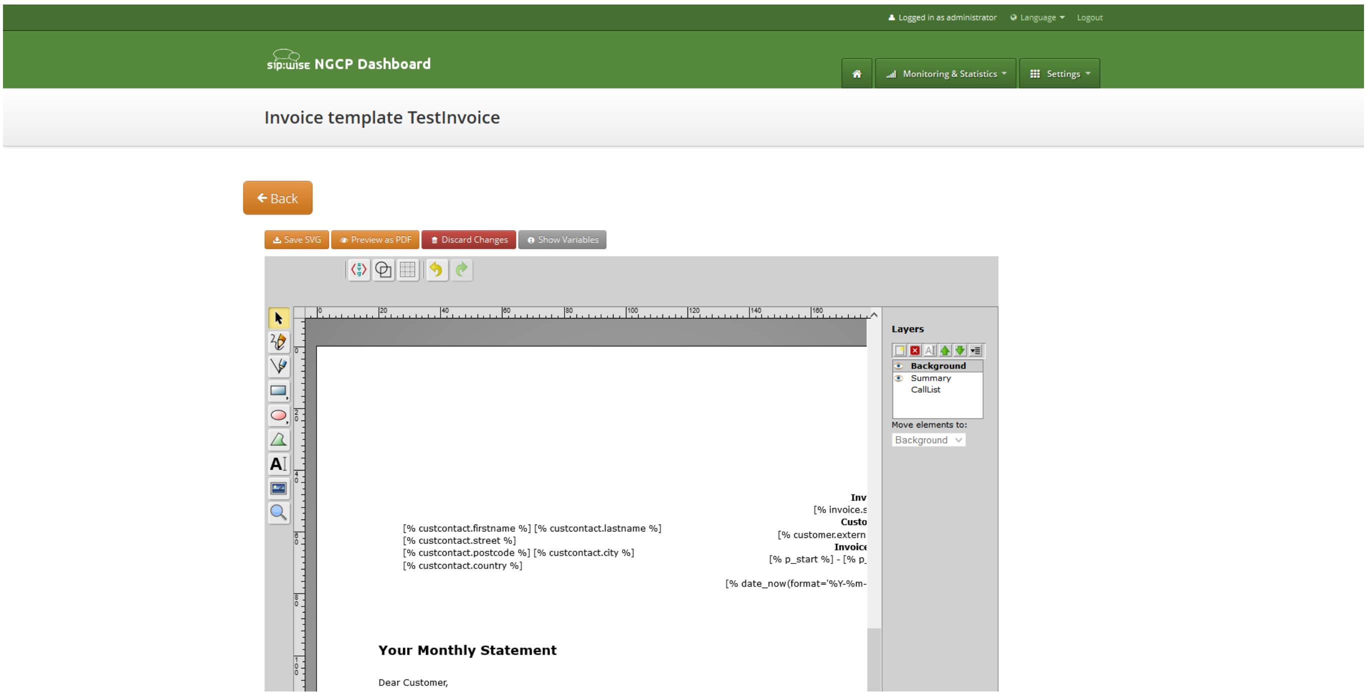
Default template contains three groups elements (<g/>), which can be thinked of as pages, or in terms of svg-edit - layers. Layers are:

- **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.

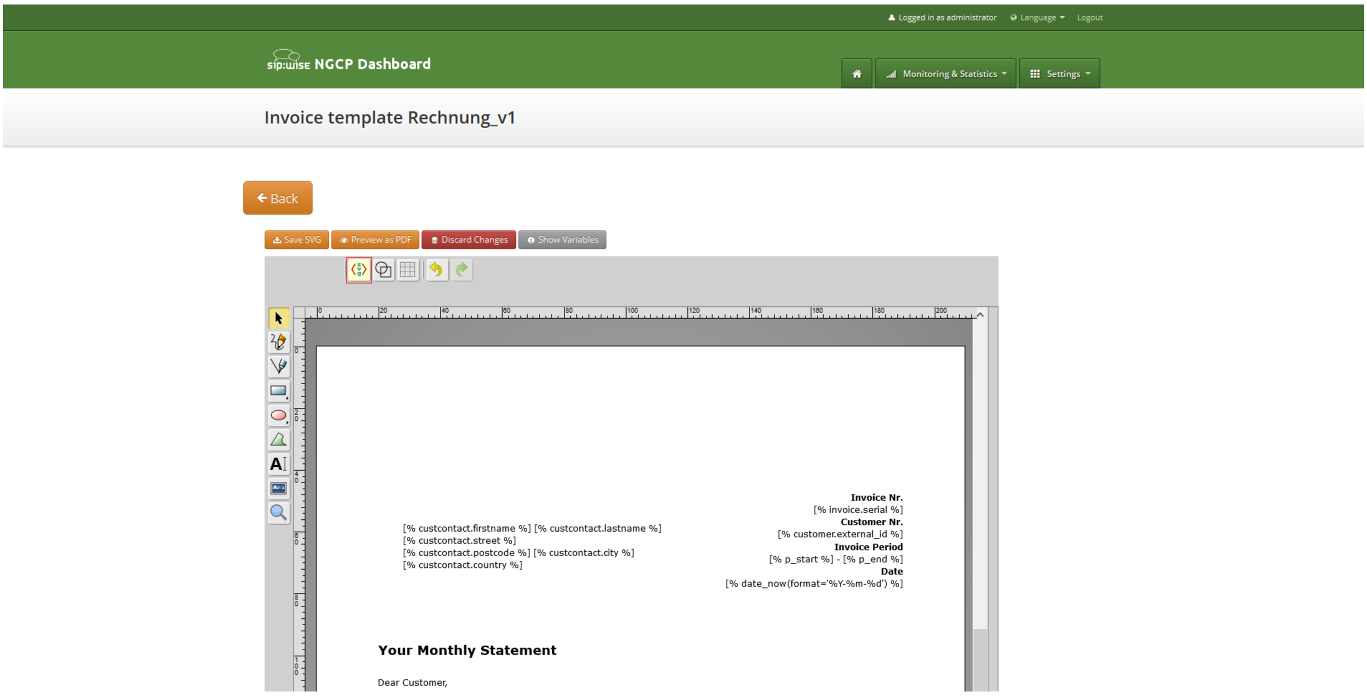


One of the layers is active, and its element can be edited in the main svg-edit window. Currently active layer's name is **bold** in the layers list. The layers may be visible or invisible. Visible layers have "eye" icon left of their names in the layers list.

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.

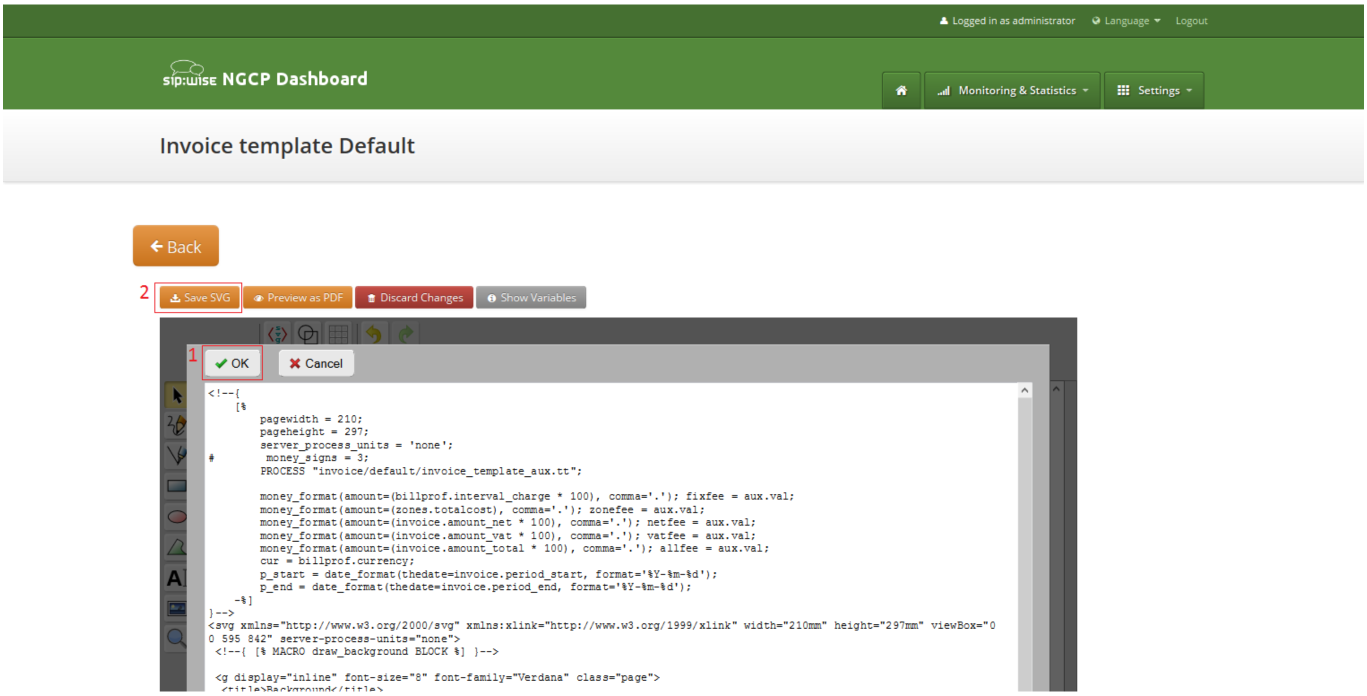
Note

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:



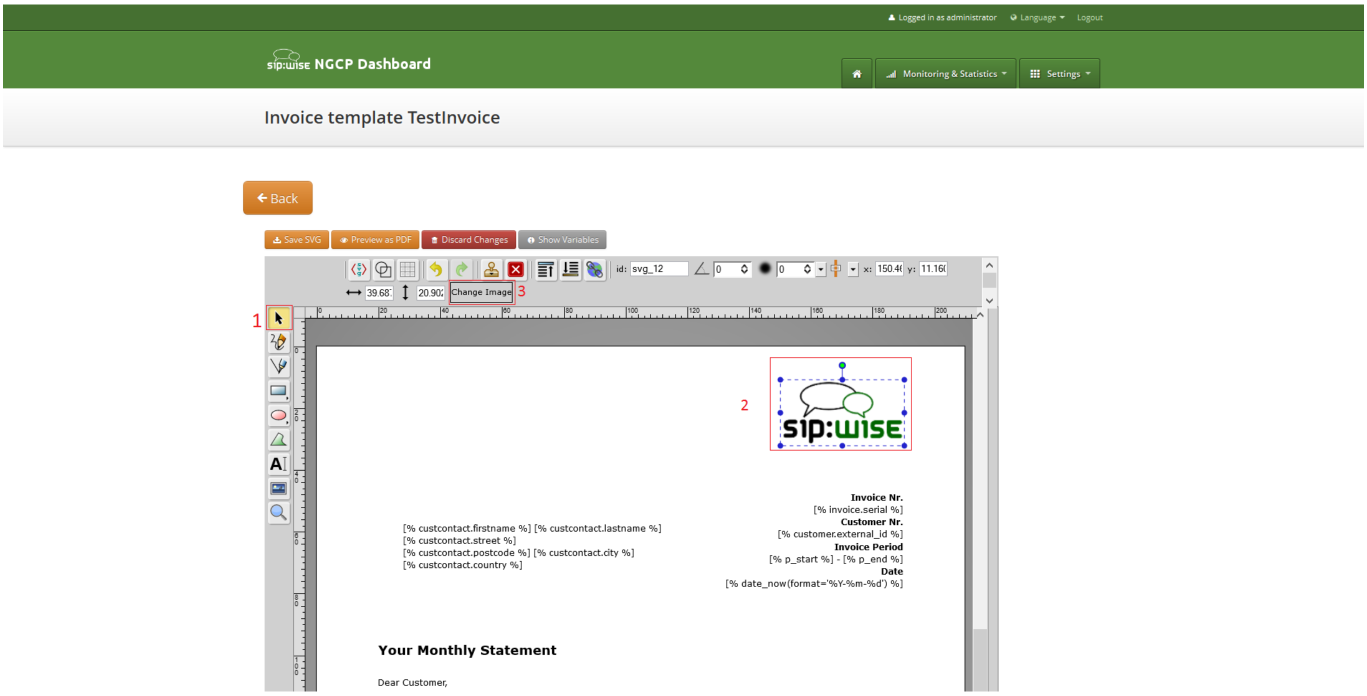
And then save invoice template changes Section 5.18.2.3.

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

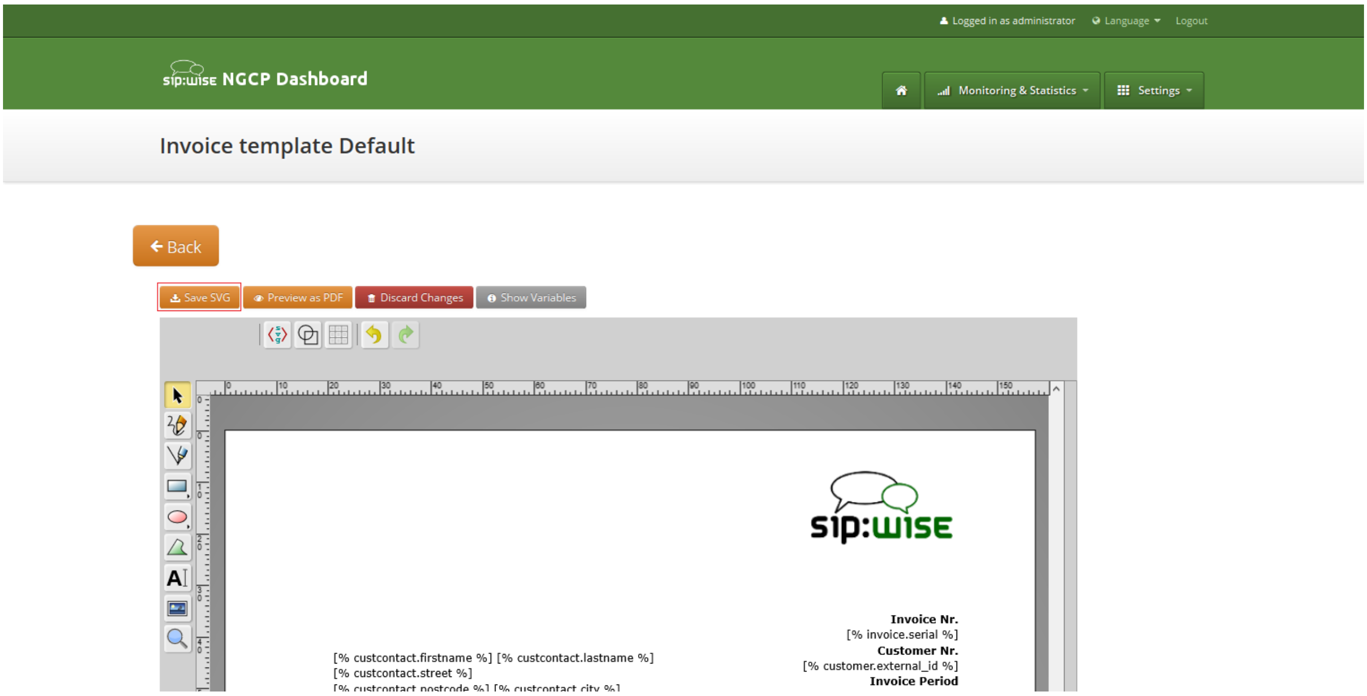
- Make sure that "Select tool" is active.
- Select default logo, clicking on the logo image.
- Press "Change image" button, which should appear on the top toolbar.



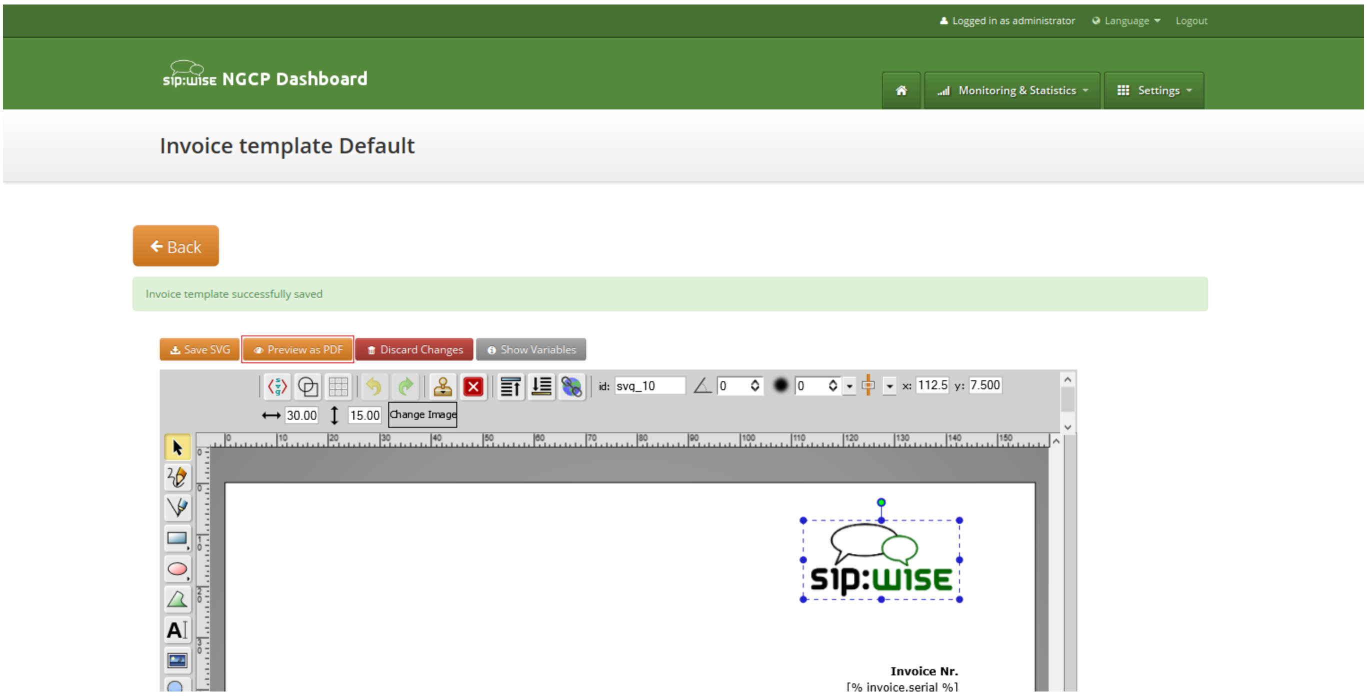
After image uploaded save invoice template changes Section 5.18.2.3.

5.18.2.3 Save and preview invoice template content

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



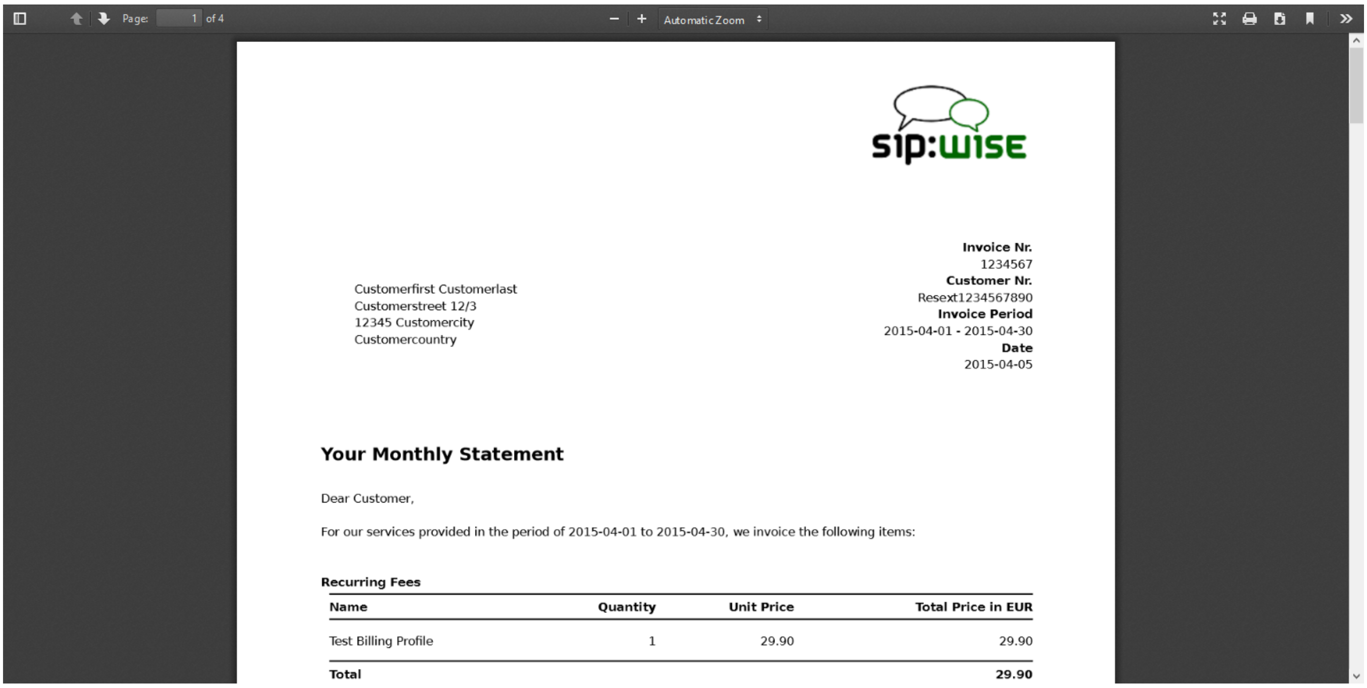
You will see message about successfully saved template. You can preview your invoice look in PDF format. Press on "Preview as PDF" button.



Invoice preview will be opened in the new window.

Note

Example fake data will be used for preview generation.



5.18.3 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 5.18.1.

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.

5.19 Email Reports and Notifications

5.19.1 Email events

The sip:provider PRO 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.

5.19.2 Initial template values and template variables

Default email templates for each of the email events are inserted on the initial sip:provider PRO 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 PRO 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 PRO Panel code.

5.19.3 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.

5.19.4 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.

5.19.5 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

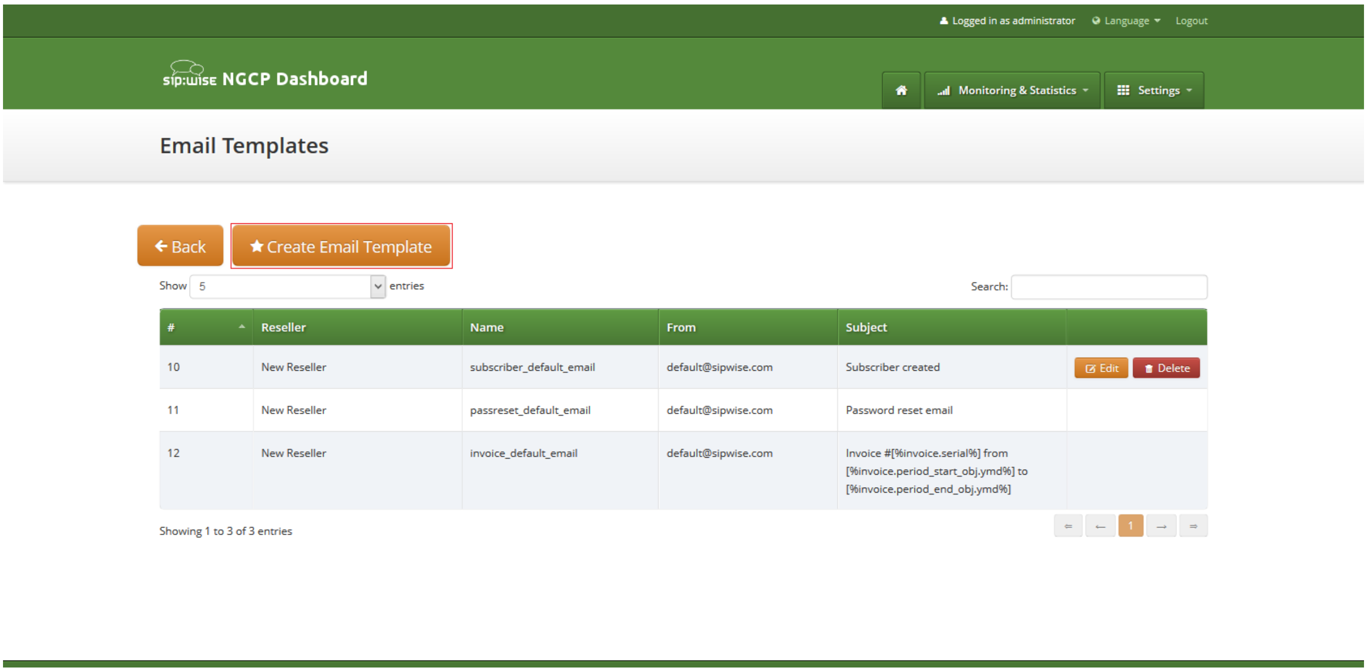
- [%**provider**%]: container variable for the reseller contact. All database contact values will be available.
- [%**client**%]: container variable for the customer contact.

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

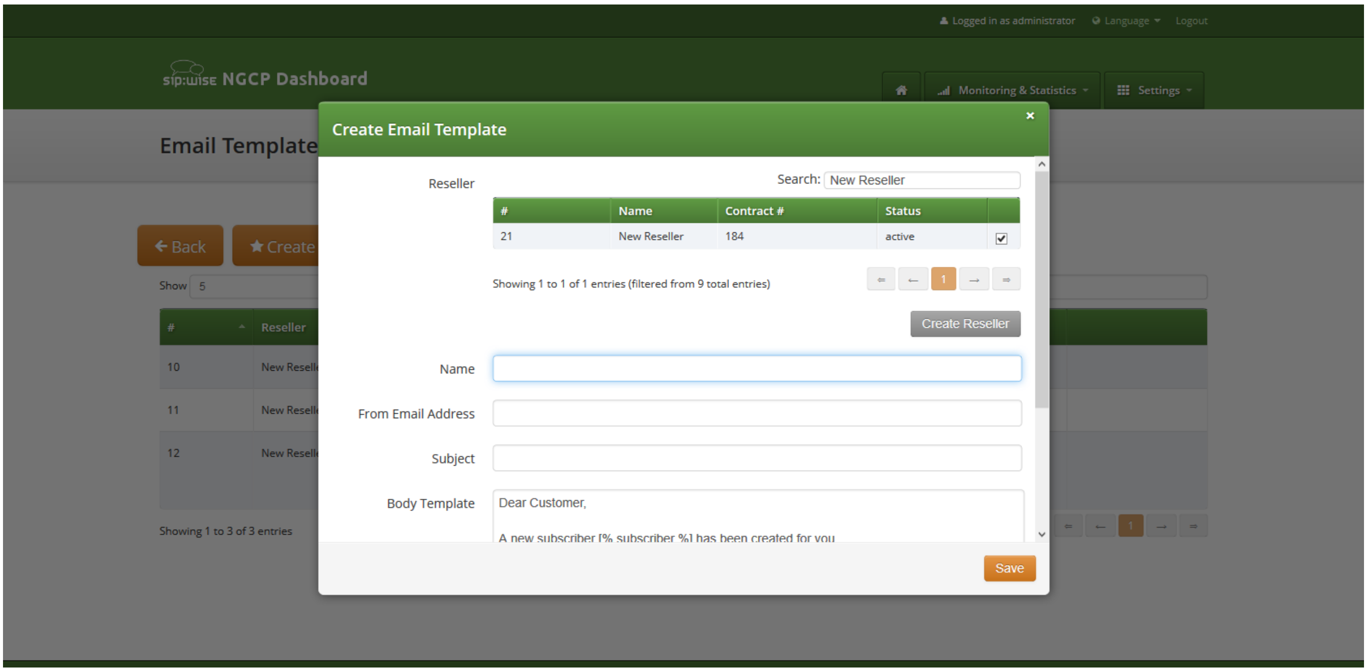
- [%provider.**gender**%]
- [%provider.**firstname**%]
- [%provider.**lastname**%]
- [%provider.**comregnum**%]
- [%provider.**company**%]
- [%provider.**street**%]
- [%provider.**postcode**%]
- [%provider.**city**%]
- [%provider.**country**%]
- [%provider.**phonenumber**%]
- [%provider.**mobilenumber**%]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

5.19.6 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 PRO on the appropriate event Section 5.19.1 :
  - 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 PRO.
- **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.

5.20 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 4.6.

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.

5.20.1 Vertical Service Codes for PBX customers

Subscribers under the same PBX customer can enjoy some PBX-specific features by means of special VSCs.

NGCP provides the following PBX-specific VSCs:

- **97** - *Call Parking:* during a conversation the subscriber can park the call with his phone to a "parking slot" and later on continue the conversation from another phone. To do that, a destination must be dialled as follows: `*97*3`; this will park the call to slot no. 3.

  PLEASE NOTE:

  – Cisco IP phones provide a softkey for Call Parking, that means the subscriber must only dial the parking slot number after pressing "Park" softkey on the phone.

  – Other IP phones can perform Call Parking as a *blind transfer*, where the destination of the transfer must be dialled in the format described above.

  – Both the caller and the callee can park the call.

- **98** - *Call Unparking:* if a call has been parked, a subscriber may continue the conversation from any extension (phone) under the same PBX customer. To do that, the subscriber must dial the following sequence: `*98*3`; this will pick up the call that was parked at slot no. 3.

- **99** - *Directed Call Pickup:* if a subscriber's phone is ringing (e.g. extension 23) and another subscriber wants to answer the call instead of the original callee, he may pick up the call by dialling `*99*23` on his phone.

5.20.2 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.

5.20.3 Voice Prompts for Vertical Service Code Configuration

Table 9: 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.

Table 9: (continued)

Prompt Handle	Related VSC	Message
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.

5.21 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.

5.22 XMPP and Instant Messaging

Instant Messaging (IM) based on XMPP comes with sip:provider PRO out of the box. sip:provider PRO 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`.

6 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.

6.1 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://<ce-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.

6.1.1 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 4.2. 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.

6.1.2 Site Customization

As an operator (as well as a Reseller), you can change the branding logo of the CSC panel by modifying the CSS via web interface. For changing the branding logo you just need to access the web interface as asministrator and move to *Reseller_ menu. Once there click on* Details *button for "default" reseller. Then on* Branding → Edit Branding*. Now you can upload your logo and copy/paste the CSS code line in the* CSS__ field. The logo will be visible into the Customer Self Care interface.

Also Reseller can customize their web page (CSC and Admin interface) by uploading their logo and change the CSS. To do that, just access the Admin interface with the Reseller web credentials and then access the *Panel Branding* menu. From them you can upload the logo as explained before. The logo will appearn in the CSC web page related to that reseller as well as to the Admin page of the reseller.

You can also enable/disable specific languages a user can choose from in the CSC panel. Currently, English (`en`), German (`de`), Spanish (`es`) and Russian (`ru`) are supported and English is activated by default. You can change the default language provided by CSC by changing the parameter *force_language* in `config.yml`.

6.2 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 4.6.) 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.

7 Billing Configuration

This chapter describes the steps necessary to rate calls and export rated CDRs (call detail records) to external systems.

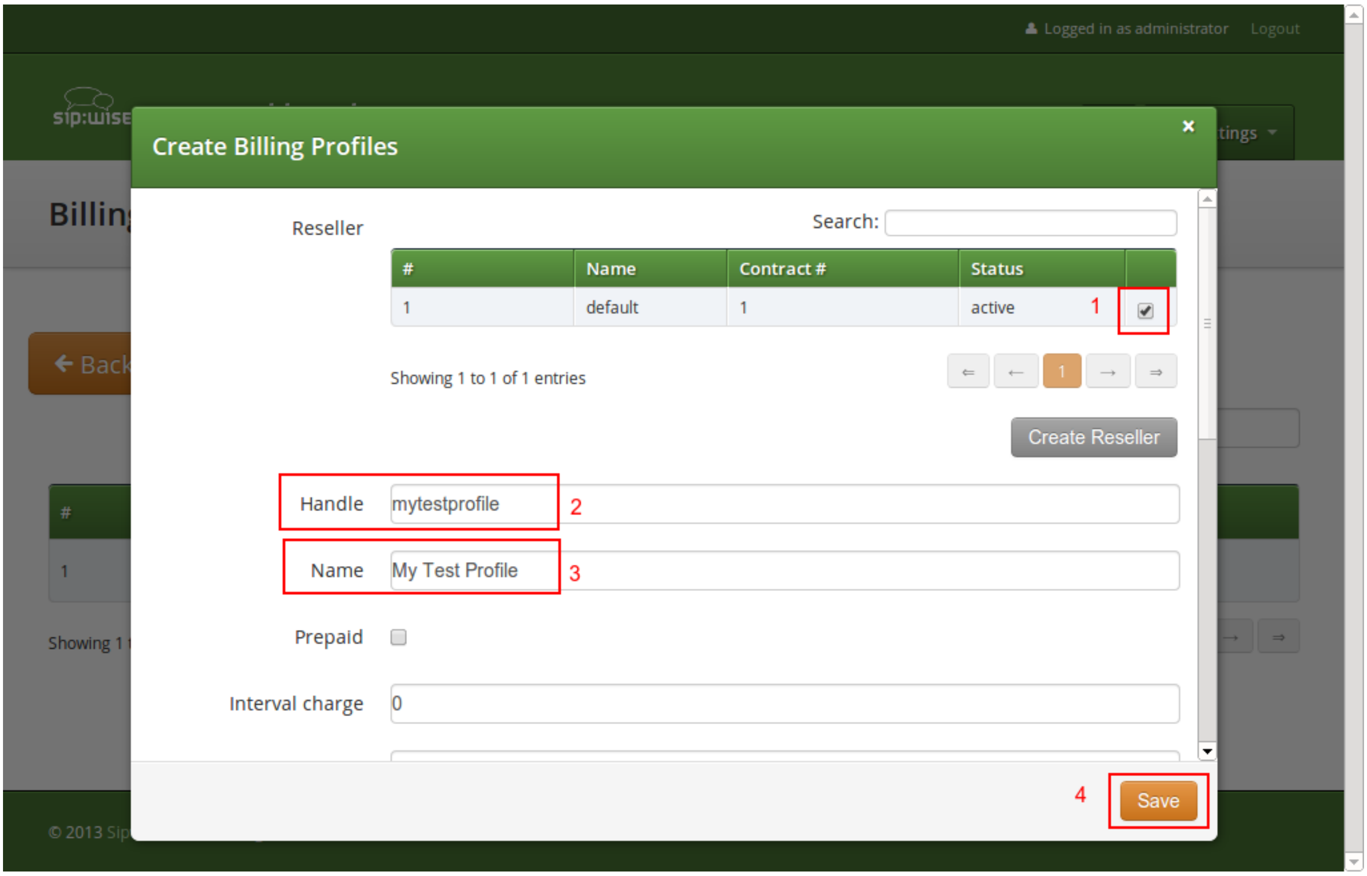
7.1 Billing Profiles

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.)

7.1.1 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.
- **Prepaid**: Enables prepaid accounting for this profile as opposed to normal post-paid mode.
- **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.

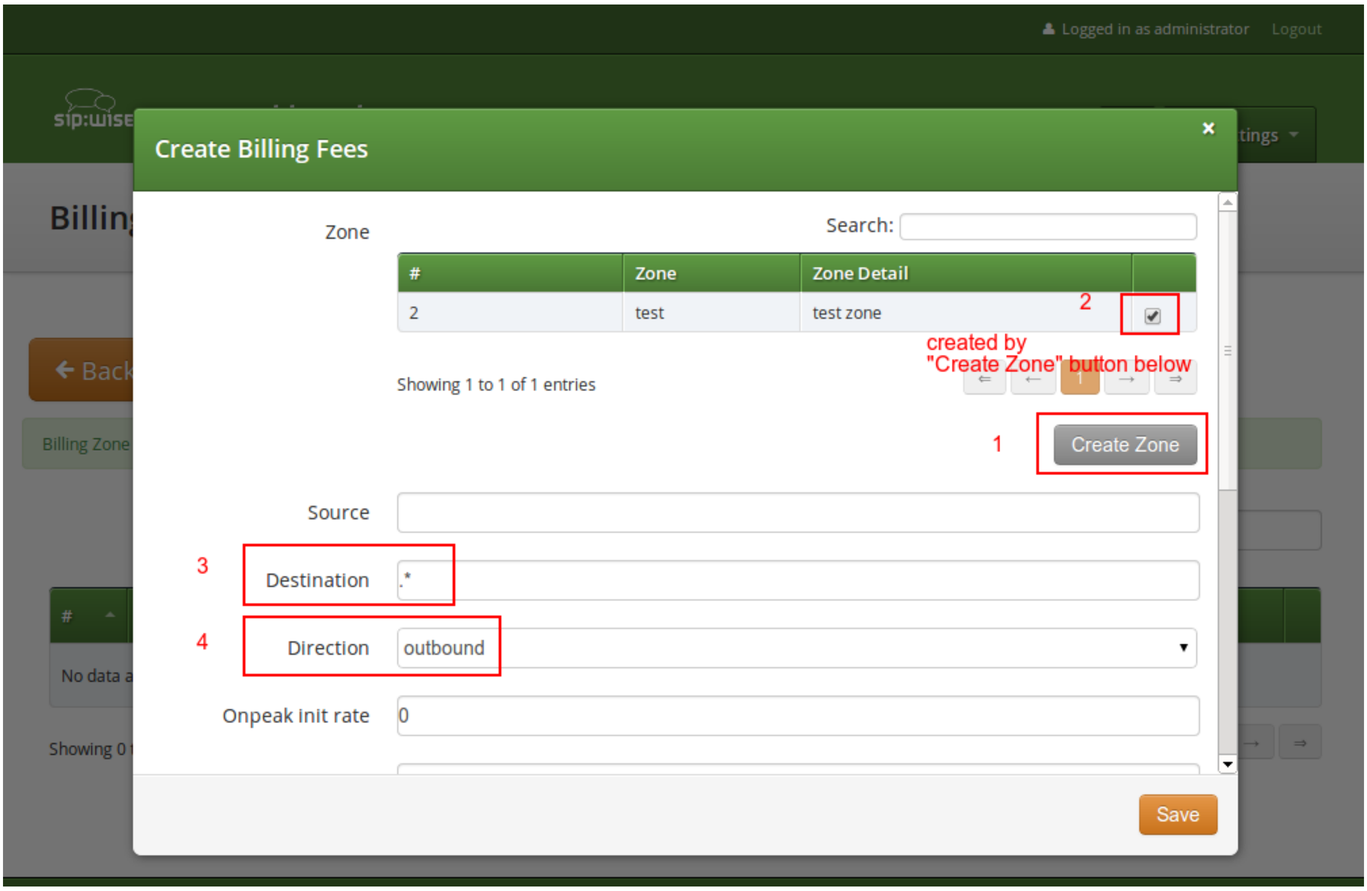
7.1.2 Creating Billing Fees

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

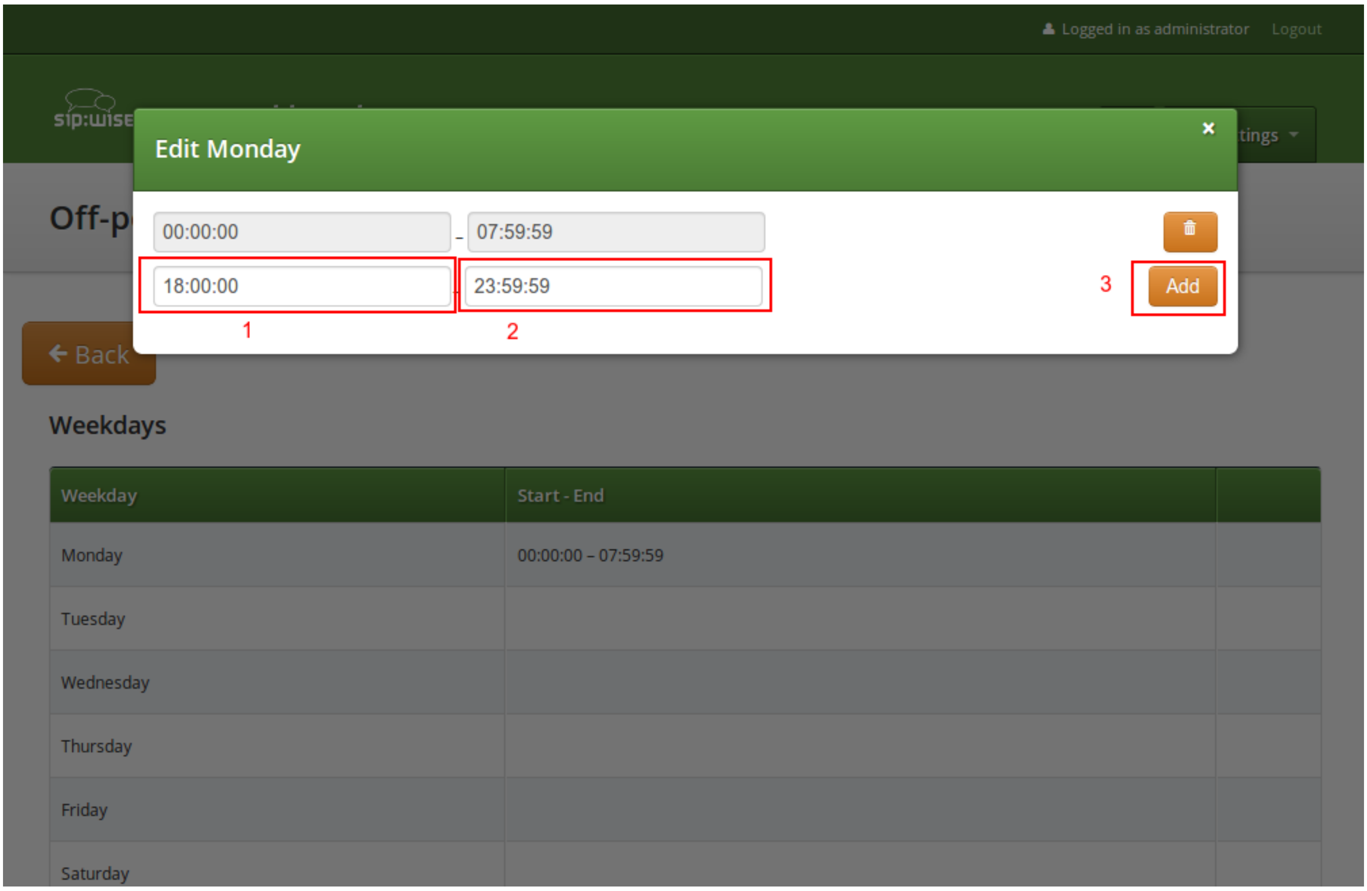
There are several internal services (vsc, conference, voicebox, fax2mail) which will need a specific destination entry with a domain-based destination. If you don't want to charge the same (or nothing) for those services, add a fee for destination `\.local$` there. If you want to charge different amounts for those services, break it down into separate fee entries for `@fax2mail\.local$`, `@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.

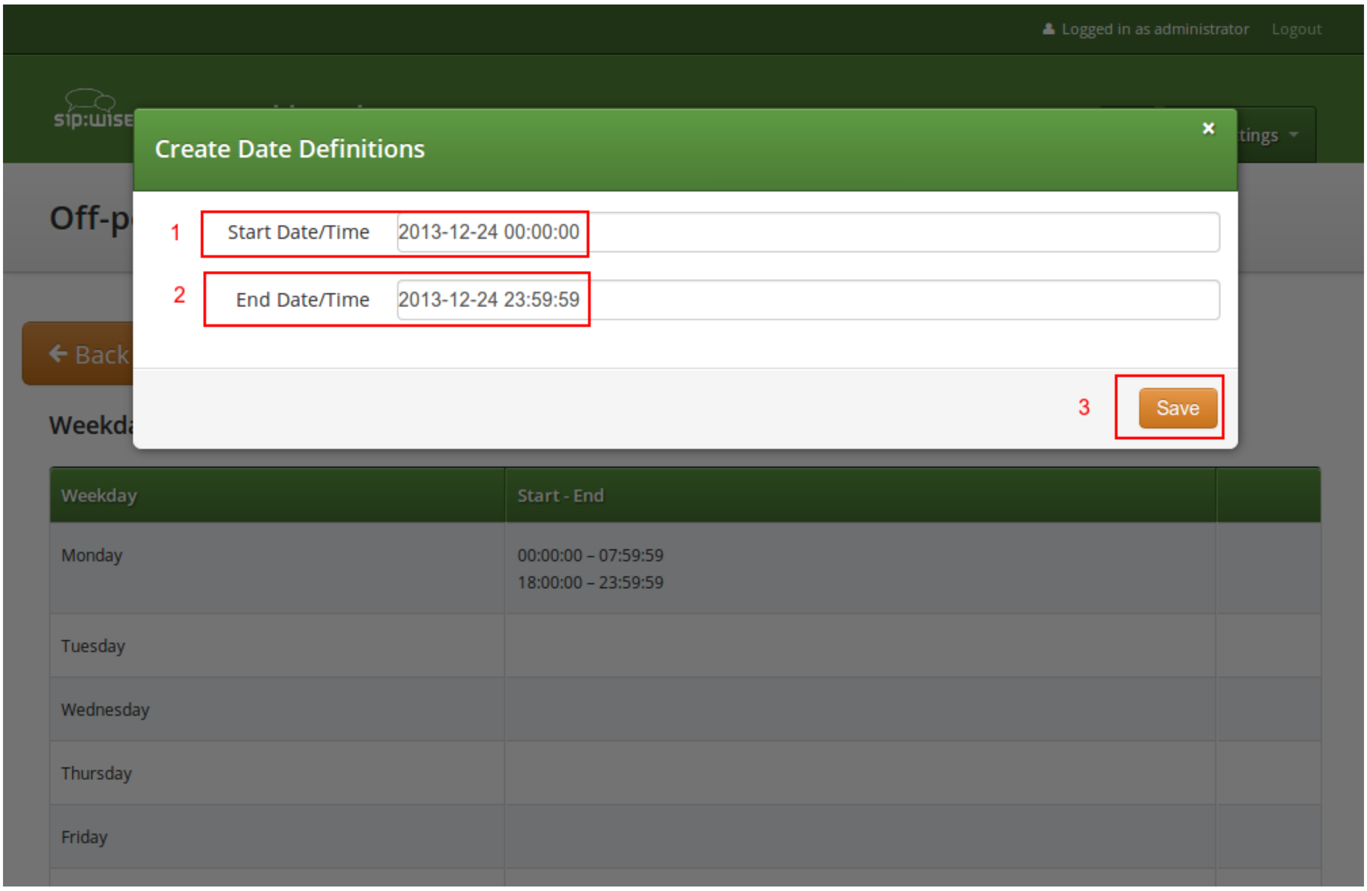
7.1.3 Creating Off-Peak Times

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.



7.2 Prepaid Accounting

In a normal post-paid accounting scenario, each customer accumulates debt in their billing account, which at the end of the billing interval is then billed to the customer. A *prepaid* billing profile reverses this sequence: the customer first has to provide credit to their account balance, and the costs for all calls are then deducted from that account balance. Once the balance reaches zero, no further calls from this customer are accepted, with the exception of free calls. Additionally, if the balance drops to zero while any calls are currently active, NGCP will disconnect those calls as soon as that happens.

With prepaid billing enabled, all details of the billing profile and all details of the billing fees behave as they normally do, including interval free time. If any interval free time is given, the free time will be used before the account's credit is.

Important

For technical reasons, the system can make the distinction between on-peak and off-peak times only at call establishment time. In other words, if the currently active call fee at the moment when the call is established is an off-peak fee, then the same off-peak fee will remain active for the whole length of this call, even if the call actually transitions into an on-peak fee (and vice versa).

Important

For technical reasons, prepaid billing can't charge local endpoint calls to Voicebox, VSC calls or calls to a Conference Room.

The Sipwise NGCP platform offers advanced billing features which are especially designed for pre-paid billing scenarios. For details please visit Billing Customizations Section 7.4 section of the handbook.

7.3 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 7.1.1, 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.

7.3.1 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`.

Note

Apart from the daily fraud detection check service, NGCP also provides instant, "hard" locking for prepaid use cases, by means of billing profile packages. See Billing Profile Packages Section 7.4.3 for reference.

7.4 Billing Customizations

The standard way of doing the billing — i.e. having fixed billing intervals of a calendar month, starting on the 1st day of month — may not fit all billing profiles and intervals that sip:provider PRO platform operators would like to use.

The sip:provider PRO supports — starting from its mr4.2.1 version — alternate ways of defining billing profiles and intervals which are especially worthy for pre-paid scenarios. New functionality is covered by the following titles:

1. Billing Networks Section 7.4.1
2. Profile Mappings Schedule Section 7.4.2
3. Profile Packages Section 7.4.3
4. Vouchers Section 7.4.4
5. Top-up Section 7.4.5
6. Balance Overviews Section 7.4.6

7. Usage Examples Section 7.4.7

Subsequent sections will provide an introduction and configuration instructions to these advanced features of sip:provider PRO.

7.4.1 Billing Networks

The idea is to dynamically select billing profiles (including fees) depending on the IP network the caller‘s SIP client is using to connect. The caller‘s IP is populated in a call‘s CDR, and effectively processed by:

- the rating engine component („rate-o-mat“) and the
- prepaid interception module (libswrate).

The billing profile for rating a call is identified by matching the source IP against network ranges linked to the customer contract‘s billing mappings records. This feature is sometimes also referred to as *roaming*.

A *Billing Network* is defined as a series of *network blocks* where each network block consists of *a single IP address* or *an IP subnet*. Blocks of a particular billing network can be defined by either IPv4, or IPv6 addresses but not mixed.

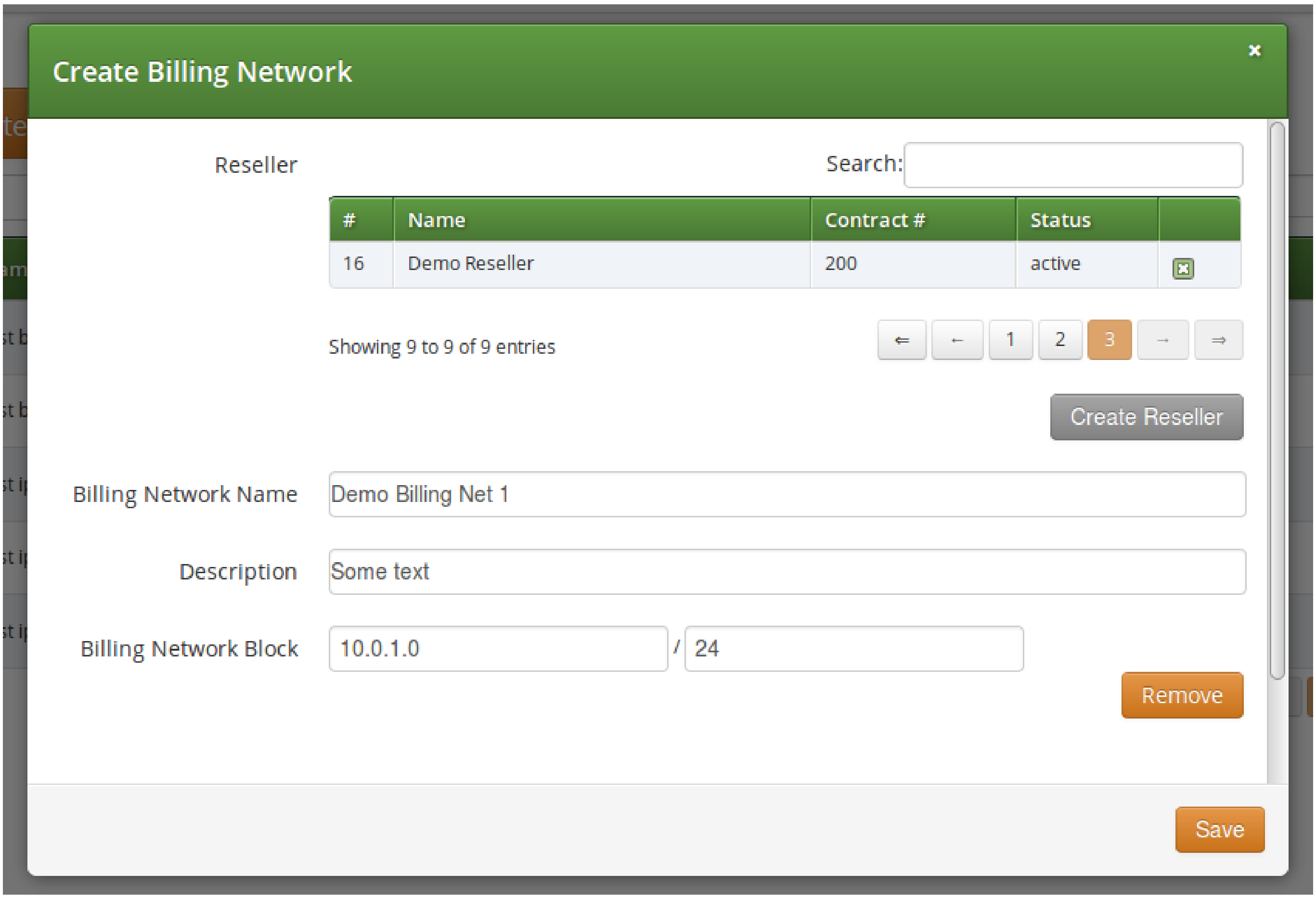


Figure 23: Creation of Billing Network

The new /api/billingnetworks/ **REST API** resource allows to manage billing networks. The example billing network that is shown in the figure above may be defined through the API with this JSON structure:

```
{ "blocks" : [ { "ip" : "10.0.1.0", // subnet: 10.0.1.0 .. 10.0.1.255
                 "mask" : 24
               },
               { "ip" : "10.0.2.2" // single ip
               }
             ],
  "description" : "Some text",
  "name" : "Demo Billing Net 1", //unique per reseller
  "reseller_id" : 1
}
```

Input validation of the network blocks is automatically performed by sip:provider PRO during their definition in a way that it prevents specifying overlapping blocks by means of Interval Trees; billing networks themselves may overlap though.

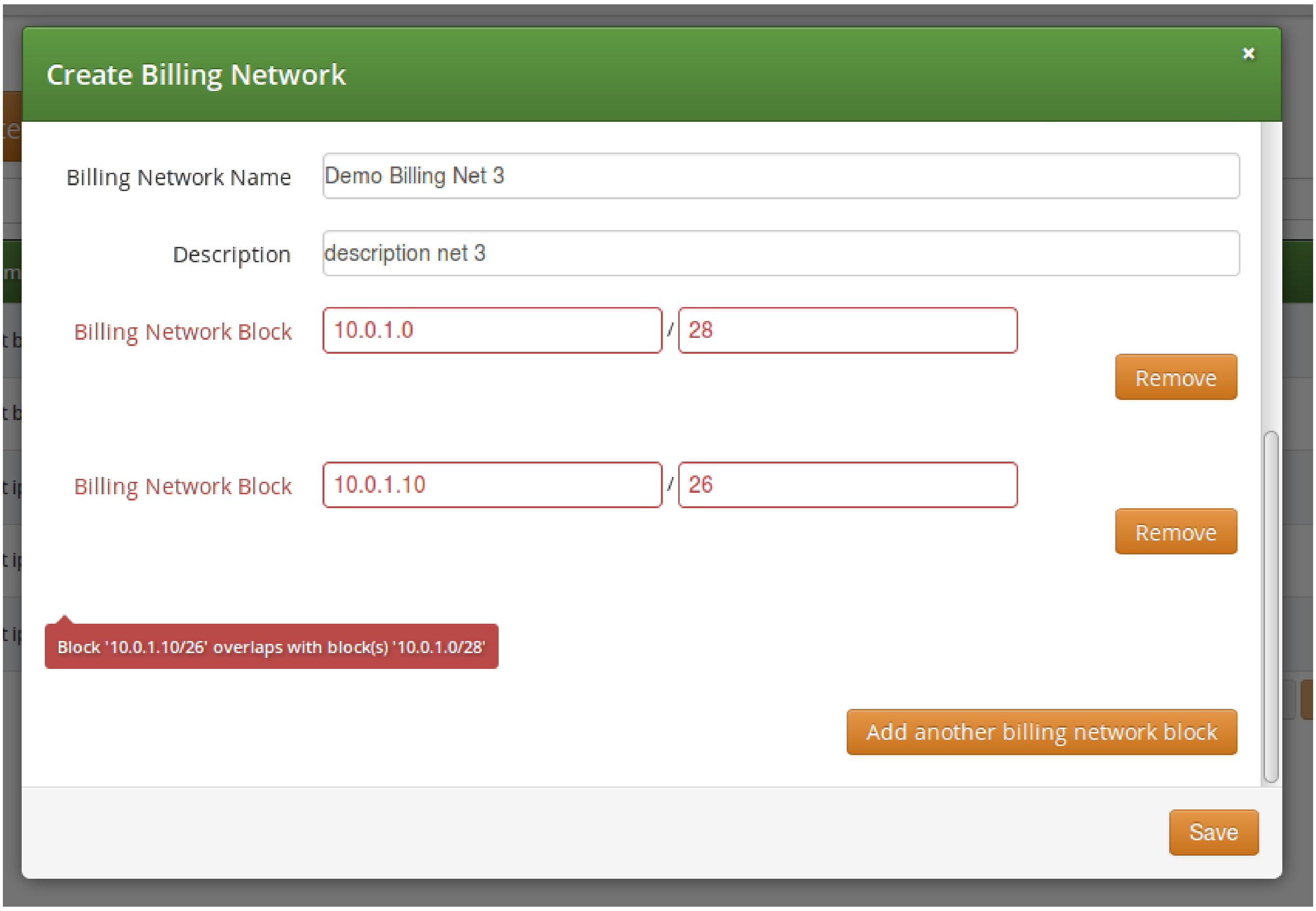


Figure 24: Overlapping Block Prevention

7.4.2 Profile Mapping Schedule

Using the default settings related to billing when creating a new *Reseller* or *Customer* on the administrative web panel results in applying the standard billing profile mapping schedule: the same billing profile is always used.

7.4.2.1 Definition of Profile Mapping Schedules

The idea of *billing profile mapping schedule* is to extend the billing mappings logic to utilize it as a schedule for billing profiles (and associated fees) for the *Customer* or *Reseller* contract. So far, billing mapping records provided only a history showing which profile was in effect at a given time in the past, which is for example required for delayed rating of calls.

Now it is also possible to define in advance, when specific billing profiles should become active in the future, e.g. to plan campaigns or special offers.

Billing profile mappings represent a schedule of overlapping time intervals with *Billing Profiles* and *Billing Networks*, which are assigned to (customer) contracts when creating or editing them.

Mapping intervals can be of type:

- open: no start time + no end time
- half-open:
  - left-open: no start time + definite end time
  - right-open: definite start time + no end time
- closed: definite start time + definite end time

7.4.2.2 Schedule Example

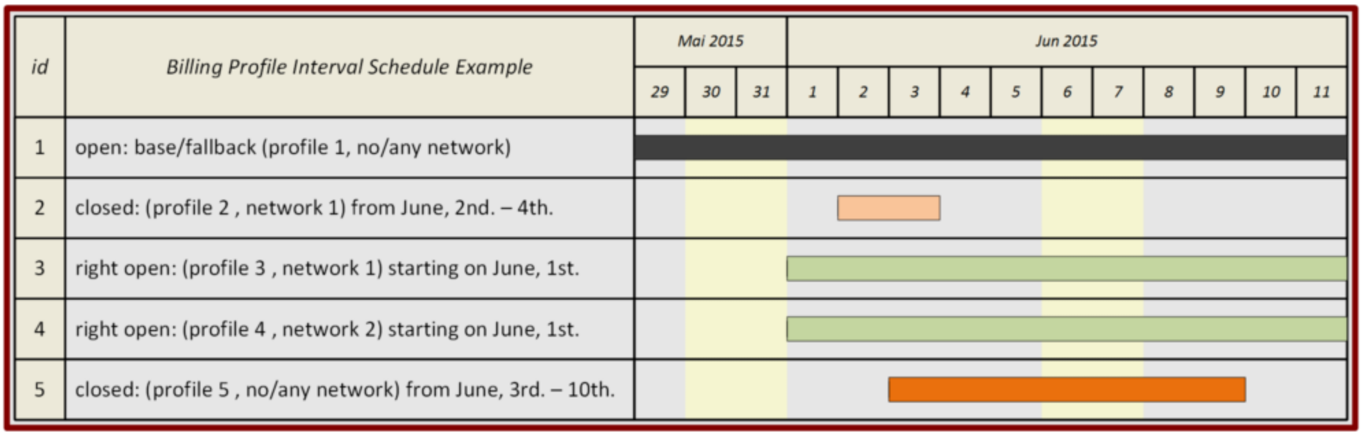


Figure 25: Profile Mapping Schedule Example

Applying the profile mapping schedule shown in the above figure will result in billing profiles being active as provided in the table below.

Table 10: Active Billing Profiles

Time	Web Panel shows	Rating		
		Caller IP in Network 1	Caller IP in Network 2	Caller IP in other network
May 30	*Profile 1*	*Profile 1*	*Profile 1*	*Profile 1*
June 1	*Profile 4*	*Profile 3*	*Profile 4*	*Profile 1*
June 2	*Profile 2*	*Profile 2*	*Profile 4*	*Profile 1*
June 5	*Profile 5*	*Profile 3*	*Profile 4*	*Profile 5*

7.4.2.3 Configuration of Schedules

A *Customer's* default billing profile mapping can be changed to scheduled mappings when editing its properties, at the parameter "Set billing profiles", selecting: `schedule (billing mapping intervals)`

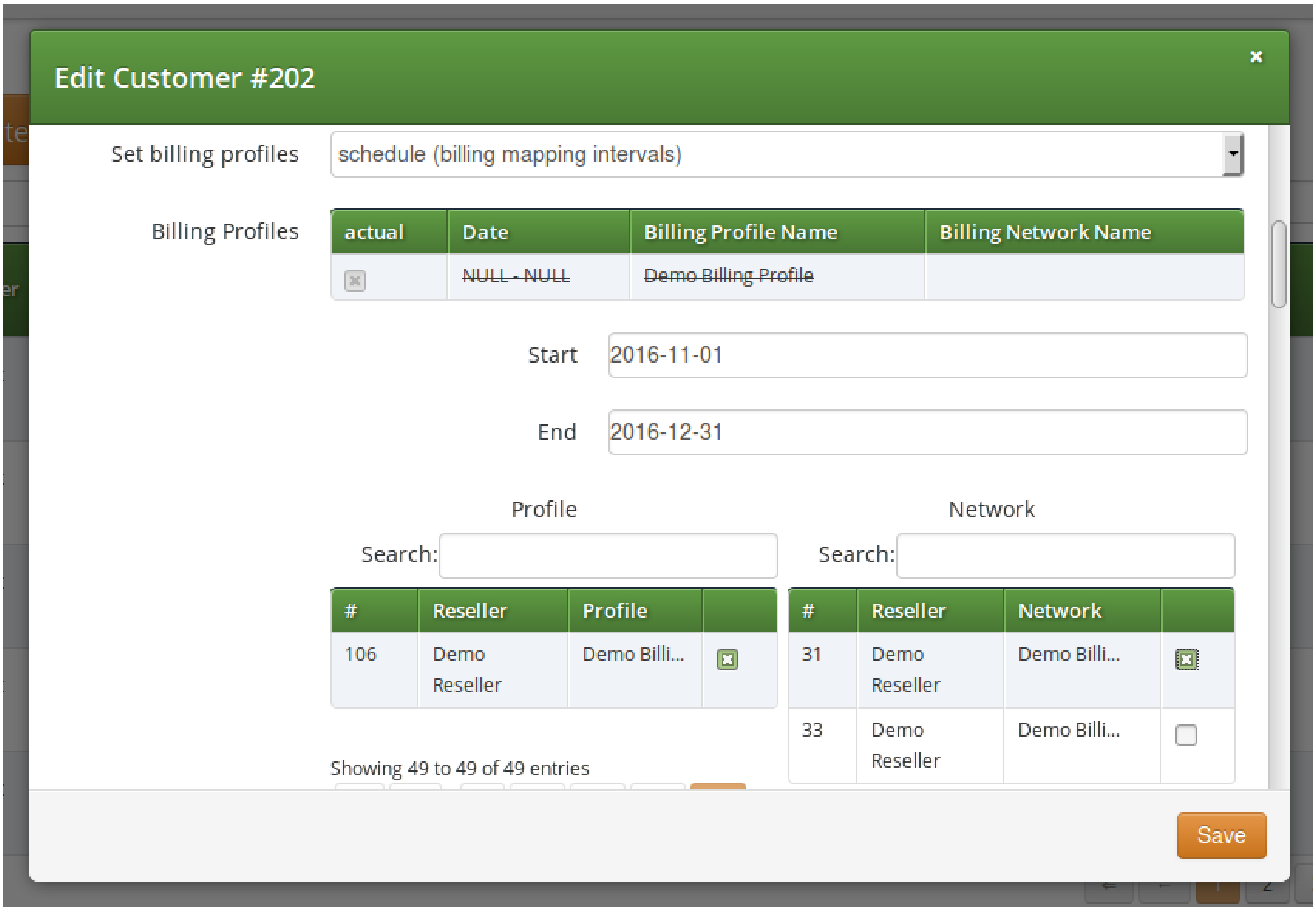


Figure 26: Profile Mapping Schedule Creation

Tip

Assigning a *Billing Network* to a billing profile mapping is optional. Without selecting the network, the *Billing Profile* will be applied to all calls.

The profile mapping schedule assigned to a *Customer* is also listed among *Customer's* properties. See *Settings* → *Customers* → *Details* → *Billing Profile Schedule*.

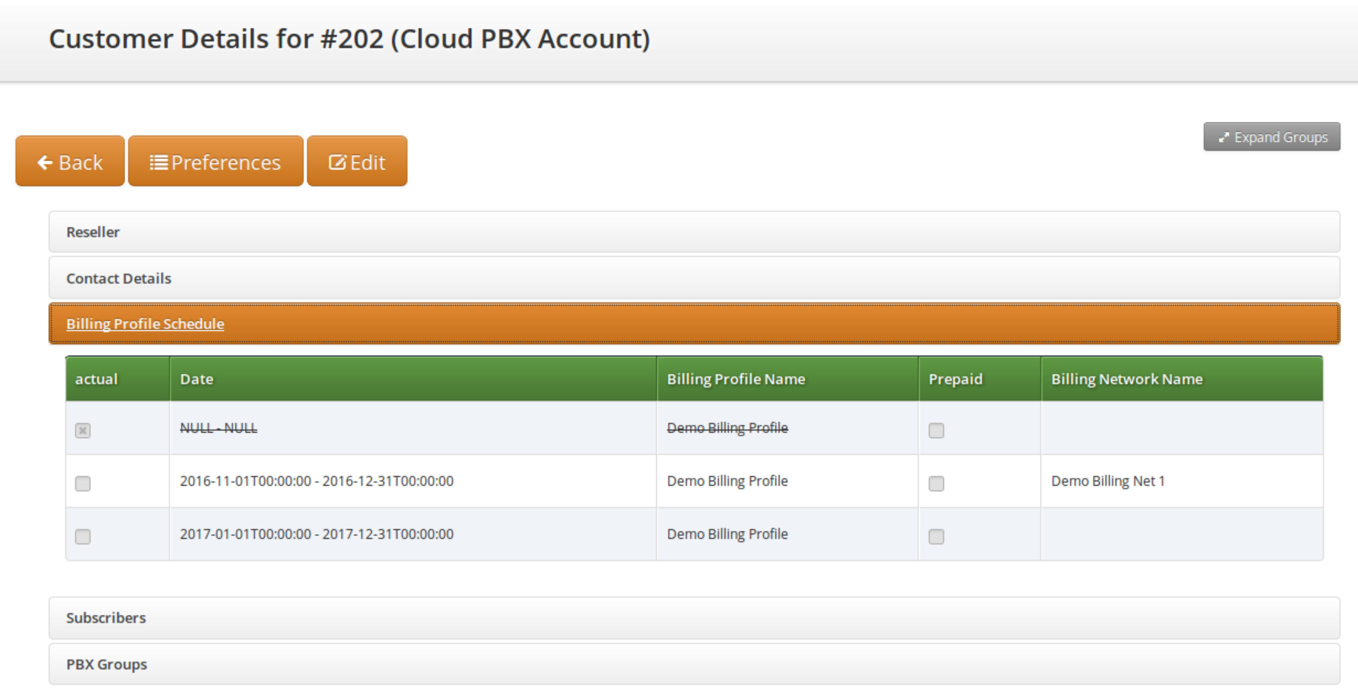


Figure 27: Profile Mapping Schedule List

Note

Profile mappings that started in the past, like the default one, are displayed with a strike-through font in order to indicate that those can not be modified.

The currently active mapping is depicted by a checked box.

7.4.2.4 REST API for Profile Mapping Schedules

The `/api/customers/` API resource was extended to provide three different modes of defining profile mappings:

1. `billing_profiles` field: explicitly declare profile mappings in form of `(billing profile, billing network, start time, stop time)` tuples
2. `billing_profile_id` field (legacy API spec): a single profile mapping interval is appended `(billing profile, no network /any caller IP respectively, starting now)`

3. `profile_package_id` field: profile mappings starting now are appended by using lists of `(billing profile, billing network)` tuples from the given profile package

With regards to *Resellers*, the `/api/contracts/` API resource was enhanced as well, but supports method 1. and 2. only, and without billing networks.

Mapping Intervals

Intervals can be of open, half-open (left-open, right-open) or closed type. When specifying profile mappings discretely, allowed interval types are restricted, depending on create/update situation:

Table 11: Allowed Mapping Intervals

Interval Type	Start	Stop	POST (create)	PUT / PATCH (update)
`open`	undefined	undefined	`1..*`	`0`
`left-open`	undefined	defined	`0`	`0`
`right-open`	> now()	undefined	`*`	`*`
`closed`	> now()	> start	`*`	`*`

Example Profile Mapping

An example JSON structure for definition of profile mapping schedules shown in Billing Profile Schedule List Figure 27 :

```
{ ...,
  "billing_profile_definition" : "profiles", // i.e. use 'billing_profiles' field
  "billing_profiles" : [ { "network_id" : "236",
                           "profile_id" : "236",
                           "start" : "2016-11-01 00:00:00",
                           "stop" : "2016-12-31 00:00:00"
                         }, // closed future interval, with network
                         { "network_id" : null,
                           "profile_id" : "237",
                           "start" : "2017-01-01 00:00:00",
                           "stop" : "2017-12-31 00:00:00"
                         } ], // closed future interval, without network
  "contact_id" : 141,
  ...
}
```

7.4.3 Profile Packages

By introducing billing profile packages, general billing parameters can be defined for a customer contract:

- Balance interval duration (regular/constant or aligned to top-up events)
- The first interval‘s start date
- The cash-balance carry-over/discard behaviour upon interval transitions
- Subscriber lock levels and profile sets to get applied upon:
  - top-up
  - balance threshold underrun
- Initial balance and billing profiles

Profile Packages are fundamental for pre-paid billing scenarios, since in such a billing scheme the traditional, fixed monthly periods prove to be insufficient to cover the business needs of the NGCP platform operator. As an example: pre-paid subscribers typically have their "billing periods" between account balance top-ups.

7.4.3.1 Elements of Profile Packages

A *Profile Package* consists of various elements that will be discussed in subsequent sections of the sip:provider PRO handbook. In order to set the parameters of a profile package one must navigate to: *Settings* → *Profile Packages* → *Create Profile Package*, or alternatively, in order to update an existing profile package: select the package and press *Edit* button.

Basic Balance Intervals Setup

- Interval duration (n hours, days, weeks, months)
- Interval start mode:
  - 1st of month (1st): billing interval is 1 calendar month; this is the default for each *Customer* created on Sipwise NGCP platform

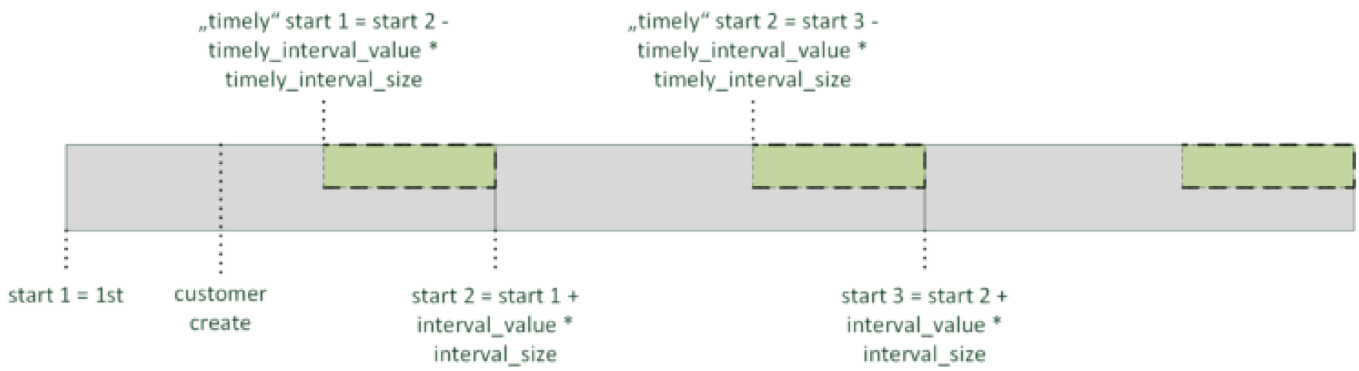


Figure 28: Interval Start Mode: 1st

  - upon customer creation (create): (the initial) billing interval starts when the *Customer* is created

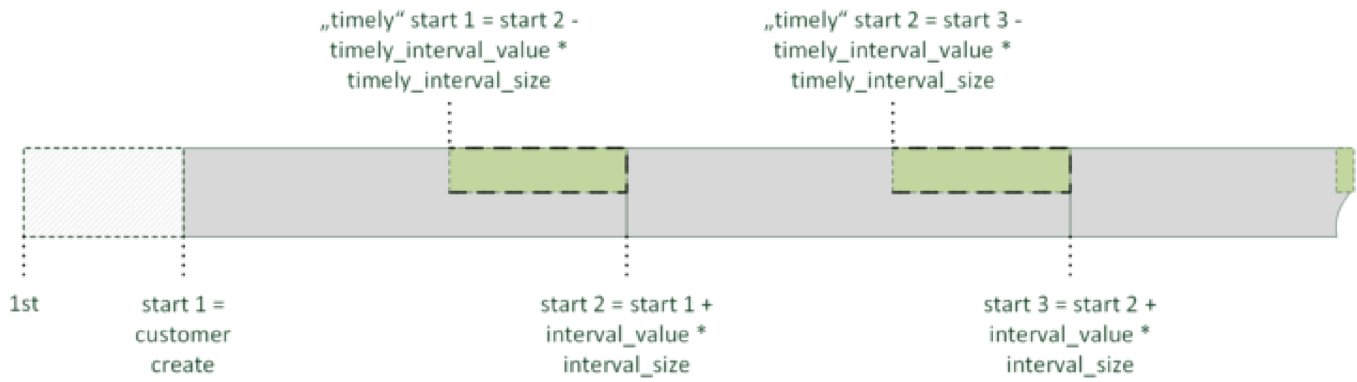


Figure 29: Interval Start Mode: create

- upon topup (topup_interval): interval starts at *first topup* event and its length is defined by `interval duration` parameter of the profile package

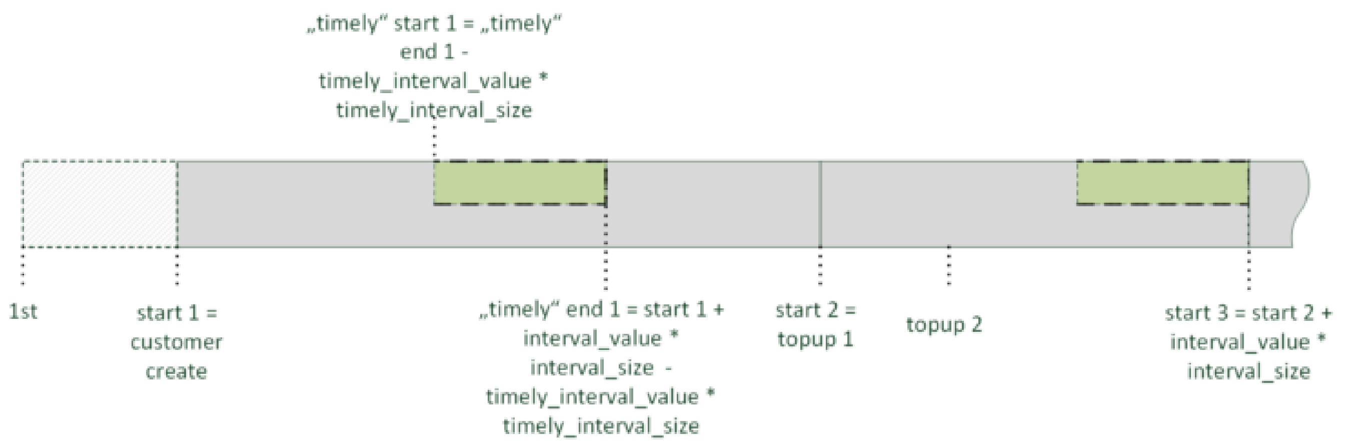


Figure 30: Interval Start Mode: topup_interval

- intervals from topup to topup (topup): interval starts at *any topup* event and its length is defined by `interval duration` parameter of the profile package; intervals can overlap in this case

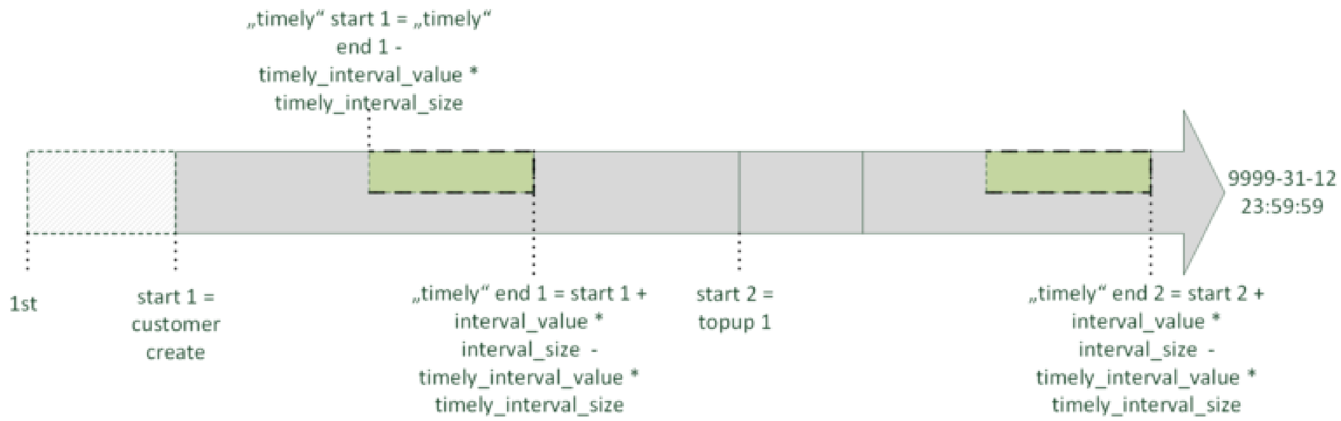


Figure 31: Interval Start Mode: topup

- Initial balance: the initial value of account balance (e.g. every new customer gets 5 Euros as a starting bonus)

Balance Carry Over

- Carry Over: balance carry over behaviour upon interval transitions:
  - `carry-over`: always keep balance
  - `carry-over only if topped-up timely`: keep balance in case of a *timely* top-up only; where **timely** means the topup happens within a pre-defined time span before the end of the balance interval
  - `discard`: discard balance at the end of each interval
- Timely Duration: duration of the *timely* period
- Discard balance after intervals: for how many balance intervals the remaining account balance is kept before its disposal

Underrun Settings

- Underrun lock threshold: when account balance reaches this amount the subscriber will be locked to a restricted set of services
- Underrun lock level: this level of services will apply when an account balance underruns
  - `don't change`: no change in the available set of services
  - `no lock`: all services are available
  - `foreign`: only calls within subscriber's own domain are allowed
  - `outgoing`: all outgoing calls are prohibited
  - `all calls`: all calls (incoming + outgoing) are prohibited
  - `global`: all calls + access to Customer Self Care web interface are prohibited
  - `ported`: only automatic call forwarding, due to number porting, is allowed
- Underrun profile threshold: when account balance reaches this amount the *Underrun Billing Profile* will be applied

Basic Top-up Settings

- Top-up lock level: subscriber lock (unlock) levels to apply upon top-up event
- Service charge: (always) subtract this value from the voucher amount, if topup happens via the usage of a voucher

Profile mappings

A lists of (billing profile, billing network) tuples for appending profile mappings:

- Initial Billing Profile: when creating or manually changing the customers package (initial_profiles)
- Underrun Billing Profile: when the balance underuns a cash threshold (underrun_profiles)
- Top-up Billing Profile: when the customer tops-up using a voucher associated with the package (topup_profiles)

7.4.3.2 Examples

Profile Package Configuration

1. Definition of basic profile package parameters

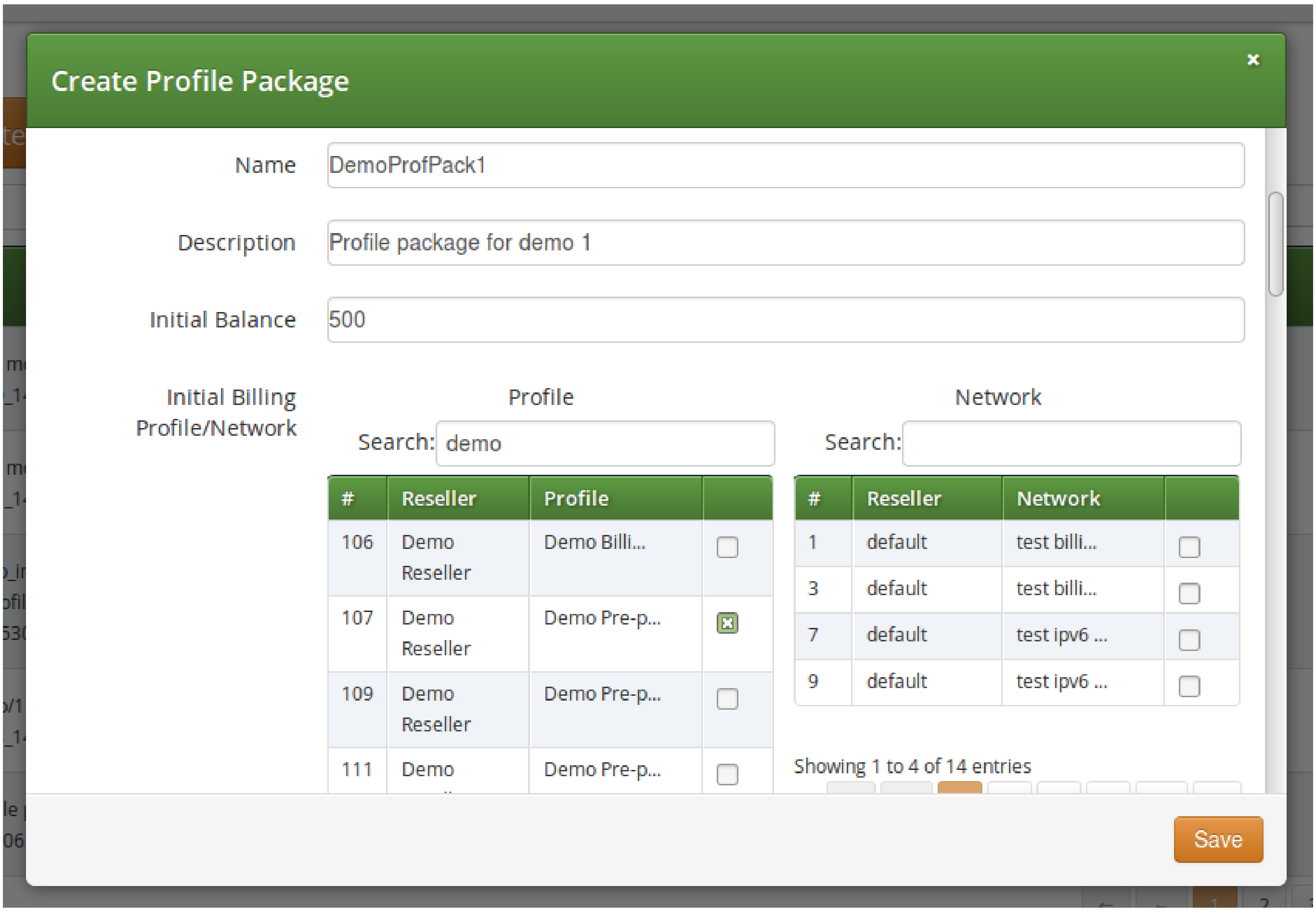


Figure 32: Basic Profile Package Parameters

2. Definition of balance interval and carry-over behaviour

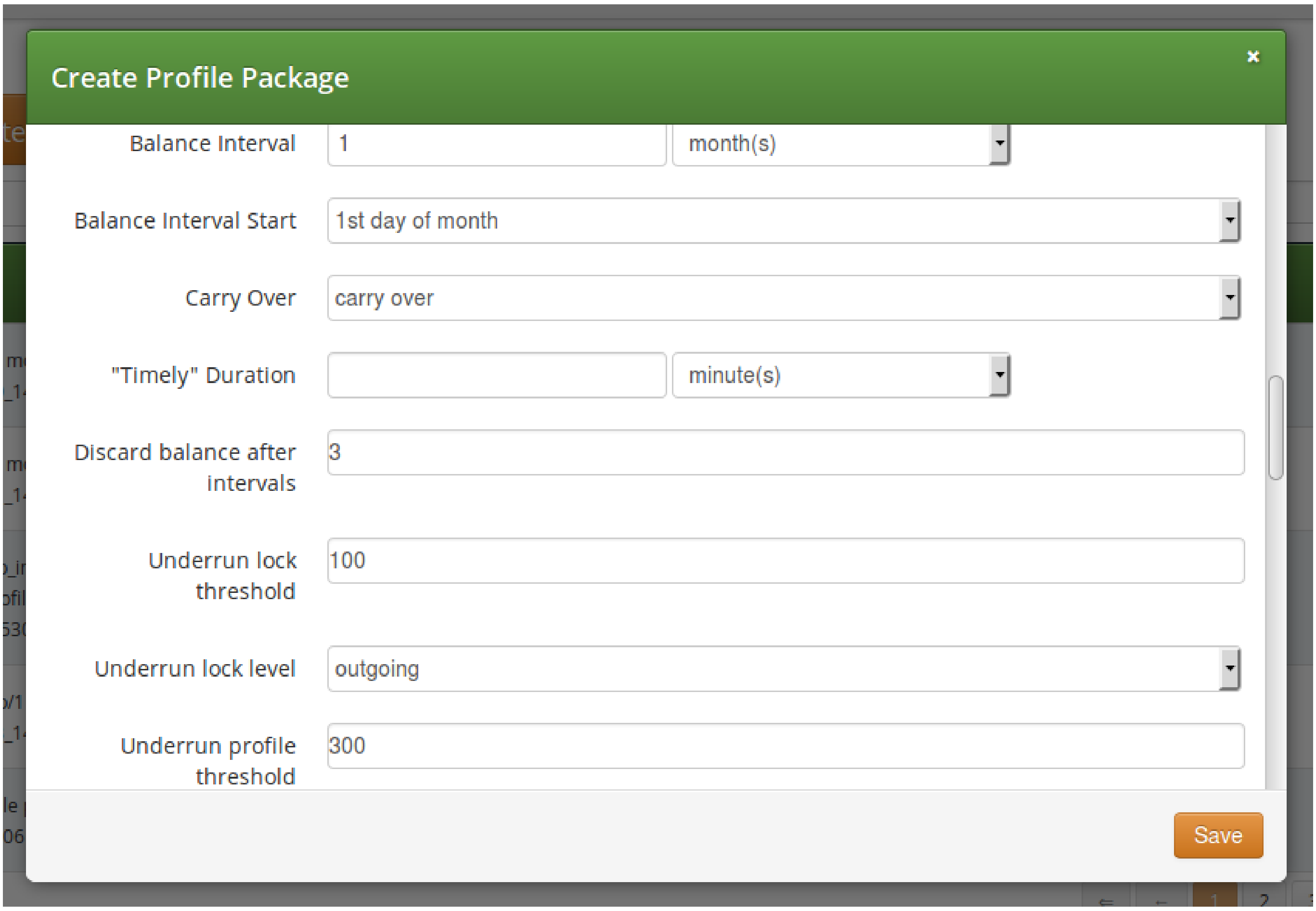


Figure 33: Balance Interval and Carry-over

3. Definition of balance underrun parameters

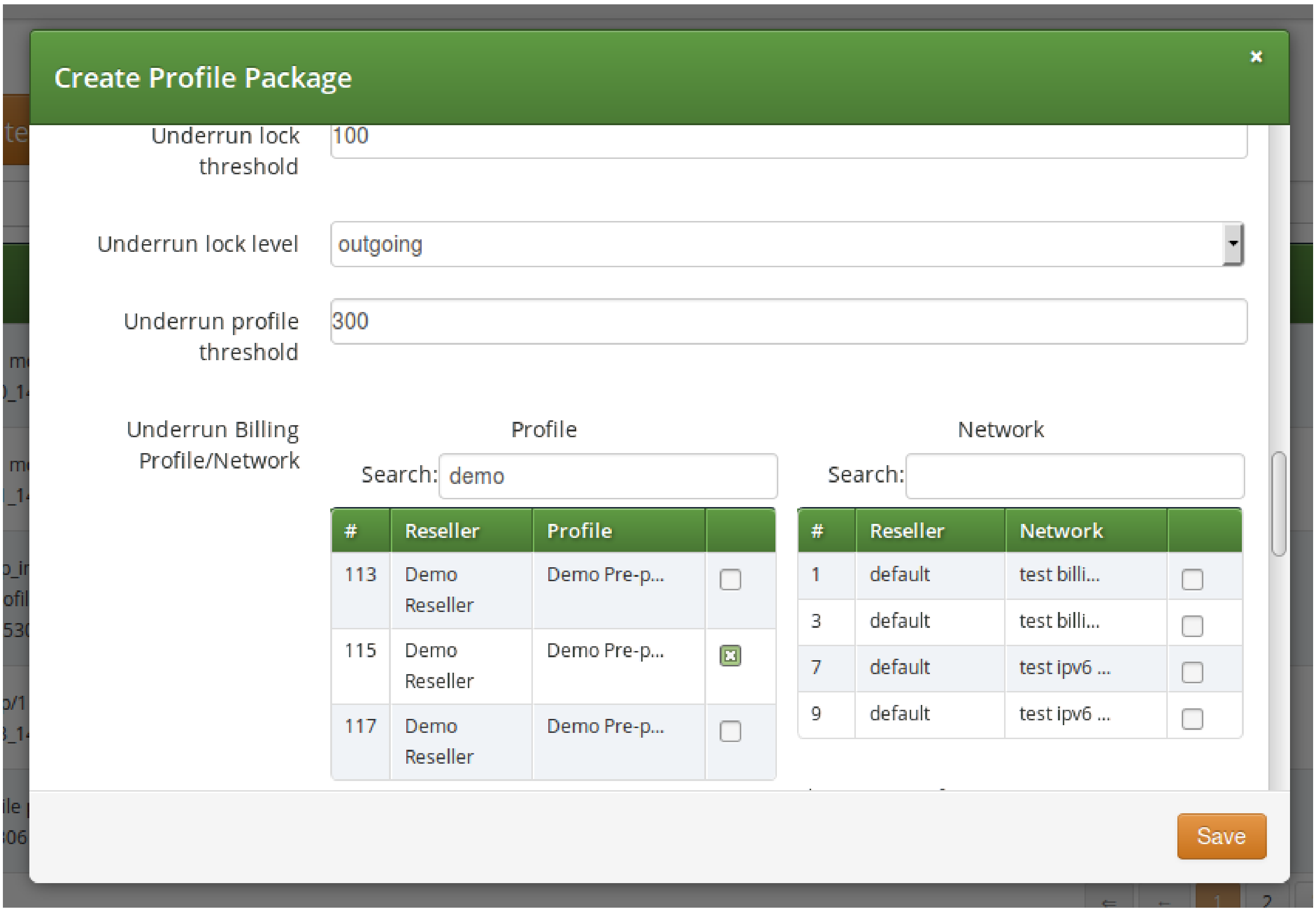


Figure 34: Balance Underrun Parameters

4. Definition of top-up settings

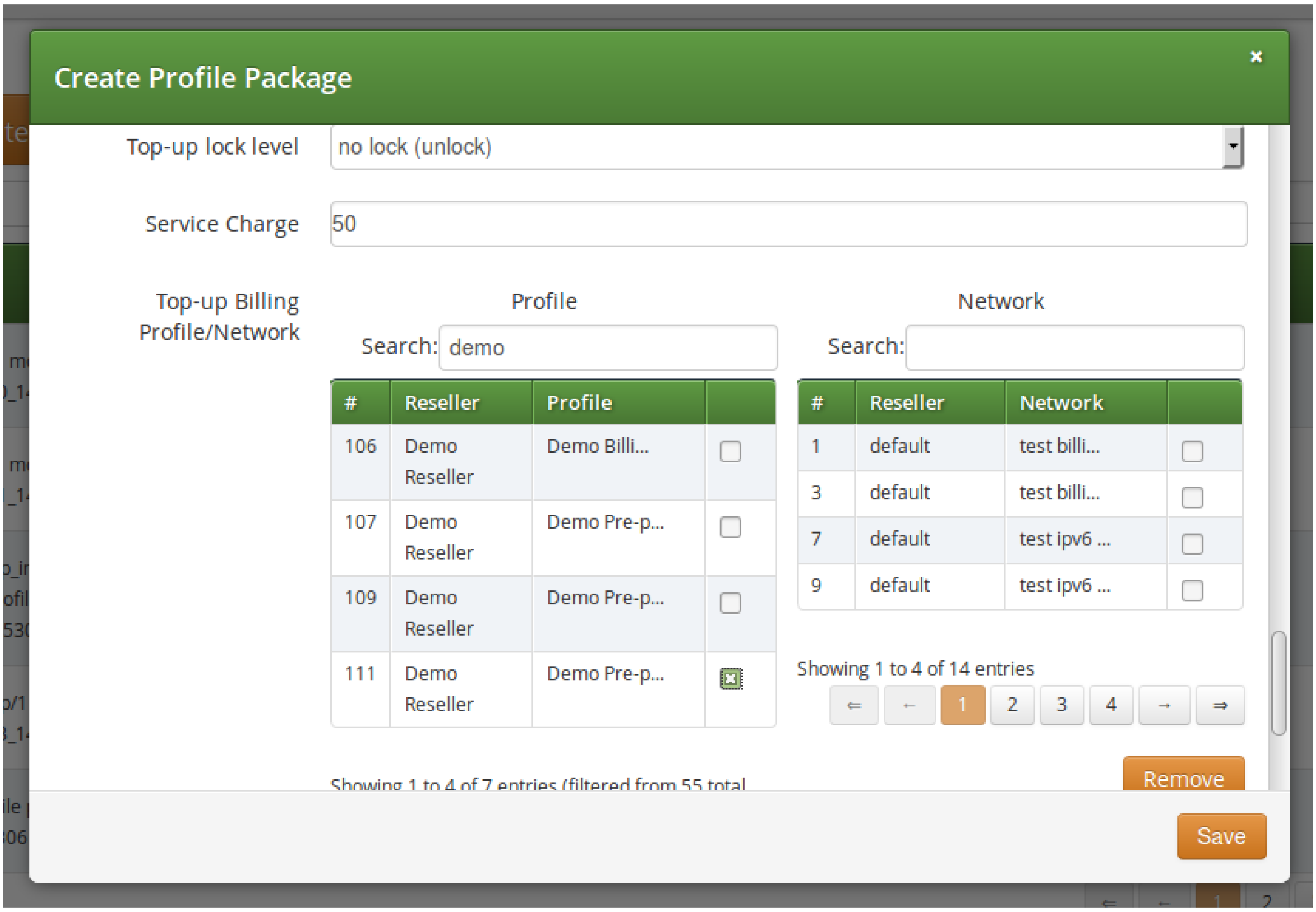


Figure 35: Balance Top-up Settings

5. Assigning a profile package to a customer

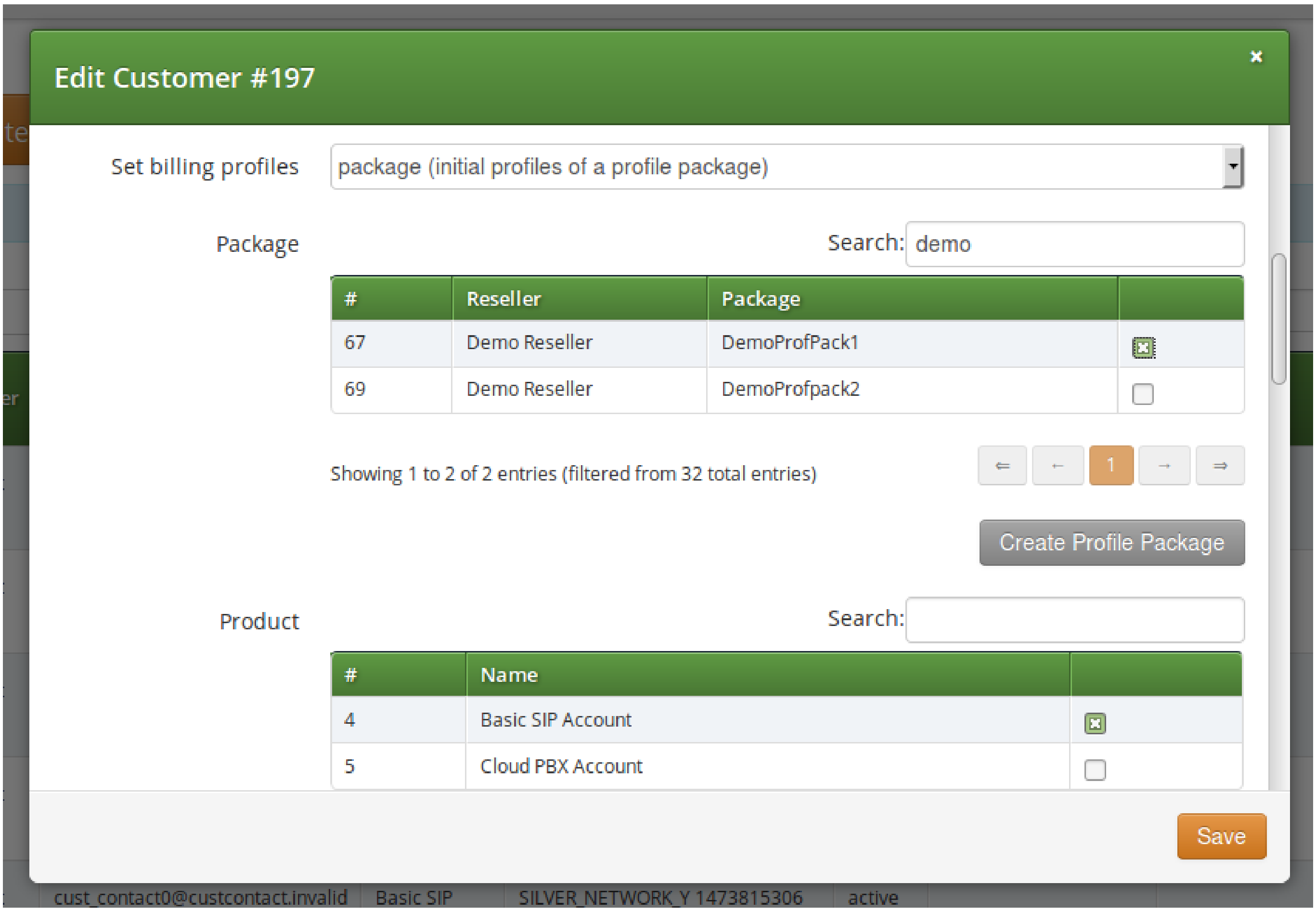


Figure 36: Assigning Profile Package to Customer

Interval start mode: top-up interval; carry-over: timely

Profile package setup:

- initial_balance: 1.0 euro
- balance_interval: 30 "day(s)“
- interval_start_mode: "topup_interval“
- carry_over_mode: "timely“
- timely_duration: 12 "day(s)“
- underrun_lock_threshold: 0.7 euro
- underrun_profile_threshold: 5.0 euro
- underrun_lock_level:. . .

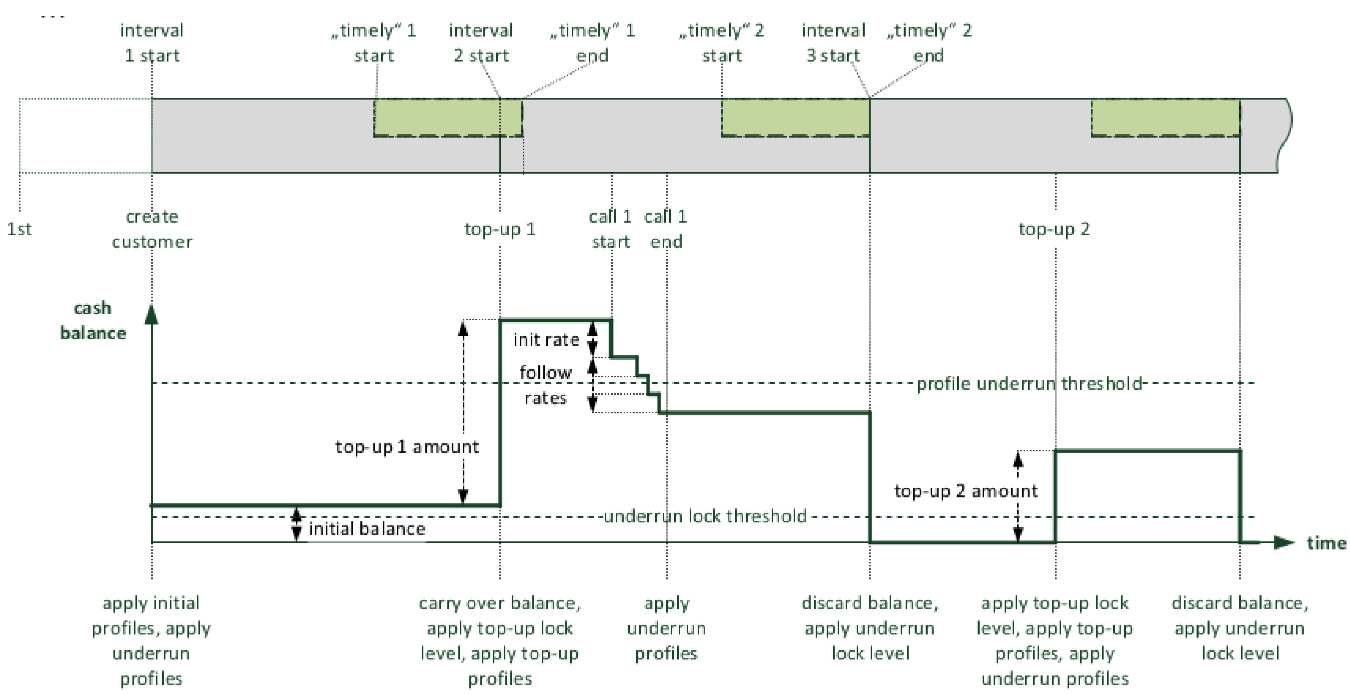


Figure 37: Example: Top-up Interval and Timely Carry-over

Interval start mode: top-up to top-up; carry-over: always

- initial_balance: 1.0 euro
- balance_interval: 30 "day(s)“
- interval_start_mode: "topup“
- carry_over_mode: "carry-over“
- notopup_discard_intervals: 1
- underrun_lock_threshold: 0.7 euro
- underrun_profile_threshold: 5.0 euro
- underrun_lock_level:. . .

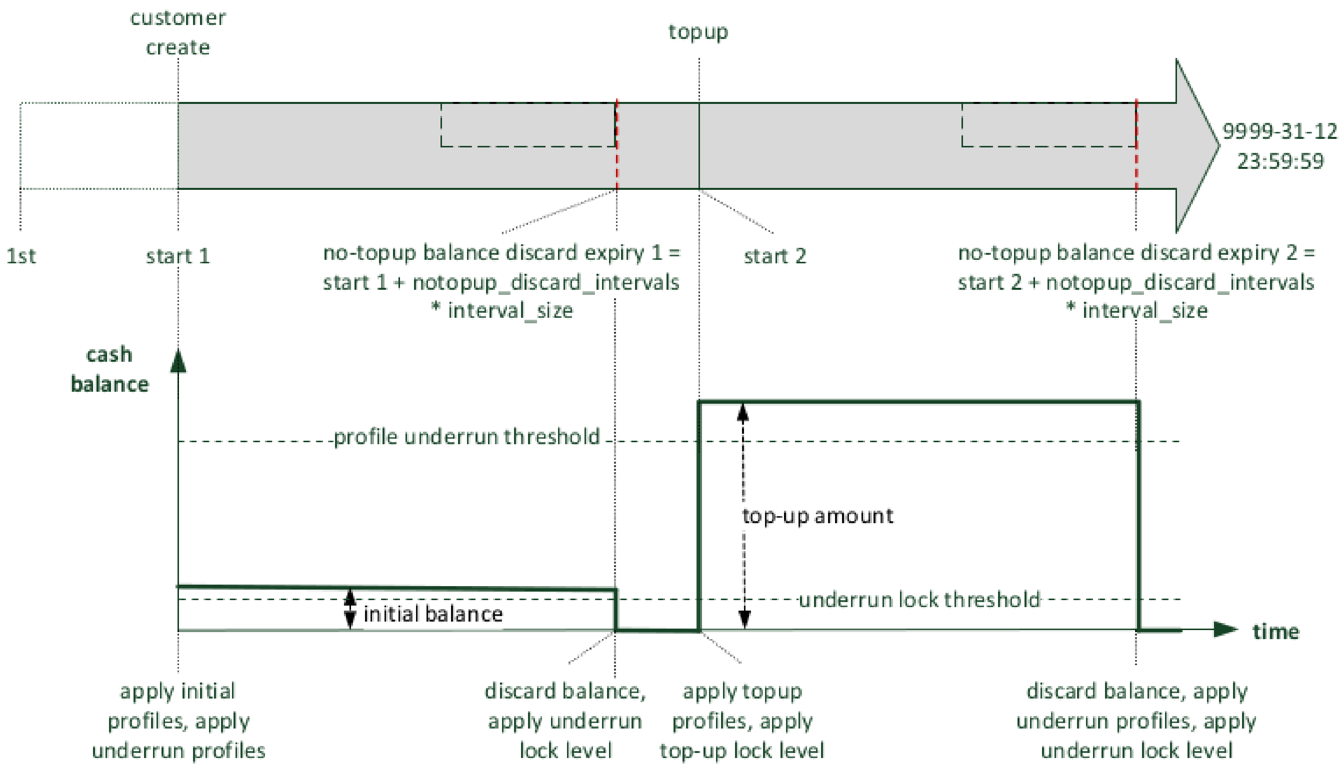


Figure 38: Example: Top-up and Always Carry-over

7.4.3.3 REST API

The new `/api/profilepackages/` REST API resource allows to manage billing profile package container entities, that aggregate settings of profile packages.

A sample JSON structure follows:

```
{
  "reseller_id" : 1,
  "status" : "active",
  "name" : "demo profile package",
  "description" : "package for 10€ ...",
  "balance_interval_start_mode" : "1st",
  "balance_interval_value" : 1,
  "balance_interval_unit" : "month",
  "carry_over_mode" : "carry_over",
  "timely_duration_unit" : null,
  "timely_duration_value" : null,
  "initial_balance" : 0,
  "initial_profiles" : [...], // required default, e.g. same as „topup_profiles‟
  "notopup_discard_intervals" : null,
  "underrun_lock_threshold" : 0,
  "underrun_lock_level" : 4,
```

```
  "underrun_profile_threshold" : 5,
  "underrun_profiles" : [...],
  "service_charge" : 10,
  "topup_lock_level" : null,
  "topup_profiles" : [ {
                         "network_id" : null, // any network
                         "profile_id" : 29
                       },
                       {
                         "network_id" : 2, // a specific billing network
                         "profile_id" : 30
                       },
                     ],
  ...
}
```

7.4.4 Vouchers

Vouchers are a typical mean of topping-up an account balance in pre-paid billing scenarios.

The definition of a voucher in the database may succeed via:

- manual entry of voucher data on the administrative web panel or through the REST API
- bulk-uploading of vouchers using a CSV (comma separated value) formatted file

In order to manage vouchers the administrator has to navigate to: *Settings* → *Vouchers* → *Create Billing Voucher* or select an existing one and press *Edit* button.

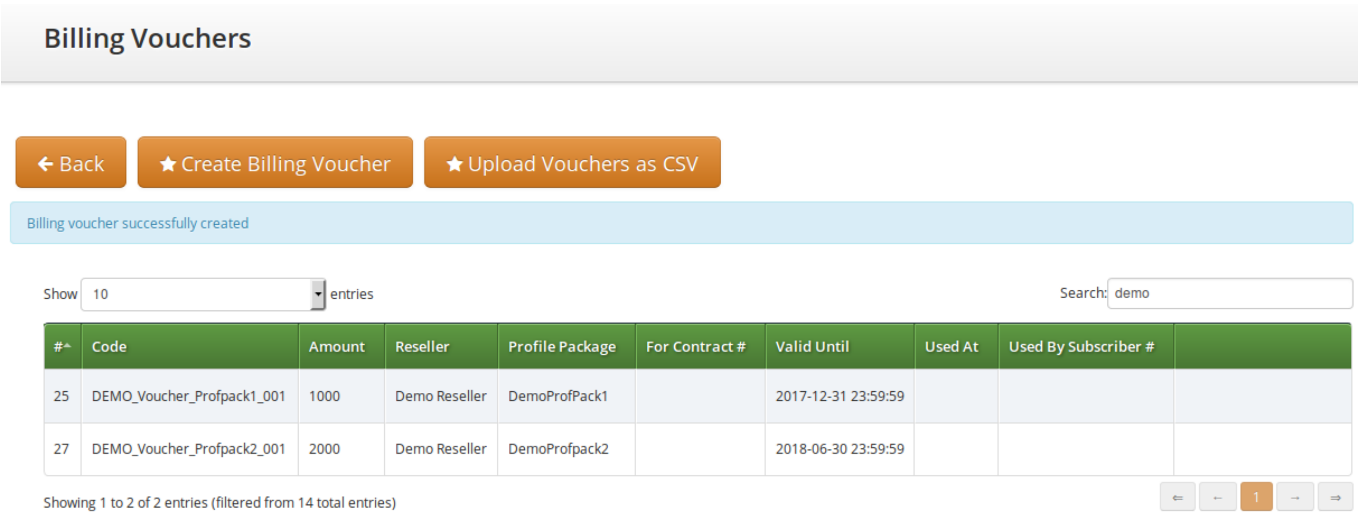


Figure 39: List of Vouchers

7.4.4.1 Properties of Vouchers

- Code: the unique code of the voucher which assures that a voucher can be used only once; this property is encrypted and displayed on the web panel to authorized users only
- Amount: the amount of money the voucher represents
- Valid until: end of validity period

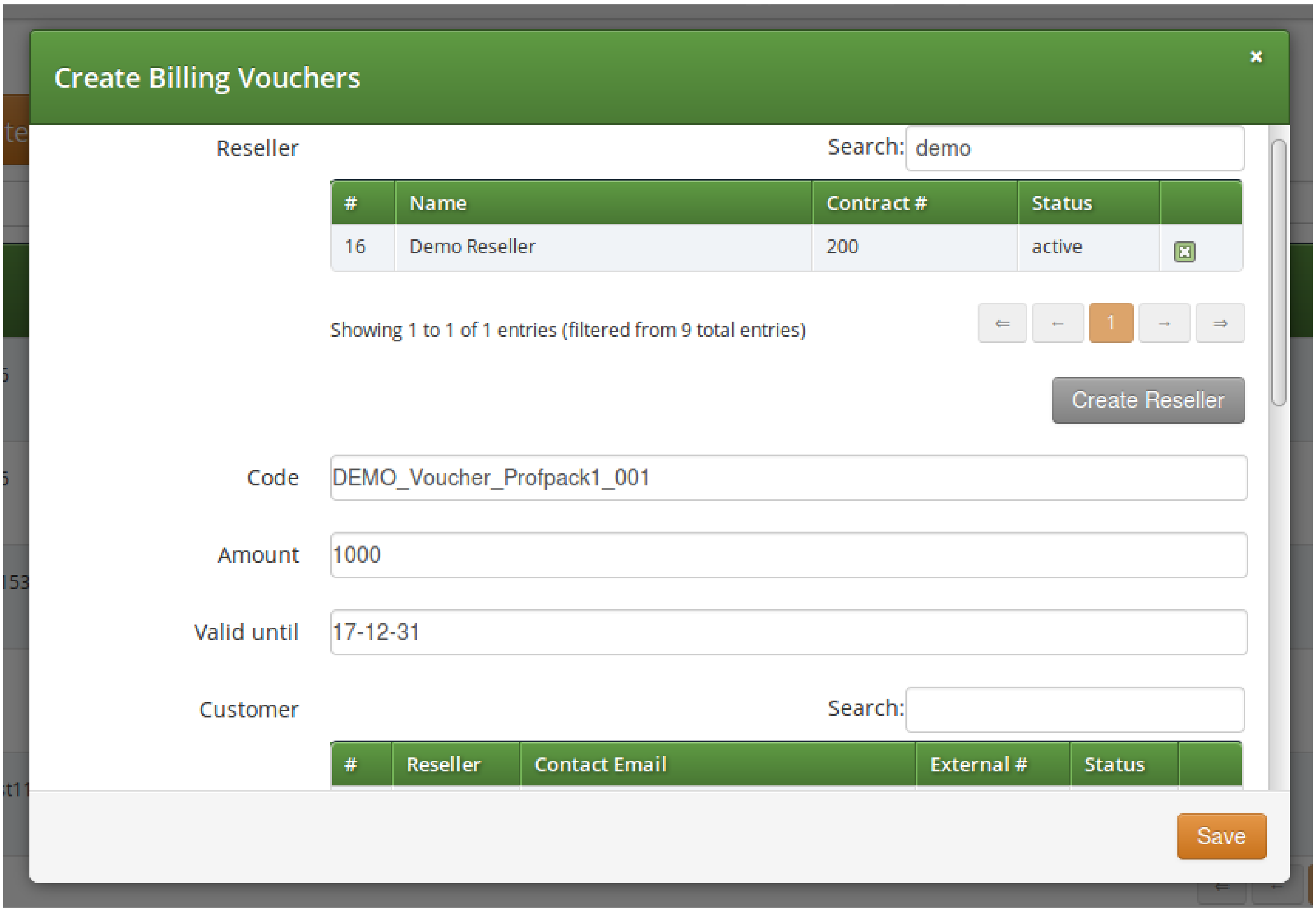


Figure 40: Voucher's Main Properties

Setting following properties of a voucher is optional:

- Customer: the *Customer* whom the voucher will be assigned to; subscribers of other customers can not redeem the voucher
- Package: vouchers may be associated with profile packages; if done so, some changes will be applied to the *Customer* for whom the voucher is redeemed with the top-up event:
  - applying top-up profile mappings starting with the time of the top-up
  - subtracting the new package's service charge from the voucher amount

– resizing the current balance interval for a gapless transition, if the new package has a different interval start mode (e.g. from "create" to "1st")

– if a new balance interval starts with the top-up, the carry-over mode of the customer‘s previous package applies

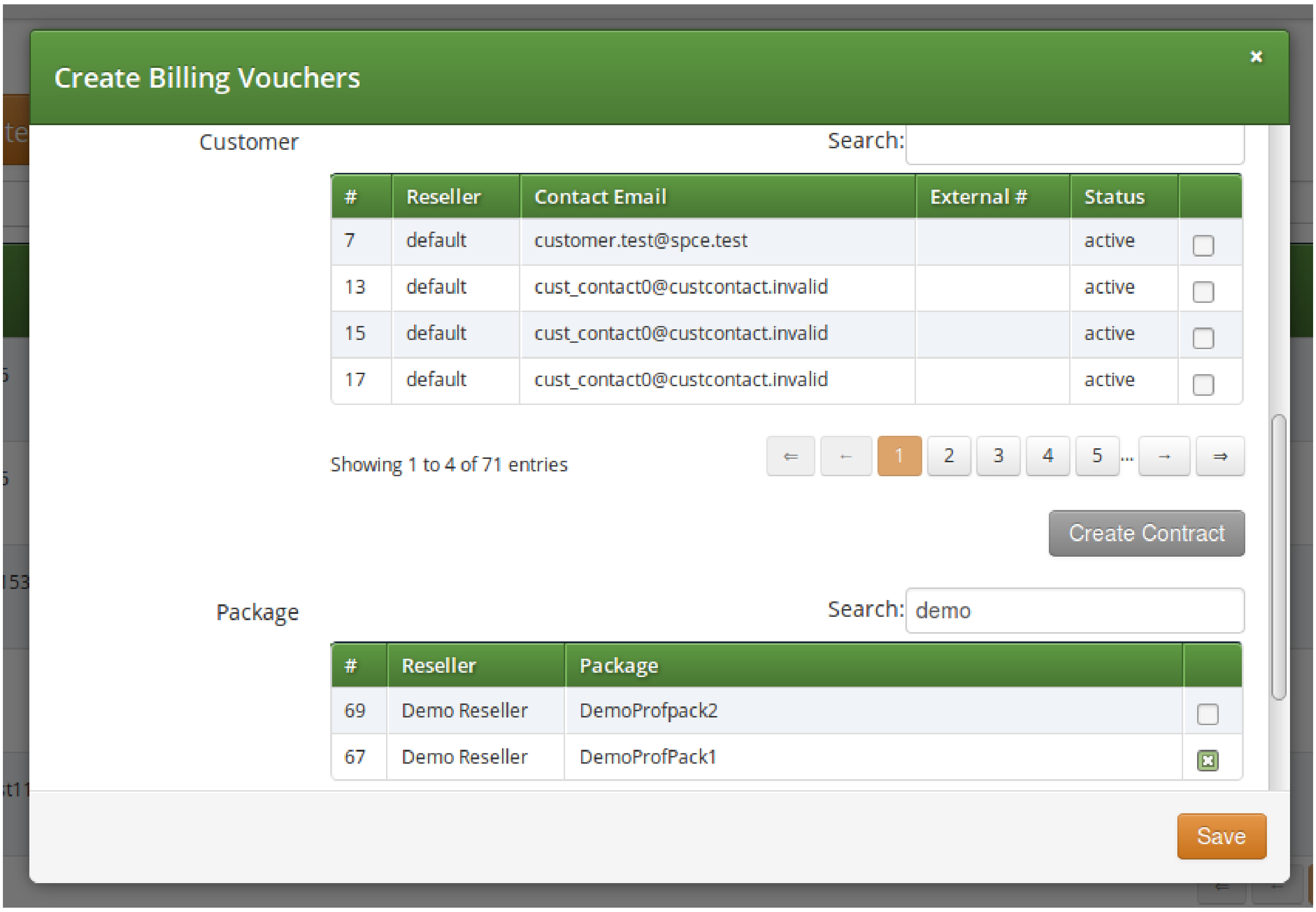


Figure 41: Voucher: Customer and Profile Package

7.4.4.2 REST API

Vouchers can be created and managed using the `/api/vouchers/` REST API resource. This resource restricts invasive operations (POST, PUT, PATCH, DELETE) to authorized users.

```
{
  "amount" : 1000,
  "customer_id" : null, //do not restrict to a specific customer
  "valid_until" : "2017-06-05 23:59:59",
  "package_id" : "571", //switch to profile package
  "reseller_id" : 1,
  "code" : "SILVER_1_1437974823"
}
```

7.4.5 Top-up

A customer's administrator or subscriber can perform a top-up to increase the contract's cash balance. The NGCP platform supports two means of topping-up the balance:

1. Top-up Cash: Directly specify the cash amount to add
2. Top-up Voucher: Specify the code of a voucher, which was set up in advance

The NGCP platform provides 2 interfaces to perform top-ups:

1. through the REST API: use a CRM or third-party REST-API Broker (which i.e. coordinates with an App-Store purchase process) to finally instruct NGCP to perform a top-up. This is the **recommended** method.
2. through the administrative web interface:

   One has to select the *Customer*, then *Details* → *Contract Balance* and finally press *Top-up Cash* or *Top-up Voucher*.

7.4.5.1 Top-up Cash

When doing top-up with cash one needs to supply the amount of top-up in the currency of the customer contract. Optionally one can assign a *Profile Package* to the top-up event which will activate that profile package for the customer.

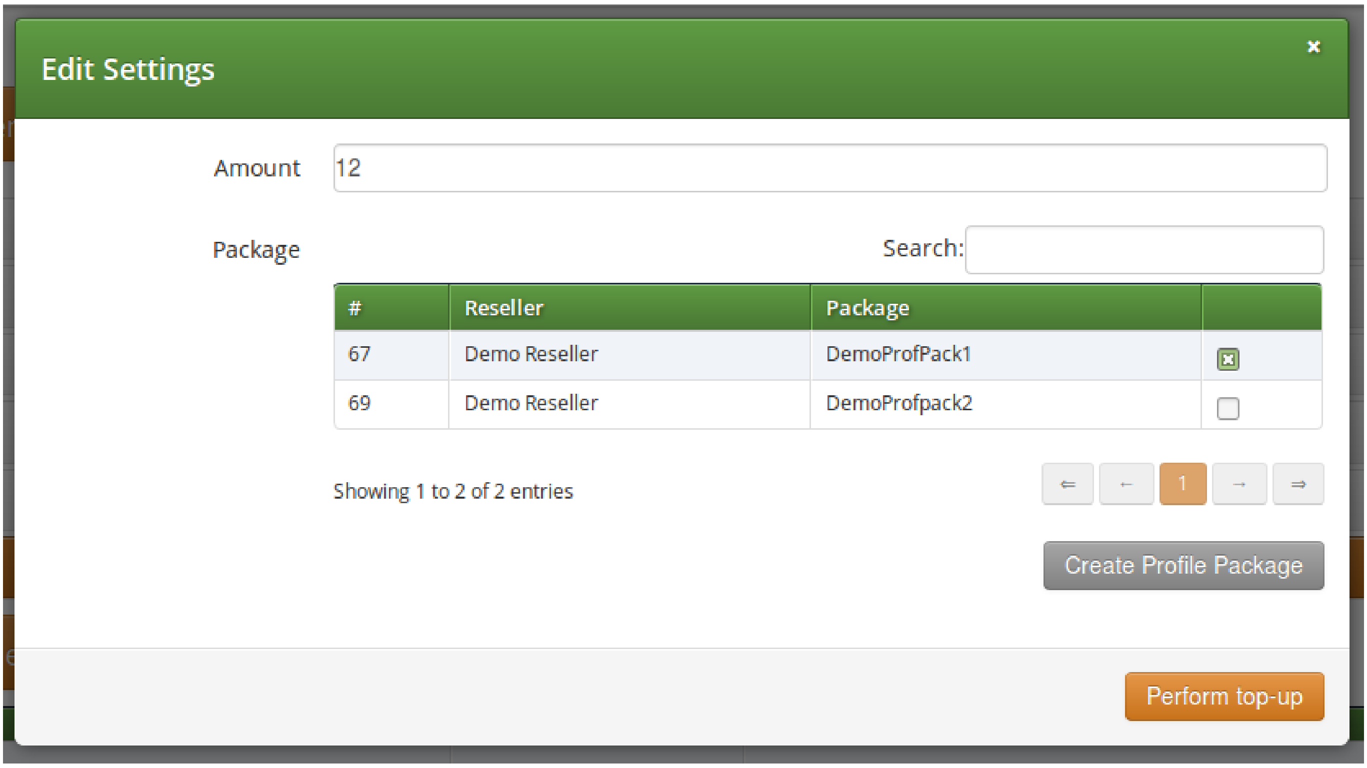


Figure 42: Balance Top-up with Cash

It is also possible to perform top-up through the **REST API**: `POST /api/topupcash`

```
{
  "subscriber_id" : "73",
  "amount" : 100,
  "package_id" : null,
}
```

7.4.5.2 Top-up Voucher

Selecting *Top-up Voucher* option will provide a simple list of available vouchers from which the administrator can choose the voucher. If a *Profile Package* is assigned to the voucher, that package will be activated for the customer on the top-up event.

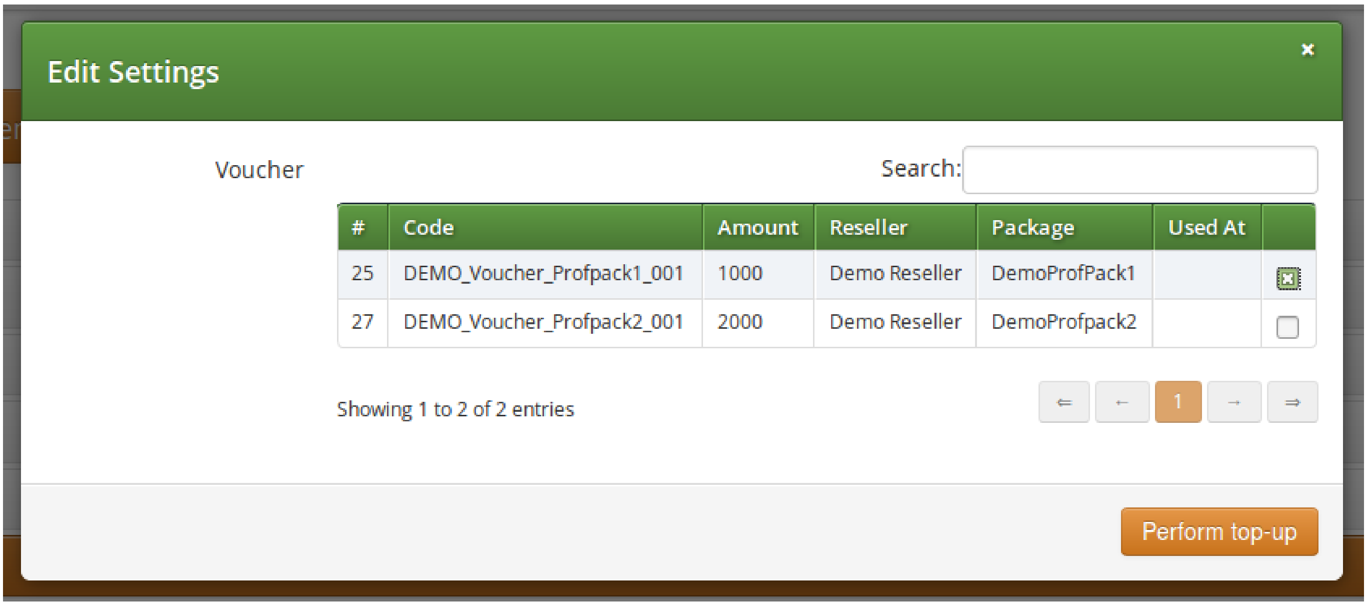


Figure 43: Balance Top-up with Voucher

It is also possible to perform top-up through the **REST API**: `POST /api/topupvouchers`

```
{
  "subscriber_id" : "73",
  "code" : "SILVER_1_1437974390“
  "request_token" : "uuid_from_3rdparty_relay" // optional request identifier
                                               // for lookups in the top-up log
}
```

7.4.6 Balance Overviews

The actual contract balance and logs of top-up or balance interval change events are a kind of financially important information and that's why those are provided on the administrative web interface for each customer. One should navigate to: *Settings* → *Customers* → *select the customer* → *Details*.

The various information details available on the web interface are discussed in subsequent sections of the handbook.

7.4.6.1 Contract Balance

This part of the overviews shows the actual financial state of the customer's balance and the current profile package and balance interval.

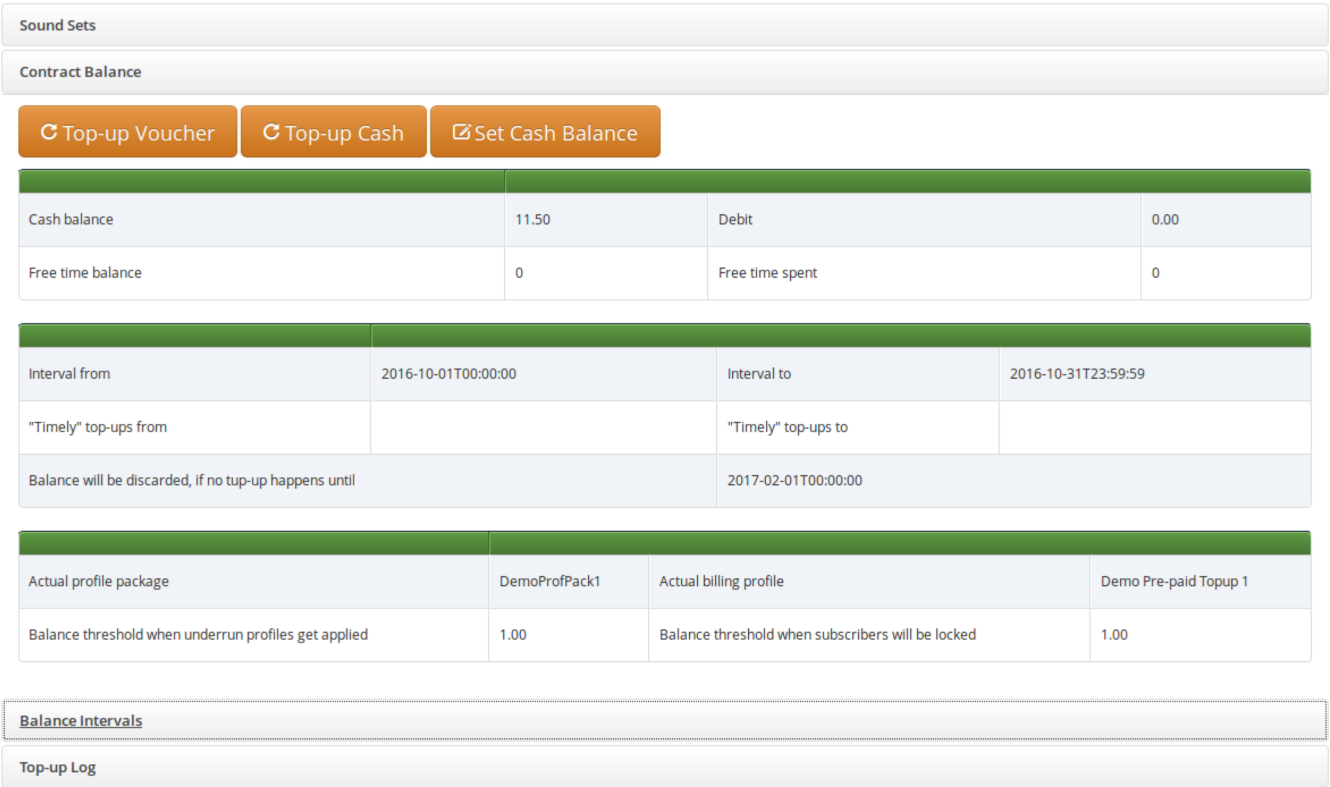


Figure 44: Contract Balance Status

Another functionality assigned to *Contract Balance* section is the manual top-up. Both top-up with cash and top-up with voucher can be performed from here.

7.4.6.2 Balance Intervals

This table shows the balance intervals that have been in use, including the current interval.

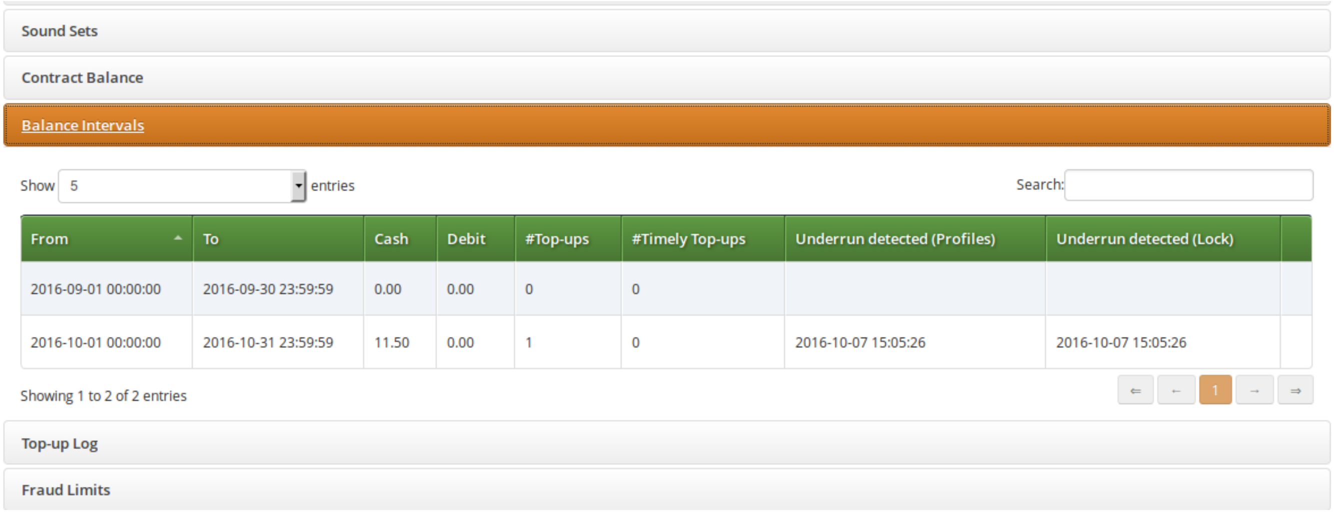


Figure 45: List of Balance Intervals

Content of the balance intervals table is:

- From, To: starting and end points of the time interval
- Cash: the contract's cash balance value at the end of the interval (former int.), or currently (actual int.)
- Debit: the total spent amount of money in the actual interval

> **Note**
> While "Cash" shows the remaining amount, "Debit" shows the spent amount. With a post-paid billing scenario only "Debit" field would be populated, with pre-paid both fields will display an amount.

- No. of Top-ups: how many top-up events happened within the interval
- No. of Timely Top-ups: how many timely top-up events happened within the interval
- Underrun detected (Profiles or Lock): the time of last underrun event when either an underrun billing profile, or a subscriber lock was activated

7.4.6.3 Top-up Log

Each successful or failing top-up request has to be logged. The log records represent an audit trail and reflect any data changes in the course of the top-up request.

In case of an error during the top-up operation the error message and any parseable fields of failed top-up attempts is recorded.

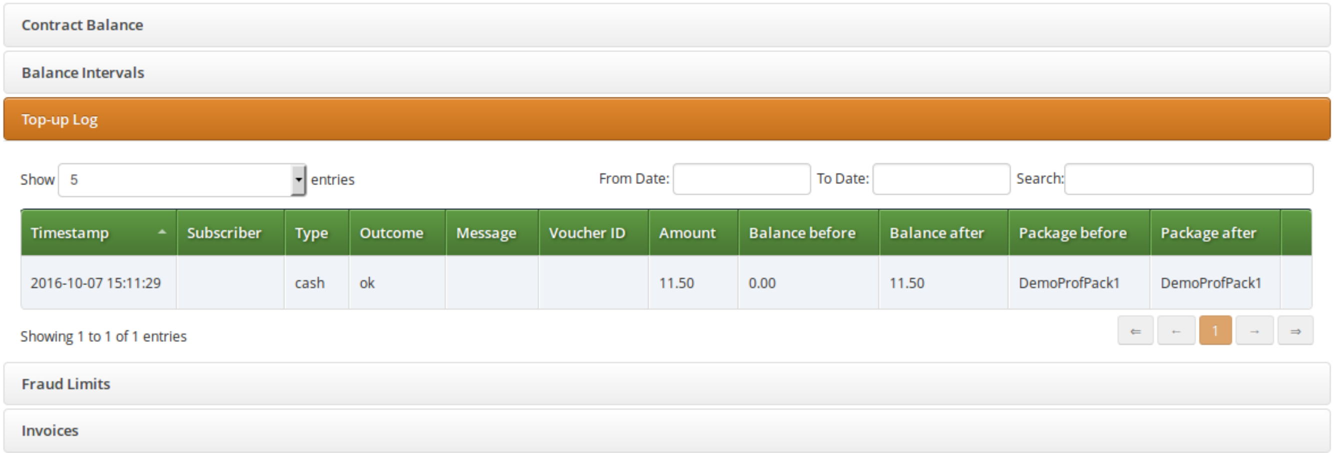


Figure 46: Balance Top-up Log

Content of the top-up log table is:

- Timestamp: when the top-up happened
- Subscriber: the ID of the subscriber who performed the top-up
- Type: `cash` or `voucher`
- Outcome: `ok` or `failed`
- Message: error message, if Outcome="failed"
- Voucher ID: ID of voucher, if Type="voucher"
- Amount: the amount by which the balance was modified (after the *Service Charge* was subtracted from the voucher's value)
- Balance before: balance's value before top-up
- Balance after: balance's value after top-up
- Package before: the name of the *Profile Package* that was active before top-up
- Package after: the name of the *Profile Package* that became active after top-up

The top-up log table can also be queried using the readonly `/api/topuplogs` **REST API** resource.

An example of the response:

```
{
  "_embedded" : {
    "ngcp:topuplogs" : [{
      "_links" : {...},
      "amount" : null,
      "cash_balance_after" : null,
      "cash_balance_before" : null,
```

```
      "contract_balance_after_id" : null,
      "contract_balance_before_id" : null,
      "contract_id" : 2565,
      "id" : 373,
      "lock_level_after" : null,
      "lock_level_before" : null,
      "message" : ..., //error reason
      "outcome" : "failed",
      "package_after_id" : null,
      "package_before_id" : null,
      "profile_after_id" : null,
      "profile_before_id" : null,
      "request_token" : "1444956281_6", // = "panel" for panel UI requests
      "subscriber_id" : 1804,
      "timestamp" : "2015-10-16 02:45:19",
      "type" : "voucher", // "cash" or "voucher"
      "username" : "administrator",
      "voucher_id" : null }]
  },
  "_links" : { ... },
  "total_count" : 1
}
```

7.4.7 Usage Examples

After getting to know the concepts of customized billing solution on sip:provider PRO platform, it's worth seeing some practical examples for the usage of those advanced features.

The starting point is the setup of *Profile Packages* for our fictive customers: A, B and C. There are 4 different packages defined, with corresponding vouchers:

- **Initial:**
  - Balance interval: 1 month
  - Timely duration: 1 month
  - Interval start mode: topup_interval
  - Carry-over mode: carry_over_timely
- **Silver:**
  - Balance interval: 1 month
  - Timely duration: 1 month
  - Interval start mode: "topup_interval"
  - Carry-over mode: "carry_over_timely"
  - Service charge: 2 EUR

- Underrun lock level: "no lock"
- Voucher value: 10 EUR

- **Gold:**
  - Balance interval: 1 month
  - Interval start mode: "topup_interval"
  - Carry-over mode: "carry_over"
  - Service charge: 5 EUR
  - Underrun lock level: "no lock"
  - Voucher value: 20 EUR

- **Extension:**
  - Balance interval: 1 month
  - Timely duration: 1 month
  - Interval start mode: "topup_interval"
  - Carry-over mode: "carry_over_timely"
  - Service charge: 2 EUR
  - Underrun lock level: "no lock"
  - Voucher value: 2 EUR

7.4.7.1 Customer A — Silver Package

1. Customer A tops up 10 EUR with a “silver” voucher. 2 EUR are deducted as service charge. Remaining balance is 8 EUR starting on the date of the top- up.
2. Customer A doesn’t top-up balance within the next month, so remaining balance is set to 0 after one month, and billing profiles and lock levels are set to the balance-underrun definition of the “silver” package.

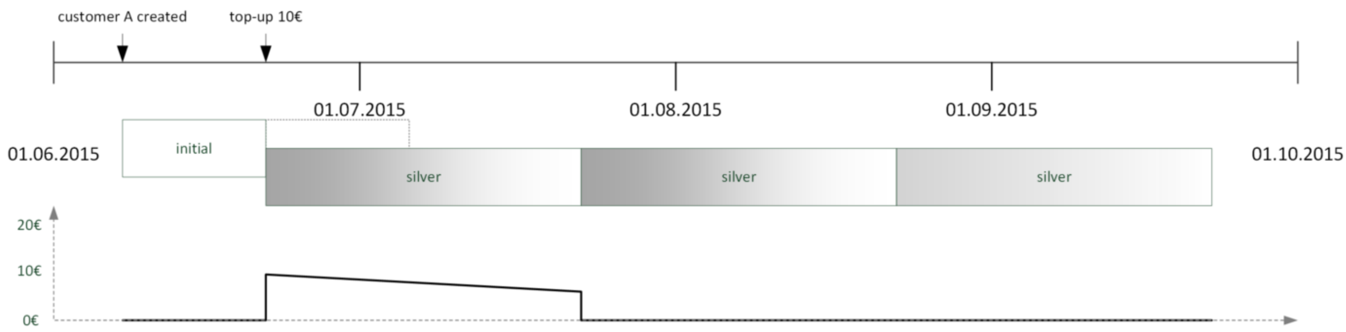


Figure 47: Usage Example: Silver Package

7.4.7.2 Customer B — Silver and Extension Package

1. Customer B tops up 10 EUR with the “silver” voucher. 2 EUR are deducted as service charge. Remaining balance is 8 EUR starting on the date of the top-up.
2. Customer B tops up 2 EUR using an “extension” voucher on the last day. 2 EUR are deducted as service charge and the interval is extended for one month, carrying over his old balance.
3. Customer B doesn’t top-up balance within the next month, so remaining balance is set to 0 after the month, and billing profiles and lock levels are set to the balance-underrun definition of the “extension” package.

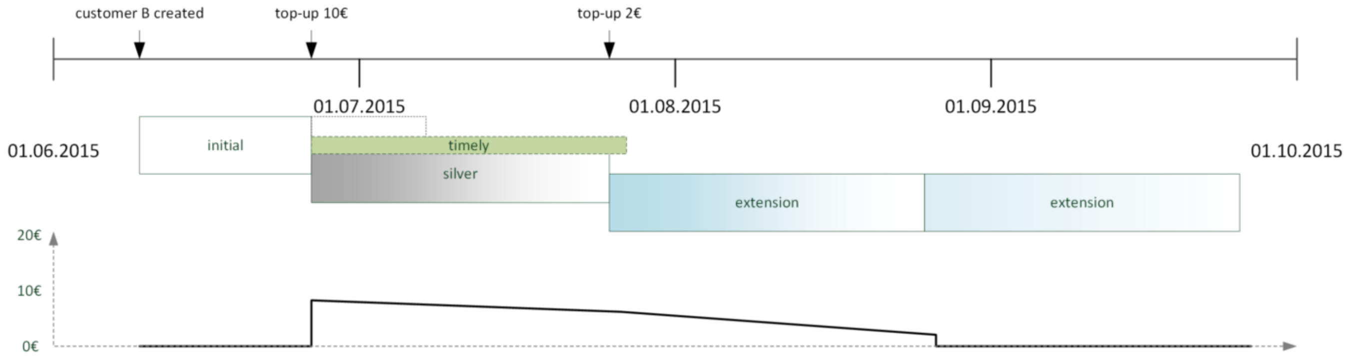


Figure 48: Usage Example: Silver + Extension Package

7.4.7.3 Customer C — Gold Package

Customer C tops up 20 EUR with the “gold” voucher. 5 EUR are deducted as service charge. Remaining balance is 15 EUR starting on the date of the top-up. Balance is carried over after each month until used up.

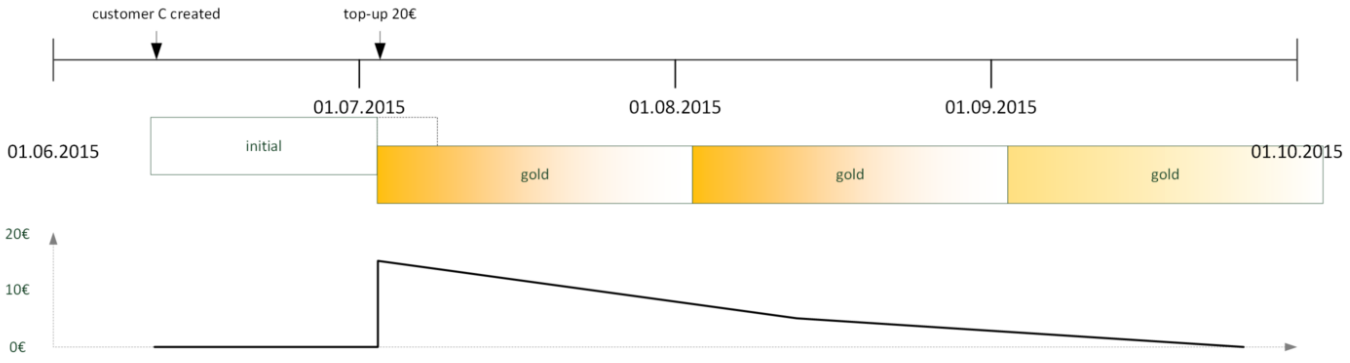


Figure 49: Usage Example: Gold Package

7.5 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.

7.5.1 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 12: 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
```

7.5.2 File Format

Each billing file consists of three parts: one header line, zero to 5000 body lines and one trailer line.

7.5.2.1 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 13: 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

7.5.2.2 File Body Format for Call Detail Records (CDR)

The body of a CDR consists of a minimum of zero and a maximum of 5000 lines. Each line holds one call detail record in CSV format and is constructed by the following fields, all of them enclosed in single quotes:

Table 14: CDR export file body line format

Body Element	Length	Type	Description
<id>	1-10	uint	Internal CDR id.
<update_time>	19	timestamp	Timestamp of last modification.
<source_user_id>	36	string	Internal UUID of calling party subscriber.
<source_provider_id>	1-255	string	Internal ID of calling party provider.
<source_ext_subscriber_id>	0-255	string	External ID of calling party subscriber.
<source_subscriber_id>	1-10	uint	Internal ID of calling party subscriber.
<source_ext_account_id>	0-255	string	External ID of calling party customer.
<source_account_id>	1-10	uint	Internal ID of calling party customer.
<source_user>	1-255	string	SIP username of calling party.
<source_domain>	1-255	string	SIP domain of calling party.
<source_cli>	1-64	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>	1 / 36	string	Internal UUID of called party subscriber or 0 if callee is not local.
<destination_provider_id>	1-255	string	Internal ID of called party provider.
<dest_ext_subscriber_id>	0-255	string	External ID of called party subscriber.

Table 14: (continued)

Body Element	Length	Type	Description
`<dest_subscriber_id>`	1-10	uint	Internal ID of called party subscriber.
`<dest_ext_account_id>`	0-255	string	External ID of called party customer.
`<destination_account_id>`	1-10	uint	Internal ID of called party customer.
`<destination_user>`	1-255	string	Final SIP username of called party.
`<destination_domain>`	1-255	string	Final SIP domain of called party.
`<destination_user_in>`	1-255	string	Incoming SIP username of called party.
`<destination_domain_in>`	1-255	string	Incoming SIP domain of called party.
`<dialed_digits>`	1-255	string	The user-part of the SIP Request URI as received by the soft-switch.
`<peer_auth_user>`	0-255	string	User to authenticate towards peer.
`<peer_auth_realm>`	0-255	string	Realm to authenticate towards peer.
`<call_type>`	3-4	string	The type of the call - one of: `call`: normal call `cfu`: call forward unconditional `cft`: call forward timeout `cfb`: call forward busy `cfna`: call forward no answer
`<call_status>`	2-7	string	The final call status - one of: `ok`: successful call `busy`: callee busy `noanswer`: no answer from callee `cancel`: cancel from caller `offline` callee offline `timeout`: no reply from callee `other`: unspecified, see `<call_code>` for details
`<call_code>`	3	uint	The final SIP status code.
`<init_time>`	23	timestamp	Timestamp of call initiation (invite received from caller). Seconds include fractional part (3 decimals).
`<start_time>`	23	timestamp	Timestamp of call establishment (final response received from callee). Seconds include fractional part (3 decimals).
`<duration>`	4-11	fixed precision	Length of call (beginning at `start_time`) in seconds with 3 decimals.
`<call_id>`	1-255	string	The SIP call-id.
`<rating_status>`	2-7	string	The internal rating status - 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	timestamp	Timestamp of rating or empty if not rated.

Table 14: (continued)

Body Element	Length	Type	Description
<source_carrier_cost>	4-11	fixed precision	The originating carrier cost or empty if not rated. In cent with two decimals. Only available in system exports, not for resellers.
<source_customer_cost>	4-11	fixed precision	The originating customer cost or empty if not rated. In cent with two decimals.
<source_carrier_zone>	0-127	string	The originating carrier billing zone or empty if not rated. Only available in system exports, not for resellers.
<source_customer_zone>	0-127	string	The originating customer billing zone or empty if not rated.
<source_carrier_destination>	0-127	string	The originating carrier billing destination or empty if not rated. Only available in system exports, not for resellers.
<source_customer_destination>	0-127	string	The originating customer billing destination or empty if not rated.
<source_carrier_free_time>	1-10	uint	The number of originating free time seconds used on carrier side or empty if not rated. Only available in system exports, not for resellers.
<source_customer_free_time>	1-10	uint	The number of originating free time seconds used from the customer's account balance or empty if not rated.
<destination_carrier_cost>	4-11	fixed precision	The termination carrier cost or empty if not rated. In cent with two decimals. Only available in system exports, not for resellers.
<destination_customer_cost>	4-11	fixed precision	The termination customer cost or empty if not rated. In cent with two decimals.
<destination_carrier_zone>	0-127	string	The termination carrier billing zone or empty if not rated. Only available in system exports, not for resellers.
<destination_customer_zone>	0-127	string	The termination customer billing zone or empty if not rated.
<destination_carrier_destination>	0-127	string	The termination carrier billing destination or empty if not rated. Only available in system exports, not for resellers.
<destination_customer_destination>	0-127	string	The termination customer billing destination or empty if not rated.
<destination_carrier_free_time>	1-10	uint	The number of termination free time seconds used on carrier side or empty if not rated. Only available in system exports, not for resellers.
<destination_customer_free_time>	1-10	uint	The number of termination free time seconds used from the customer's account balance or empty if not rated.

Table 14: (continued)

Body Element	Length	Type	Description
`<source_reseller_cost>`	4-11	fixed precision	The originating reseller cost or empty if not rated. In cent with two decimals. `Only available in system exports, not for resellers.`
`<source_reseller_zone>`	0-127	string	The originating reseller billing zone or empty if not rated. `Only available in system exports, not for resellers.`
`<source_reseller_destination>`	0-127	string	The originating reseller billing destination or empty if not rated. `Only available in system exports, not for resellers.`
`<source_reseller_free_time>`	1-10	uint	The number of originating free time seconds used from the reseller’s account balance or empty if not rated. `Only available in system exports, not for resellers.`
`<destination_reseller_cost>`	4-11	fixed precision	The termination reseller cost or empty if not rated. In cent with two decimals. `Only available in system exports, not for resellers.`
`<destination_reseller_zone>`	0-127	string	The termination reseller billing zone or empty if not rated. `Only available in system exports, not for resellers.`
`<destination_reseller_destination>`	0-127	string	The termination reseller billing destination or empty if not rated. `Only available in system exports, not for resellers.`
`<destination_reseller_free_time>`	1-10	uint	The number of termination free time seconds used from the reseller’s account balance or empty if not rated. `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 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. Reseller CDR CSV files don't contain the fields for *carrier* or *reseller* ratings, neither in *source* nor *destination* direction. Thus, the reseller CSV files have 16 fewer fields.

7.5.2.3 File Body Format for Event Detail Records (EDR)

The body of a EDR consists of a minimum of zero and a maximum of 5000 lines. Each line holds one call detail record in CSV format and is constructed by the following fields, all of them enclosed in single quotes:

Table 15: EDR export file body line format

Body Element	Length	Type	Description
<event_id>	1-10	uint	Internal EDR id.
<event_type>	1-255	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 group. `end_huntgroup`: A subscriber has been deprovisioned as group. `start_ivr`: A subscriber has a new call-forward to auto-attendant set. `end_ivr`: A subscriber has removed a call-forward to auto-attendant.
<customer_external_id>	0-255	string	The external customer ID as provisioned for the subscriber.
<contact_company>	0-255	string	The company name of the customer's contact.
<subscriber_external_id>	0-255	string	The external subscriber ID as provisioned for the subscriber.
<subscriber_number>	0-255	string	The voip number of the subscriber with the highest ID (DID or primary number).

Table 15: (continued)

Body Element	Length	Type	Description
<old_status>	0-255	string	The old status of the event. Depending on the event_type: `start_profile`: Empty. `end_profile`: The profile id of the profile which got removed from the subscriber. `update_profile`: The old profile id which got updated. `start_huntgroup`: Empty. `end_huntgroup`: The profile id of the group which got deprovisioned. `start_ivr`: Empty. `end_ivr`: Empty.
<new_status>	0-255	string	The new status of the event. Depending on the event_type: `start_profile`: The profile id which got assigned to the subscriber. `end_profile`: Empty. `update_profile`: The new profile id which got updated. `start_huntgroup`: The current profile id assigned to the group subscriber. `end_huntgroup`: The current profile id assigned to the group subscriber. `start_ivr`: Empty. `end_ivr`: Empty.
<timestamp>	0-255	string	The time when the event occured.
<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"
```

7.5.2.4 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
```

7.5.3 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.

---

8 Provisioning REST API Interface

The sip:provider PRO provides the REST API interface for interconnection with 3rd party tools.

The sip:provider PRO 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.

8.1 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";
```

9 Configuration Framework

The sip:provider PRO 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.

9.1 Configuration templates

The sip:provider PRO 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.

9.1.1 .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 PRO 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).

9.1.2 .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
```

9.1.3 .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.

9.2 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.

9.3 ngcpcfg and its command line options

The ngcpcfg utility supports the following command line options:

9.3.1 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.

9.3.2 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/
```

9.3.3 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.

9.3.4 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.

9.3.5 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.

9.3.6 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.

9.3.7 help

The *help* options displays ngcpcfg's help screen and then exits without any further actions.

9.3.8 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.

9.3.9 pull

Retrieve modifications from shared storage. Note: This option is available in the High Availability setup only.

9.3.10 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.

9.3.11 services

The *services* option executes the service handlers for any modified configuration file(s)/directory.

9.3.12 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.

10 Network Configuration

Starting with version 2.7, the sip:provider PRO 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.

10.1 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 PRO systems all hosts of the system are defined, and the names are the actual host names instead of *self*, like this:

```
hosts:

  sp1:
    peer: sp2
    role: ...
```

```
    interfaces: ...

  sp2:
    peer: sp1
    role: ...
    interfaces: ...
```

10.1.1 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 PRO 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 PRO 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 PRO 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 PRO 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 PRO 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 PRO system
- *peer*: the hostname of the peer node within the pair of nodes (e.g. "sp2" for *sp1* host). The purpose of that: each node knows its companion for providing high availability, data replication etc.

10.1.2 Interface Parameters

- `hwaddr`: MAC address of the interface

- `ip`: IPv4 address of the node
- `v6ip`: IPv6 address of the node; optional
- `netmask`: IPv4 netmask
- `shared_ip`: shared IPv4 address of the pair of nodes; this is a list of addresses
- `shared_v6ip`: shared IPv6 address of the pair of nodes; optional; this is a list of addresses
- `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
  – `aux_ext`: interface for potentially insecure external components like remote system log collection service.

    Note
    For example the *CloudPBX* module can use it to provide time services and remote logging facilities to end customer devices. The type *aux_ext* is assigned to *lo* interface by default. If it is needed to expose this type to the public, it is recommended to assign the type *aux_ext* to a separate VLAN interface to be able to limit or even block the incoming traffic easily via firewalling in case of emergency, like a (D)DoS attack on external services.

  – `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
  – `li_int`: used for LI (Lawful Interception) traffic routing
  – `ha_int`: HA (High Availability) communication interface between the nodes

  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

10.2 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.

10.2.1 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' && ngcpcfg push
```

The direction of communication through this SIP extra socket is incoming+outgoing. The sip:provider PRO 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 10.2.2 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.

10.2.2 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 --host=sp1 --ip=auto --netmask=auto --type= ↩
    sip_ext_incoming --type=rtp_int_100
ngcp-network --set-interface=eth0.100 --host=sp2 --ip=auto --netmask=auto --type= ↩
    sip_ext_incoming --type=rtp_int_100
```

The generated file should look like the following:

```
sp1:
..
..
    eth0.100:
      hwaddr: ff:ff:ff:ff:ff:ff
      ip: 192.168.1.2
      netmask: 255.255.255.0
      shared_ip:
        - 192.168.1.3
      shared_v6ip: ~
      type:
        - sip_ext_incoming
        - rtp_int_100
..
..
    interfaces:
      - lo
      - eth0
      - eth0.100
      - eth1
..
..
sp2:
..
..
    eth0.100:
      hwaddr: ff:ff:ff:ff:ff:ff
      ip: 192.168.1.4
      netmask: 255.255.255.0
      shared_ip:
        - 192.168.1.3
      shared_v6ip: ~
      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 PRO 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' && ngcpcfg push
```

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.

11 Software Upgrade

11.1 Release Notes

The sip:provider PRO version mr4.5.3 has several important changes comparing to the previous release. Please find the complete changelog in our release notes on our WEB site.

11.2 Preparing for a Software Upgrade

Warning

Make sure that all the SIP domains and peering servers have appropriate rtp_interface option (e.g. *ext*) selected in the NAT and Media Flow Control section. If you leave *default* there, then incorrect network interface may be used for sending and receiving RTP media after the software upgrade.

Make sure you're prepared to spend two hours or so upgrading the system. There can be service interruptions, so also notify the customer and get their approval.

Check the system overal status:

```
ngcp-status --all
```

Check the system for locally modified files (move them to appropriate customtt.tt2 files if necessary):

```
ngcp-status --integrity
```

Try to find local changes to the template files by issuing:

```
find /etc/ngcp-config -name \*customtt.tt2
```

You will also need to find the dpkg-dist files under the templates files because people sometimes forget about creating customtt files and edit tt2 files directly. That makes upgrades not to replace the tt2 files. If so, you need to treat the tt2 files as if they were customtt files and make sure you merge the new templates with the changes of the old ones.

```
find /etc/ngcp-config -name \*.tt2.dpkg-dist
```

Also, please check/clean old dpkg backup files (just in case if previous person did the previous step not carefully enough). Normally the list should be empty:

```
find /etc/ngcp-config -name \*.tt2.dpkg\*
```

You will have to understand why the changes are there and if they are still needed after the upgrade. You should create a ticket in the bugtracker if there isn't one yet.

Log into the two servers. Use their real IPs so you can switch the cluster forth and back later on. Make sure the cluster status is ok - on **both** nodes issue:

- **monit summary** - one should be running **all** services, the other all but rtpengine, lb, proxy, sbc, mediator and rate-o-mat
- **cl_status rscstatus** - one (with all services running) should print "all", the other "none"
- **mysql -e "show slave status\G"** - look for the following:

```
Slave_IO_Running: Yes
Slave_SQL_Running: Yes
Seconds_Behind_Master: 0
```

- **ngcpcfg status** - should print OK all the times
- **ngcp-collective-check** - should not report any problems.

A cluster fail-over could be a good idea to see if everything works on the second node too. On the standby node issue:

```
/usr/share/heartbeat/hb_takeover
```

Afterwards again check monit, cl_status and ngcp-collective-check.

Create two test subscribers, or retrieve the credentials for two of them. Register a client to the platform and perform a test call between the two to ensure call routing works.

Run "apt-get update", ensure you have no warnings/errors here.

> **Warning**
> Installation may use locally specified mirrors. Discuss with a customer possibility to switch on Sipwise APT repositories (at least for the time of upgrades), the public Debian mirrors may not provide packages for old Releases anymore or be at least outdated!

11.3 Upgrade sip:provider PRO from previous mr4.4/mr4.5 versions to mr4.5.3 LTS version

> **Warning**
> ensure you have read and performed all the steps described in the section "Preparing for a Software Upgrade" above!

The sip:provider PRO system upgrade to mr4.5.3 will perform a couple of fundamental steps:

- Upgrade NGCP software packages
- Upgrade NGCP configuration templates
- Upgrade NGCP DB schema
- Upgrade the base system within Debian (v8) to the latest package versions

So assuming you have a running sip:provider PRO system and want to upgrade it, start on the inactive node by upgrading software, then take over from the other node and then upgrade the other (now inactive) node, as detailed in the steps below.

11.3.1 Switch to new repositories

For upgrading the sip:provider PRO to the latest mr4.5.3 release, execute the following commands on **both** nodes:

```
NGCP_CURRENT_VERSION=$(cat /etc/ngcp_version)
sed -i "s/$NGCP_CURRENT_VERSION/mr4.5.3/" /etc/apt/sources.list.d/sipwise.list
ngcp-approx-cache-helper --auto --node localhost
apt-get update
apt-get install ngcp-upgrade-pro
```

11.3.2 Upgrade the first PRO node

Execute `ngcp-upgrade` in inactive node as *root*:

```
ngcp-upgrade
```

Note

sip:provider PRO can be upgraded to mr4.5.3 from previous release or previous build only. The script ngcp-upgrade will find all the possible target releases for the upgrade and allow to choose the proper one.

Note

If there is an error during upgrade, the ngcp-upgrade script will request you to solve it. Once you've fixed the problem just re-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).

11.3.3 The customtt files handling (if necessary)

Merge/add the custom configuration templates if needed. Apply the changes to configuration templates if any and send them to the shared storage and the other node:

```
ngcpcfg apply 'upgrade node'
ngcpcfg push --nobuild --noapply
```

11.3.4 Promote inactive node to active.

Execute on inactive node as *root*:

```
/usr/share/heartbeat/hb_takeover
```

11.3.5 Upgrade the second PRO node

Go to the new inactive node. Run ngcp-upgrade, as *root*:

```
ngcp-upgrade
```

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).

When all finishes successfully check that replication is running. Check `ngcp-status --all`. Finally, do a basic functionality test. Check 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-mr4.5.3-** (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-mr4.5.3-*/upgrade.log*.

11.4 Upgrade sip:provider PRO from previous mr3.8 LTS version to mr4.5.3 LTS version

> **Warning**
> ensure you have read and performed all the steps described in the section "Preparing for a Software Upgrade" above!

The sip:provider PRO system upgrade to mr4.5.3 will perform a couple of fundamental steps:

- Verify APT source lists
- Upgrade Debian from version 7 (wheezy) to version 8 (jessie)
- Upgrade NGCP software packages
- Upgrade NGCP configuration templates
- Upgrade NGCP DB schema
- Upgrade the base system within Debian 8 (jessie) to the latest package versions

So assuming you have a running sip:provider PRO system and want to upgrade it, start on the inactive node by upgrading software, then take over from the other node and then upgrade the other (now inactive) node, as detailed in the steps below.

11.4.1 Verify APT source lists

Warning

ensure you are using Sipwise APT repositories. Public Debian mirrors may not provide packages for old Debian releases anymore. Also they can be outdated. Consider to use Sipwise repositories for the time of upgrade. Discuss with a customer the situation here and ensure Sipwise IPs are open on firewalls.

To switch on sipwise APT source lists, 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
## custom sources.list, deployed via upgrade ${NGCP_CURRENT_VERSION}->mr4.5.3
#
# Debian repositories
deb http://debian.sipwise.com/debian/ wheezy main contrib non-free
#deb-src http://debian.sipwise.com/debian/ wheezy main contrib non-free
#
deb http://debian.sipwise.com/debian-security/ wheezy-security main contrib non-free
#deb-src http://debian.sipwise.com/debian-security/ wheezy-security main contrib non-free
#
deb http://debian.sipwise.com/debian/ wheezy-updates main contrib non-free
#deb-src http://debian.sipwise.com/debian/ wheezy-updates main contrib non-free
EOF

NGCP_CURRENT_VERSION=$(cat /etc/ngcp_version)
cat > /etc/apt/sources.list.d/sipwise.list << EOF
# NGCP_MANAGED_FILE - do not remove this line if it should be automatically handled
#
# Sipwise repository
deb http://deb.sipwise.com/sppro/${NGCP_CURRENT_VERSION}/ wheezy main
#deb-src http://deb.sipwise.com/sppro/${NGCP_CURRENT_VERSION}/ wheezy main
EOF
```

Run "apt-get update" and ensure you have no warnings/errors here.

11.4.2 Switch to new repositories

For upgrading the sip:provider PRO to the latest mr4.5.3 release, execute the following commands on **both** nodes:

```
NGCP_CURRENT_VERSION=$(cat /etc/ngcp_version)
sed -i "s/$NGCP_CURRENT_VERSION/mr4.5.3/" /etc/apt/sources.list.d/sipwise.list
```

```
sed -i 's/wheezy/jessie/g' /etc/apt/sources.list.d/sipwise.list
rm -f /etc/apt/sources.list.d/dell.list
apt-get update
apt-get install ngcp-upgrade-pro
```

11.4.3 Upgrade the first PRO node

Execute `ngcp-upgrade` in inactive node as *root*:

```
ngcp-upgrade
```

Note

sip:provider PRO can be upgraded to mr4.5.3 from previous release or previous build only. The script ngcp-upgrade will find all the possible target releases for the upgrade and allow to choose the proper one.

Note

If there is an error during upgrade, the ngcp-upgrade script will request you to solve it. Once you've fixed the problem just re-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).

11.4.4 The customtt files handling (if necessary)

Merge/add the custom configuration templates if needed. Apply the changes to configuration templates if any and send them to the shared storage and the other node:

```
ngcpcfg apply 'upgrade node'
ngcpcfg push --nobuild --noapply
```

11.4.5 Promote inactive node to active.

Execute on inactive node as *root*:

```
/usr/share/heartbeat/hb_takeover
```

11.4.6 Upgrade the second PRO node

Go to the new inactive node. Run ngcp-upgrade, as *root*:

```
ngcp-upgrade
```

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).

When all finishes successfully check that replication is running. Check `ngcp-status --all`. Finally, do a basic functionality test. Check 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-mr4.5.3-** (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-mr4.5.3-*/upgrade.log*.

---

12 Backup, Recovery and Database Maintenance

12.1 sip:provider PRO 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. Although the design of sip:provider PRO implies data duplication and high availability of services, we still strongly suggest you to configure a backup procedure. The sip:provider PRO has a built-in solution that can help you back up the most crucial data. Alternatively, it can be integrated with any Debian compatible backup software.

12.1.1 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 on corresponding NGCP nodes.

- 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.

12.1.2 The built-in backup solution

The sip:provider PRO comes with an easy-to-use solution that creates everyday backups of the most important data:

- The system configuration files. The whole /etc directory is backed up.
- Exported CDRs. The */home/jail/home/cdrexport* directory with csv files.
- All required databases on corresponding servers.

This functionality is disabled by default and can be enabled and configured in the backuptools subsection in the config.yml file. Please, refer to the “C.1.3 backup tools” section of the “NGCP configs overview” chapter for the backup configuration options.

Once you set the required configuration options, apply the changes:

```
ngcpcfg apply ’enabled the backup feature’
ngcpcfg push
```

Once you activate the feature, the sip:provider PRO will create backups in the off-peak time on the standby nodes and put them to the */var/backup/ngcp_backup* directory. You can copy these files to your backup server using scp or ftp.

Note
make sure that you have enough free disk space to store the backups for the specified number of days.

12.2 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 PRO 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'
ngcpcfg push
```

12.3 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.

12.4 Synchronize database

In case of unresolvable database replication issues or to copy mysql data between a pair of hosts (usually a pair of sp1 and sp2 nodes).

There is a script for that: *ngcp-sync-db*.

To synchronize databases you need to run the script on your target host.

- Definitions:

- *master* - remote/master host (the database is dumped from there)
- *local* - target/local host (the database is imported onto)

• Usage:

Important
Your existing database on *local* will be completely wiped. The script provides a possibility to backup both *master* and *local* databases during the procedure.

You can run the script with `-h` or `--help` to check its options or use `man ngcp-sync_db`

If you run it without any options it automatically calculates *master* hostname (e.g. if you run it on *sp2* then *sp2==local* and *sp1==master*).

The script also requires mysql credentials and if none provided it uses username=*sipwise* and the password is picked from /etc/mysql/sipwise.cnf. You can specify user and/or password for both *master* and *local*.

Before the actual start it produces a summary with settings used to the procedure and a confirmation prompt to prevent accidental usage. Making use of `--force` option" however suppresses the confirmation prompt. By default no messages are printed on STDOUT (compliant to be integrated into another tools) and with `-v` or `--verbose` options you enable debugging where all the ongoing steps will be printed to STDOUT.

There are 2 modes available for synchronization, *online* and *backup*. By default *online* is used where the procedure does not create any backups and everything goes on the fly. That is useful for large databases where creating backups would require solid amounts of available free disk space. With the *backup* mode *master* db is dumped into a backup file on *local* first (default directory: */var/backup/ngcp-sync-db*) and imported upon the backup completion.

Mysql database connection to the *master* db and the *local* db is the essential part and by default the script tries to establish direct mysql connection however that may not be possible due to the access restrictions. To overcome that you can use `--ssh-tunnel` option and specifying there a local custom free port (e.g. `--ssh-tunnel=33125`) in this case an ssh tunnel will be created to *master* and used to establish the db connection on the *localhost* behalf (NOTE: Public key based ssh negotiation is required for the tunnel as the script does not suppot ssh credentials for security reasons).

Backups may be a subject to create during synchornization for possible rollbacks. To create the *local* db backup you should add `--local-backup`. The *master* db backup is automatically created only using `--sync-mode=backup`. Upon completion all those created backups are deleted and if you need to keep them please use `--keep-backups` option (NOTE: In case of errors during synchronization and when backups are created they are NOT automatically deleted. Therefore, if the script had failed with an error and afterwards completed successully you may want to manually remove the remaining backups from */var/backup/ngcp-sync-db*).

• Examples:

Normal online mode synchronization *sp1* → *sp2*.

```
sp2> ngcp-sync-db
```

Normal backup mode synchronization *sp1* → *sp2*.

```
sp2> ngcp-sync-db --sync-mode=backup
```

Forced online mode synchronization *sp1* → *sp2*. USE WITH CARE as there will be no confirmation prompts.

```
sp2> ngcp-sync-db --force
```

Direct mysql db access is not possible. SSH tunnel is initialised to local port 33125 and forwards all connections 127.0.0.1:33125 → sp1:3306.

```
sp2> ngcp-sync-db --ssh-tunnel=33125
```

Custom mysql credentials for the *master* db connection (by default: sipwise:/etc/mysql/sipwise.cnf)

```
sp2> ngcp-sync-db --master-user=frank --master-pass=dbconnect
```

Normal online mode synchronization *sp1* → *sp2* with the *local* db backup and retaining the backup. (no *master* backup in this case as it is only available with `--sync-mode=backup`).

```
sp2> ngcp-sync-db --local-backup --keep-backups
```

Normal online mode synchronization *custom-node* → *sp2* with ssh tunnel

```
sp2> ngcp-sync-db --master-host=custom-node --ssh-tunnel=45001
```

Forced syncrhonization *custom-node* → *sp2* with ssh tunnel, backup sync mode, local backup, custom *master* and *local* db credentials and ports as well as a different backup dir

```
sp2> ngcp-sync-db --force --sync-mode=backup --master-host=custom-node --master-port=3308  ↩
    --ssh-tunnel=45001 --master-user=frank --master-pass=dbconnect --local-user=john --local ↩
    -pass=dblocal --local-backup --keep-backups --backup-dir=/home/barry/backups
```

12.5 Accounting Data (CDR) Cleanup

Sipwise sip:provider PRO 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.

12.5.1 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'
> ngcpcfg push
```

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`

12.5.2 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>`

12.5.2.1 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.

12.5.2.2 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.

12.5.2.3 Cleanup CDRs

The database cleanup tool may do database table cleanup without performing backup. In order to do that, the statement: `clean up <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.

12.5.3 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

13 Platform Security, Performance and Troubleshooting

Once the sip:provider PRO 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.

13.1 Sipwise SSH access to sip:provider PRO

The sip:provider PRO 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 PRO 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.
```

13.2 Firewalling

The sip:provider PRO runs a wide range of services. Some of them need to interact with the user, while some others need to interact with the administrator or with nobody at all. Assuming that we trust the sip:provider PRO server for outgoing connections, we'll focus only on incoming traffic to define the services that need to be open for interaction.

Table 16: Subscribers

Service	Default port	Config option
Customer self care interface	443 TCP	www_admin→http_csc→port
SIP	5060 UDP, TCP	kamailio→lb→port
SIP over TLS	5061 TCP	kamailio→lb→tls→port + kamailio→lb→tls→enable (Disabled by default)
RTP	30000-40000 UDP	rtpproxy→minport + rtpproxy→maxport
XCAP	1080 TCP	kamailio→proxy→presence→enable + nginx→xcap_port (Disabled by default)
XMPP	5222 and 5269 TCP	None, standard XMPP ports for clients (5222) and federation (5269)

Table 17: Administrators

Service	Default port	Config option
SSH/SFTP	22 TCP	NA
Administrator interface	1443 TCP	www_admin→http_admin→port
Provisioning interfaces	2443 TCP	ossbss→apache→port

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`

13.3 Password management

The sip:provider PRO 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.

- 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*.

- The *root* user in MySQL has no default password. A password should be set using the *mysqladmin password* command.
- The administrative web interface has a default user *administrator* with password *administrator*. It should be changed within this interface.
- Generate new password for user *ngcpsoap* to access the provisioning interfaces, see the details in Section 8.

> **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.

13.4 SSL certificates.

The sip:provider PRO 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 PRO 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'*.

13.5 Securing your sip:provider PRO against SIP attacks

The sip:provider PRO 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.

13.5.1 Denial of Service

As soon as you have packets arriving on your sip:provider PRO 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 PRO 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 PRO 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 PRO 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.

13.5.2 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 PRO 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
```

13.6 System Requirements and Performance

The sip:provider PRO 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 PRO 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 PRO administrator tune the system requirements for optimum performance.

Table 18: 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	1/2 * Total system RAM	The installer will calculate the total system RAM and dedicate 50% to the mysql innodb buffer. This value won't be changed in case the system RAM changes so it's up to the administrator to adjust it. 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.
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.

Table 18: (continued)

Option	Default value	Requirement impact
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.
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 PRO 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:~#
```

13.7 Troubleshooting

The sip:provider PRO 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 PRO 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 PRO 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

Log file	Content	Estimated size
/var/log/ngcp/ha.log	fail-over related logs in case a node in a pair loses connection to the other side, when a standby node takes over or an active node goes standby due to intra-node communication issues or external ping node connection issues	small
/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
/var/log/ngcp/kamailio-lb.log	Overview of SIP requests and replies along with network source and destination information flowing through the platform	huge

Log file	Content	Estimated size
/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.

13.7.1 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=364e46767766210349779 34e055d19ea@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
```

13.7.2 Collecting SIP traces

The sip:provider PRO platform provides several tools to collect SIP traces. It can be used the sip:provider PRO *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
```

The sip:provider PRO platform provides also the native Voip sniffer, called *voisniff-ng*, which provide a graphic view of all the calls passing through the platform. It can be enabled via __/etc/ngcp-config/config.yml:

```
voisniff:
  admin_panel: 'yes'
  daemon:
    bpf: 'port 5060 or 5062 or ip6 proto 44 or ip[6:2] & 0x1fff != 0'
    external_interfaces: 'eth0 eth1'
    filter:
      exclude:
        -
          active: 1
          case_insensitive: 1
          pattern: '\ncseq: *\d+ +(register|notify|options|subscribe)'
```

```
    include: []
  internal_interfaces: lo
  mysql_dump_threads: 4
  start: 'yes'
  threads_per_interface: 10
partitions:
  increment: 700000
  keep: 10
```

`admin_panel` should be set to *yes* as well as `start`. Also `filter.exclude.active` should be set to *1* in order to avoid to sniff REGISTER, NOTIFY, OPTIONS and SUBSCRIBE messages. Then run:

```
ngcpcfg apply 'enable voisniff' && ngcpcfg push
```

Warning

Please notice that enabling voisniff, specially under a huge amount of traffic, may affect the system performance due to the fact that voisniff needs to save all the traffic into the database.

14 Monitoring and Alerting

14.1 Internal Monitoring

The platform uses the internal *monit* service to monitor all essential services. Since the sip:provider PRO runs in an active/standby mode, not all services are always running on both nodes, some of them will only run on the active node and be stopped on the standby node. The following commands show the most critical services on the platform: * `monit summary` - to get the list of services and their current status, * `monit status` - to get the list of services with detailed status.

> **Important**
> When you perform a stop/start/monitor/unmonitor operation on a service, *monit* affects other services that depend on the initial one. Hence, if you stop or unmonitor a service all services that depend on it will be stopped or unmonitored as well.

For example, `monit stop mysql` operation will stop kamailio, sbc, asterisk, prosody and some other services.

If any service ever fails for whatever reason the *monit* daemon quickly restarts it. When that happens, the daemon will send a notification email to the address specified in the `config.yml` file under the `general.adminmail` key. It will also send warning emails to this address under certain abnormal conditions, such as high memory consumption (> 75% is used) or high CPU load.

> **Important**
> In order for *monit* to be able to send emails to the specified address, the local MTA (*exim4*) must be configured correctly. If you haven't done so already, run `dpkg-reconfigure exim4-config` to do this. The CE edition's handbook contains more information about this in the *Installation* chapter.

14.2 Statistics Dashboard

The platform's administration interface (described in Section 4) provides a simple graphical overview 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.

14.3 External Monitoring Using SNMP

14.3.1 Overview and Initial Setup

The sip:provider PRO exports a variety of cluster health data and statistics over the standard SNMP interface. By default, the SNMP interface can only be accessed locally. To make it possible to provide the SNMP data to an external system, the `config.yml` file needs to be edited and the list of allowed community names and allowed hosts/IP ranges must be populated. This list can be found under the `checktools.snmpd.communities` key and it consists of one or more `community/source` value pairs. The `community` is the allowed community name, while `source` is an IP address or an IP block where to allow the requests from.

The `public` entry with the `localhost` source is used for local testing of SNMP functionality. It is recommended that you leave this entry in place. Other legal `sources` can be formed as single IP addresses or IP blocks in IP/prefix notation, for example `192.168.115.0/24`.

Tip

To locally check if SNMP is working correctly, execute the command `snmpwalk -v2c -cpublic localhost .` (note the trailing dot). This will generate a long list of raw SNMP OIDs and their values, provided that the `default` SNMP community key has been left in place.

Tip

SNMP version 1 and version 2c are supported.

14.3.2 Details

There are two types of information that can be retrieved from SNMP. The first one is the native NGCP cluster overview from the Sipwise MIBs (Management Information Bases). The second is the legacy ad-hoc information using the Net-SNMP extension OIDs, and detailed information for the node running the SNMP daemon using standard OIDs (Object Identifiers).

14.3.2.1 Sipwise NGCP OIDs

The entire NGCP cluster can be monitored by using the `SIPWISE-NGCP-MIB`, `SIPWISE-NGCP-MONITOR-MIB` and `SIPWISE-NGCP-STATS-MIB`. These OIDs are rooted at the Sipwise NGCP slot `.1.3.6.1.4.1.34274.1.*`.

The MIBs are self-documented, and can be found as part of the *ngcp-snmp-mibs* package. The NGCP SNMP Agent is a part of the *ngcp-snmp-agent* package, which when installed it should work out-of-the-box as long as the snmpd has been properly configured.

The `SIPWISE-NGCP-MIB` acts as the root MIB and provides information about the cluster licensing and layout (which is mostly static data about each node, such as node name, its IP address, its roles, etc.) and information required to access the OIDs from the other MIBs.

The `SIPWISE-NGCP-MONITOR-MIB` provides current monitoring information, global health conditions, the number of provisioned and registered subscribers and devices. It also provides per node information (independently of the number of nodes or their names) on their filesystem, processes, databases, system load, memory, heartbeat status, MTA queues, etc.

The `SIPWISE-NGCP-STATS-MIB` provides accumulated statistics on billing, performance and processed SIP messages.

NOTICE: OIDs under the following trees are not yet implemented: *ngcpMonitorFraud*, *ngcpMonitorPerformance.perfCAPSCurTable* and *ngcpStats*.

Tip

The NGCP SNMP Agent uses *Redis* and the *collectd* RRD files as data sources. This data is essential for accurate and complete monitoring data in the SNMP OID tree. In addition, the *Redis* database must be available on a shared IP address, so that *collectd* can always write to it. Otherwise, *collectd* may not work correctly or even crash.

14.3.2.2 Legacy OIDs

Note

The following OIDs have been superseded by the Sipwise NGCP OIDs, but they are still provided for backwards compatibility.

All basic system health variables (such as memory, disk, swap, CPU usage, network statistics, process lists, etc.) for the *mgmt_ node can be found in standard OID slots from standard MIBs. For example, memory statistics can be found through the* UCD-SNMP-MIB__ in OIDs such as `memTotalSwap.0`, `memAvailSwap.0`, `memTotalReal.0`, `memAvailReal.0`, etc., which translate to numeric OIDs `.1.3.6.1.4.1.2021.4.*`. In fact, `UCD-SNMP-MIB` is the most useful MIB for overall system health checks.

Additionally, there's a list of specially monitored processes, also found through the `UCD-SNMP-MIB`. `UCD-SNMP-MIB::prNames` (`.1.3.6.1.4.1.2021.2.1.2`) gives the list of monitored processes, `prCount` (`.1.3.6.1.4.1.2021.2.1.5`) is how many of each process are running and `prErrorFlag` (`.1.3.6.1.4.1.2021.2.1.100`) gives a 0/1 error indication (with `prErrMessage` (`.1.3.6.1.4.1.2021.2.1.101`) providing an explanation of any error).

Tip

Some of these processes are not supposed to be running on the standby node, so you'll see the error flag raised there. A possible solution is to run these SNMP checks against the shared service IP of the cluster. See in Section 2.4 below for more information.

Furthermore, `UCD-SNMP-MIB` provides a list of custom external checks. The names of these can be found under the `UCD-SNMP-MIB::extNames` (`.2`) tree, with `extOutput` (`.101`) providing the output (one line) from each check and `extResult` (`.100`) the exit code from each check.

The first of these external checks called `collective_check` provides a combined and overall system health status indicator. It gathers information from both nodes and returns *0* in `extResult.1` (`.100.1`) if everything is OK and running as it should. If it finds a problem somewhere, but with the system still operational (e.g. a service is stopped on the inactive node), `extResult.1` will return *1* and `extOutput.1` will be set to a string that can be used to diagnose the problem. In case the system is found in a critical and non-operational state, `extResult.1` will return *2*, again with an error message set. If you want to keep it really simple, you can just monitor this one OID and raise an alarm if it ever goes to non-zero.

Tip

The *0/1/2* status codes allow for easy integration with *Nagios*.

The remaining external checks simply return statistics on the system, they all return a number in `extOutput` and have `extResult` always set to zero.

The full list of such checks is below. All of these checks have three modes: the first returns the statistics from `sp1` (the first node in the sip:provider PRO pair), the second - from `sp2`, and the third - from whichever node is being queried (which is useful when querying the shared service IP). For example, the local SIP response time from `sp1` is in `sip_check_sp1`, from `sp2` - is in `sip_check_sp2`, and from the host itself - is in `sip_check_self`.

The base OID of the Result and Output OIDs is always `.1.3.6.1.4.1.2021.8.1`, so if you read `.100.1`, the full OID is `.1.3.6.1.4.1.2021.8.1.100.1`.

Name in MIB	Result OID	Output OID	Name	Description
UCD-SNMP-MIB::extNames.1	.100.1	.101.1	collective_check	Summarized platform check
UCD-SNMP-MIB::extNames.2	.100.2	.101.2	sip_check_sp1	SIP response time in seconds on sp1
UCD-SNMP-MIB::extNames.3	.100.3	.101.3	sip_check_sp2	SIP response time in seconds on sp2
UCD-SNMP-MIB::extNames.4	.100.4	.101.4	mysql_check_sp1	Average number of MySQL queries per second on sp1
UCD-SNMP-MIB::extNames.5	.100.5	.101.5	mysql_check_sp2	Average number of MySQL queries per second on sp2
UCD-SNMP-MIB::extNames.6	.100.6	.101.6	mysql_replication_check_sp1	MySQL replication delay in seconds on sp1
UCD-SNMP-MIB::extNames.7	.100.7	.101.7	mysql_replication_check_sp2	MySQL replication delay in seconds on sp2
UCD-SNMP-MIB::extNames.8	.100.8	.101.8	mpt_check_sp1	RAID status on sp1
UCD-SNMP-MIB::extNames.9	.100.9	.101.9	mpt_check_sp2	RAID status on sp2
UCD-SNMP-MIB::extNames.10	.100.10	.101.10	exim_queue_check_sp1	Number of mails undelivered in MTA queue on sp1
UCD-SNMP-MIB::extNames.11	.100.11	.101.11	exim_queue_check_sp2	Number of mails undelivered in MTA queue on sp2
UCD-SNMP-MIB::extNames.12	.100.12	.101.12	provisioned_subscribers_check_sp1	Number of subscribers provisioned on sp1

Name in MIB	Result OID	Output OID	Name	Description
UCD-SNMP-MIB::extNames.13	.100.13	.101.13	provisioned_subscribers_check_sp2	Number of subscribers provisioned on sp2
UCD-SNMP-MIB::extNames.14	.100.14	.101.14	kam_dialog_active_check_sp1	Number of active calls on sp1
UCD-SNMP-MIB::extNames.15	.100.15	.101.15	kam_dialog_active_check_sp2	Number of active calls on sp2
UCD-SNMP-MIB::extNames.16	.100.16	.101.16	kam_dialog_early_check_sp1	Number of calls in Early Media state on sp1
UCD-SNMP-MIB::extNames.17	.100.17	.101.17	kam_dialog_early_check_sp2	Number of calls in Early Media state on sp2
UCD-SNMP-MIB::extNames.18	.100.18	.101.18	kam_dialog_type_local_check_sp1	Number of active calls local on sp1
UCD-SNMP-MIB::extNames.19	.100.19	.101.19	kam_dialog_type_local_check_sp2	Number of active calls local on sp2
UCD-SNMP-MIB::extNames.20	.100.20	.101.20	kam_dialog_type_relay_check_sp1	Number of active calls routed via peers on sp1
UCD-SNMP-MIB::extNames.21	.100.21	.101.21	kam_dialog_type_relay_check_sp2	Number of active calls routed via peers on sp2
UCD-SNMP-MIB::extNames.22	.100.22	.101.22	kam_dialog_type_incoming_check_sp1	Number of incoming calls on sp1
UCD-SNMP-MIB::extNames.23	.100.23	.101.23	kam_dialog_type_incoming_check_sp2	Number of incoming calls on sp2
UCD-SNMP-MIB::extNames.24	.100.24	.101.24	kam_dialog_type_outgoing_check_sp1	Number of outgoing calls on sp1
UCD-SNMP-MIB::extNames.25	.100.25	.101.25	kam_dialog_type_outgoing_check_sp2	Number of outgoing calls on sp2
UCD-SNMP-MIB::extNames.26	.100.26	.101.26	kam_usrloc_regusers_check_sp1	Number of subscribers with at least one active registration on sp1
UCD-SNMP-MIB::extNames.27	.100.27	.101.27	kam_usrloc_regusers_check_sp2	Number of subscribers with at least one active registration on sp2
UCD-SNMP-MIB::extNames.28	.100.28	.101.28	kam_usrloc_regdevices_check_sp1	Total number of registered end devices on sp1

Name in MIB	Result OID	Output OID	Name	Description
UCD-SNMP-MIB::extNames.29	.100.29	.101.29	kam_usrloc_regdevices_check_sp2	Total number of registered end devices on sp2
UCD-SNMP-MIB::extNames.30	.100.30	.101.30	mysql_replication_discrepancies_check_sp1	Number of MySQL tables not in sync between sp1 and sp2
UCD-SNMP-MIB::extNames.31	.100.31	.101.31	mysql_replication_discrepancies_check_sp2	Number of MySQL tables not in sync between sp1 and sp2
UCD-SNMP-MIB::extNames.32	.100.32	.101.32	sip_check_self	Summarized platform check on active node
UCD-SNMP-MIB::extNames.33	.100.33	.101.33	mysql_check_self	Average number of MySQL queries per second on active node
UCD-SNMP-MIB::extNames.34	.100.34	.101.34	mysql_replication_check_self	MySQL replication delay in seconds on active node
UCD-SNMP-MIB::extNames.35	.100.35	.101.35	mpt_check_self	RAID status on active node
UCD-SNMP-MIB::extNames.36	.100.36	.101.36	exim_queue_check_self	Number of mails undelivered in MTA queue on active node
UCD-SNMP-MIB::extNames.37	.100.37	.101.37	provisioned_subscribers_check_self	Number of subscribers provisioned on active node
UCD-SNMP-MIB::extNames.38	.100.38	.101.38	kam_dialog_active_check_self	Number of active calls on active node
UCD-SNMP-MIB::extNames.39	.100.39	.101.39	kam_dialog_early_check_self	Number of calls in Early Media state on active node
UCD-SNMP-MIB::extNames.40	.100.40	.101.40	kam_dialog_type_local_check_self	Number of active calls local on active node
UCD-SNMP-MIB::extNames.41	.100.41	.101.41	kam_dialog_type_relay_check_self	Number of active calls routed via peers on active node
UCD-SNMP-MIB::extNames.42	.100.42	.101.42	kam_dialog_type_incoming_check_self	Number of incoming calls on active node
UCD-SNMP-MIB::extNames.43	.100.43	.101.43	kam_dialog_type_outgoing_check_self	Number of outgoing calls on active node

Name in MIB	Result OID	Output OID	Name	Description
UCD-SNMP-MIB::extNames.44	.100.44	.101.44	kam_usrloc_regusers_check_self	Number of subscribers with at least one active registration on active node
UCD-SNMP-MIB::extNames.45	.100.45	.101.45	kam_usrloc_regdevices_check_self	Total number of registered end devices on active node
UCD-SNMP-MIB::extNames.46	.100.46	.101.46	mysql_replication_discrepancies_check_self	Number of MySQL tables not in sync between sp1 and sp2

Tip

Some of the checks can be disabled (most are enabled by default) through the `config.yml` file, and those checks will then return an error message or an empty string in their `extOutput`. Enable those checks in the config file to get their output in the SNMP OID tree. The enable/disable flags can be found in the `checktools` section.

15 Extensions and Additional Modules

15.1 Cloud PBX

The sip:provider PRO comes with a commercial Cloud PBX module to provide B2B features for small and medium sized enterprises. The following chapters describe the configuration of the PBX features.

15.1.1 Configuring the Device Management

The *Device Management* is used by admins and resellers to define the list of device models, firmwares and configurations available for end customer usage. These settings are pre-configured for the default reseller up-front by Sipwise and have to be set up for every reseller separately, so a reseller can choose the devices he'd like to serve and potentially tweak the configuration for them. List of available pre-configured devices Section 15.1.12.

End customers choose from a list of *Device Profiles*, which are defined by a specific *Device Model*, a list of *Device Firmwares* and a *Device Configuration*. The following sections describe the setup of these components.

To do so, go to *Settings*→*Device Management*.

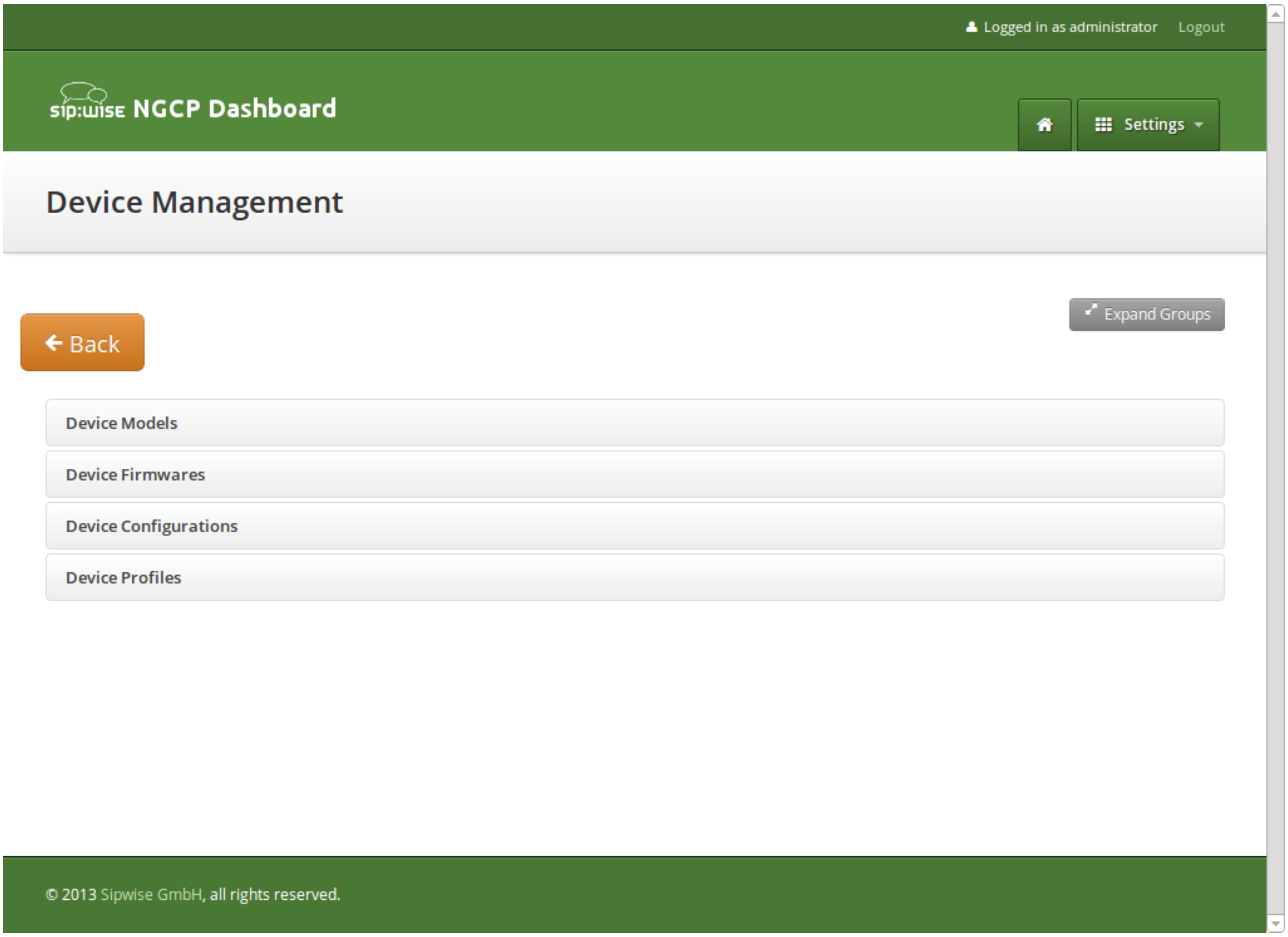


Figure 50: Device Management

15.1.1.1 Setting up Device Models

A *Device Model* defines a specific hardware device, like the vendor, model name, the number of keys and their capabilities. For example a Cisco SPA504G has 4 keys, which can be used for private lines, shared lines (SLA) and busy lamp field (BLF). If you have an additional attendant console, you get 32 more buttons, which can only do BLF.

In this example, we will create a Cisco SPA504G with an additional Attendant Console.

Expand the *Device Models* row and click *Create Device Model*.

First, you have to select the reseller this device model belongs to, and define the vendor and model name.

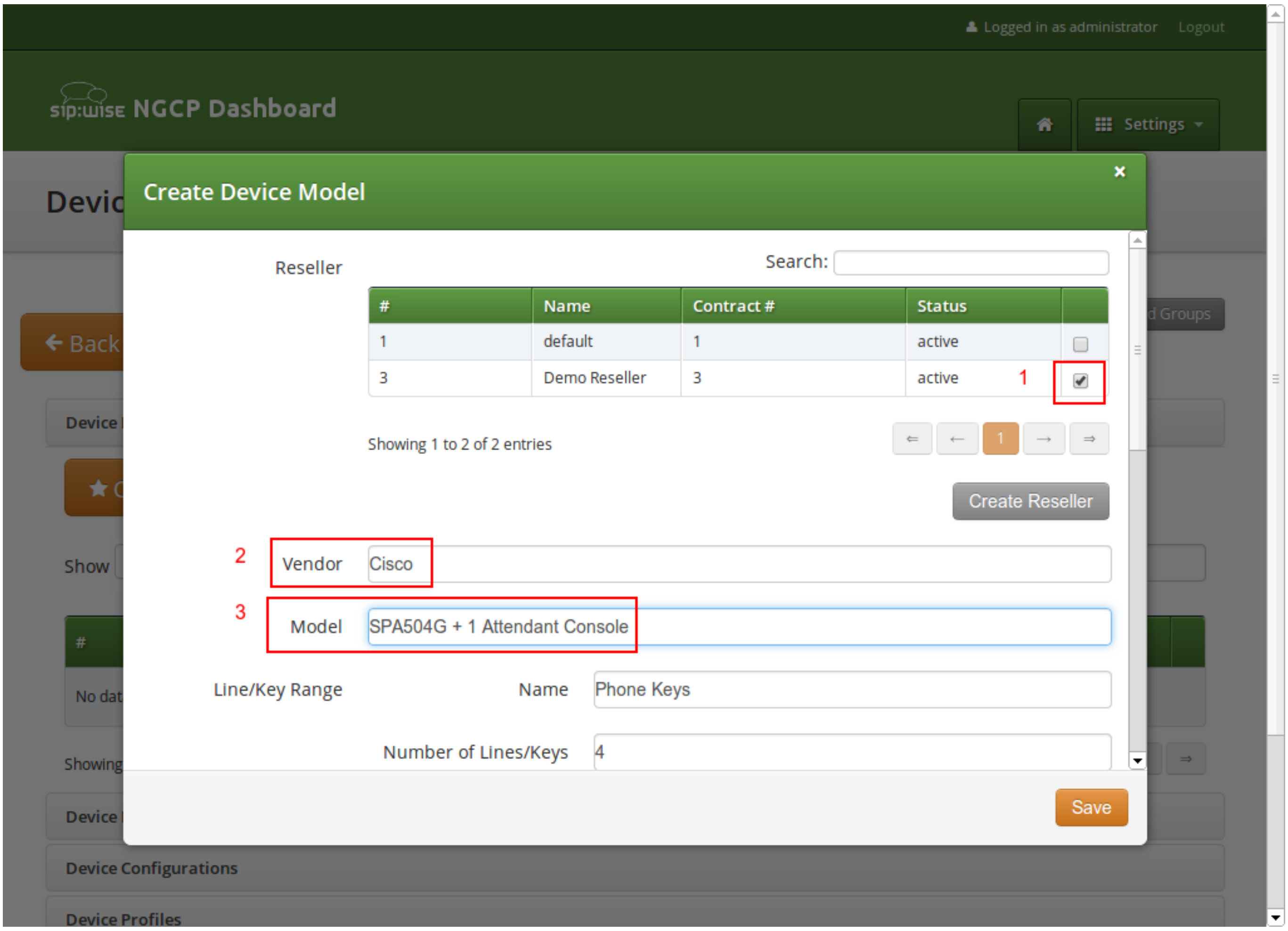


Figure 51: Create Device Model Part 1

In the *Line/Key Range* section, you can define the first set of keys, which we will label `Phone Keys`. The name is important, because it is referenced in the configuration file template, which is described in the following sections. The SPA504G internal phone keys support private lines (where the customer can assign a normal subscriber, which is used to place and receive standard phone calls), shared lines (where the customer can assign a subscriber which is shared across multiple people) and busy lamp field (where the customer can assign other subscribers to be monitored when they get a call, and which also acts as speed dial button to the subscriber assigned for BLF), so we enable all 3 of them.

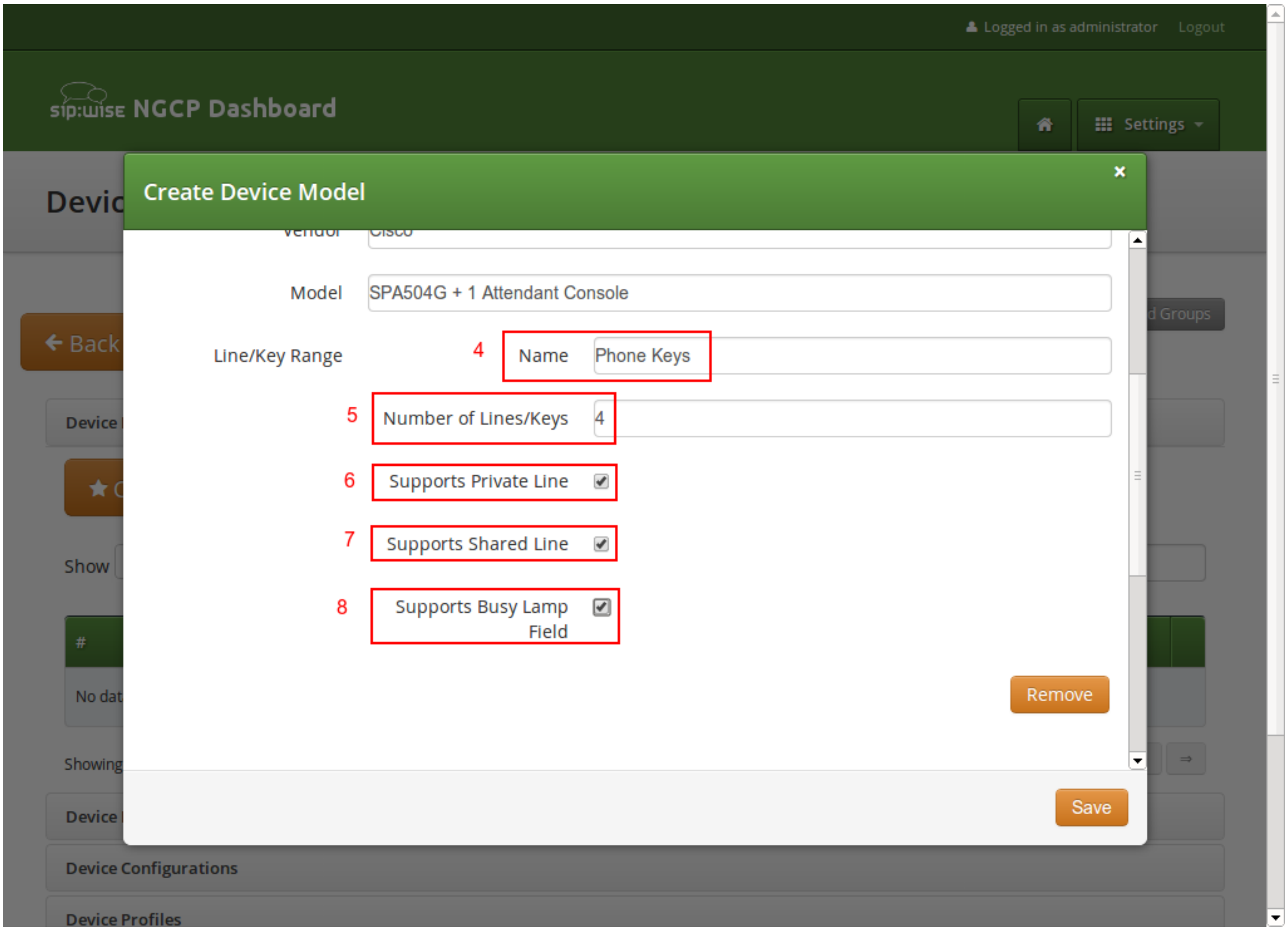


Figure 52: Create Device Model Part 2

In order to also configure the attendant console, press the *Add another Line/Key Range* button to specify the attendant console keys.

Again provide a name for this range, which will be `Attendant Console 1` to match our configuration defined later. There are 32 buttons on the attendant console, so set the number accordingly. Those 32 buttons only support BLF, so make sure to **uncheck** the private and shared line options, and only check the `busy lamp field` option.

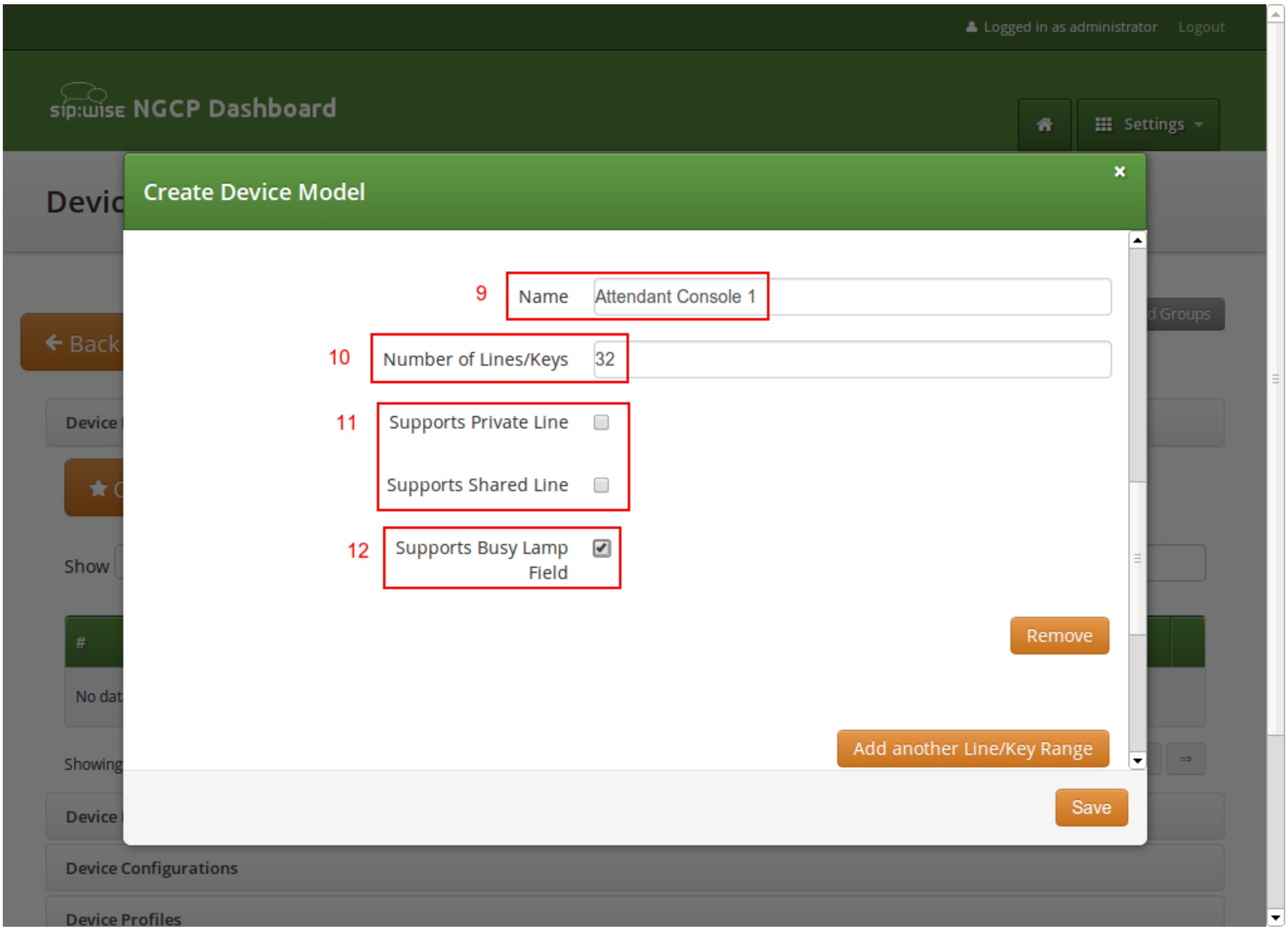


Figure 53: Create Device Model Part 3

The last two settings to configure are the *Front Image* and *MAC Address Image* fields. Upload a picture of the phone here in the first field, which is shown to the customer for him to recognize easily how the phone looks like. The MAC image is used to tell the customer where he can read the MAC address from. This could be a picture of the back of the phone with the label where the MAC is printed, or an instruction image how to get the MAC from the phone menu.

The rest of the fields are left at their default values, which are set to work with Cisco SPAs. Their meaning is as follows:

- *Bootstrap Sync URI*: If a stock phone is plugged in for the first time, it needs to be provisioned somehow to let it know where to fetch its configuration file from. Since the stock phone doesn't know about your server, you have to define an HTTP URI here, where the customer is connected with his web browser to set the according field.
- *Bootstrap Sync HTTP Method*: This setting defines whether an HTTP GET or POST is sent to the Sync URI.
- *Bootstrap Sync Params*: This setting defines the parameters appended to the Sync URI in case of a GET, or posted in the request body in case of POST, when the customer presses the *Sync* button later on.

Finally press *Save* to create the new device model.

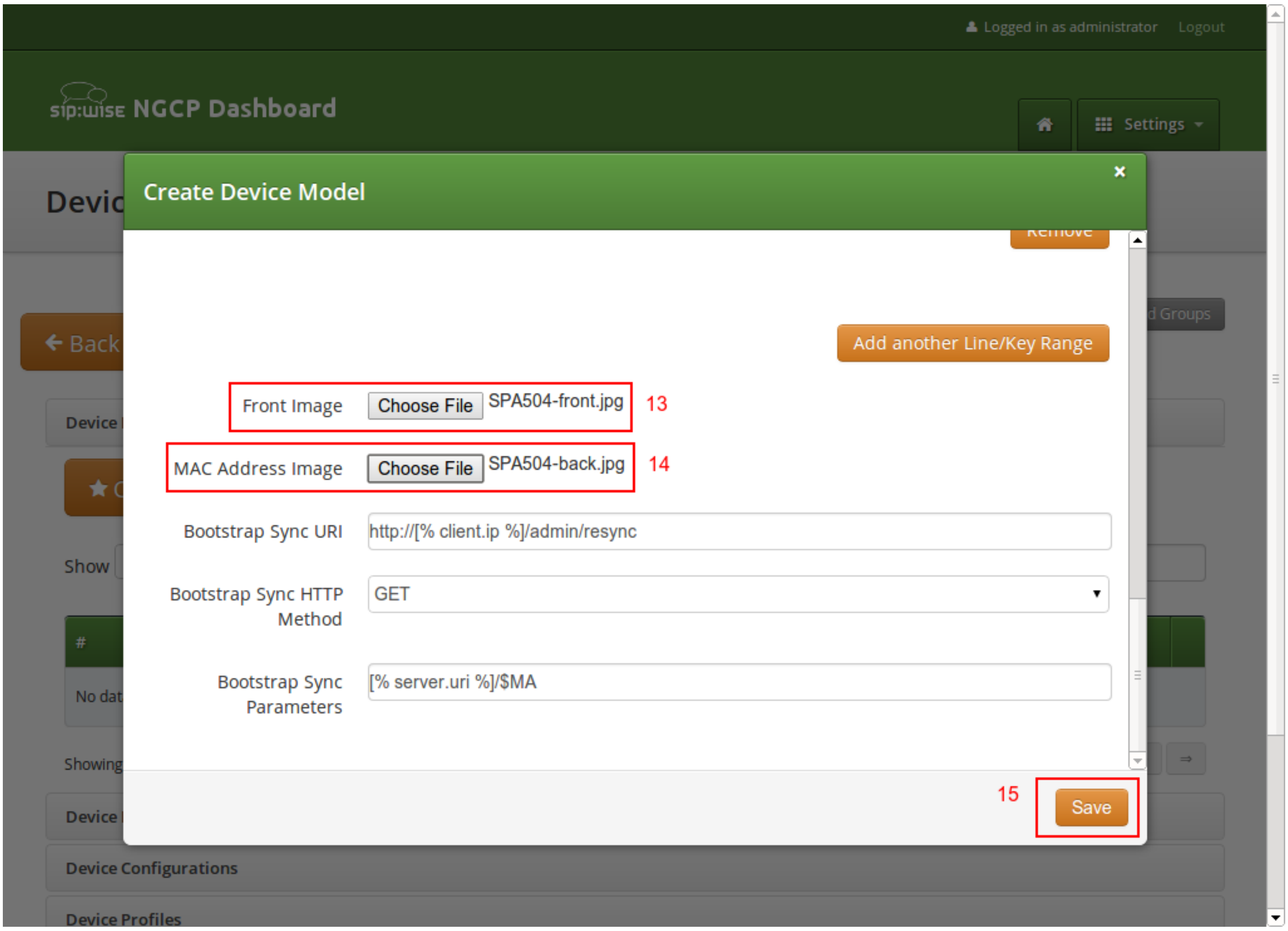


Figure 54: Create Device Model Part 4

15.1.1.2 Uploading Device Firmwares

A device model can optionally have one or more device firmware(s). Some devices like the Cisco SPA series don't support direct firmware updates from an arbitrary to the latest one, but need to go over specific firmware steps. In the device configuration discussed next, you can return the *next* supported firmware version, if the phone passes the current version in the firmware URL.

Since a stock phone purchased from any shop can have an arbitrary firmware version, we need to upload all firmwares needed to get from any old one to the latest one. In case of the Cisco SPA3x/SPA5x series, that would be the following versions, if the phone starts off with version 7.4.x:

- spa50x-30x-7-5-1a.bin
- spa50x-30x-7-5-2b.bin
- spa50x-30x-7-5-5.bin

So to get an SPA504G with a firmware version 7.4.x to the latest version 7.5.5, we need to upload each firmware file as follows.

Open the *Device Firmware* row in the *Device Management* section and press *Upload Device Firmware*.

Select the device model we're going to upload the firmware for, then specify the firmware version and choose the firmware file, then press *Save*.

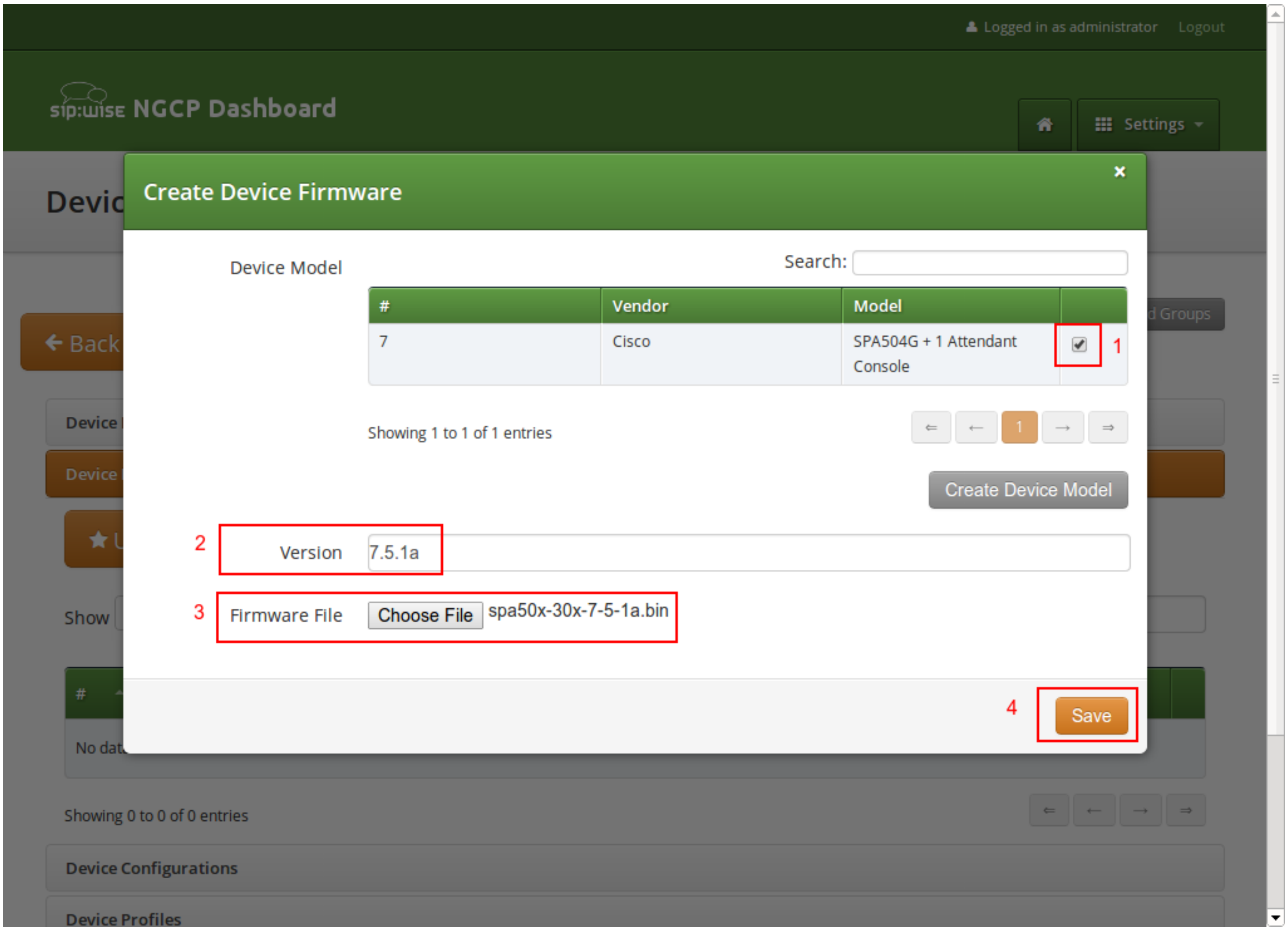


Figure 55: Upload Device Firmware

Repeat this step for every firmware in the list above (and any new firmware you want to support when it's available).

15.1.1.3 Creating Device Configurations

Each customer device needs a configuration file, which defines the URL to perform firmware updates, and most importantly, which defines the subscribers and features configured on each of the lines and keys. Since these settings are different for each physical phone at all the customers, the Cloud PBX module provides a template system to specify the configurations. That way, template variables can be used in the generic configuration, which are filled in by the system individually when a physical device fetches its configuration file.

To upload a configuration template, open the *Device Configuration* row and press *Create Device Configuration*.

Select the device model and specify a version number for this configuration (it is only for your reference to keep track of different versions). For Cisco SPA phones, keep the *Content Type* field to `text/xml`, since the configuration content will be served to the phone as XML file.

For devices other than the Cisco SPA, you might set `text/plain` if the configuration file is plain text, or `application/octet-stream` if the configuration is compiled into some binary form.

Finally paste the configuration template into the *Content* area and press *Save*.

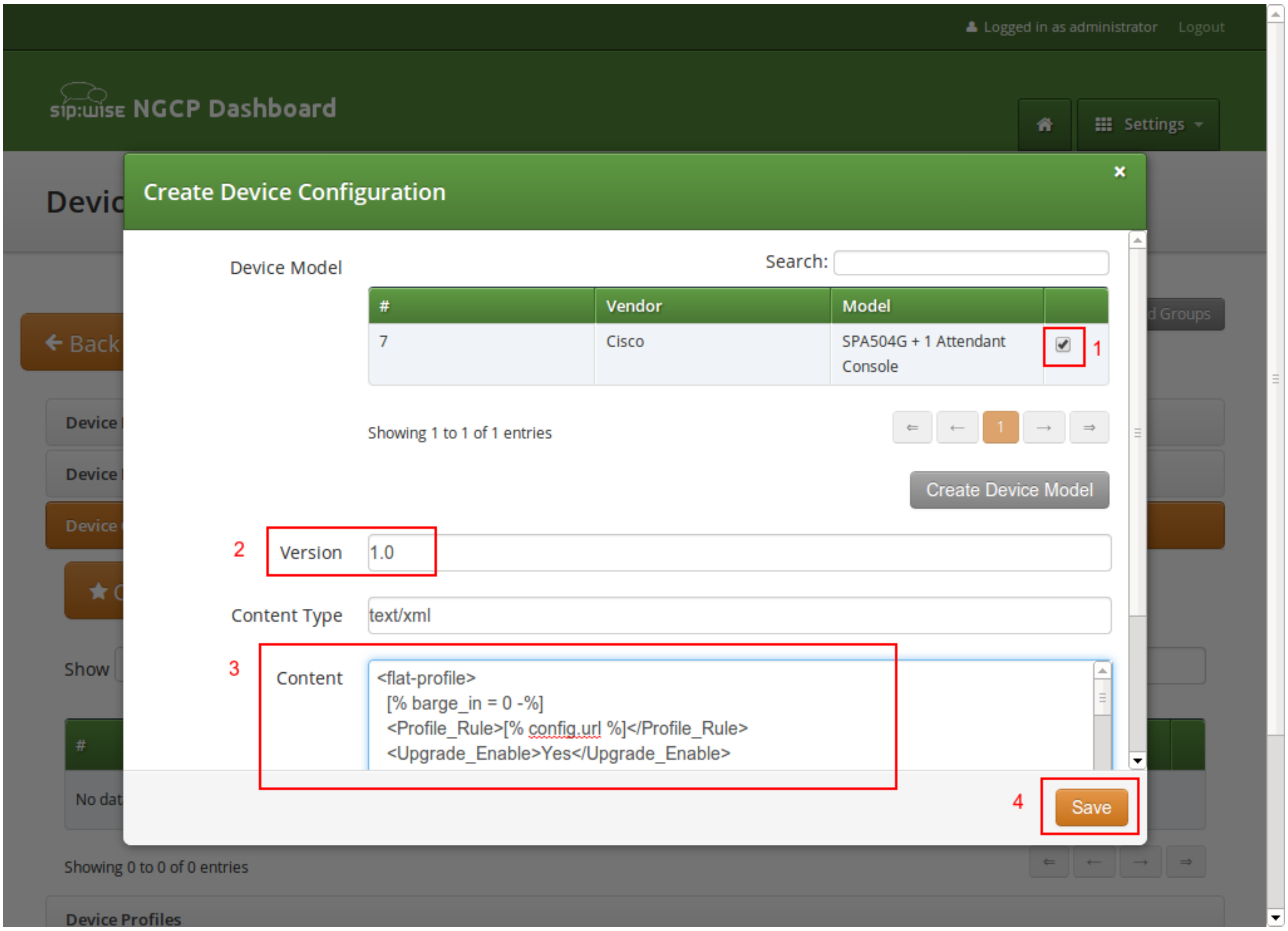


Figure 56: Upload Device Configuration

The templates for certified device models are provided by Sipwise, but you can also write your own. The following variables can be used in the template:

- `config.url`: The URL to the config file, including the device identifier (e.g. `http://sip.example.org:1444/device/autoprov/config/001122334455`).
- `firmware.maxversion`: The latest firmware version available on the system for the specific device.
- `firmware.baseurl`: The base URL to download firmwares (e.g. `http://sip.example.org:1444/device/autoprov/firmware`). To fetch the next newer firmware for a Cisco SPA, you can use the template line `[% firmware.baseurl %]/$MA/from/$SWVER/next`.

- `phone.stationname`: The name of the station (physical device) the customer specifies for this phone. Can be used to show on the display of the phone.
- `phone.lineranges`: An array of lines/keys as specified for the device model. Each entry in the array has the following keys:
  - `name`: The name of the line/key range as specified in the *Device Model* section (e.g. `Phone Keys`).
  - `num_lines`: The number of lines/keys in the line range (e.g. `4` in our `Phone Keys` example, or `32` in our `Attendant Console 1` example).
  - `lines`: An array of lines (e.g. subscriber definitions) for this line range. Each entry in the array has the following keys:
    - `keynum`: The index of the key in the line range, starting from 0 (e.g. `keynum` will be 3 for the 4th key of our `Phone Keys` range).
    - `rangenum`: The index of the line range, starting from 0. The order of line ranges is as you have specified them (e.g. `Phone Keys` was specified first, so it gets rangenum 0, `Auto Attendant 1` gets rangenum 1).
    - `type`: The type of the line/key, on of `private`, `shared` or `blf`.
    - `username`: The SIP username of the line.
    - `domain`: The SIP domain of the line.
    - `password`: The SIP password of the line.
    - `displayname`: The SIP Display Name of the line.

Within the configuration template itself, you can use any Template Toolkit directive and any own variables you like (just make sure to not override any of the ones specified above). For documentation on the syntax, please refer to the Template Toolkit Manual.

Tip

In order to change the provisioning base IP and port (default 1444), you have to access `/etc/ngcp-config/config.yml` and change the value `host` and `port` under the `autoprov.server` section.

15.1.1.4 Creating Device Profiles

When the customer configures his own device, he doesn't select a *Device Model* directly, but a *Device Profile*. A device profile specifies which model is going to be used with which configuration version. This allows the operator to create new configuration files and assign them to a profile, while still keeping older configuration files for reference or roll-back scenarios. It also makes it possible to test new firmwares by creating a test device model with the new firmware and a specific configuration, without impacting any existing customer devices.

To create a *Device Profile* for our phone, open the *Device Profile* row in the *Device Management* section and press *Create Device Profile*.

Select the device configuration (which implicitly identifies a device model) and specify a *Profile Name*. This name is what the customer sees when he is selecting a device he wants to provision, so pick a descriptive name which clearly identifies a device. Press *Save* to create the profile.

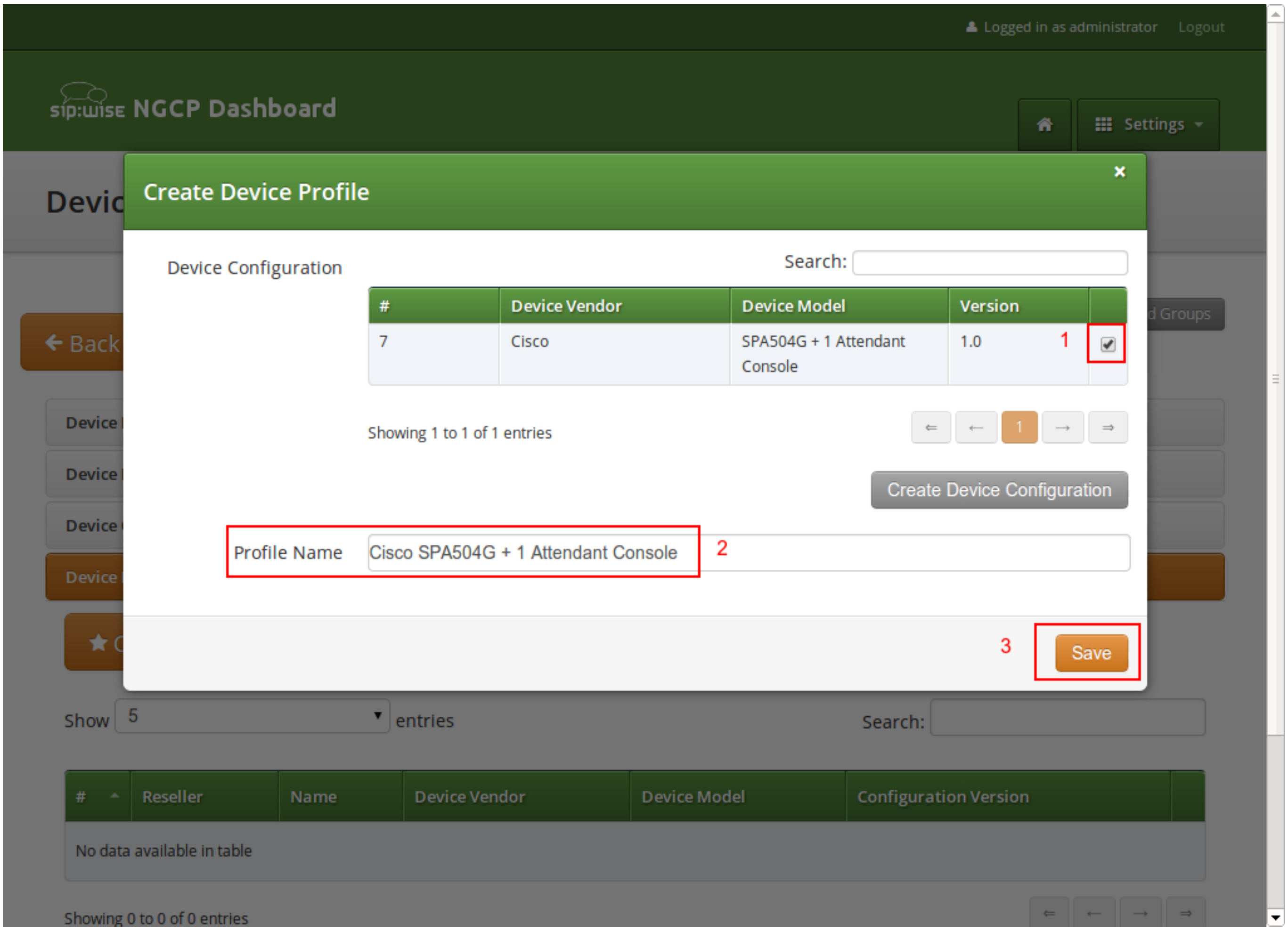


Figure 57: Create Device Profile

Repeat the steps as needed for every device you want to make available to customers.

15.1.2 Preparing PBX Rewrite Rules

In a PBX environment, the dial-plans usually looks different than for normal SIP subscribers. PBX subscribers should be able to directly dial internal extensions (e.g. 100) instead of the full number to reach another PBX subscriber in the same PBX segment. Therefore, we need to define specific *Rewrite Rules* to make this work.

The PBX dial plans are different from country to country. In the Central European area, you can directly dial an extension (e.g. `100`), and if you want to dial an international number like `0049 1 23456`, you have to dial a break-out digit first (e.g. `0`), so the number to be dialed is `0 0049 1 23456`. Other countries are used to other break-out codes (e.g. `9`), which then results in `9 0049 1 23456`. If you dial a national number like `01 23456`, then the number to actually be dialled is `9 01 23456`.

Since all numbers must be normalized to E.164 format via inbound rewrite rules, the rules need to be set up accordingly.

Let's assume that the break-out code for the example customers created below is `0`, so we have to create a *Rewrite Rule Set* with the following rules.

15.1.2.1 Inbound Rewrite Rules for Caller

- **Match Pattern**: `^([1-9][0-9]+)$`
- **Replacement Pattern**: `${caller_cloud_pbx_base_cli}\1`
- **Description**: `extension to e164`
- **Direction**: `Inbound`
- **Field**: `Caller`

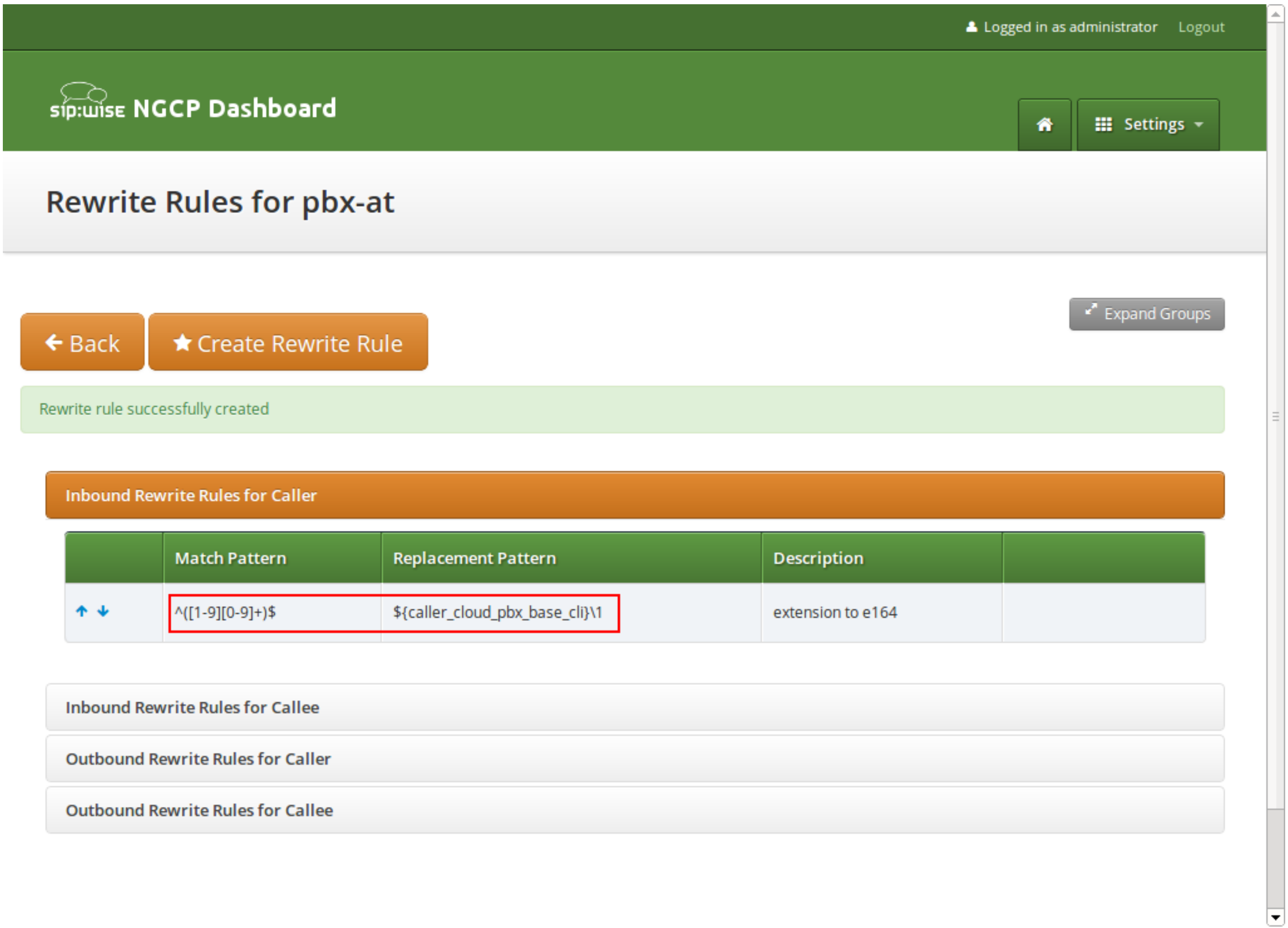


Figure 58: Inbound Rewrite Rule for Caller

15.1.2.2 Inbound Rewrite Rules for Callee

These rules are the most important ones, as they define which number formats the PBX subscribers can dial. For the break-out code of `0`, the following rules are necessary e.g. for German dialplans to allow pbx internal extension dialing, local area calls without area codes, national calls with area code, and international calls with country codes.

PBX INTERNAL EXTENSION DIALIN

- **Match Pattern**: `^([1-9][0-9]+)$`
- **Replacement Pattern**: `${caller_cloud_pbx_base_cli}\1`
- **Description**: `extension to e164`
- **Direction**: `Inbound`
- **Field**: `Callee`

LOCAL DIALING WITHOUT AREA CODE (USE BREAK-OUT CODE 0)

- **Match Pattern**: `^0([1-9][0-9]+)$`
- **Replacement Pattern**: `${caller_cc}${caller_ac}\1`
- **Description**: `local to e164`
- **Direction**: `Inbound`
- **Field**: `Callee`

NATIONAL DIALING (USE BREAK-OUT CODE 0 AND PREFIX AREA CODE BY 0)

- **Match Pattern**: `^00([1-9][0-9]+)$`
- **Replacement Pattern**: `${caller_cc}\1`
- **Description**: `national to e164`
- **Direction**: `Inbound`
- **Field**: `Callee`

INTERNATIONAL DIALING (USE BREAK-OUT CODE 0 AND PREFIX COUNTRY CODE BY 00)

- **Match Pattern**: `^000([1-9][0-9]+)$`
- **Replacement Pattern**: `\1`
- **Description**: `international to e164`
- **Direction**: `Inbound`
- **Field**: `Callee`

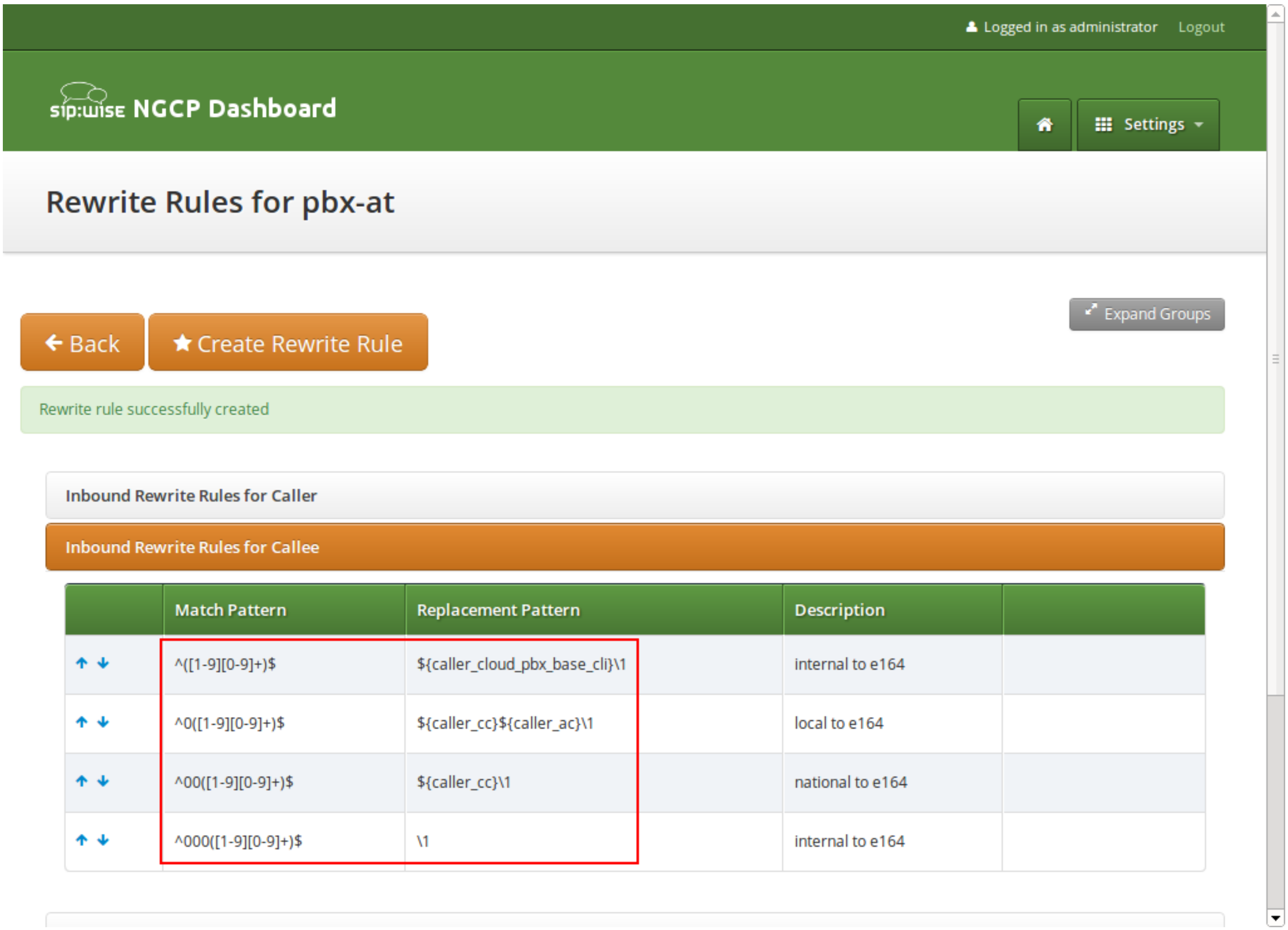


Figure 59: Inbound Rewrite Rule for Callee

15.1.2.3 Outbound Rewrite Rules for Caller

When a call goes to a PBX subscriber, it needs to be normalized in a way that it's call-back-able, which means that it needs to have the break-out code prefixed. We create a rule to show the calling number in international format including the break-out code. For PBX-internal calls, the caller name will be shown (this is handled by implicitly setting domain preferences accordingly, so you don't have to worry about that in rewrite rules).

ADDING A BREAK-OUT CODE (USE BREAK-OUT CODE `0` AND PREFIX COUNTRY CODE BY `00`)

- **Match Pattern**: `^([1-9][0-9]+)$`
- **Replacement Pattern**: `000\1`
- **Description**: `e164 to full international`
- **Direction**: `Outbound`
- **Field**: `Caller`

Displaying the extension in the caller number for PBX-internal calls

- **Match Pattern**: `^@{callee_cloud_pbx_account_cli_list}$`
- **Replacement Pattern**: `${caller_cloud_pbx_ext}`
- **Description**: `e164 to full international`
- **Direction**: `Outbound`
- **Field**: `Caller`

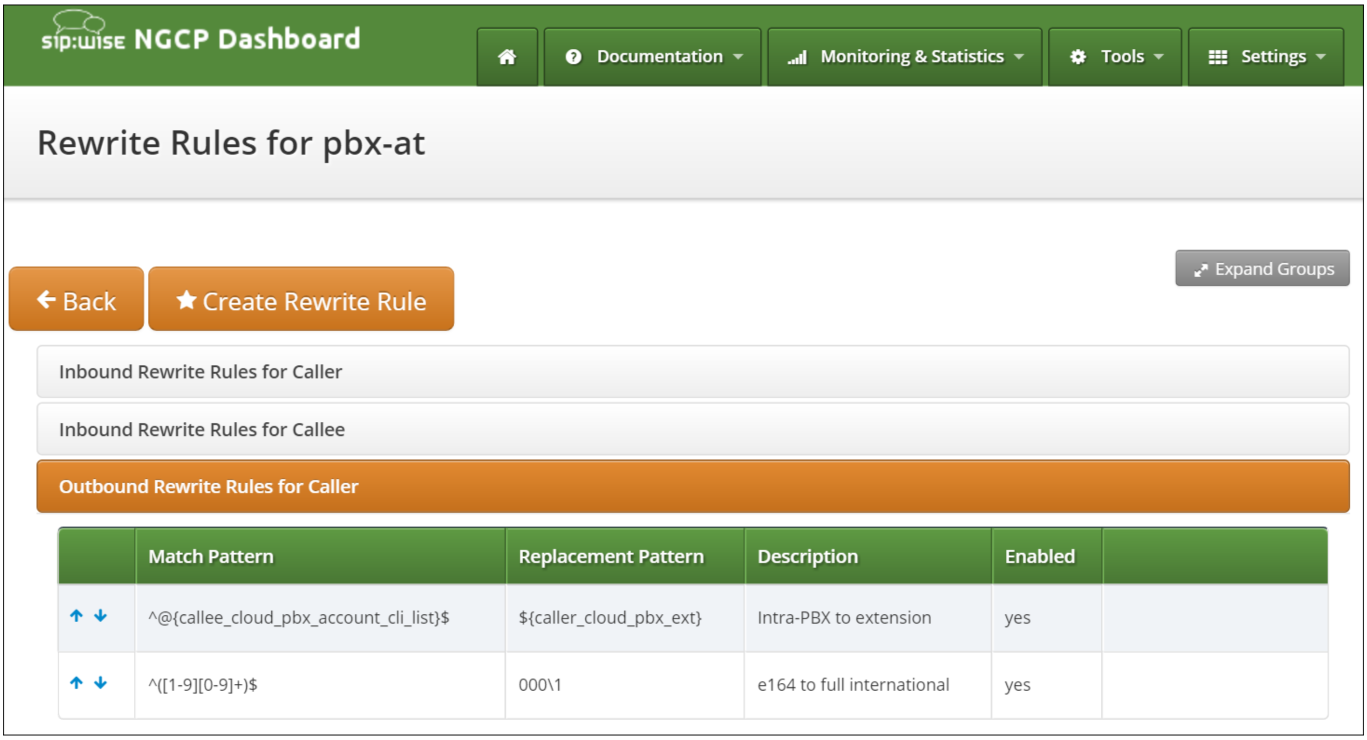


Figure 60: Outbound Rewrite Rule for Caller

Create a new *Rewrite Rule Set* for each dial plan you'd like to support. You can later assign it to customer domains and even to subscribers, if a specific subscriber of a PBX customer would like to have his own dial plan.

15.1.3 Creating Customers and Pilot Subscribers

As with a normal SIP Account, you have to create a *Customer* contract per customer, and one *Subscriber*, which the customer can use to log into the web interface and manage his PBX environment.

15.1.3.1 Creating a PBX Customer

Go to *Settings*→*Customers* and click *Create Customer*. We need a *Contact* for the customer, so press *Create Contact*.

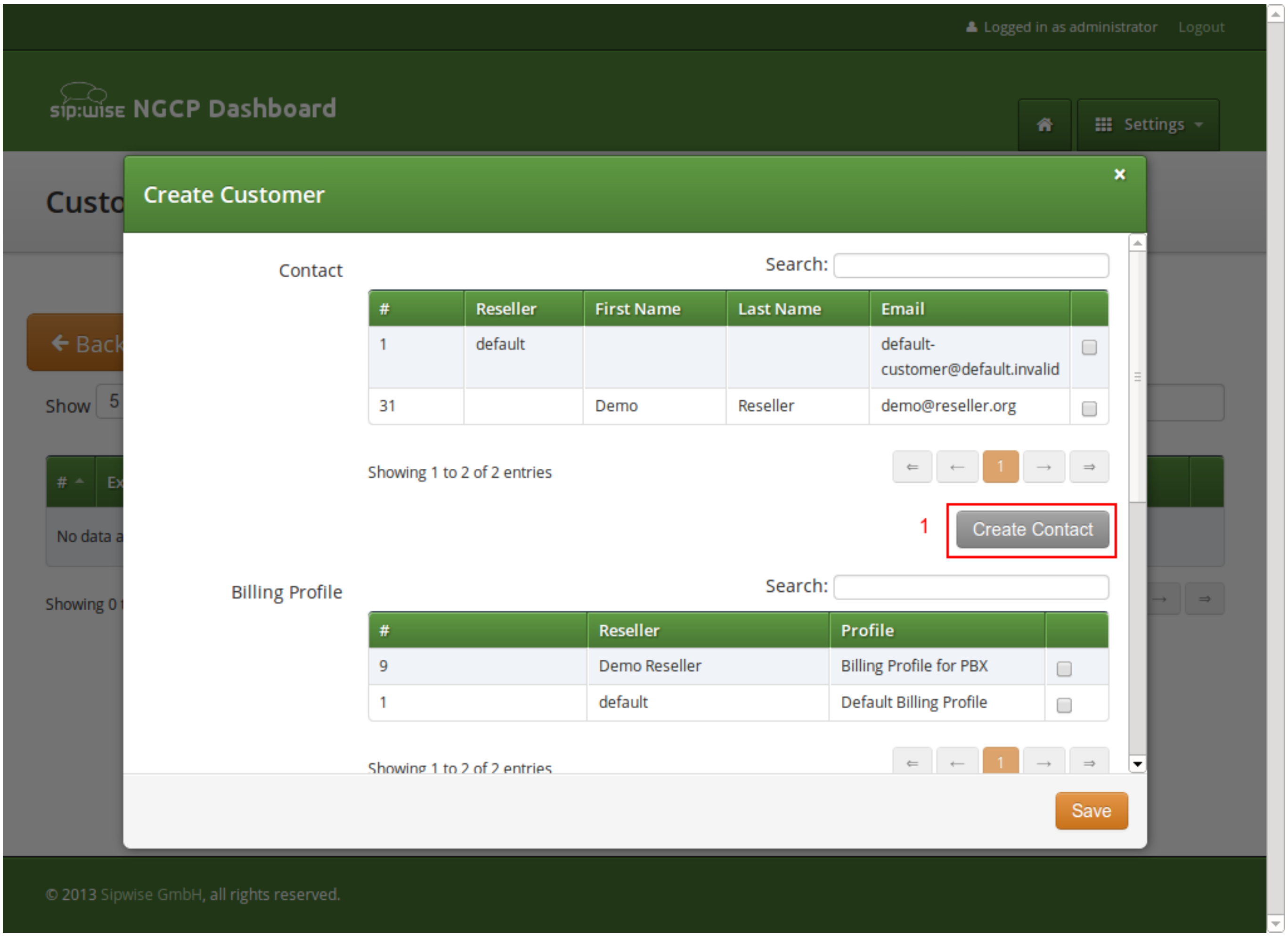


Figure 61: Create PBX Customer Part 1

Fill in the desired fields (you need to provide at least the *Email Address*) and press *Save*.

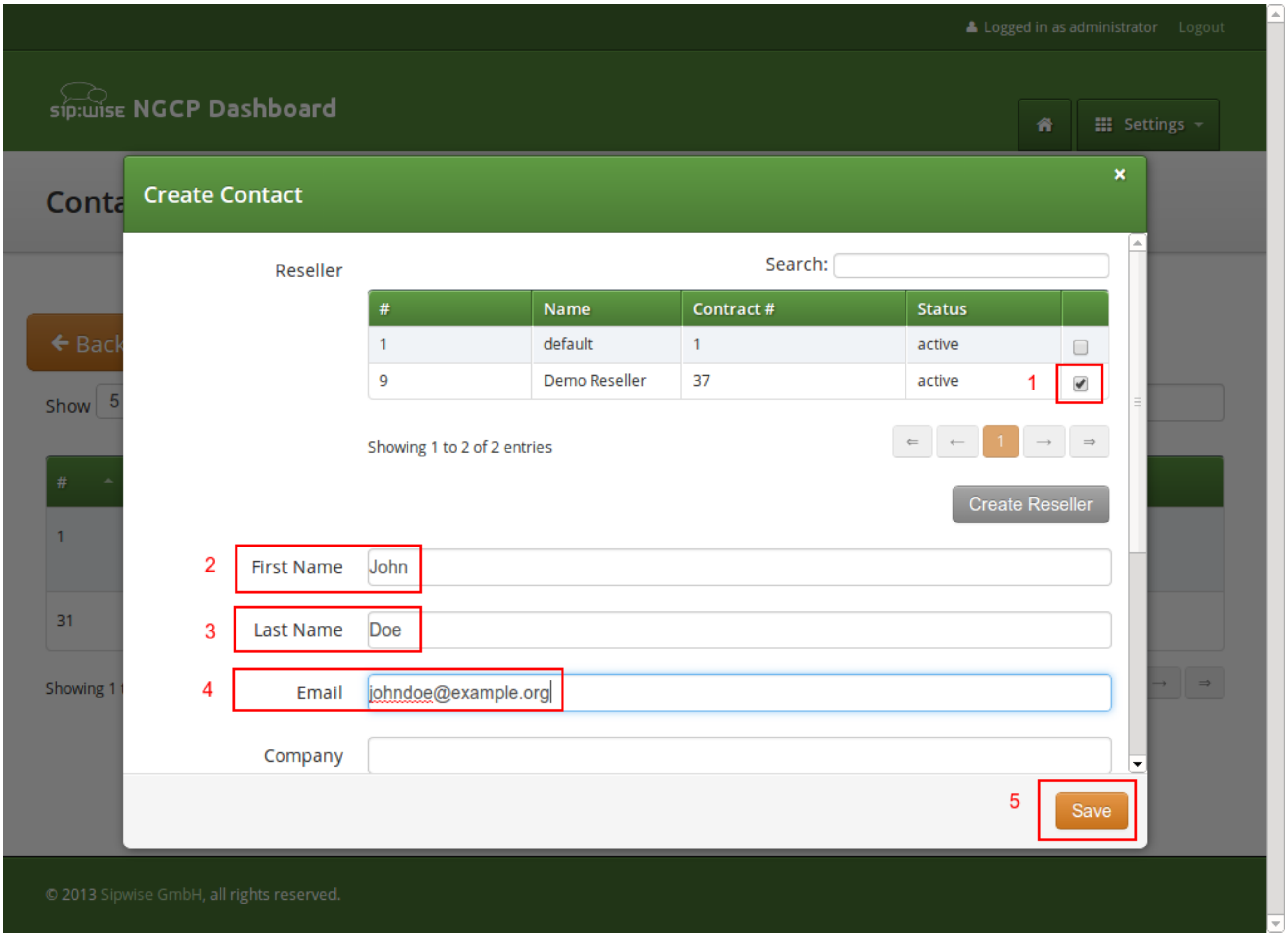


Figure 62: Create PBX Customer Contact

The new *Contact* will be automatically selected now. Also select a *Billing Profile* you want to use for this customer. If you don't have one defined yet, press *Create Billing Profile*, otherwise select the one you want to use.

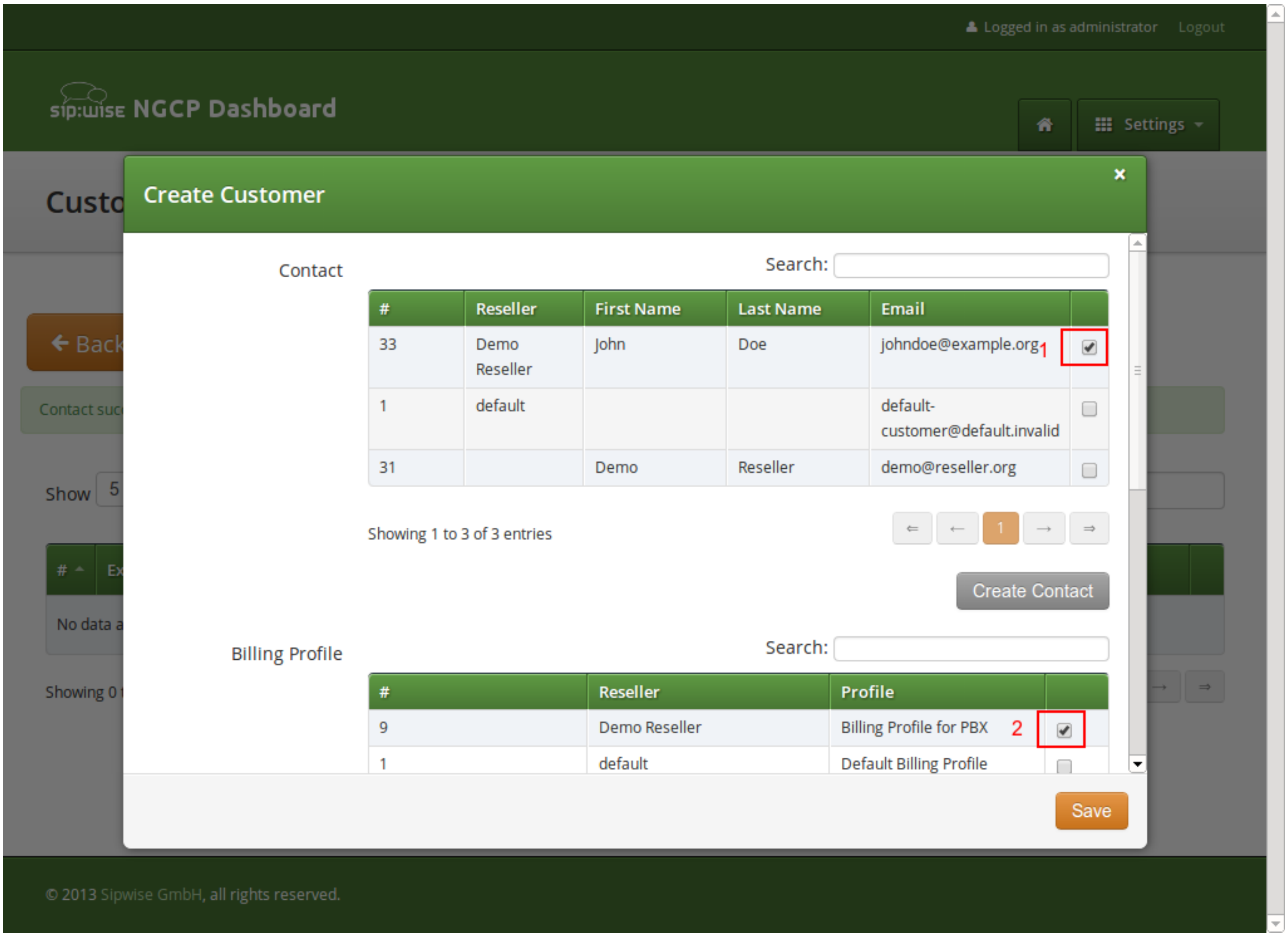


Figure 63: Create PBX Customer Part 2

Next, you need to select the *Product* for the PBX customer. Since it's going to be a PBX customer, select the product *Cloud PBX Account*.

Since PBX customers are supposed to manage their subscribers by themselves, they are able to create them via the web interface. To set an upper limit of subscribers a customer can create, define the value in the *Max Subscribers* field.

Important

As you will see later, both PBX subscribers and PBX groups are normal subscribers, so the value defined here limits the overall amount of subscribers **and** groups. A customer can create an unlimited amount of subscribers if you leave this field empty.

Press *Save* to create the customer.

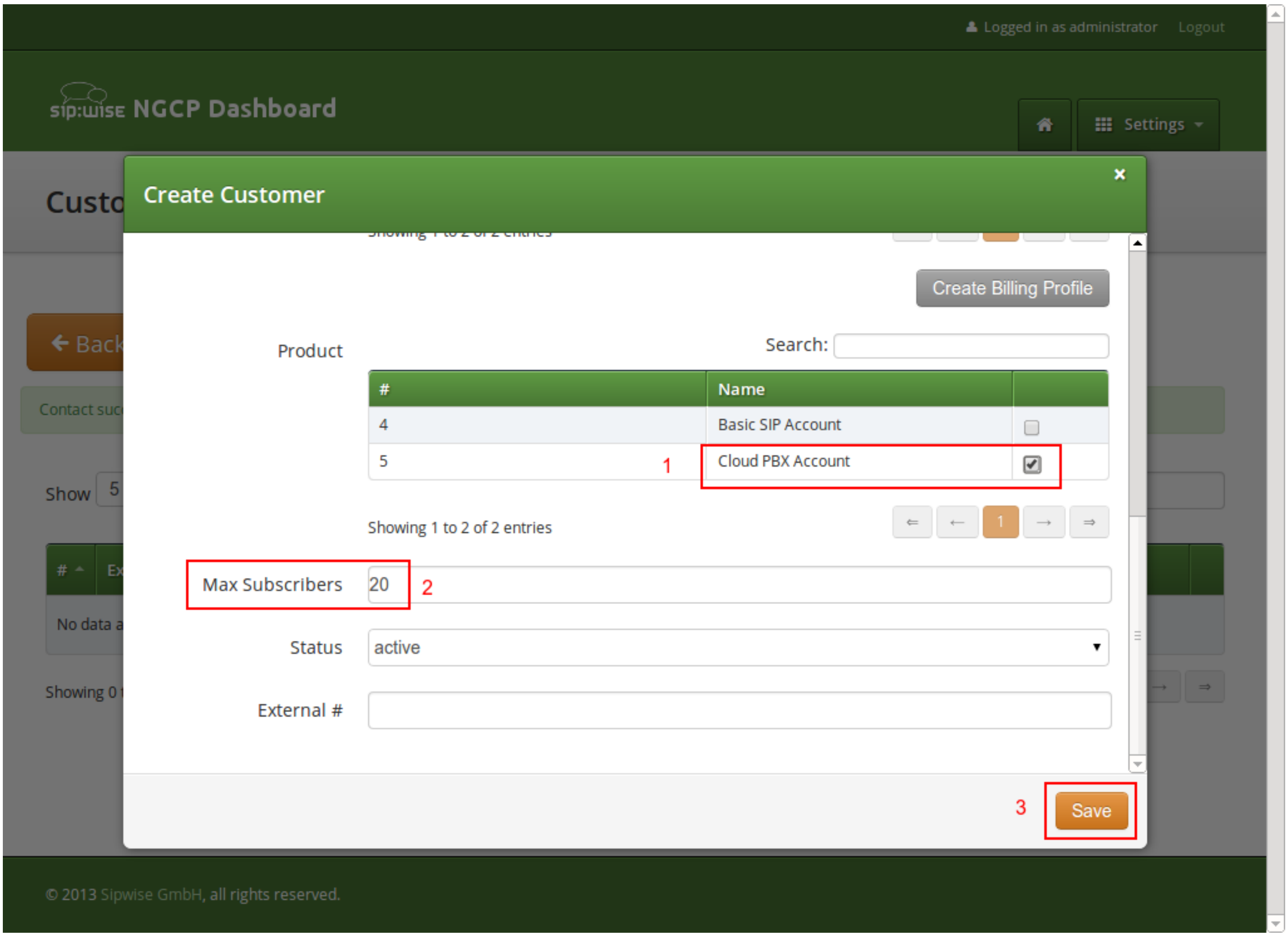


Figure 64: Create PBX Customer Part 3

15.1.3.2 Creating a PBX Pilot Subscriber

Once the customer is created, you need to create at least one *Subscriber* for the customer, so he can log into the web interface and manage the rest by himself.

Click the *Details* button on the newly created customer to enter the detailed view.

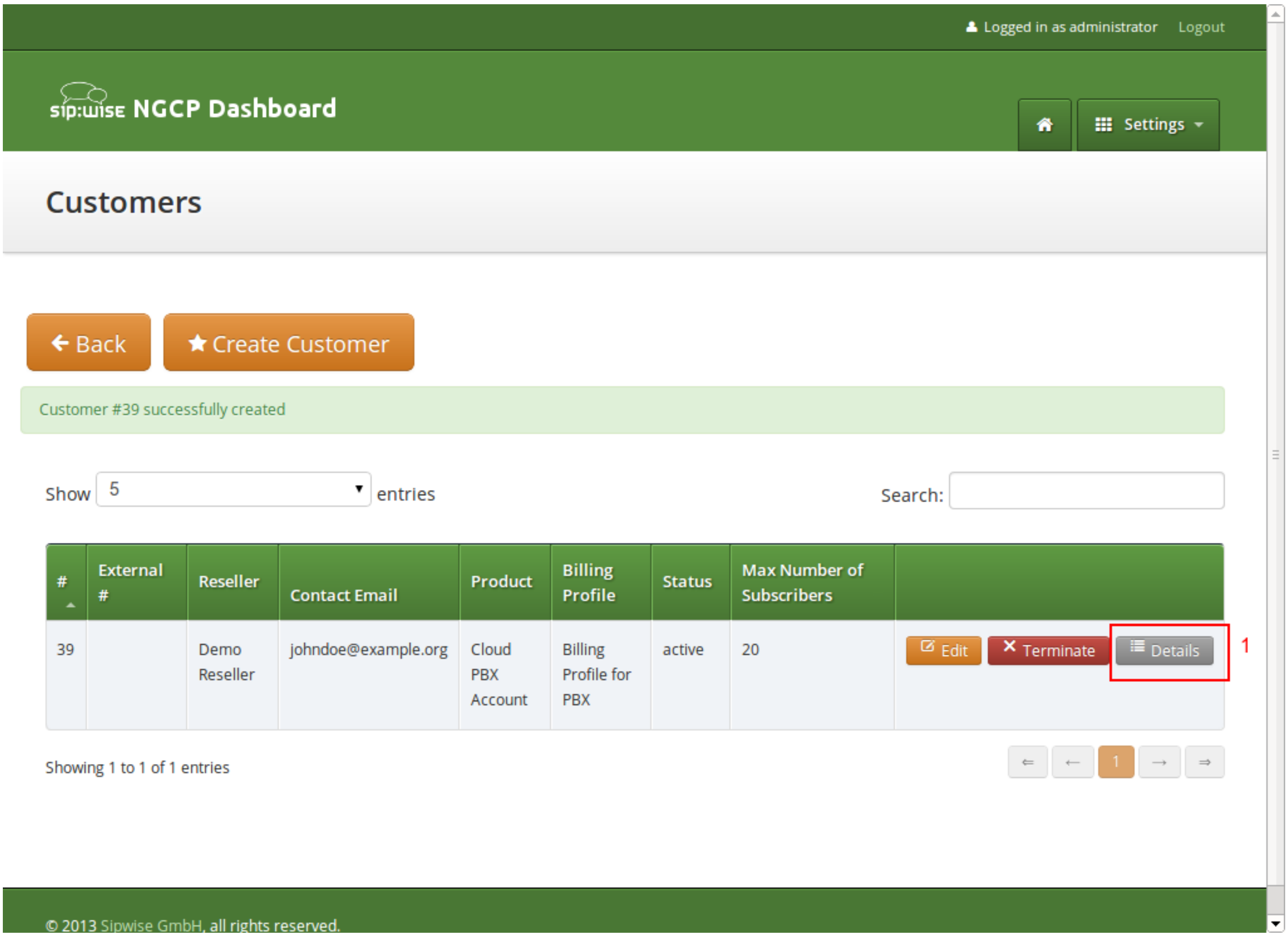


Figure 65: Go to Customer Details

To create the subscriber, open the *Subscribers* row and click *Create Subscriber*.

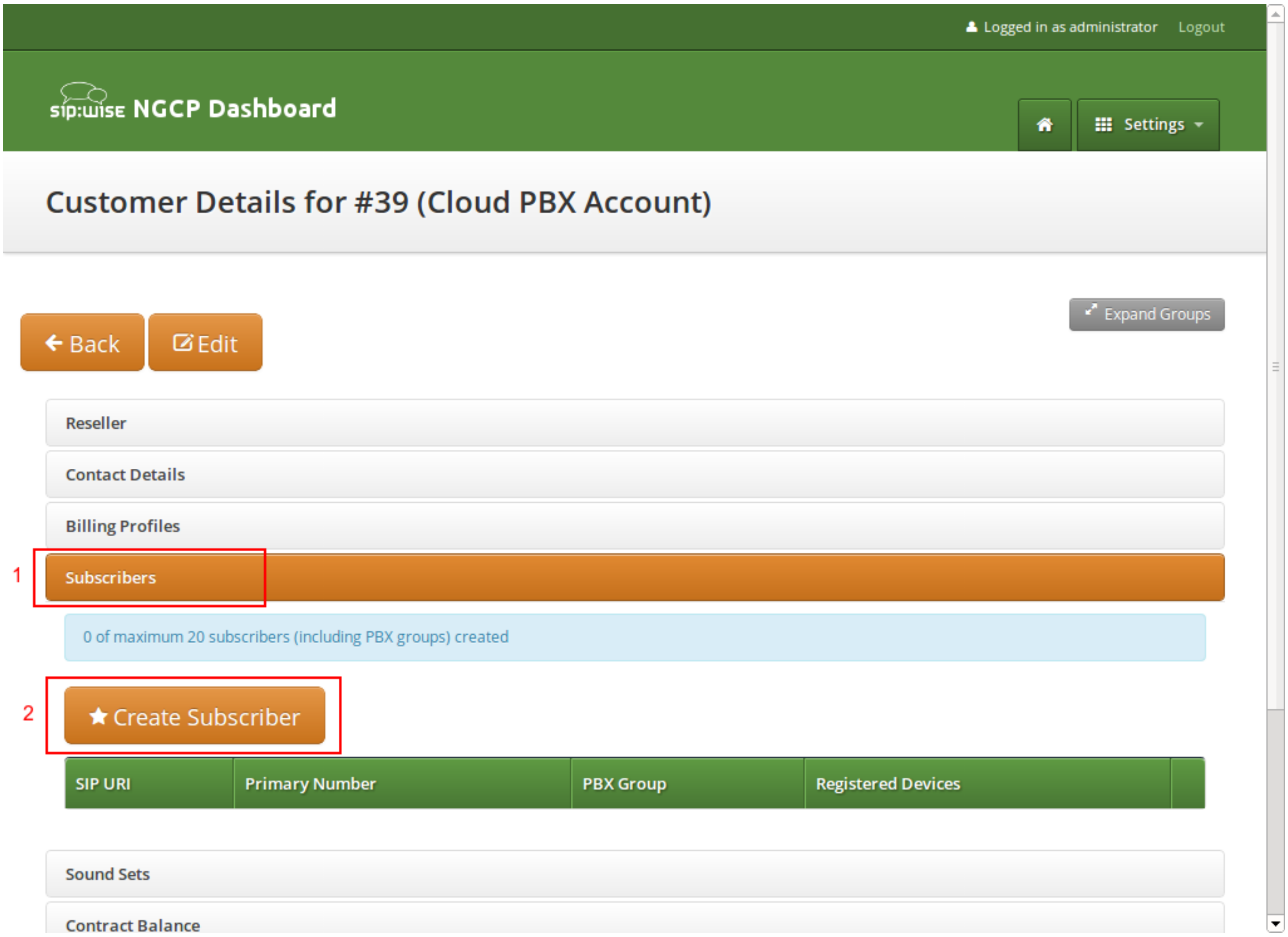


Figure 66: Go to Create Subscriber

For your pilot subscriber, you need a SIP domain, a pilot number (the main number of the customer PBX), the web credentials for the customer to log into the web interfaces, and the SIP credentials to authenticate via a SIP device.

Important

In a PBX environment, customers can create their own subscribers. As a consequence, each PBX customer should have its own SIP domain, in order to not collide with subscribers created by other customers. This is important because two customers are highly likely to create a subscriber (or group, which is also just a subscriber) called `office`. If they are in the same SIP domain, they'd both have the SIP URI `office@pbx.example.org`, which is not allowed, and the an end customer will probably not understand why `office@pbx.example.org` is already taken, because he (for obvious reasons, as it belongs to a different customer) will not see this subscriber in his subscribers list.

Tip

To handle one domain per customer, you should create a wild-card entry into your DNS server like `*.pbx.example.org`, which points to the IP address of `pbx.example.org`, so you can define SIP domains like `customer1.pbx.example.org` or `customer2.pbx.example.org` without having to create a new DNS entry for each of them. For proper secure access to the web interface and to the SIP and XMPP services, you should also obtain a SSL wild-card certificate for `*.pbx.example.org` to avoid certification warnings on customers' web browsers and SIP/XMPP clients.

So to create a new domain for the customer, click *Create Domain*.

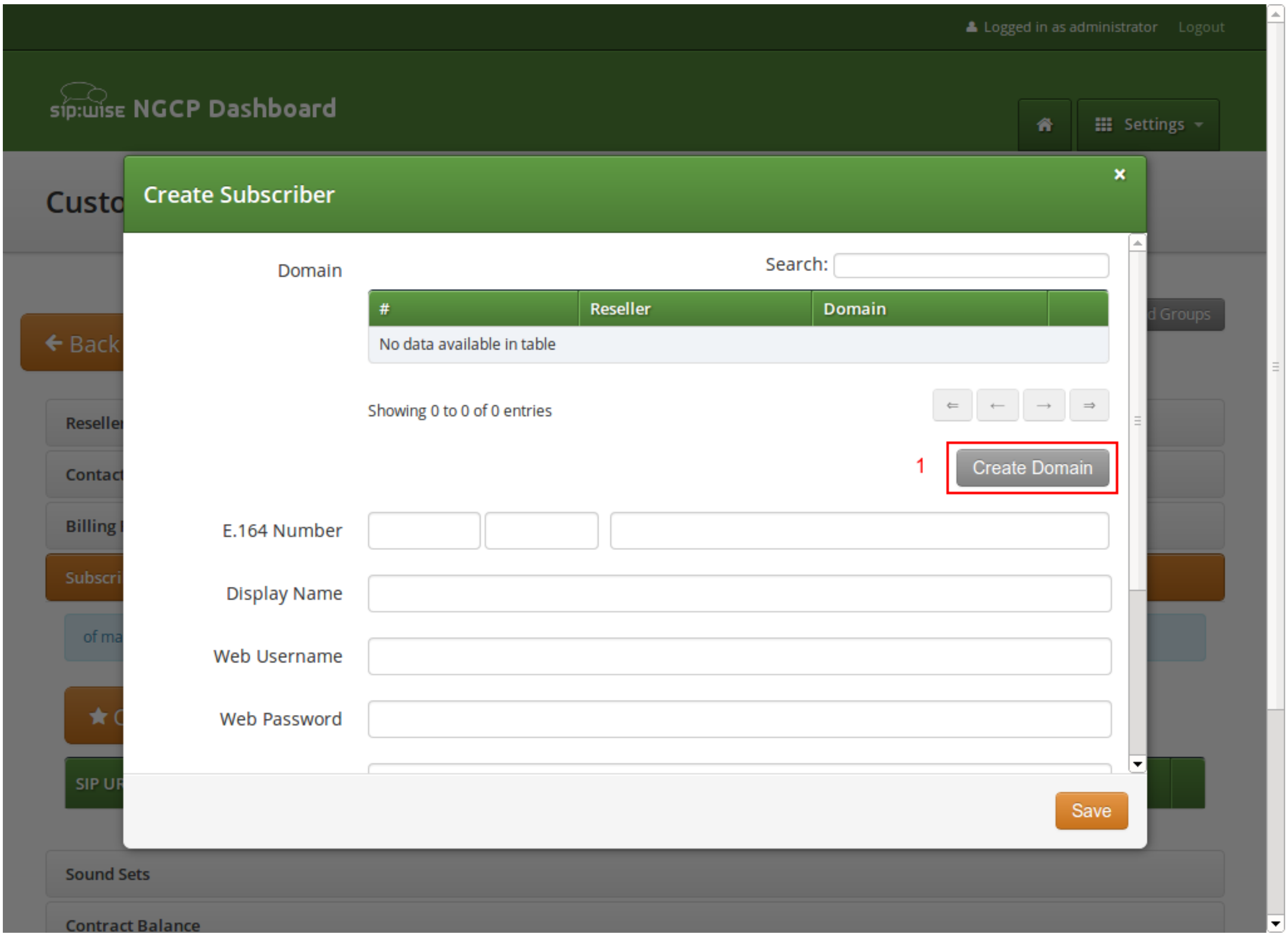


Figure 67: Go to Create Customer Domain

Specify the domain you want to create, and select the PBX *Rewrite Rule Set* which you created in Section 15.1.2, then click *Save*.

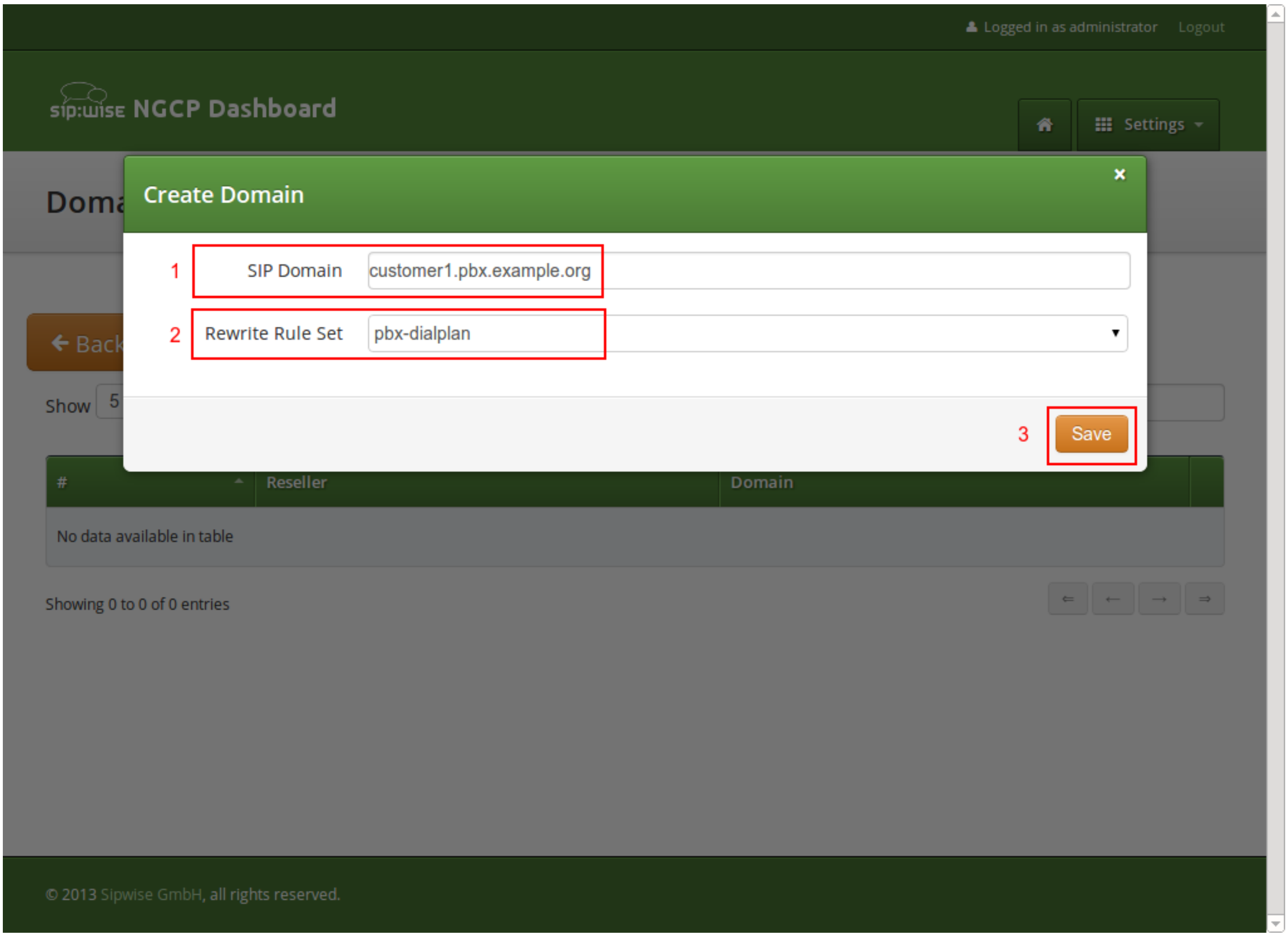


Figure 68: Create Customer Domain

Finish the subscriber creation by providing an E.164 number, which is going to be the base number for all other subscribers within this customer, the web username and password for the pilot subscriber to log into the web interface, and the sip username and password for a SIP device to connect to the PBX.

The parameters are as follows:

- **Domain**: The domain in which to create the pilot subscriber. *Each customer should get his own domain as described above to not collide with SIP usernames between customers*.
- **E.164 Number**: The primary number of the PBX. Calls to this number are routed to the pilot subscriber, and each subsequent subscriber created for this customer will use this number as its base number, suffixed by an individual extension. You can later assign alias numbers also for DID support.
- **Display Name**: This field is used on phones to identify subscribers by their real names instead of their number or extension. On outbound calls, the display name is signalled in the Display-Field of the From header, and it's used as a name in the XMPP contact lists.
- **Web Username**: The username for the subscriber to log into the customer self-care web interface. This is optional, if you don't

want a subscriber to have access to the web interface.

- **Web Password**: The password for the subscriber to log into the customer self-care web interface.
- **SIP Username**: The username for the subscriber to authenticate on the SIP and XMPP service. It is automatically used for devices, which are auto-provisioned via the *Device Management*, or can be used manually by subscribers to sign into the SIP and XMPP service with any arbitrary clients.
- **SIP Password**: The password for the subscriber to authenticate on the SIP and XMPP service.

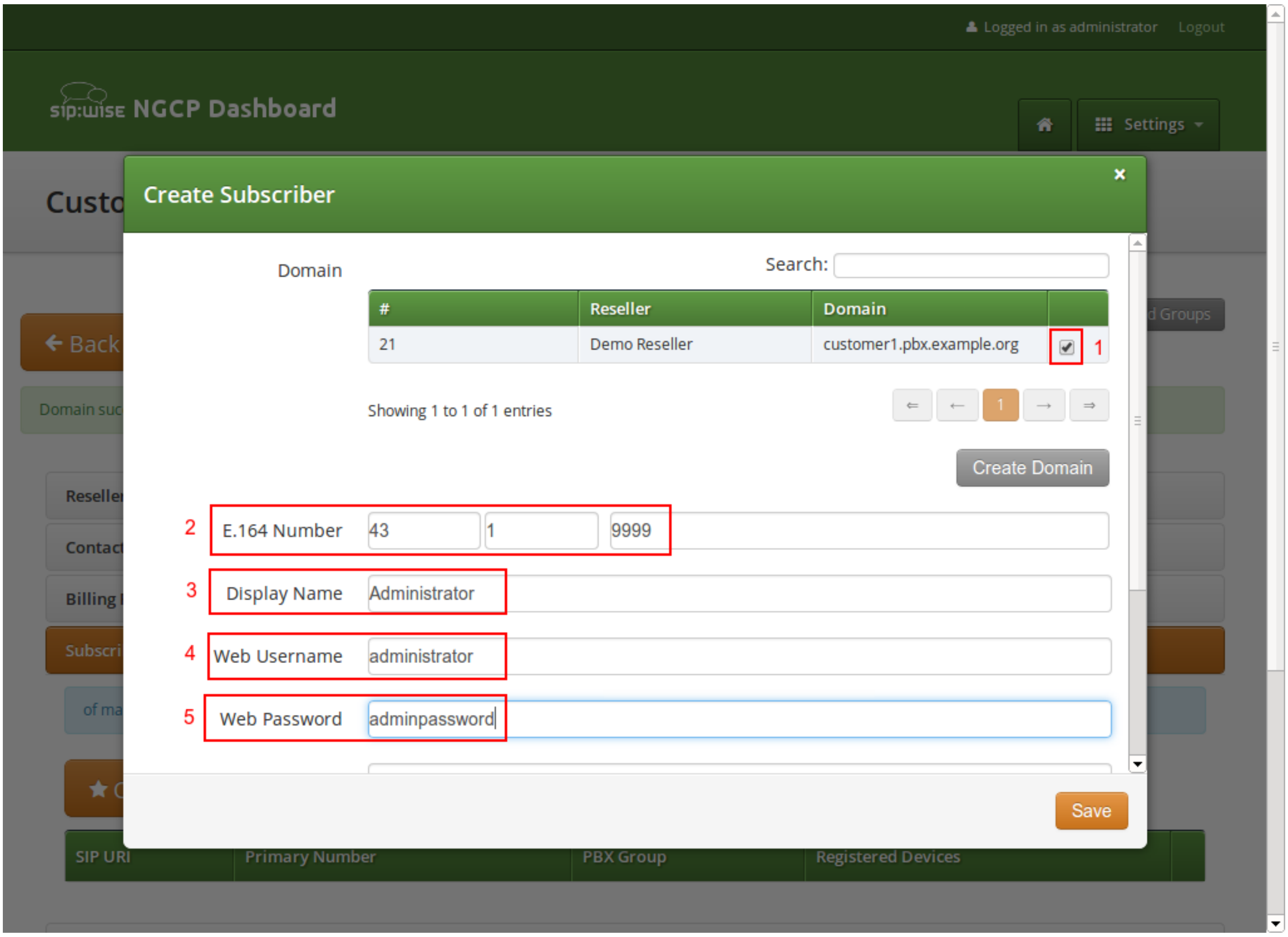


Figure 69: Create Pilot Subscriber Part 1

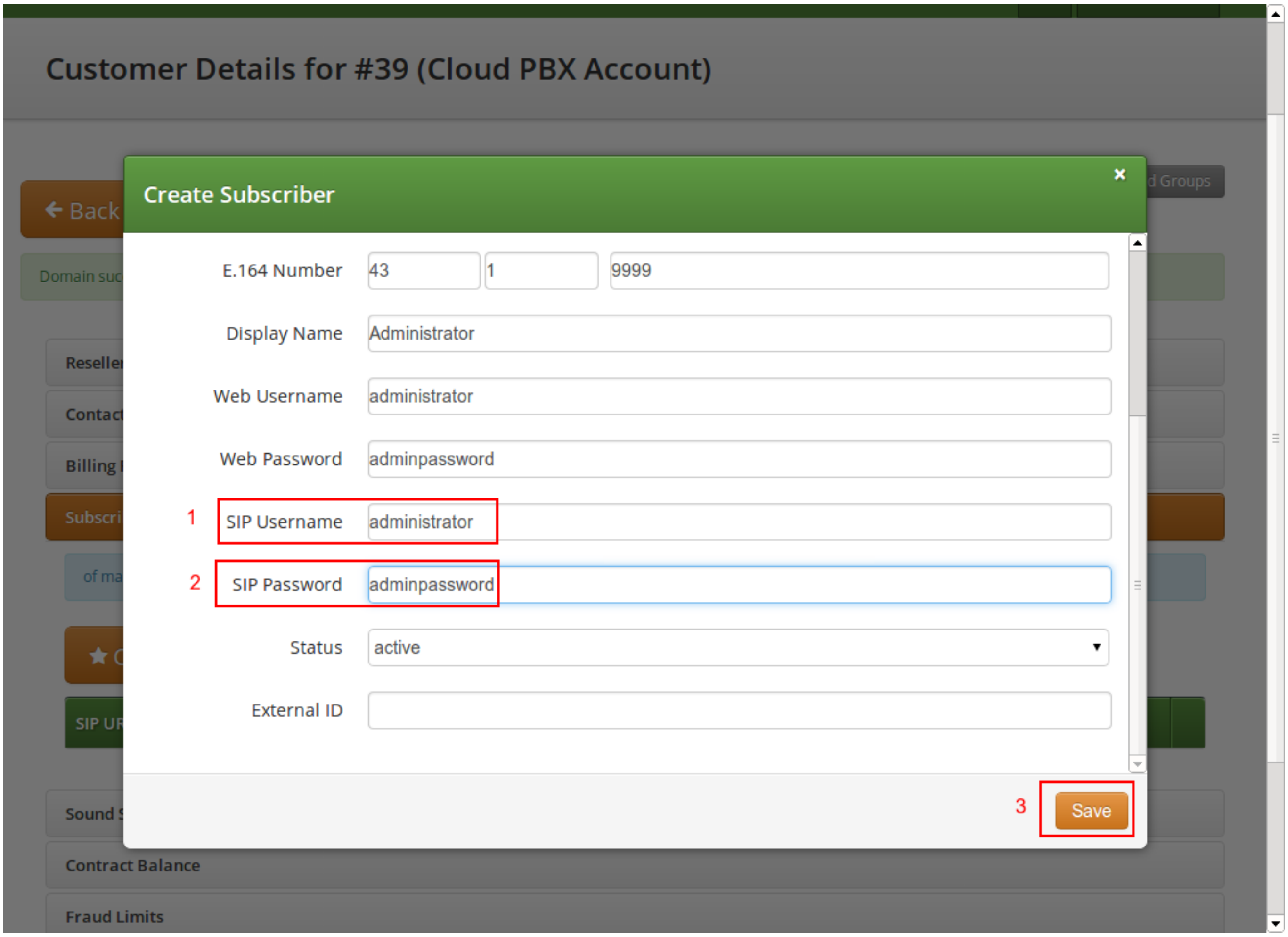


Figure 70: Create Pilot Subscriber Part 2

Once the subscriber is created, he can log into the customer self-care interface at `https://<your-ip>/login/subscriber` and manage his PBX, like creating other users and groups, assigning Devices to subscribers and configure the Auto Attendant and more.

As an administrator, you can also do this for the customer, and we will walk through the typical steps as an administrator to configure the different features.

Go the the *Customer Details* of the PBX customer you want to configure, e.g. by navigating to *Settings*→*Customers* and clicking the *Details* button of the customer you want to configure.

15.1.4 Creating Regular PBX Subscribers

Since we already created a pilot subscriber, more settings now appear on the *Customer Details* view. The sections we are interested in for now are the *Subscribers* and *PBX Groups* sections.

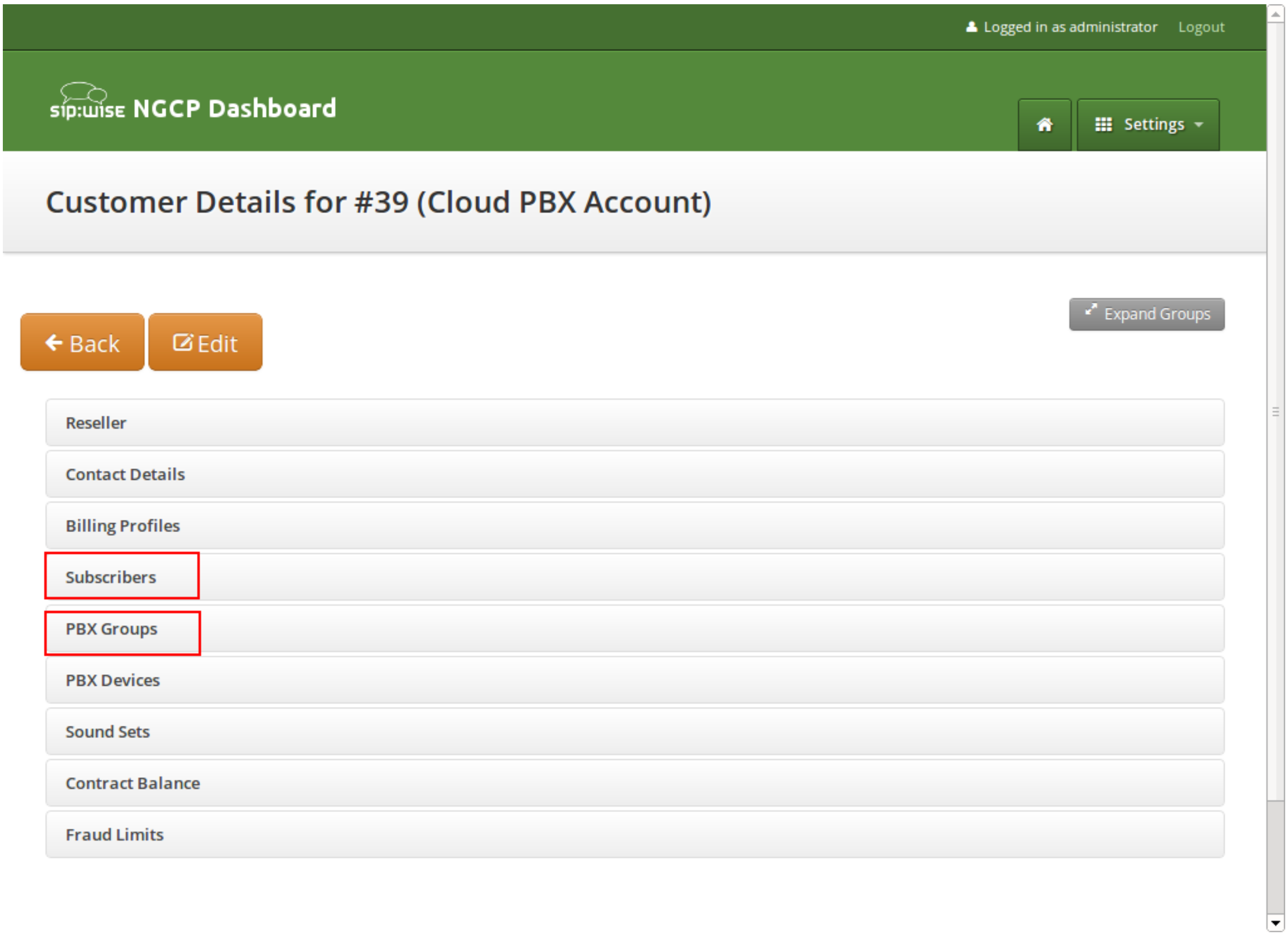


Figure 71: Subscribers and PBX Groups

To create another subscriber for the customer PBX, open the *Subscribers* row and click *Create Subscriber*.

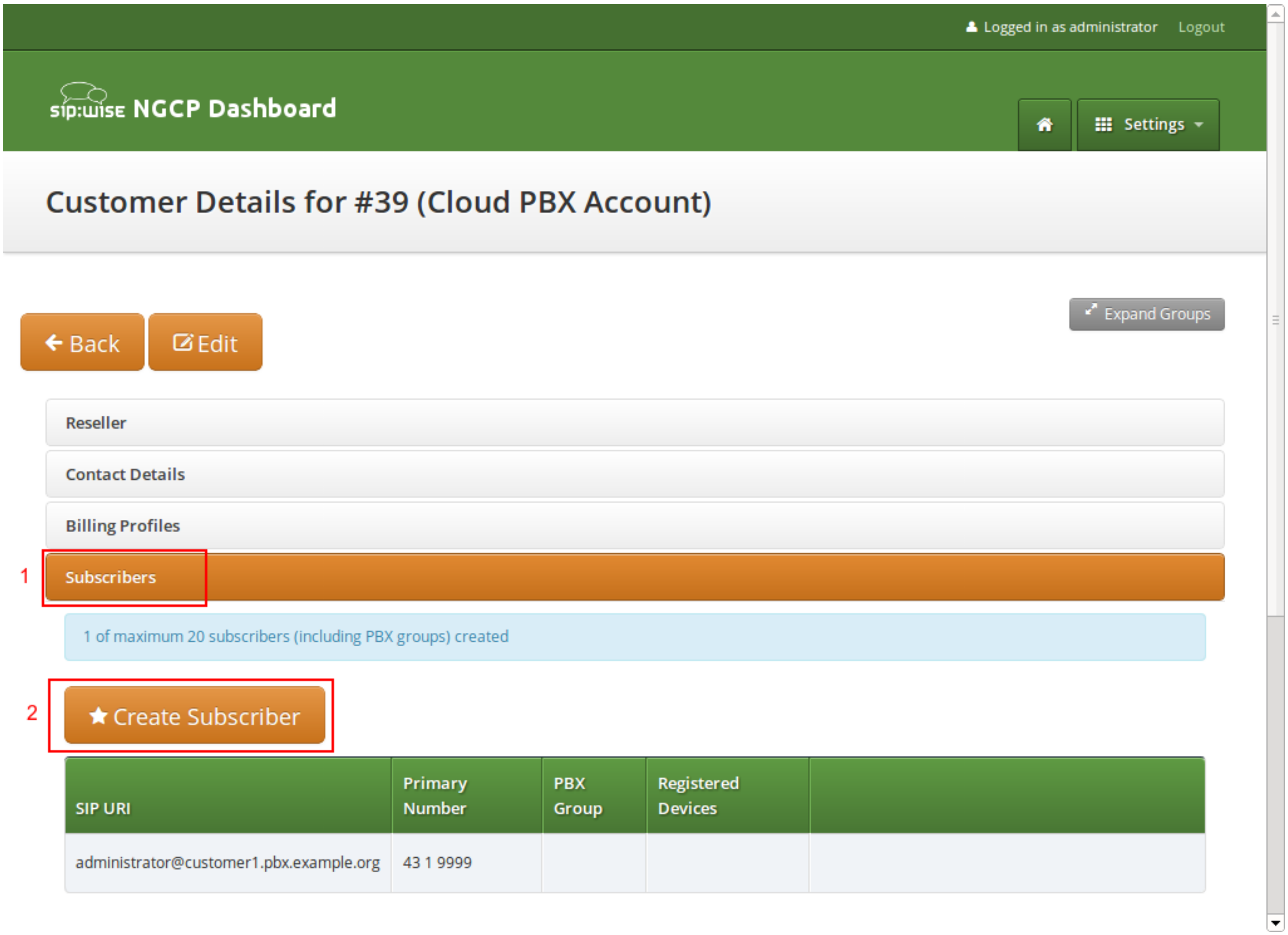


Figure 72: Create a Subscriber Extension

When creating another subscriber in the PBX after having the pilot subscriber, some fields are different now, because the *Domain* and *E.164 Number* are already pre-defined at the pilot subscriber level.

What you need to define for a new subscriber is the *Group* the subscriber is supposed to be in. We don't have a group yet, so create one by clicking *Create Group*.

A *PBX Group* has four settings:

- **Name**: The name of the group. This is used to identify a group when assigning it to subscribers on one hand, and also subscribers are pushed as server side contact lists to XMPP clients, where they are logically placed into their corresponding groups.
- **Extension**: The extension of the group, which is appended to the primary number of the pilot subscriber, so you can actually call the group from the outside. If our pilot subscriber number is `43 1 9999` and the extension is `100`, you can reach the group from the outside by dialing `43 1 9999 100`. Since PBX Groups are actually just normal subscribers in the system, you can assign *Alias Numbers* to it for DID later, e.g. `43 1 9998`.
- **Hunting Policy**: If you call a group, then all members in this group are ringing based on the policy you choose. `Serial`

`Ringing` causes each of the subscribers to be tried one after another, until one of them picks up or all subscribers are tried. `Parallel Ringing` causes all subscribers in the group to be tried in parallel. Note that a subscriber can have a call-forward configured to some external number (e.g. his mobile phone), which will work as well.

- **Serial Hunting Timeout**: This value defines for how long to ring each member of a group in case of serial hunting until the next subscriber is being tried.

We will only fill in the *Name* and *Extension* for now, as the hunting policy can be changed later if needed. Click *Save* to create the group.

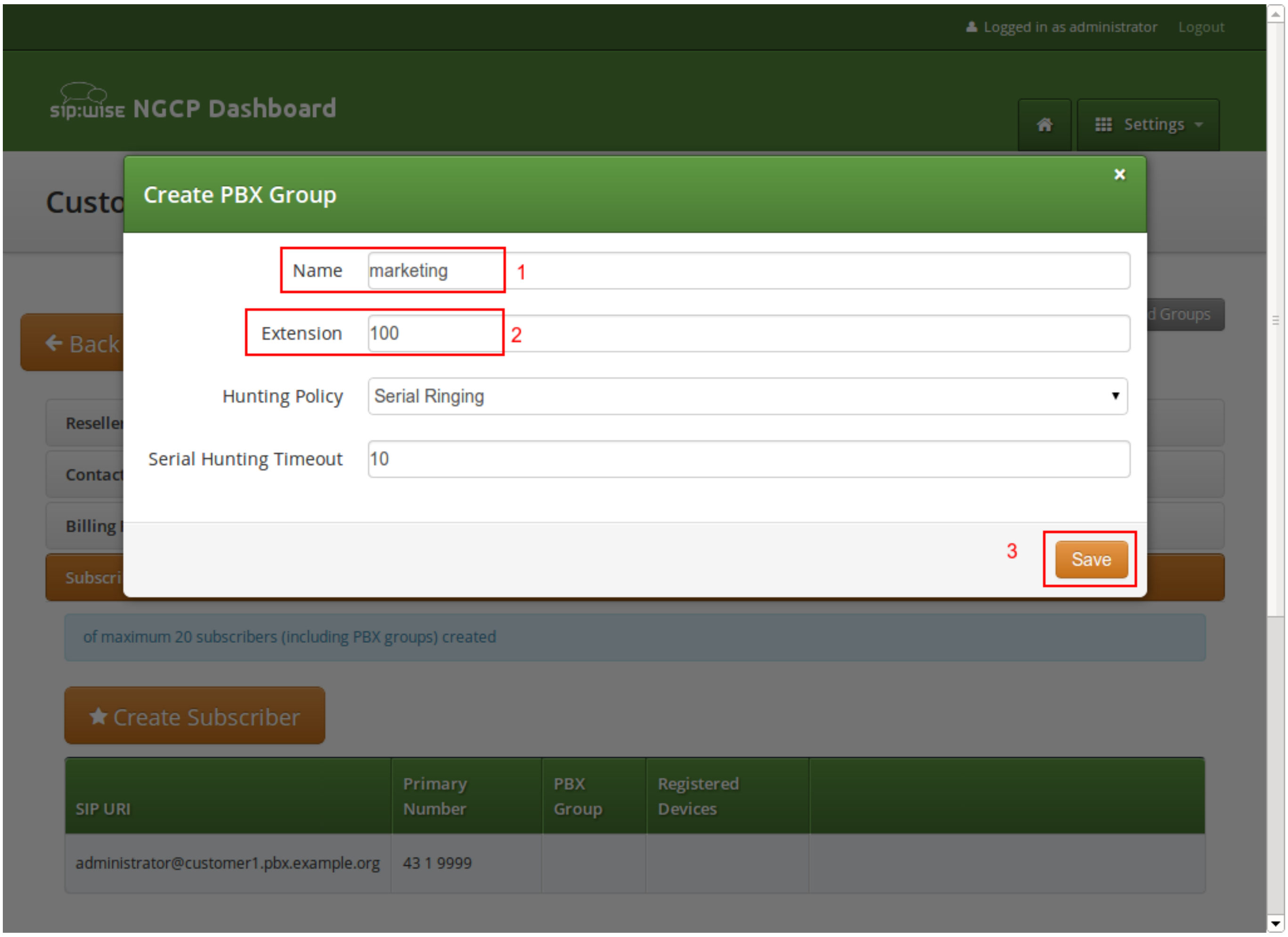


Figure 73: Create a PBX Group

Once the group is created and selected, fill out the rest of the form as needed. Instead of the *E.164 Number*, you can now only choose the *Extension*, which is appended to the primary number of the pilot subscriber and is then used as primary number for this particular subscribers. Again, if your pilot number is `43 1 9999` and you choose extension `101` here, the number of this subscriber is going to be `43 1 9999 101`. Also, you can again later assign more alias numbers (e.g. `43 1 9997`) to this subscriber for DID.

The rest of the fields is as usual, with *Display Name* defining the real name of the user, *Web Username* and *Web Password* allowing the subscriber to log into the customer self-care interface, and the *SIP Username* and *SIP Password* to allow signing into the SIP and XMPP services.

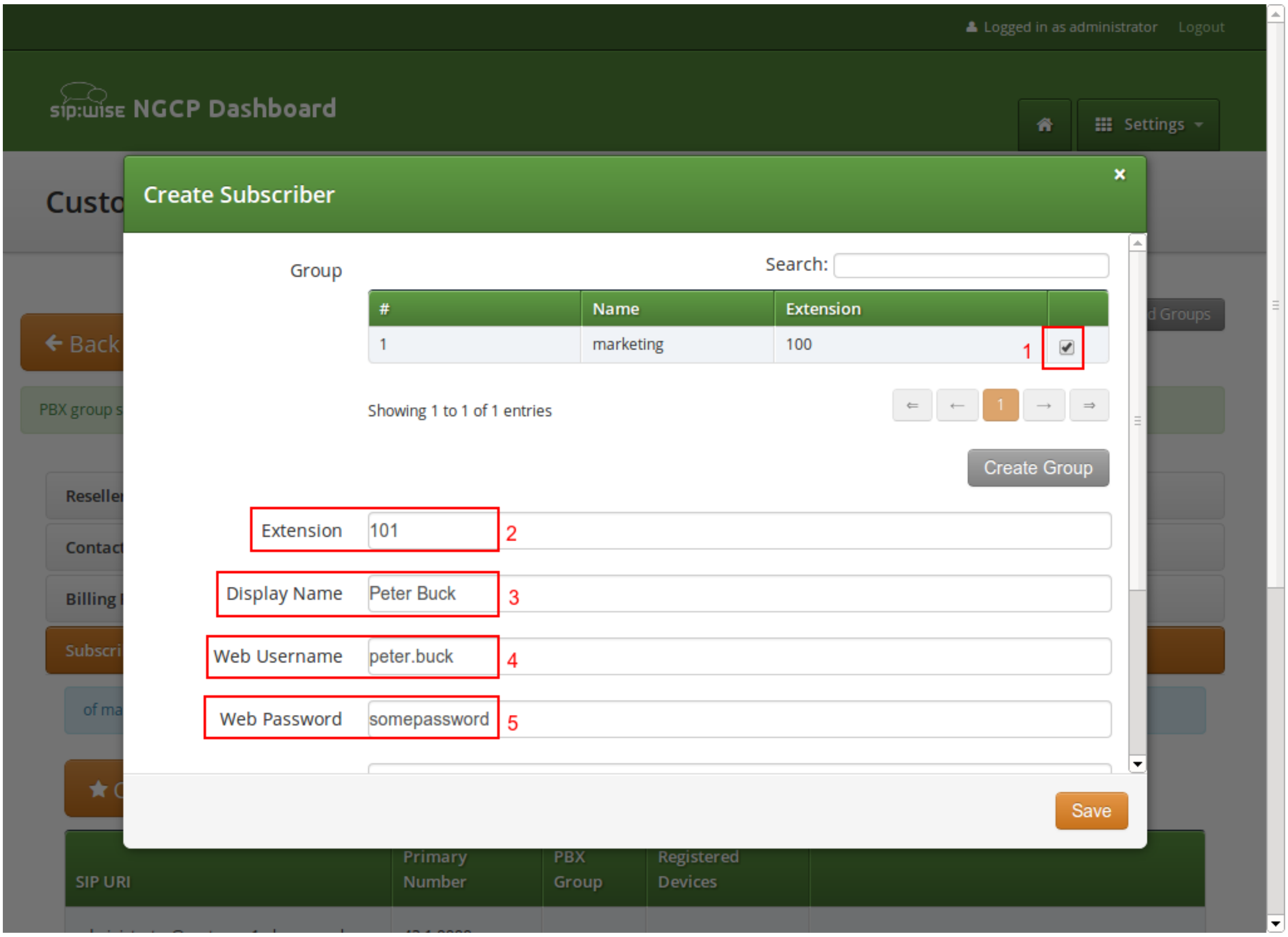


Figure 74: Finish PBX Subscriber Creation Part 1

Click *Save* to create the subscriber.

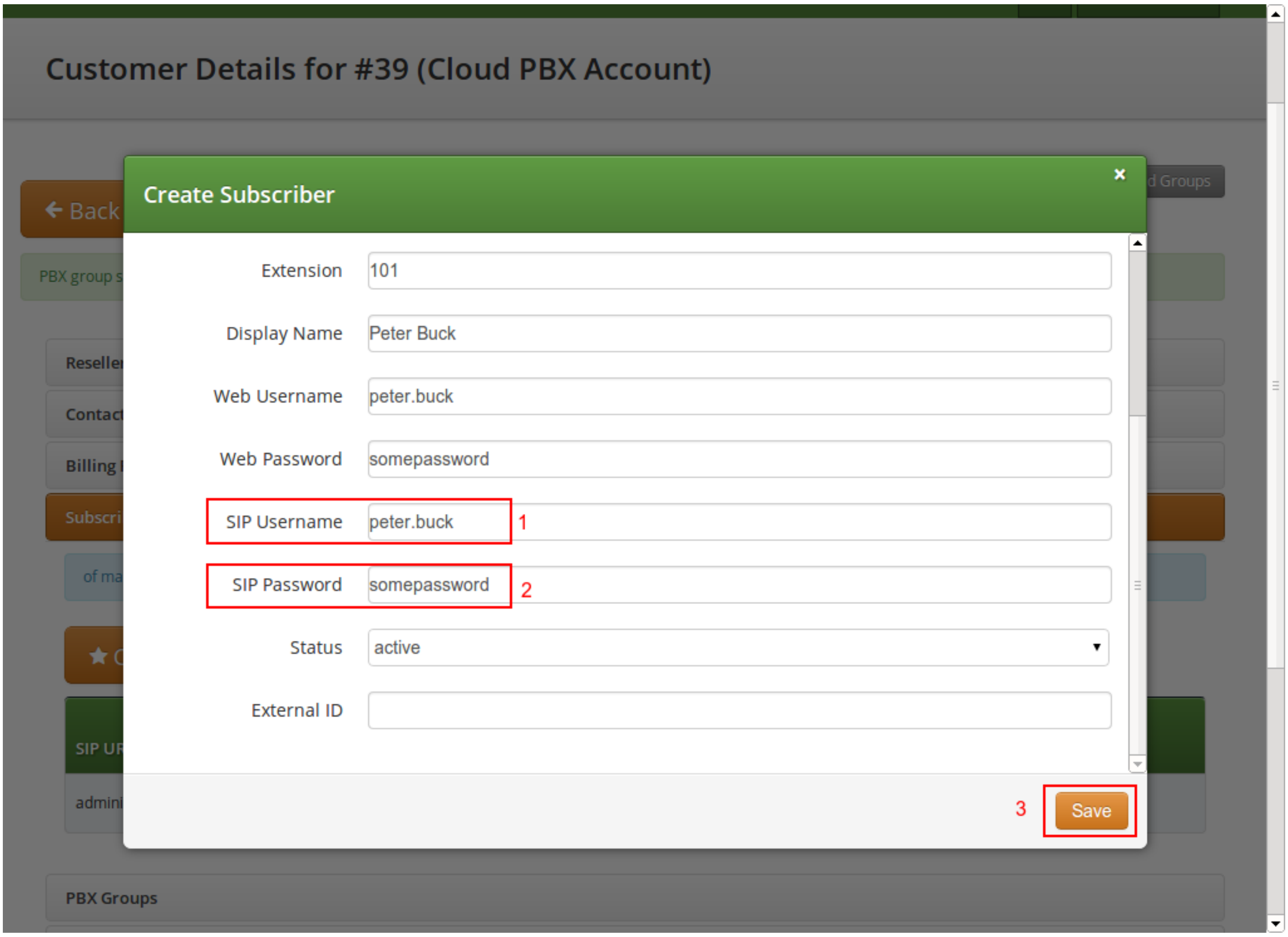


Figure 75: Finish PBX Subscriber Creation Part 2

Repeat the steps to create all the subscribers and groups as needed. An example of a small company configuration in terms of subscribers and groups might look like this:

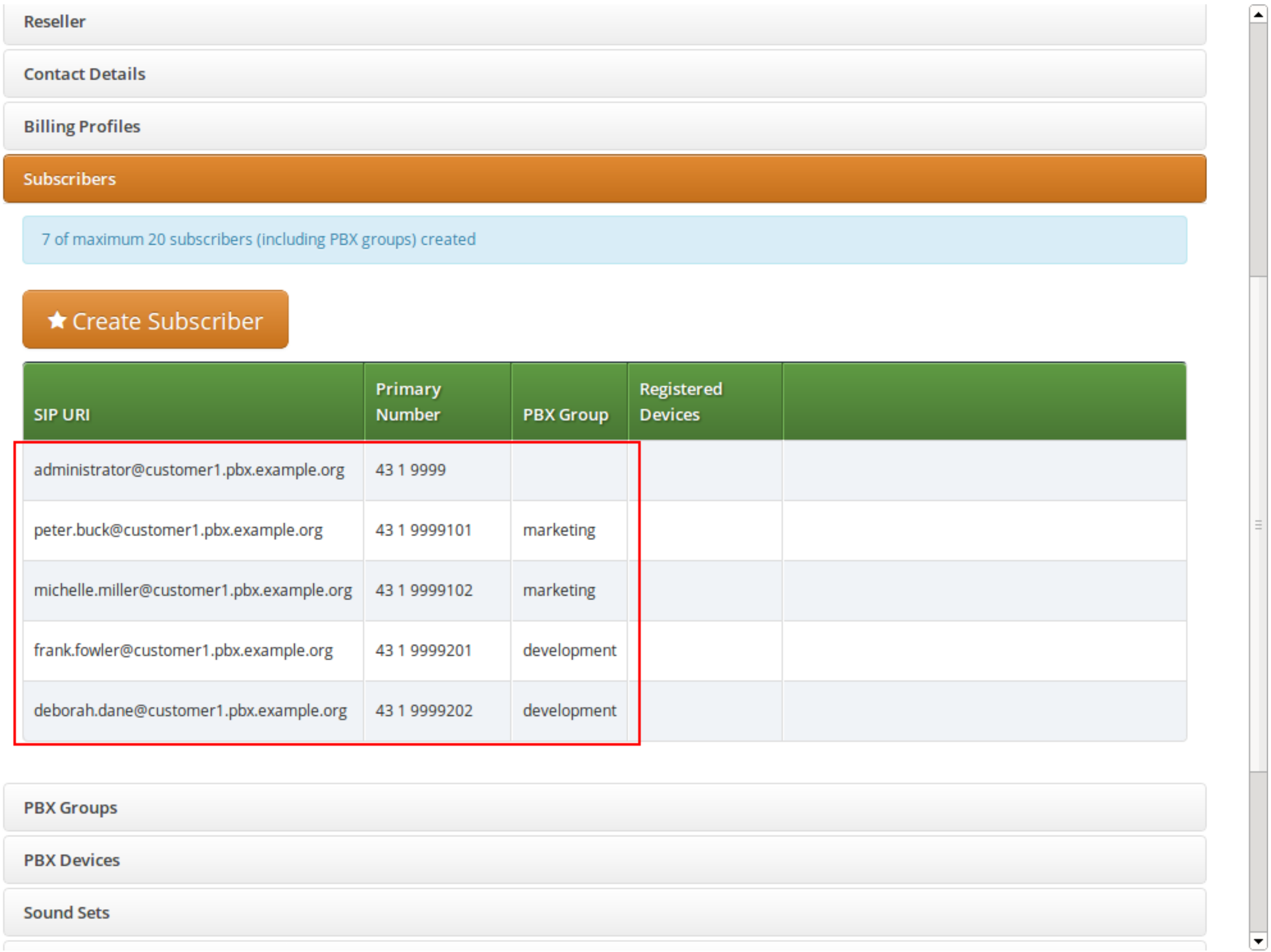


Figure 76: Example of Subscribers List

Tip

The subscribers can be reached via 3 different ways. First, you can call them by their SIP URIs (e.g. by dialing `frank.fowler@customer1.pbx.example.org`) from both inside and outside the PBX. Second, you can dial by the full number (e.g. `43 1 9999 201`; depending on your rewrite rules, you might need to add a leading `\+` or `00` or leave out the country code when dialing from the outside, and adding a `0` as break-out digit when dialing from the inside) from both inside and outside the PBX. Third, you can dial just the extension (e.g. `201`) from inside the PBX. If the subscriber also has an alias number assigned, you can dial that number also, according to your dial-plan in the rewrite rules.

15.1.5 Assigning Subscribers to Devices

Basically you can register any SIP phone to the system using the SIP credentials of your subscribers. However, the platform supports *Device Provisioning* of certain vendors and models, as described in Section 15.1.1.

To configure a physical device, open the *PBX Devices* row in the *Customer Details* view and click *Create Device*.

You have to set three general parameters for your new device, which are:

- **Device Profile**: The actual device profile you want to use. This has been pre-configured in the *Device Management* by the administrator or reseller, and the customer can choose from the list of profiles (which is a combination of an actual device plus its corresponding configuration).
- **MAC Address/Identifier**: The MAC address of the phone to be added. The information can usually either be found on the back of the phone, or in the phone menu itself.
- **Station Name**: Since you can (depending on the actual device) configure more lines on a phone, you can give it a station name, like `Reception` or the name of the owner of the device.

In addition to that information, you can configure the lines (subscribers) you want to use on which key, and the mode of operation (e.g. if it's a normal private phone line, or if you want to monitor another subscriber using BLF, or if you want it to act as shared line using SLA).

For example, a *Cisco SPA504G* has 4 keys you can use for private and shared lines as well as BLF on the phone itself, and in our example we have an *Attendant Console* attached to it as well, so you have 32 more keys for BLF.

The settings per key are as follows:

- **Subscriber**: The subscriber to use (for private/shared lines) or to monitor (for BLF).
- **Line/Key**: The key where to configure this subscriber to.
- **Line/Key Type**: The mode of operation for this key, with the following options (depending on which options are enabled in the *Device Model* configuration for this device:
  - **Private Line**: Use the subscriber as a regular SIP phone line. This means that the phone will register the subscriber, and you can place and receive phone calls with/for this subscriber.
  - **Shared Line**: The subscriber is also registered on the system and you can place and receive calls. If another phone has the same subscriber also configured as shared line, both phones will ring on incoming calls, and you can pick the call up on either of them. You cannot place a call with this subscriber though if the line is already in use by another subscriber. However, you can "steal" a running call by pressing the key where the shared line is configured to barge into a running call. The other party (the other phone where the shared line is configured too) will then be removed from the call (but can steal the call back the same way).
  - **BLF Key**: The *Busy Lamp Field* monitors the call state of another subscriber and provides three different functionalities, depending on the actual state:
    - **Speed Dial**: If the monitored subscriber is on-hook, the user can press the button and directly call the monitored subscriber.
    - **Call Pickup**: If the monitored subscriber is ringing, the user can press the button to pick up the call on his own phone.
    - **State Indication**: It the monitored subscriber is on the phone, the key is indicating that the monitored subscriber is currently busy.

In our example, we will first configure a private line on the first key, and BLF for another subscriber on the second key.

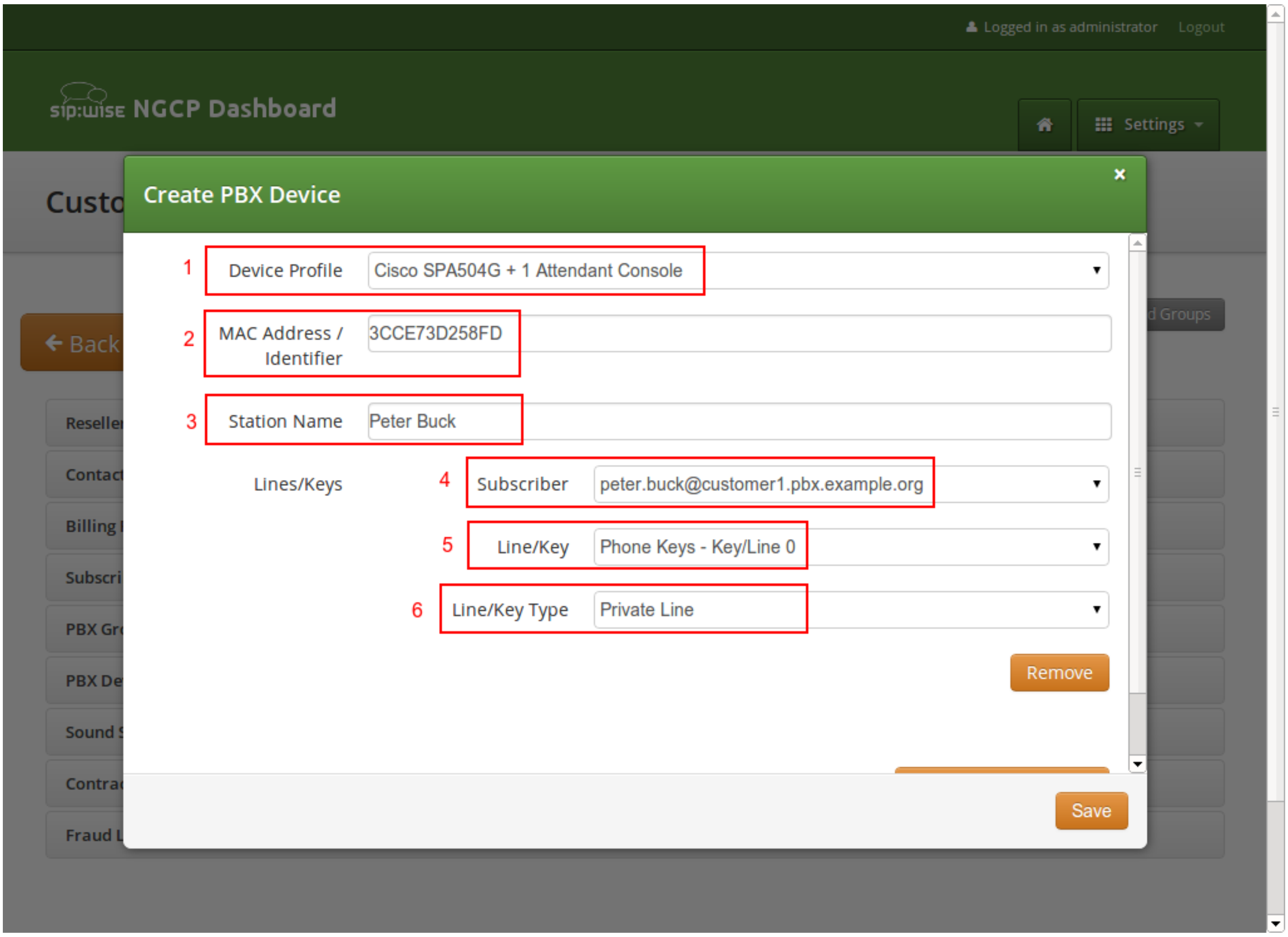


Figure 77: Configuring a PBX Device Part 1

This configures the general options plus the first key. To configure the second key, click *Add another Line/Key* and fill out the second line config accordingly. Click *Save* to save your PBX device configuration.

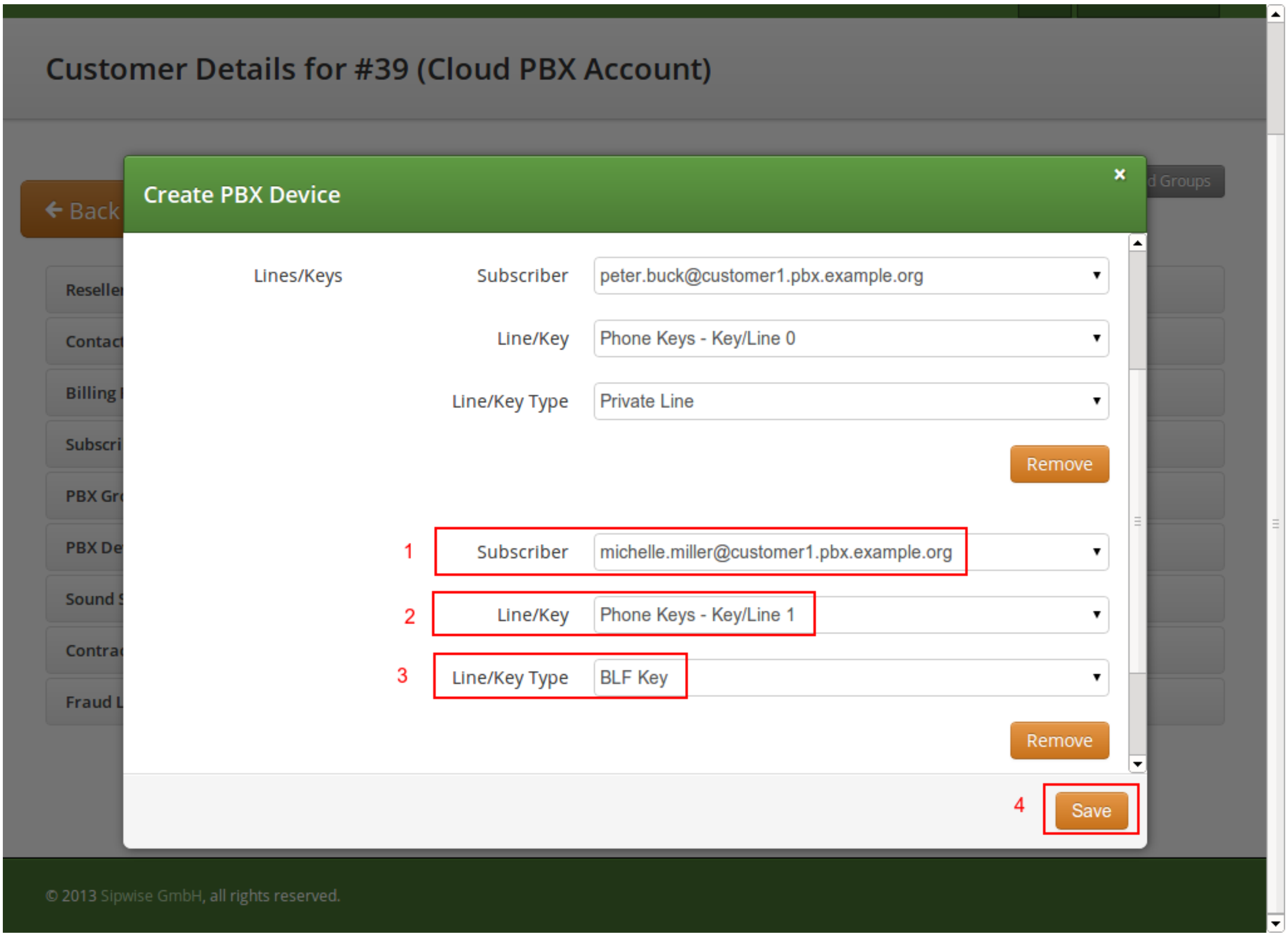


Figure 78: Configuring a PBX Device Part 2

Once the PBX device is saved, you will see it in the list of *PBX Devices*.

15.1.5.1 Synchronizing a PBX Device for Initial Usage

Since a stock device obtained from an arbitrary distributor doesn't know anything about your system, it can't fetch its configuration from there. For that to work, you need to push the URL of where the phone can get the configuration to the phone once.

In order to do so, click the *Sync Device* button on the device you want to configure for the very first time.

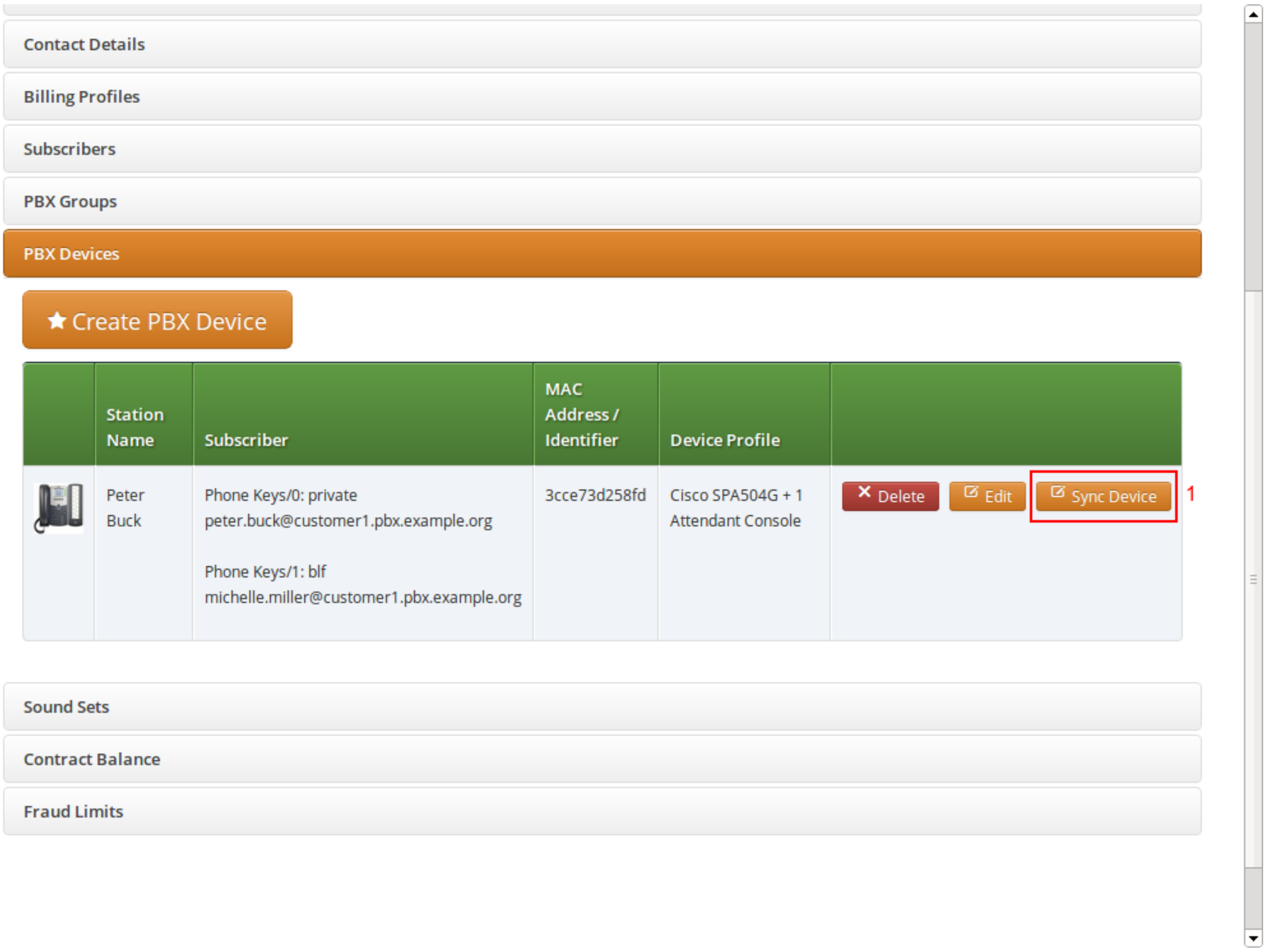


Figure 79: Go to Sync Device

Important

As you will see in the next step, you need the actual IP address of the phone to push the provisioning URL onto it. That implies that you need access to the phone to get the IP, and that your browser is in the same network as the phone in order to be able to connect to it, in case the phone is behind NAT.

Enter the IP Address of the phone (on Cisco SPAs, press `Settings 9`, where `Settings` is the paper sheet symbol, and note down the `Current IP` setting), then click *Push Provisioning URL*.

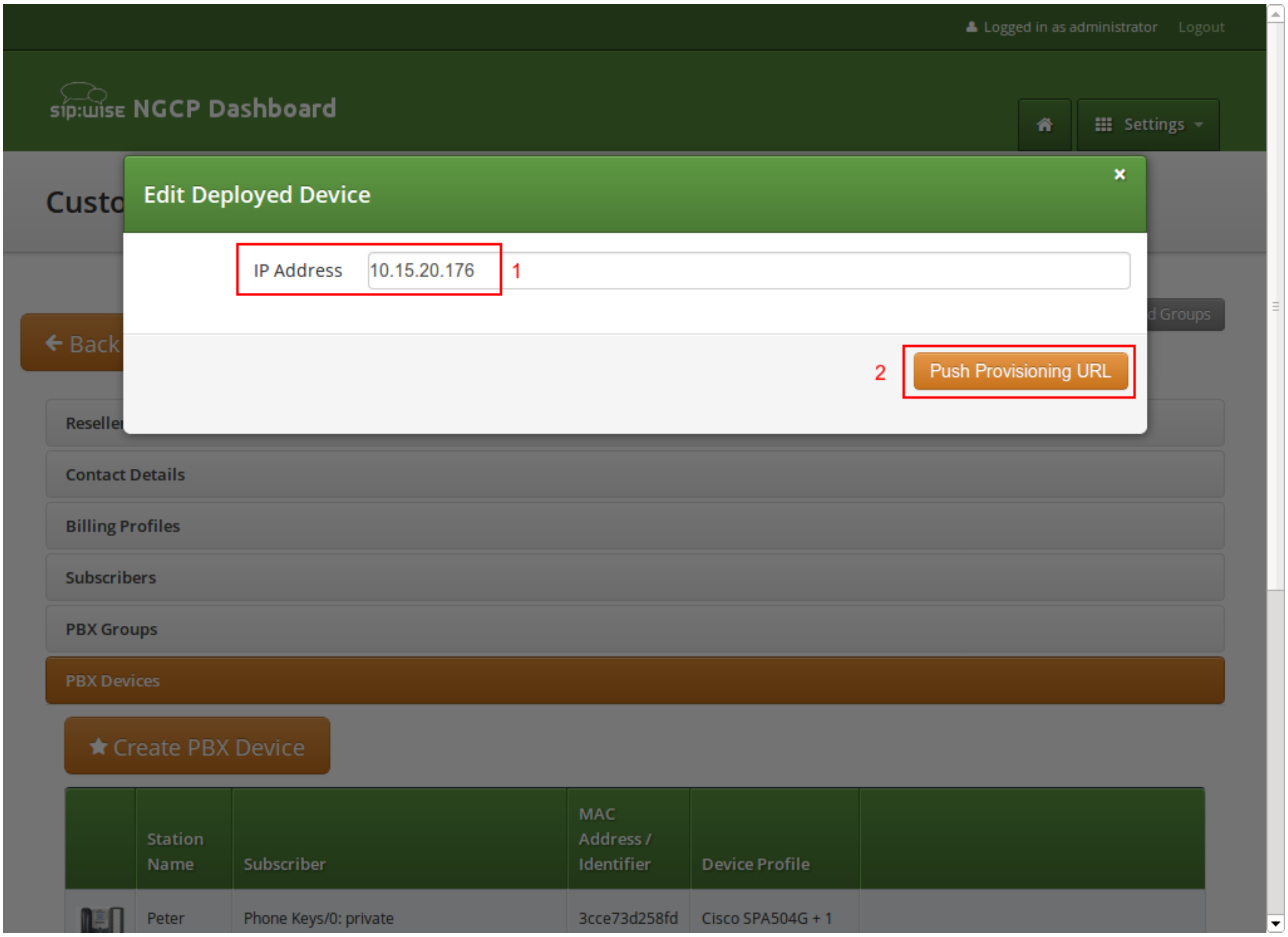


Figure 80: Sync Device

You will be redirected directly to the phone, and the Provisioning URL is automatically set. If everything goes right, you will see a confirmation page from the phone that it's going to reboot.

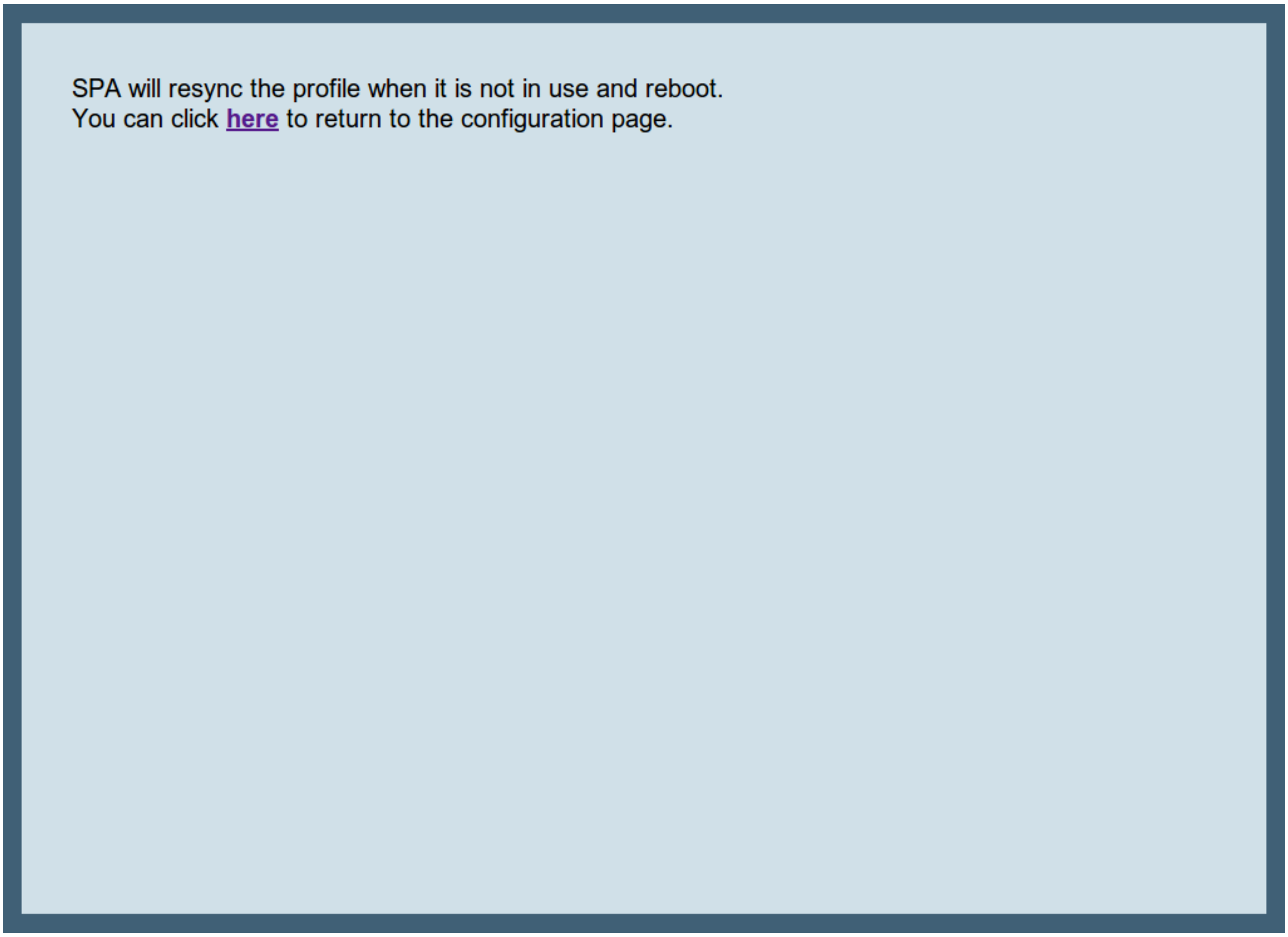


Figure 81: Device Sync Confirmation from Phone

You can close the browser window/tab and proceed to sync the next subscriber.

Tip

You only have to do this step once per phone to tell it the actual provisioning URL, where it can fetch the configuration from. From there, it will regularly sync with the server automatically to check for configuration changes, and applies them automatically.

15.1.6 Configuring Sound Sets for the Customer PBX

In the *Customer Details* view, there is a row *Sound Sets*, where the customer can define his own sound sets for *Auto Attendant*, *Music on Hold* and the *Office Hours Announcement*.

To create a new sound set, open the *Sound Sets* row and click *Create Sound Set*.

If you do this as administrator or reseller, the Reseller and/or Customer is pre-selected, so keep it as is. If you do this as customer, you don't see any *Reseller* or *Customer* fields.

So the important settings are:

- **Name**: The name of the sound set as it will appear in the *Subscriber Preferences*, where you can assign the sound set to a subscriber.
- **Description**: A more detailed description of the sound set.
- **Default for Subscribers**: If this setting is enabled, then the sound set is automatically assigned to all already existing subscribers which do NOT have a sound set assigned yet, and also for all newly created subscribers.

Fill in the settings and click *Save*.

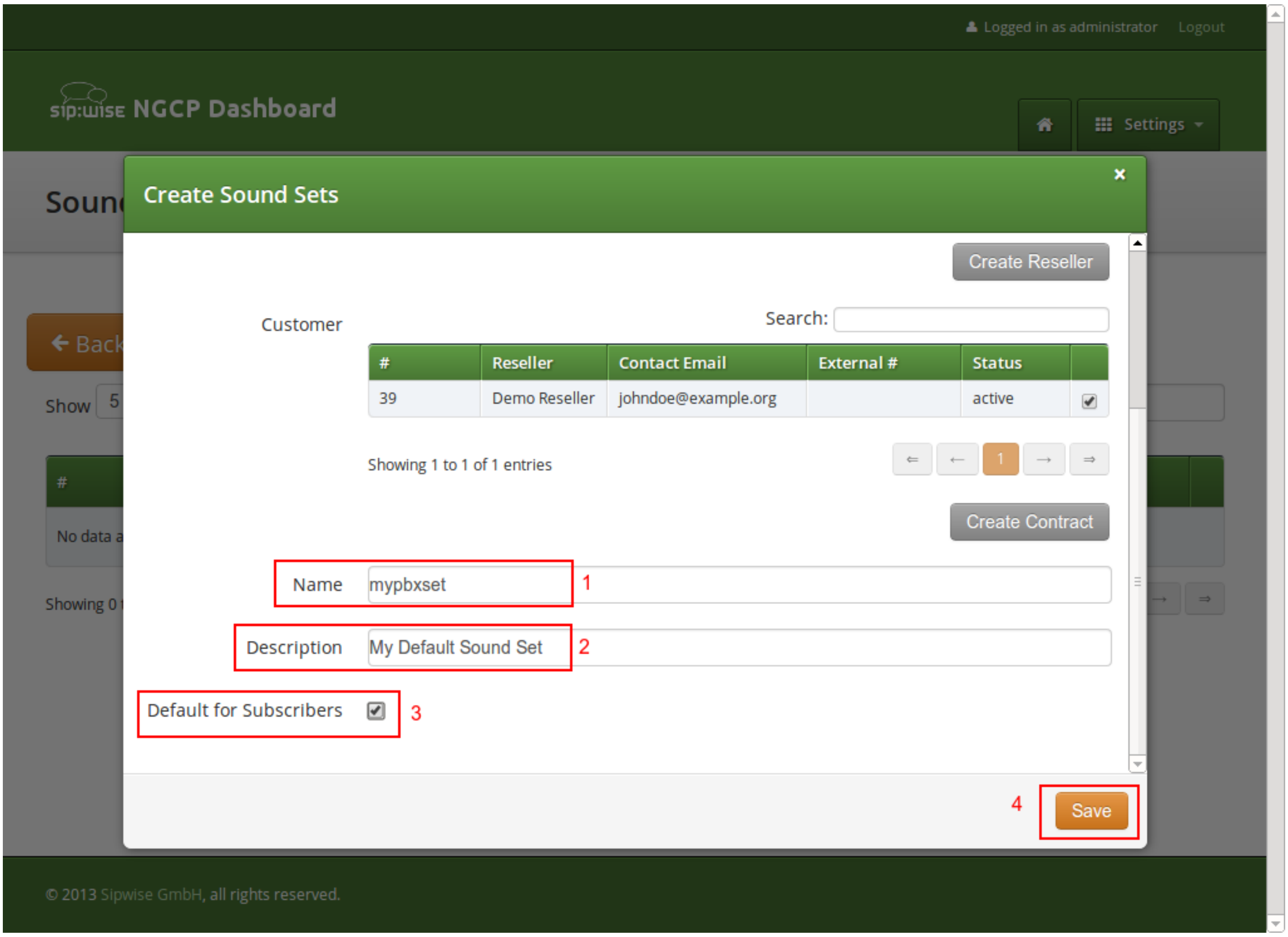


Figure 82: Create Customer Sound Set

To upload files to your Sound Set, click the *Files* button for the Sound Set.

15.1.6.1 Uploading a Music-on-Hold File

Open the *music_on_hold* row and click *Upload* on the *music_on_hold* entry. 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.

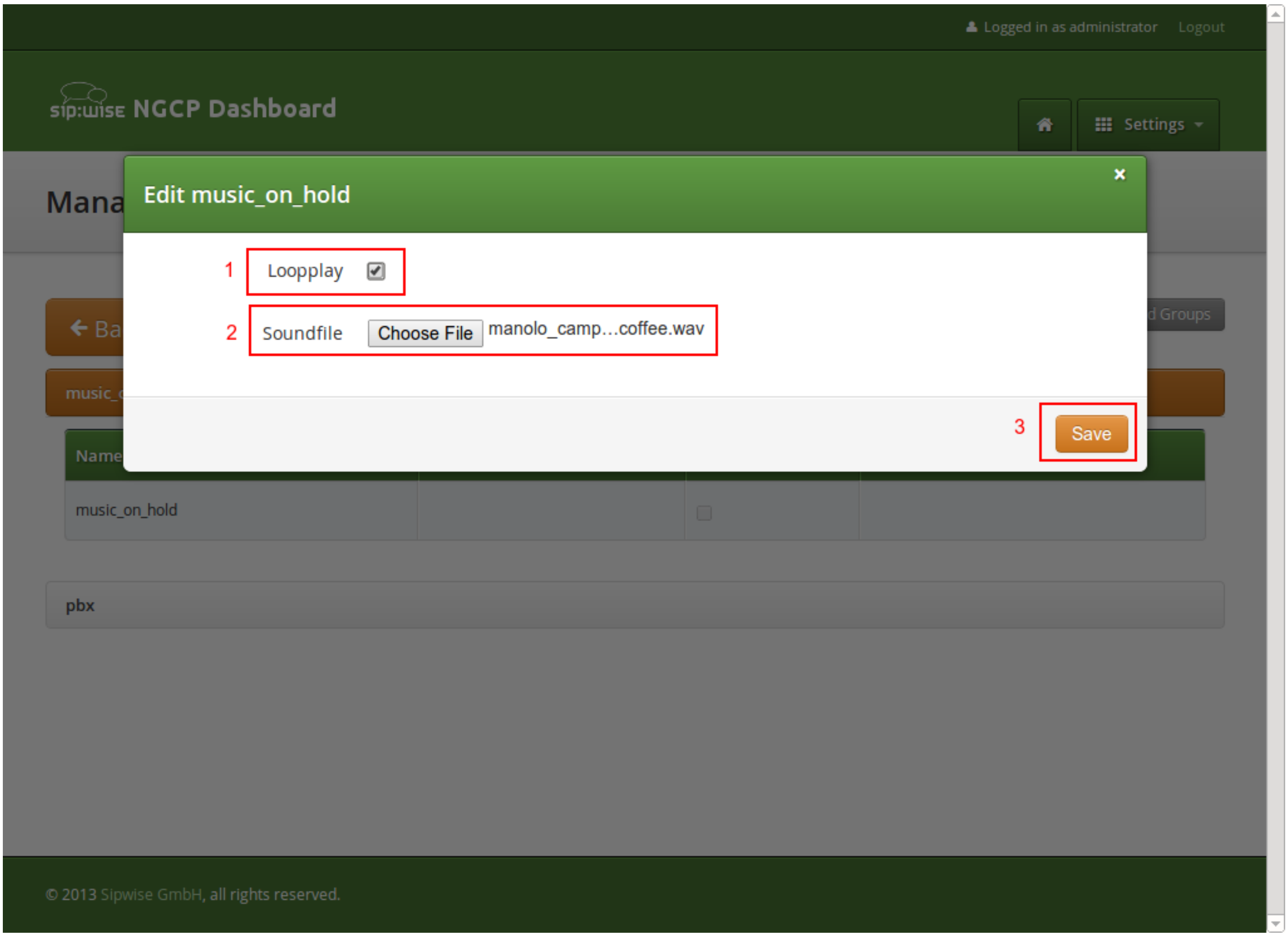


Figure 83: Upload MoH Sound File

15.1.7 Configuring Auto Attendant

The *Auto Attendant* is a built-in IVR feature that is available to Cloud PBX subscribers. It provides an automatic voice menu that enables the caller to select from a number of destinations — which are other PBX subscribers or groups — when the called subscriber is not available.

The Auto Attendant feature can be activated for any subscriber in the Customer PBX individually. There are three steps involved:

1. You have to prepare a *Sound Set* to have Auto Attendant sound files.
2. You have to configure the destinations for the various options you provide (e.g. pressing `1` should go to the `marketing` subscriber, `2` to `development` and `3` to some external number).
3. You have to set a Call Forward to the Auto Attendant.

To do so, go to *Customer Details* and in the *Subscribers* section, click the *Preferences* button of the subscriber, where the Auto Attendant should be set.

15.1.7.1 Preparing the Sound Set

Create a Sound Set and upload the Sound Files for it as described below. Afterwards in the *Subscriber Preferences* view, set the *Customer Sound Set* preference to the Sound Set to be used. To do so, click *Edit* on the *Customer Sound Set* preference and assign the set to be used.

Uploading Auto-Attendant Sound Files

When configuring a Call Forward to the *Auto Attendant*, it will play the following files:

- `aa_welcome`: This is the welcome message (the greeting) which is played when someone calls the Auto Attendant.
- each available pair of `aa_X_for`/`aa_X_option`: Each menu item in the Auto Attendant consists of two parts. The `for` part, which plays something like *Press One for*, and the `option` part, which play something like *Marketing*. The Auto Attendant only plays those menu options where both the `for` part and the `option` part is present, so if you only have 3 destinations you'd like to offer, and you want them to be on keys 1, 2 and 3, you have to upload files for `aa_1_for`, `aa_1_option`, `aa_2_for`, `aa_2_option` and `aa_3_for` and `aa_3_option`.

Important

The sound files only define the general structure of what is being played to the caller. The actual destinations behind your options are configured separately in Configuring the Auto Attendant Slots Section 15.1.7.3.

An example configuration could look like this:

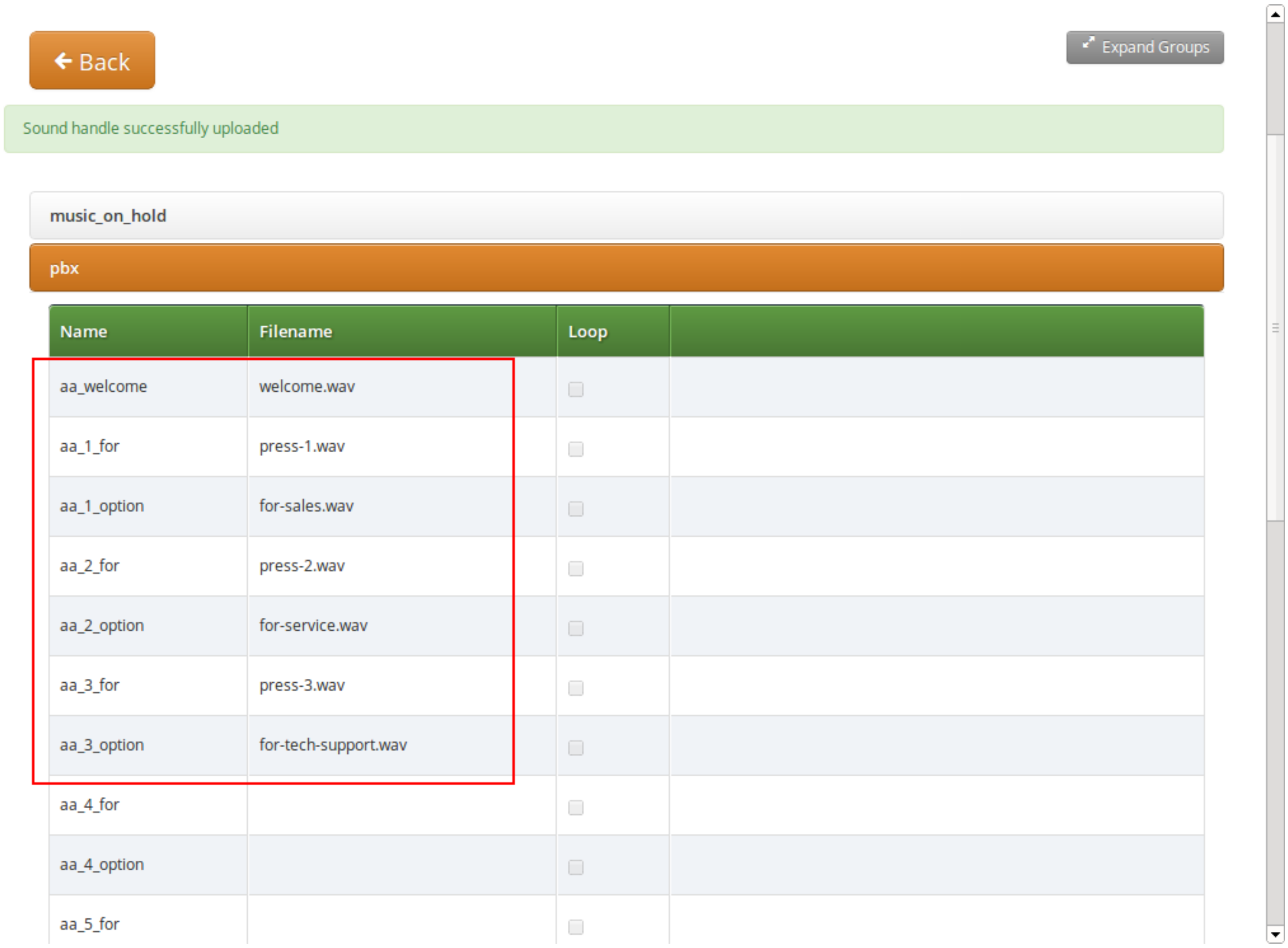


Figure 84: Upload Auto Attendant Sound File

15.1.7.2 Auto Attendant Flowchart with Voice Prompts

The illustration below shows the sequence of voice prompts played when Auto Attendant feature is activated and a caller listens the IVR menu.

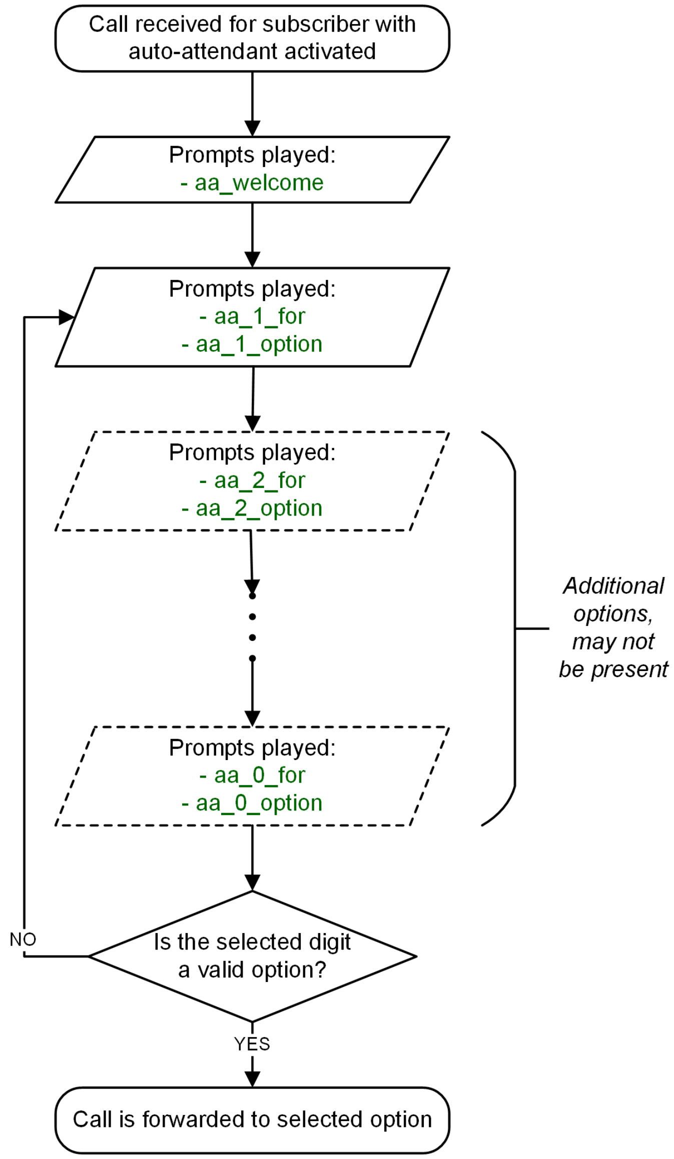


Figure 85: Flowchart of Auto Attendant

15.1.7.3 Configuring the Auto Attendant Slots

In the *Auto Attendant Slots* section, click the *Edit Slots* button to configure the destination options.

Click *Add another Slot* to add a destination option, select the Key the destination should be assigned to, and enter a Destination. The destination can be a subscriber username (e.g. `marketing`), a full SIP URI (e.g. `sip:michelle.miller@customer1.pbx.example.org` or any external SIP URI) or a number or extension (e.g. `491234567` or `101`).

Repeat the step for every option you want to add, then press *Save*.

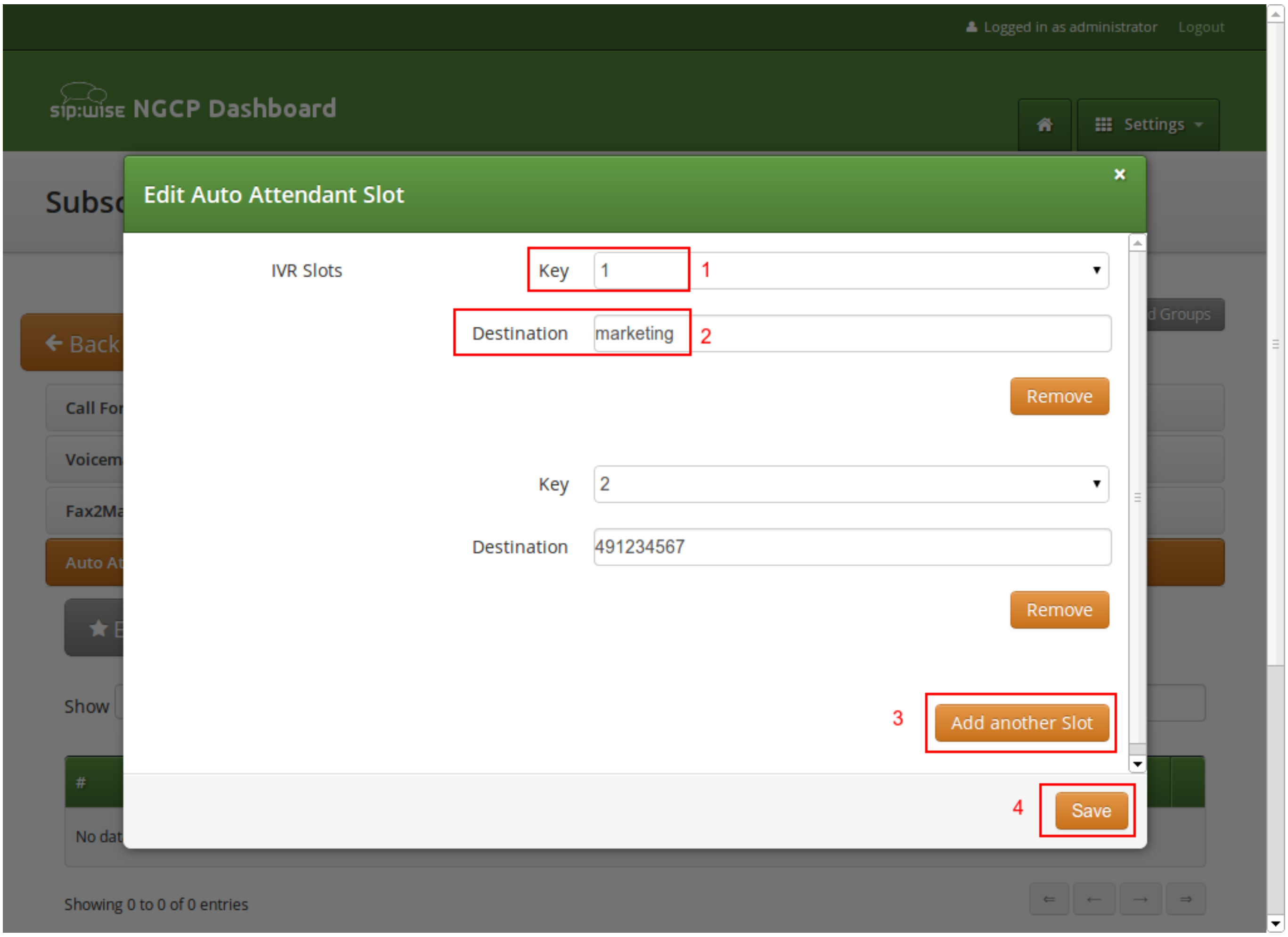


Figure 86: Define the Auto Attendant Slots

15.1.7.4 Activating the Auto Attendant

Once the Sound Set and the Slots are configured, activate the Auto Attendant by setting a Call Forward to Auto Attendant.

To do so, open the *Call Forwards* section in the *Subscriber Preferences* view and press *Edit* on the Call Forward type (e.g. *Call Forward Unconditional* if you want to redirect callers unconditionally to the Auto Attendant).

Select *Auto Attendant* and click *Save* to activate the Auto Attendant.

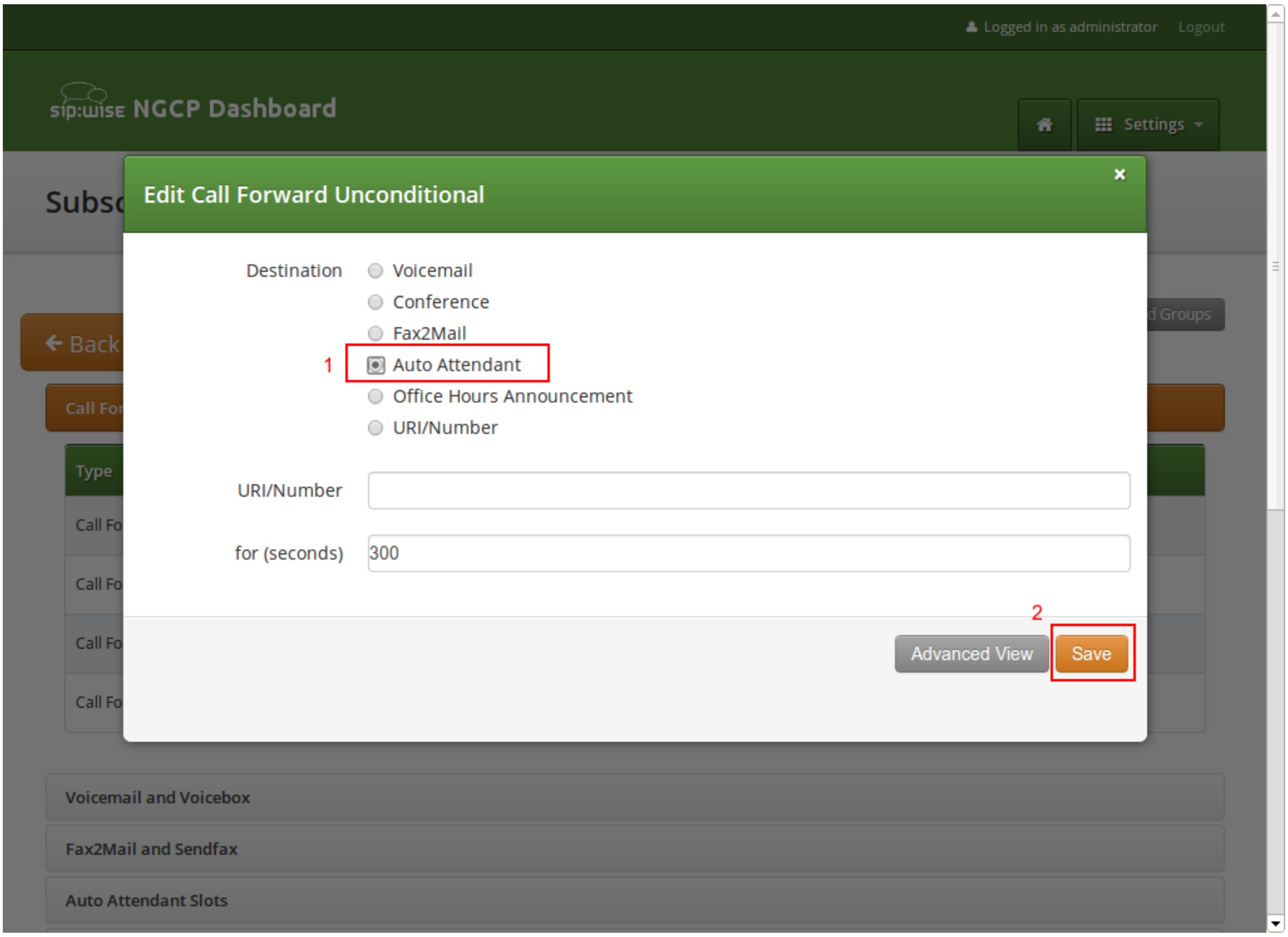


Figure 87: Set a Call Forward to Auto Attendant

Tip

As with any other Call Forward, you can define more complex forwarding rules in the *Advanced View* to only forward the call to the Auto Attendant during specific time periods, or as a fallback if no one picks up the office number.

15.1.8 Configuring Call Queues

The sip:provider PRO platform offers call queueing feature for Cloud PBX subscribers. For any subscriber within the PBX the NGCP system administrator or the subscriber himsef may activate the *Call Queue*. This is done individually for each subscriber on demand.

If call queue activation has been done and the subscriber receives more than 1 call at a time, then the second and all further callers will be queued until the subscriber finishes his call with the first caller and gets free.

15.1.8.1 Activating the Call Queue

The call queue configuration is available at the path: *Subscribers → select one → Details → Preferences → Cloud PBX*.

Following configuration parameters may be set for call queueing:

- `cloud_pbx_callqueue` : shows the status of call queueing (enabled / disabled); by default it is disabled
- `max_queue_length` : the length of call queue, i.e. the maximum number of callers in a queue; the default is 5
- `queue_wrap_up_time` : the delay in seconds between the ending of the previous call and the connection of the next queued caller with the subscriber; the default is 10

In order to change the actual setting, press the *Edit* button in the relevant row.

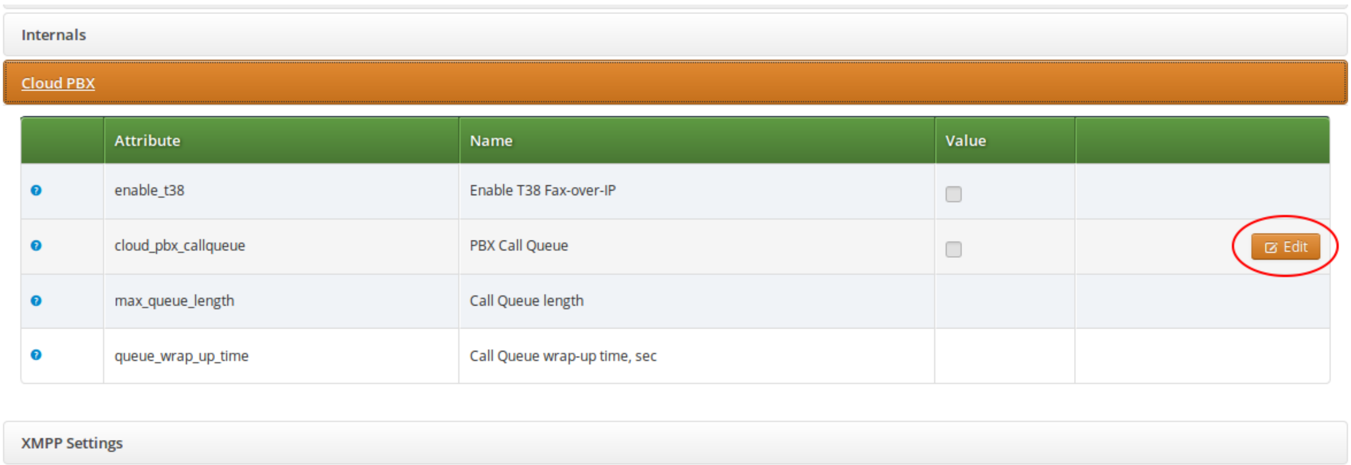


Figure 88: Call Queue Configuration

15.1.8.2 Call Queue Voice Prompts

Queued callers first hear a greeting message then information about their position in the queue and finally a waiting music / signal.

Table 19: Call Queue Voice Prompts

Prompt handle	Prompt content
`queue_greeting`	All lines are busy at the moment, you are being queued.
`queue_prefix`	You are currently number. . .
`queue_suffix`	. . . in the queue, please hold the line.
`queue_full`	All lines are busy at the moment, please try again later.
`queue_waiting_music`	<waiting music>

15.1.8.3 Call Queue Flowchart with Voice Prompts

The following illustration shows which voice prompts are played to the caller when the call gets into a queue.

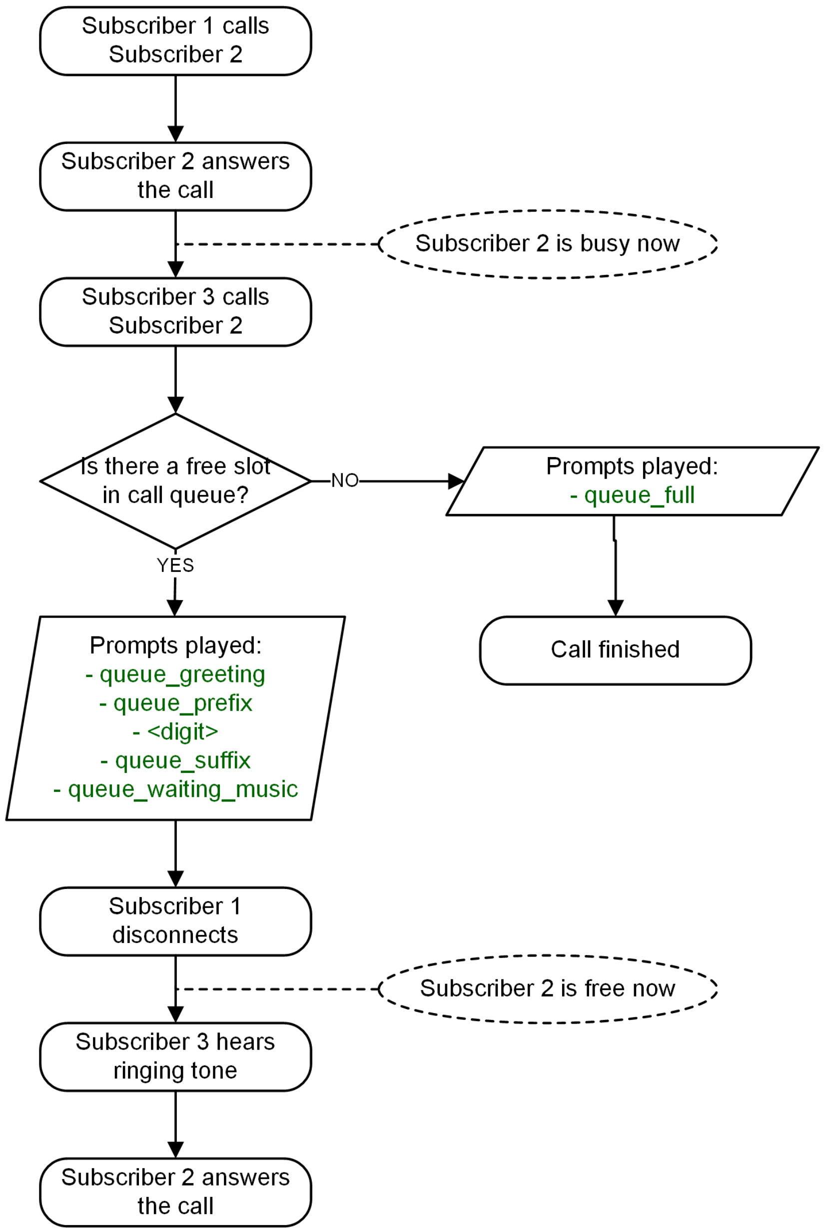


Figure 89: Flowchart of Call Queue

15.1.9 Device Auto-Provisioning Security

15.1.9.1 Server Certificate Authentication

The Cisco SPA phones can connect to the provisioning interface of the PBX via HTTP and HTTPS. When perform secure provisioning over HTTPS, the phones validate the server certificate to check if its a legitimate Cisco provisioning server. To pass this check, the provisioning interface must provide a certificate signed by Cisco for that exact purpose.

The following steps describe how to obtain such a certificate.

First, a new SSL key needs to be generated:

```
$ openssl genrsa -out provisioning.key 2048
Generating RSA private key, 2048 bit long modulus
...+++
................................................................+++
e is 65537 (0x10001)
```

Next, a certificate signing request needs to be generated as follows. Provide your company details.

> **Important**
> The **Common Name (e.g. server FQDN or YOUR name)** field is crucial here. Provide an FQDN which the phones will later use via DNS to connect to the provisioning interface, for example *pbx.example.org*. Cisco does **NOT** support wild-card certificates.

> **Important**
> Leave the password empty when asked for it (press Enter without entering anything).

```
$ openssl req -new -key provisioning.key -out provisioning.csr
You are about to be asked to enter information that will be incorporated
into your certificate request.
What you are about to enter is what is called a Distinguished Name or a DN.
There are quite a few fields but you can leave some blank
For some fields there will be a default value,
If you enter '.', the field will be left blank.

Country Name (2 letter code) [AU]:AT
State or Province Name (full name) [Some-State]:Vienna
Locality Name (eg, city) []:Vienna
Organization Name (eg, company) [Internet Widgits Pty Ltd]:Sipwise GmbH
Organizational Unit Name (eg, section) []:Operations
Common Name (e.g. server FQDN or YOUR name) []:pbx.example.org
Email Address []:office@sipwise.com

Please enter the following 'extra' attributes
to be sent with your certificate request
A challenge password []:
An optional company name []:
```

Finally, compress the `provisioning.csr` file via ZIP and send it to our Cisco sales representative. If in doubt, you can try to send it directly to `ciscosb-certadmin@cisco.com` asking them to sign it.

Important
Only send the CSR file. **Do NOT send the key file, as this is your private key!**

Important
Ask for both the signed certificate AND a so-called *combinedca.crt* which is needed to perform client authentication via SSL. Otherwise you can not restrict access to Cisco SPAs only.

You will receive a signed CRT file, which Sipwise can use to configure the PBX provisioning interface.

15.1.9.2 Client Certificate Authentication

If a client connects via HTTPS, the server also checks for the client certificate in order to validate that the device requesting the configuration is indeed a legitimate Cisco phone, and not a fraudulent user with a browser trying to fetch user credentials.

15.1.10 Device Bootstrap and Resync Workflows

The IP phones supported by the PBX need to initially be configured to fetch their configuration from the system. Since the phones have no initial information about the system and its provisioning URL, they need to be boot-strapped. Furthermore, changes for a specific device might have to be pushed to the device immediately instead of waiting for it to re-fetch the configuration automatically.

The following sections describe the work-flows how this is accomplished without having the customer directly accessing the phone.

15.1.10.1 Cisco SPA Device Bootstrap

Initial Bootstrapping

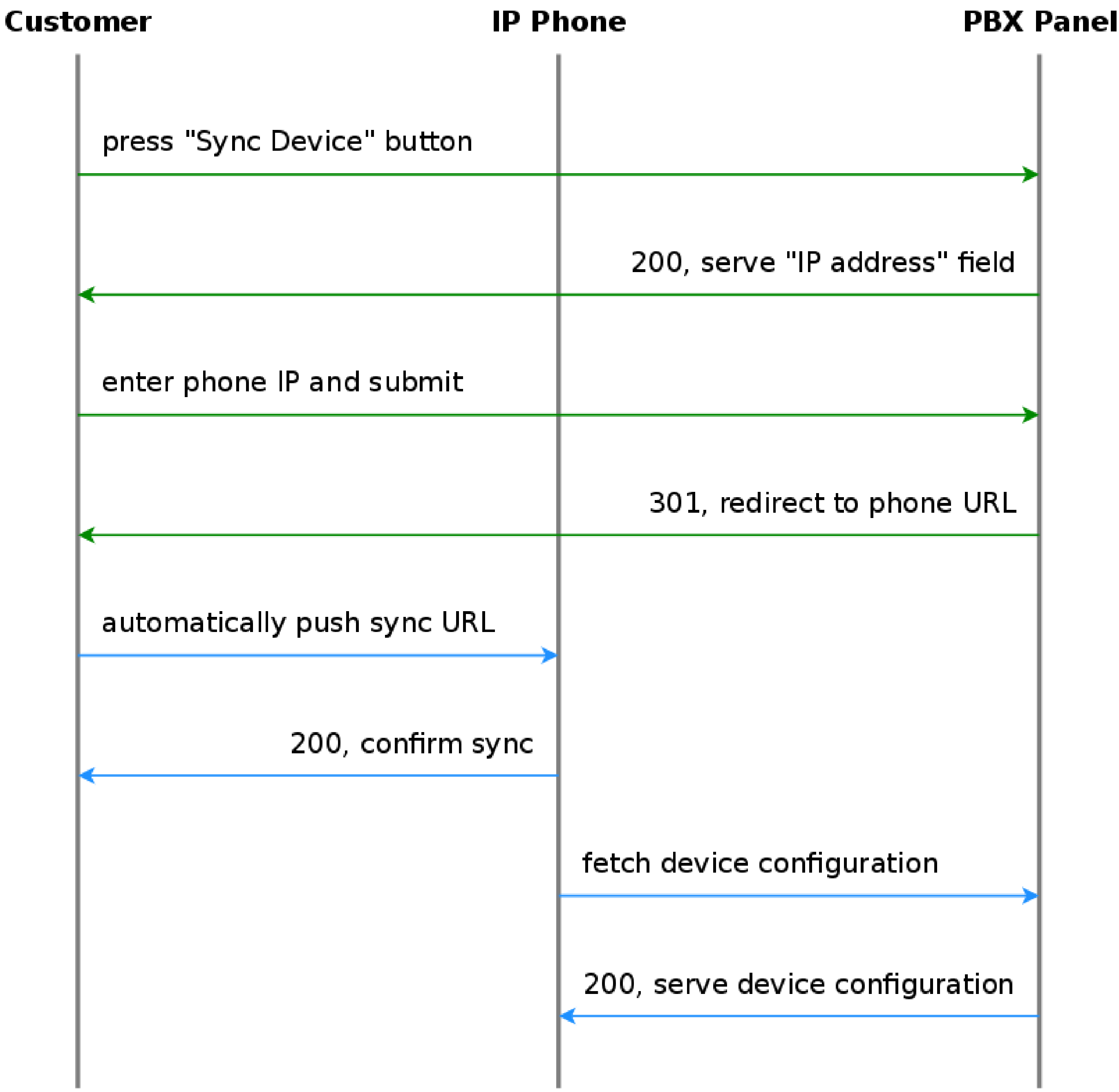


Figure 90: Initially bootstrap a PBX device

Subsequent Device Resyncs

If one of the subscribers configured on a PBX device is registered via SIP, the system can trigger a re-sync of the phone directly via SIP without having the customer enter the IP of the phone again. This is accomplished by sending a special `NOTIFY` message to the subscriber:

```
NOTIFY sip:subscriber@domain SIP/2.0
To: <sip:subscriber@domain>
From: <sip:subscriber@domain>;tag=some-random-tag
```

```
Call-ID: some-random-call-id
CSeq: 1 NOTIFY
Subscription-State: active
Event: check-sync
Content-Length: 0
```

In order to prevent unauthorized re-syncs, the IP phone challenges the request with its own SIP credentials, so the NOTIFY is sent twice, once without authentication, and the second time with the subscriber's own SIP credentials.

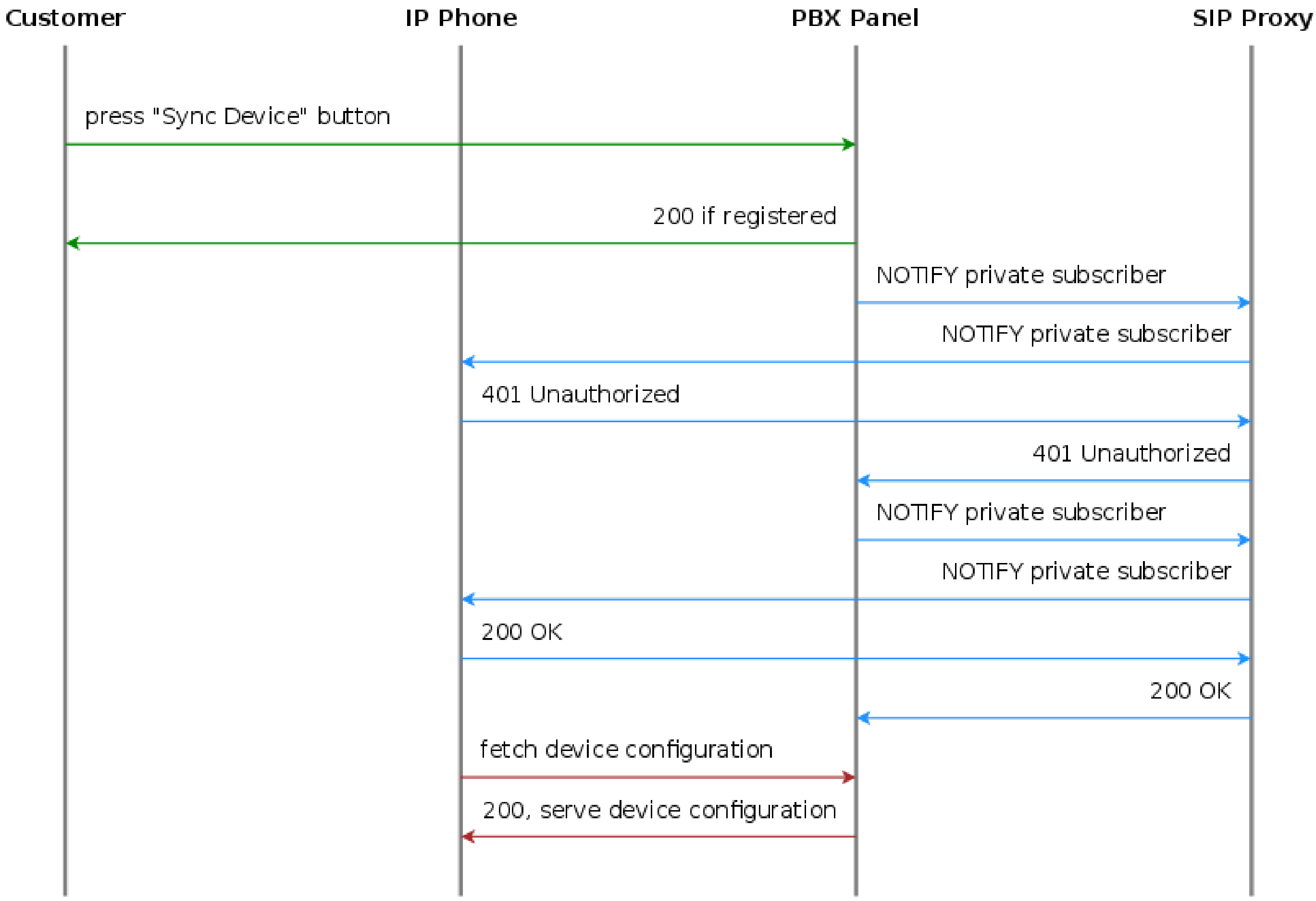


Figure 91: Resync a registered PBX device

15.1.10.2 Panasonic Device Bootstrap

Initial Bootstrapping

Panasonic provides a zero-touch provisioning mechanism in their firmwares, which causes the factory-reset phones to connect to a Panasonic web service at *https://provisioning.e-connecting.net* to check if a custom provisioning URL is configured for the MAC address of the phone. If an association between the MAC and a provisioning URL is found, the web service redirects the phone to the provisioning URL, where the phone connects to in order to obtain the configuration file.

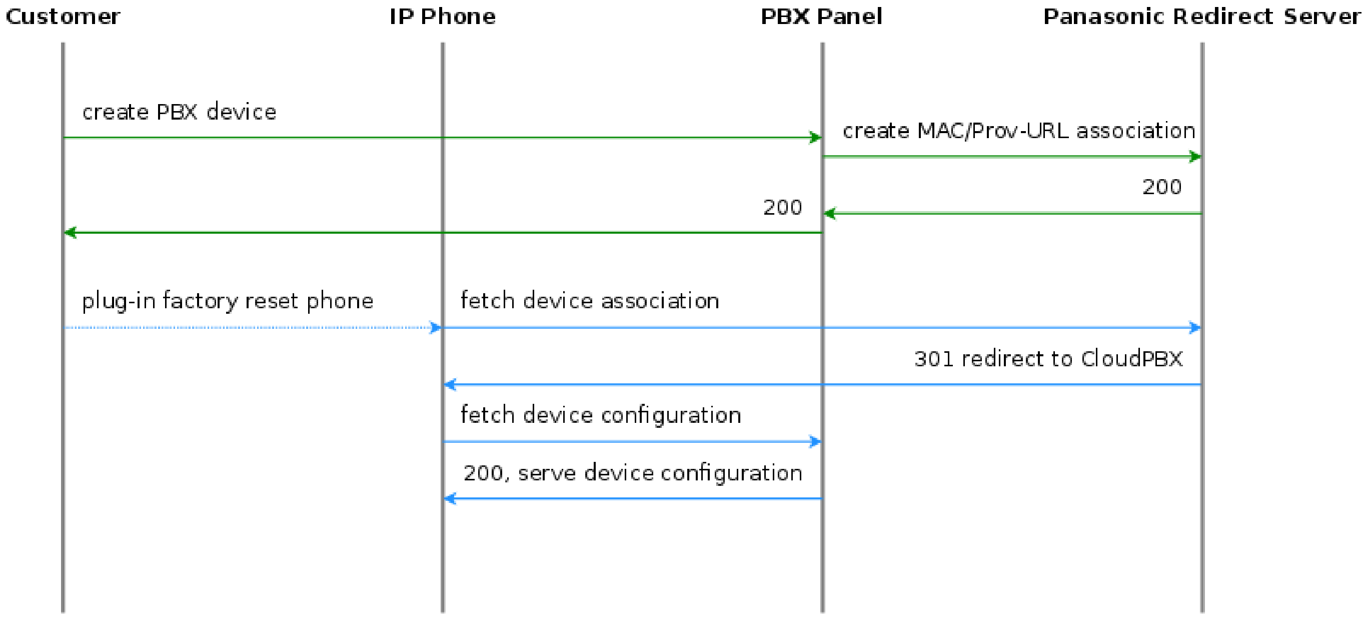


Figure 92: Initially bootstrap a Panasonic phone

The CloudPBX module ensures that when an end customer creates a Panasonic device, the MAC address is automatically provisioned on the Panasonic web service via an API call, so the customer's phone can use the correct provisioning URL to connect to the auto-provisioning server of the CloudPBX.

As a result, no customer interaction is required to bootstrap Panasonic phones, other than just creating the phone with the proper MAC on the CloudPBX web interface.

Factory Reset

For already provisioned phones, the end customer might need to perform a factory reset:

- Press *Settings* or *Setup*
- Enter *#136*
- Select *Factory Setting* and press *Enter*
- Select *Yes* and press *Enter*
- Select *Yes* and press *Enter*

The default username for factory-reset phones is *admin* with password *adminpass*.

Subsequent Device Resyncs

The same procedure as with Cisco SPA phones applies, once a subscriber configured on the phone is registered.

15.1.10.3 Yealink Device Bootstrap

Initial Bootstrapping

Yealink provides a zero-touch provisioning mechanism in their firmwares, which causes the factory-reset phones to connect to a Yealink web service at *https://rps.yealink.com* to check if a custom provisioning URL is configured for the MAC address of the phone. If an association between the MAC and a provisioning URL is found, the web service redirects the phone to the provisioning URL, where the phone connects to in order to obtain the configuration file.

If both Cisco SPA and Yealink phones are used, an issue with the Cisco-signed server certificate configured on the provisioning port (1444 by default) of the CloudPBX provisioning server arises. Yealink phones by default only connect to trusted server certificates, and the Cisco CA certificate used to sign the server certificate is not trusted by Yealink. Therefore, a two-step approach is used to disable the trusted check via a plain insecure http port (1445 by default) first, where only device-generic config options are served. No user credentials are provided in this case, because no SSL client authentication can be performed. The generic configuration disables the trusted check, and at the same time changes the provisioning URL to the secure port, where the Yealink phone is now able to connect to.

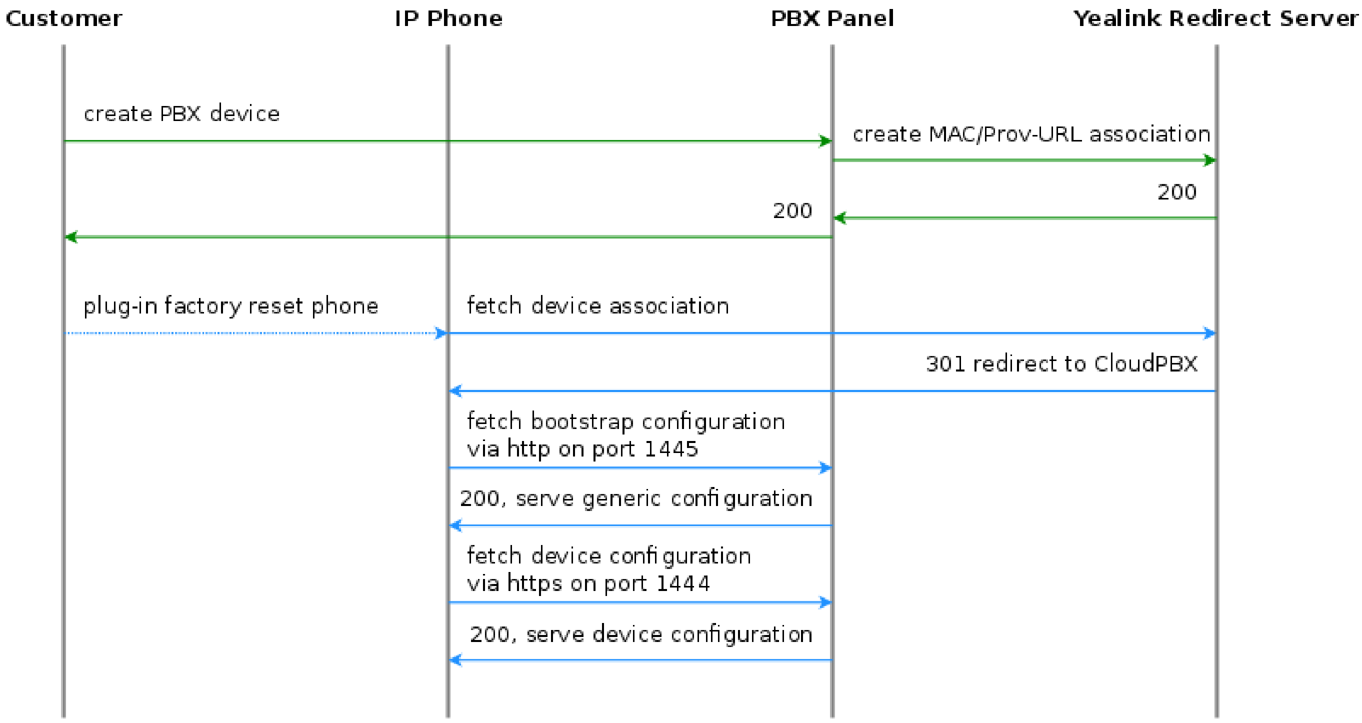


Figure 93: Initially bootstrap a Yealink phone

The CloudPBX module ensures that when an end customer creates a Yealink device, the MAC address is automatically provisioned on the Yealink web service via an API call, so the customer's phone can use the correct insecure bootstrap provisioning URL to connect to the auto-provisioning server of the CloudPBX for the generic configuration, which in turn provides the information on where to connect to for the secure, full configuration.

As a result, no customer interaction is required to bootstrap Yealink phones, other than just creating the phone with the proper MAC on the CloudPBX web interface.

Factory Enable Yealink Auto-Provisioning

Older Yealink firmwares don't automatically connect to the Yealink auto-provisioning server on initial boot, so it needs to be enabled manually by the end customer.

- Log in to *http://phone-ip/servlet?p=hidden&q=load* using *admin* and *admin* as user/password when prompted
- Change *Redirect Active* to *Enabled*
- Press *Confirm* and power-cycle phone

Subsequent Device Resyncs

The same procedure as with Cisco SPA phones applies, once a subscriber configured on the phone is registered.

15.1.10.4 Audiocodes Mediant Device Bootstrap and Configuration

Initial Bootstrapping

An Audiocodes device provides a zero-touch provisioning mechanism in its firmware which causes a factory-reset device to connect to the URL built into the firmware. This URL is pointing to the NGCP provisioning server (in case of NGCP Carrier: `web01` node) listening on TCP port 1444 for HTTPS sessions.

The prerequisites for the device provisioning are that the device has a routable IP address and can reach the IP address of the NGCP provisioning interface.

The Audiocodes device should request the firmware file or CLI configuration file from the NGCP platform. The firmware versions and CLI config versions are decoupled from each other; the NGCP can not enforce specific version of the firmware on the device. Instead, it should be requested by the device itself. In other words, provisioning is a *pull* and not a *push* process.

NGCP expects the provisioning request from the Audiocodes device after SSL handshake and serves the requested file to the device if the device provides valid MAC address as the part of the URL. The MAC address is used to identify the device to the NGCP platform. The firmware and CLI config files are provided at the following URLs:

- the base URL to download firmwares: `https://<NGCP_IP>:1444/device/autoprov/firmware/001122334455/from/0/latest`
- the base URL to download CLI config: `https://<NGCP_IP>:1444/device/autoprov/config/001122334455`

where 001122334455 should be replaced with the actual device's MAC address and <NGCP_IP> with IP address of the NGCP provisioning interface.

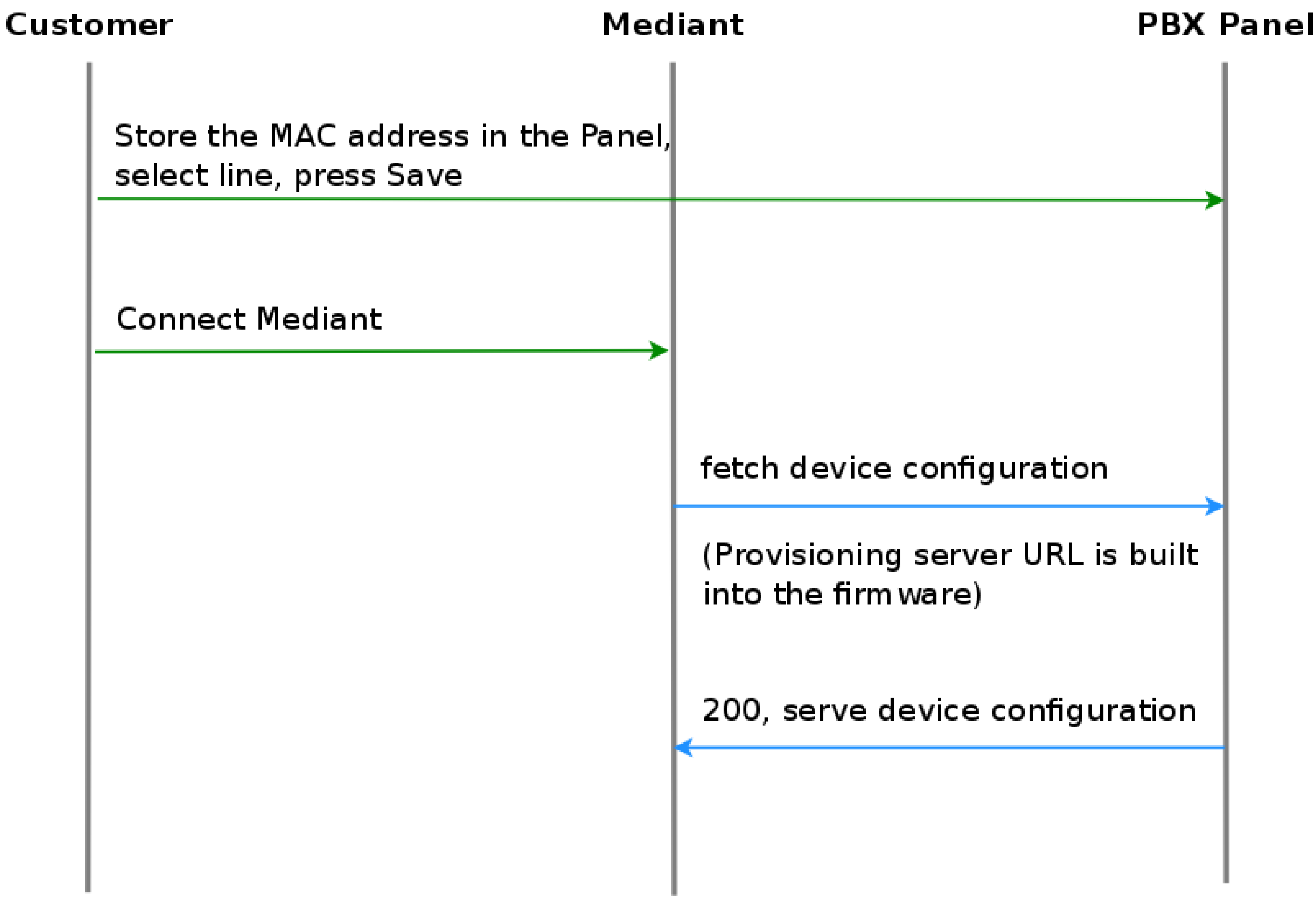


Figure 94: Initially bootstrap a Mediant gateway

Device management basics

The list of device models, firmwares and configurations are global to a reseller and are available for end customer. This data is initially provided by Sipwise as bulk upload of all supported phone models. The firmwares and settings are stored in the database on the DB node pair(s). The NGCP leverages the Cloud PBX module with its template system to generate the configurations and firmware files from database on the fly. Please refer to the following chapters in NGCP handbook for the current information on how to perform device management:

- Uploading device firmwares Section 15.1.1.2
- Creating device configuration Section 15.1.1.3
- Creating device profiles Section 15.1.1.4

Parameterizing the Device Configuration Template

The device-specific parameters are filled in by the system individually when a physical device fetches its configuration file. Parameters from the NGCP panel:

- `username`: Subscriber Details → Master Data → SIP Username

- `password`: Subscriber Details → Master Data → SIP Password
- `domain`: Subscriber Details → Master Data → Domain
- `extension`: Subscriber Details → Master Data → Extension
- `area code`: Subscriber Preferences → Number Manipulations → ac
- `country code`: Subscriber Preferences → Number Manipulations → cc

The produced **CLI config file** has the following structure:

1. SIP account credentials:

   `"sip-definition account 0"`

   - user-name `[username]`
   - password `[password]`
   - host-name `[domain]`
   - register reg
   - contact-user `"[country code][area code][extension]"`

2. IP Groups:

   `"voip-network ip-group 1" and "voip-network ip-group 2"`

   - sip-group-name `[domain]`

3. Proxy and registration settings:

   `"sip-definition proxy-and-registration"`

   - set gw-name `[domain]`

4. Manipulations:

   - manipulation-name "from trunk domain":

     `"sbc manipulations message-manipulations 3"`

     – action-value `"[% line.domain %]"`

   - manipulation-name "clip no screening":

     `"sbc manipulations message-manipulations 8"`

     – action-value `"'<sip:+[country code][area code][extension]@' + param.ipg.dst.host + '>'"`

Specific CLI parameters are:

- `[IPPBX_Hostname]`
- `[IPPBX_server_IP]`

which are used at the following configuration parameters:

- Proxy settings:

  `"voip-network proxy-ip 1"`

  – proxy-address `[IPPBX_Hostname]`

- Manipulations:

  `"sbc manipulations message-manipulations 1"`

  – action-value `[IPPBX_Hostname]`

15.1.11 Device Provisioning and Deployment Workflows

This chapter provides information and hints for preparing and performing the deployment of certain VoIP devices at customer sites, that have a customer-facing interface which also needs customisation.

15.1.11.1 Audiocodes Mediant Device Provisioning Workflow

Audiocodes ISDN gateways and eSBCs are devices used to connect legacy (ISDN) PBX and IP-PBX to the Sipwise NGCP platform and maintain their operations within the Operator's network. Sipwise NGCP offers a *SipConnect 1.1* compliant signaling and media interface to connect SIP trunks to the platform. In addition to this interface, the Sipwise NGCP provides an auto-provisioning mechanism to configure SIP endpoints like IP phones, media gateways and eSBCs.

Provisioning URL

An Audiocodes device needs to obtain the provisioning URL of the Sipwise NGCP in one way or the other to request its device configuration and subsequently download specific firmwares, obtain SIP credentials to connect to the network facing side, and configure the customer facing side for customer devices to connect either via ISDN or SIP. Typical ways of obtaining the provisioning URL for a SIP endpoint are:

- using DHCP option-66 (in a pre-staging environment or directly at the customer premise) where vendor-specific Redirect Servers are configured in the default configuration or firmware
- getting pre-configured per deployment from the SIP endpoint vendor
- getting pre-configured per deployment by a 3rd party distributor

The assumption is that Audiocodes devices are supplied with a firmware (and all required SSL certificates) being pre-configured and the provisioning URL pointing to an Operator URL the Sipwise NGCP is serving, before handing the devices over to field service engineers doing the truck rolls.

Field Configuration

The Sipwise NGCP provides a SipConnect 1.1 compliant interface on the network side for the Audiocodes devices. This interface clearly defines the numbering formats of the calling and called party, the SIP header mechanisms to provide CLI restriction, the RTP codecs, etc.

On the customer facing side, however, those variables might be different from deployment to deployment:

- An IP-PBX might choose to only send its extension as calling party number, or might choose to send the full number in national format.
- It might choose to use the SIP From-header mechanisms to suppress displaying of the CLI, or use the SIP Privacy header.
- The same uncertainty exists to some extent for a legacy PBX connecting via ISDN to the Audiocodes device.

The assumption here is that a field service engineer is NOT supposed to change the Audiocodes configuration in order to make the customer interface work, as this will lead to big issues in maintaining those local changes, especially if a replacement of the device is necessary. Instead, the Audiocodes configuration must ensure that all different kinds of variants in terms of SIP headers, codecs and number formats are translated correctly to the network side and vice versa. If it turns out that there are scenarios in the field which are not handled correctly, temporary local changes might be performed to finish a truck roll, but those changes MUST be communicated to the platform operator, and the server-side configuration templates must be adapted to handle those scenarios gracefully as well.

For deployments with ISDN interfaces on the customer facing side of the Audiocodes, different *Device Profiles* with specific *Device Configurations* per *Device Model* must exist to handle certain scenarios, specifically whether the ISDN interface is operating in Point-to-Point or Point-to-Multipoint mode. Configuration options like which side is providing the clock-rate are to be defined up-front, and the PBX must be reconfigured to adhere to the configuration.

Network Configuration

On the network facing side, both the ISDN and eSBC style deployments have to be designed to obtain an IP address via DHCP. The definition of the IP address ranges is up to the Operator. It may or may not be NAT-ed, but it is advised to use a private IP range directly routed in the back-bone to avoid NAT.

On the customer facing side, networking is only relevant for the eSBC deployment. In order to make the IP-PBX configuration as stream-lined as possible, a pre-defined network should be established on the customer interface of the Audiocodes device.

Tip

The proposal is to define a network 192.168.255.0/24 with the Audiocodes device using the IP 192.168.255.2 (leaving the 192.168.255.1 to a possible gateway). The IP-PBX could obtain its IP address via DHCP from a DHCP server running on the Audiocodes device (e.g. serving IP addresses in the range of 192.168.255.100-254), or could have it configured manually (e.g. in the range of 192.168.255.3-99). Since the Audiocodes device IP on the customer side is always fixed at 192.168.255.2, the IP-PBX for each customer can be configured the same way, pointing the SIP proxy/registrar or outbound proxy always to this IP.

The customer facing side is outside the Sipwise demarcation line, that’s why the network configuration mentioned above only serves as proposal and any feedback is highly welcome. However, it must be clearly communicated how the customer facing network is going to be configured, because the Sipwise NGCP needs to incorporate this configuration into the Audiocodes configuration templates.

15.1.11.2 Audiocodes Mediant Device Deployment Workflow

Pre-Configuration on Sipwise NGCP platform

1. Before connecting a customer to a SIP trunk, it must be clear which Audiocodes *Device Model* is going to be used (depending on if, which and how many ISDN ports are necessary) and which *Device Profile* for the *Device Model* is required (eSBC mode, ISDN P-to-P or P-to-MP mode). Based on that, the correct physical device must be picked.
2. Next, the customer has to be created on the Sipwise NGCP. This step requires the creation of the customer, and the creation of a subscriber within this customer. For the subscriber, the proper E.164 numbers or number blocks must be assigned, and the correct subscriber preferences must be set for the network interface to adhere to the SipConnect 1.1 interface. This step is automated by a script provided by Sipwise until the provisioning work-flow is fully integrated with Operator's OSS/BSS systems. *Required parameters are:*
   - an external customer id to relate the customer entity on the Sipwise NGCP with a customer identifier in Operator's IT systems
   - a billing profile name
   - a subscriber username and password, the domain the subscriber is configured for
   - the numbers or number blocks assigned to the subscriber, and the network provided number of the subscriber
   - optional information is geographic location information and IP network information to properly map emergency calls
3. Finally, the association between the MAC address of the Audiocodes device and the SIP subscriber to be used on the SIP trunk must be established. This step is also automated by a script provided by Sipwise. *Required parameters are:*
   - the subscriber id
   - the Device Profile to be used
   - and the MAC address of the Audiocodes device

Installation

Once the above requirements are fulfilled and the customer is created on the Sipwise NGCP, the Audiocodes device can be installed at the customer premise.

When the Audiocodes device boots, it requests the configuration file from the Sipwise NGCP by issuing a GET request via HTTPS.

For **authentication and authorization** purposes, the Sipwise NGCP requests an SSL client certificate from the device and will check whether it's signed by a Certificate Authority known to the Sipwise NGCP. Therefore, Audiocodes must provide the CA certificate used to sign the devices' client certificates to Sipwise to allow for this process. Also, the Sipwise NGCP will provide an SSL server certificate to the device. The device must validate this certificate in order to prevent man-in-the-middle attacks. Options here are to have:

- Sipwise provide a self-signed certificate to Audiocodes for Audiocodes or a 3rd party distribution partner to configure it as trusted CA in the pre-staging process

- the Operator provide a certificate signed by a CA which is already in the trust store of the Audiocodes devices.

Once the secured HTTPS connection is established, the Sipwise NGCP will provide a CLI style configuration file, with its content depending on the pre-configured *Device Profile* and subscriber association to the device's MAC address.

The configuration includes the firmware version of the latest available firmware configured for the *Device Model*, and a URL defining from where to obtain it. The configuration details on how the Audiocodes devices manage the scheduling of firmware updates are to be provided by Audiocodes or its partners, since this is out of scope for Sipwise. Ideally, firmware updates should only be performed if the device is idle (no calls running), and within a specific time-frame (e.g. between 1 a.m. and 5 a.m. once a certain firmware version is reached, including some random variation to prevent all devices to download a new firmware version at the same time).

Device Replacement

If a customer requires the replacement of a device, e.g. due to hardware issues or due to changing the number or type of ISDN interfaces, a new association of the new device MAC, its *Device Profile* and the subscriber must be established.

In order to make the change as seamless as possible for the customer, a new device is created for the customer with the new MAC, a proper *Device Profile*, but the same subscriber as used on the old device. Once the new device boots at the customer premise, it will obtain its configuration and will register with the same subscriber as the old device (in case it's still operational). For inbound calls to the customer, this will cause parallel ringing to take place, and it's up to the customer or the field engineer when to re-configure or re-cable the PBX to connect to one or the other device.

Once the old device is decommissioned, the old MAC association can be deleted on the Sipwise NGCP.

15.1.12 List of available pre-configured devices

Vendor	Model	Available from release
Audiocodes	Mediant800	mr4.1.1.1
Cisco	ATA112	mr3.4.1.1
Cisco	ATA122	mr3.4.1.1
Cisco	SPA232D	mr3.4.1.1
Cisco	SPA301	mr3.4.1.1
Cisco	SPA303	mr3.4.1.1
Cisco	SPA501G	mr3.4.1.1
Cisco	SPA502G	mr3.4.1.1
Cisco	SPA512G	mr3.4.1.1
Cisco	SPA504G	mr3.4.1.1
Cisco	SPA504G + SPA500S	mr3.7.1.4
Cisco	SPA504G + two SPA500S	mr3.7.1.4
Cisco	SPA514G	mr3.4.1.1
Cisco	SPA508G	mr3.4.1.1
Cisco	SPA509G	mr3.4.1.1
Cisco	SPA525G	mr3.4.1.1
Innovaphone	IP2X2X	mr3.8.3.3

Vendor	Model	Available from release
Innovaphone	IP230-X	mr3.8.3.3
Innovaphone	IP232	mr3.8.3.3
Innovaphone	IP222	mr3.8.3.3
Innovaphone	IP240	mr3.8.3.3
Innovaphone	IP22	mr3.8.3.3
Innovaphone	IP111	mr3.8.3.3
Panasonic	KX-UT113	mr3.7.1.1
Panasonic	KX-UT123	mr3.7.1.1
Panasonic	KX-UT133	mr3.7.1.1
Panasonic	KX-UT136	mr3.7.1.1
Panasonic	KX-UT248	mr3.7.1.1
Yealink	SIP-T19P	mr3.7.1.1
Yealink	SIP-T20P	mr3.7.1.1
Yealink	SIP-T21P	mr3.7.1.1
Yealink	SIP-T22P	mr3.7.1.1
Yealink	SIP-T23P	mr3.7.1.1
Yealink	SIP-T23G	mr3.7.1.1
Yealink	SIP-T26P	mr3.7.1.1
Yealink	SIP-T28P	mr3.7.1.1
Yealink	SIP-T32G	mr3.7.1.1
Yealink	SIP-T38G	mr3.7.1.1
Yealink	SIP-T41P	mr3.7.1.1
Yealink	SIP-T42G	mr3.7.1.1
Yealink	SIP-T46G	mr3.7.1.1
Yealink	SIP-T48G	mr3.7.1.1
Yealink	SIP-T28P + EXP39	mr3.8.1.1
Yealink	SIP-T28P + two EXP39	mr3.8.1.1
Yealink	W52P	mr3.7.1.6

15.1.12.1 Cisco Devices

IP Phones

Model	IPv6	TLS	SRTP	Auto provisioning	Private Line	Shared Line	Busy Lamp	Extension Boards
SPA301	N	Y	Y	http	1	1	0	N
SPA303	N	Y	Y	http	1-3	1-3	1-2	N
SPA501G	N	Y	Y	http	1-8	1-8	1-7	N
SPA502G	N	Y	Y	http	1	1	0	N
SPA512G	N	N	Y	http	1	1	0	N
SPA504G	N	Y	Y	http	1-4	1-4	1-3	2
SPA514G	N	N	Y	http	1-4	1-4	1-3	N
SPA508G	N	Y	Y	http	1-8	1-8	1-7	N

Model	IPv6	TLS	SRTP	Auto provisioning	Private Line	Shared Line	Busy Lamp	Extension Boards
SPA509G	N	Y	Y	http	1-12	1-12	1-11	N
SPA525G	N	Y	N	http	1-5	1-5	1-4	N

Analog Adapters

Model	IPv6	TLS	SRTP	Auto provisioning	Private Line	Shared Line	Busy Lamp
SPA232D	N	Y	Y	http	1-6	0	0
ATA112	Y	Y	Y	http	1-2	0	0
ATA122	Y	Y	Y	http	1-2	0	0

Extension Boards

Model	Ports	Buttons	Busy Lamp	Supported phones
SPA500S	2	32	1-32	SPA500

15.1.12.2 Panasonic Devices

IP Phones

Model	IPv6	TLS	SRTP	Auto provisioning	Private Line	Shared Line	Busy Lamp	Extension Boards
KX-UT113	N	N	N	redirect	1-2	1-2	0	N
KX-UT123	N	N	N	redirect	1-2	1-2	0	N
KX-UT133	N	N	N	redirect	1-4	1-4	1-23	N
KX-UT136	N	N	N	redirect	1-4	1-4	1-23	N
KX-UT248	N	N	Y	redirect	1-6	1-6	1-23	N

15.1.12.3 Yealink Devices

IP Phones

Model	IPv6	TLS	SRTP	Auto provisioning	Private Line	Shared Line	Busy Lamp	Extension Boards
SIP-T19P	Y	Y	Y	redirect	1	1	0	N
SIP-T20P	Y	Y	Y	redirect	1	1	0	N
SIP-T21P	Y	Y	Y	redirect	1-2	1-2	1	N
SIP-T22P	Y	Y	Y	redirect	1-3	1-3	1-2	N
SIP-T23P	Y	Y	Y	redirect	1-3	1-3	1-2	N
SIP-T23G	Y	Y	Y	redirect	1-3	1-3	1-2	N
SIP-T26P	Y	Y	Y	redirect	1-3	1-3	1-12	N

Model	IPv6	TLS	SRTP	Auto provisioning	Private Line	Shared Line	Busy Lamp	Extension Boards
SIP-T28P	Y	Y	Y	redirect	1-6	1-6	1-15	2
SIP-T32G	Y	Y	Y	redirect	1-3	1-3	1-2	N
SIP-T38G	Y	Y	Y	redirect	1-6	1-6	1-15	N
SIP-T41P	Y	Y	Y	redirect	1-3	1-3	1-14	N
SIP-T42G	Y	Y	Y	redirect	1-3	1-3	1-14	N
SIP-T46G	Y	Y	Y	redirect	1-6	1-6	1-26	N
SIP-T48G	Y	Y	Y	redirect	1-6	1-6	1-28	N
W52P	N	Y	Y	redirect	1-5	1-5	0	N

15.1.12.4 Innovaphone Devices

IP Phones

Model	IPv6	TLS	SRTP	Auto provisioning	Private Line	Shared Line	Busy Lamp	Extension Boards
IP232	N	Y	Y	dhcp	1	0	1-16	2
IP222	N	Y	Y	dhcp	1	0	1-16	2
IP240	N	N	N	dhcp	1	0	1-15	2
IP111	N	Y	Y	dhcp	1	0	1-16	0

Analog Adapters

Model	IPv6	TLS	SRTP	Auto provisioning	Private Line	Shared Line	Busy Lamp
IP22	N	Y	Y	dhcp	1	0	0

Extension Boards

Model	Ports	Buttons	Busy Lamp	Supported phones
IP2X2X	2	64	1-32	IP2x2
IP230-X	2	30	1-30	IP230

15.1.12.5 Audiocodes Devices

Model	IPv6	TLS	SRTP	Auto provisioning	Private Line	Shared Line	Busy Lamp	Speed Dial
Mediant800	Y	Y	Y	dhcp	1	0	0	N

15.1.13 Phone features

15.1.13.1 Cisco phones

SPA301

1) Soft keys

Not available.

2) Hard keys

- vm
- hold/unhold

3) Line keys

Not available.

4) VSC

- directed pickup
- park/unpark

SPA303

1) Soft keys

Idle:

redial	lcr	dir	dnd >
< cfwd	unpark		

Idle with missed calls:

lcr			miss

Call:

hold	endCall	conf	xfer >
< bxfer	park		

Call on hold:

resume	endCall	newCall	redial >
< dir	cfwd	dnd	

Ringing:

answer	ignore		

2) Hard keys

- vm
- hold/unhold

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- directed pickup

SPA501G

1) Soft keys

Idle:

redial	lcr	dir	dnd >
< cfwd	unpark		

Idle with missed calls:

lcr			miss

Call:

hold/resume	endCall	conf	xfer >
< bxfer	park		

Call on hold:

resume	endCall	newCall	redial >
< dir	cfwd	dnd	

Ringing:

answer	reject		

2) Hard keys

- vm
- hold/unhold

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- directed pickup

SPA502G

1) Soft keys

Idle:

redial	lcr	dir	dnd >
< cfwd	unpark		

Idle with missed calls:

lcr			miss

Call:

hold/resume	endCall	conf	xfer >
< bxfer	park		

Call on hold:

resume	endCall	newCall	redial >
< dir	cfwd	dnd	

Ringing:

answer	reject		

2) Hard keys

- vm
- hold/unhold

3) Line keys

Not available.

4) VSC

- directed pickup

SPA504G

1) Soft keys

Idle:

redial	lcr	dir	dnd >
< cfwd	unpark		

Idle with missed calls:

lcr			miss

Call:

hold/resume	endCall	conf	xfer >
< bxfer	park		

Call on hold:

resume	endCall	newCall	redial >
< dir	cfwd	dnd	

Ringing:

answer	reject		

2) Hard keys

- vm
- hold/unhold

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- directed pickup

SPA512G

1) Soft keys

Idle:

redial	lcr	dir	dnd >
< cfwd	unpark		

Idle with missed calls:

lcr			miss

Call:

hold/resume	endCall	conf	xfer >
< bxfer	park		

Call on hold:

resume	endCall	newCall	redial >
< dir	cfwd	dnd	

Ringing:

answer	reject		

2) Hard keys

- vm
- hold/unhold

3) Line keys

Not available.

4) VSC

- directed pickup

SPA514G

1) Soft keys

Idle:

redial	lcr	dir	dnd >
< cfwd	unpark		

Idle with missed calls:

lcr			miss

Call:

hold/resume	endCall	conf	xfer >
< bxfer	park		

Call on hold:

resume	endCall	newCall	redial >
< dir	cfwd	dnd	

Ringing:

answer	reject		

2) Hard keys

- vm
- hold/unhold

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- directed pickup

SPA509G

1) Soft keys

Idle:

redial	lcr	dir	dnd >
< cfwd	unpark		

Idle with missed calls:

lcr			miss

Call:

hold/resume	endCall	conf	xfer >
< bxfer	park		

Call on hold:

resume	endCall	newCall	redial >
< dir	cfwd	dnd	

Ringing:

answer	reject		

2) Hard keys

- vm
- hold/unhold

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- directed pickup

SPA508G

1) Soft keys

Idle:

redial	lcr	dir	dnd >
< cfwd	unpark		

Idle with missed calls:

lcr			miss

Call:

hold/resume	endCall	conf	xfer >
< bxfer	park		

Call on hold:

resume	endCall	newCall	redial >
< dir	cfwd	dnd	

Ringing:

answer	reject		

2) Hard keys

- vm
- hold/unhold

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- directed pickup

SPA525G

1) Soft keys

Idle:

Redial	call Rtn	Directory	DND >
< Forward	Unpark		

Idle with missed calls:

Call Rtn			Miss

Call:

Hold	End Call	Conf	Transfer >
BlindXfer	Park		

Call on hold:

Resume	EndCall	EewCall	Redial >
< Directory	Forward	DND	

Ringing:

Answer	Ignore		

2) Hard keys

- vm
- hold/unhold

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- directed pickup

15.1.13.2 Yealink phones

T19P

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

Not available.

4) VSC

- transfer park
- directed pick up
- park/unpark

T20P

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- transfer park
- park/unpark

T21P

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- transfer park
- park/unpark

T22P

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- park/unpark
- transfer park

T23P

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- park/unpark
- transfer park

T23G

1) Soft keys

Idle:

History	Dir	DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	EndCall

Call on hold:

Tran	Resume	NewCall	EndCall

Ringing:

Answer	FWD		Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- unpark
- transfer park

T26P

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- unpark
- transfer park

T28P

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- park/unpark
- transfer park

T32G

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- unpark
- transfer park

T38G

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- unpark
- transfer park

T41P

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- park/unpark
- transfer park

T42G

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial

- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- park/unpark
- transfer park

T46G

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- park/unpark
- transfer park

T48G

1) Soft keys

Idle:

History		DND	Menu

Idle with missed calls:

Exit			View

Call:

Tran	Hold	Conf	Cancel

Call on hold:

Tran	Resume	NewCall	Cancel

Ringing:

Answer	FWD	Silence	Reject

2) Hard keys

- vm
- redial
- transfer

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- park/unpark
- transfer park

W52P

1) Soft keys

Idle:

History	Line

Idle with missed calls:

Exit	View

Call:

Ext. Call	Options

Call on hold:

Resume	Line

Ringing:

Accept	

2) Hard keys

- vm
- redirect

3) VSC

- park/unpark
- transfer park

15.1.13.3 Panasonic phones

KX-UT113

1) Soft keys

Idle:

Settings	Call Log	Phone book	

Call:

Blind		Phone book	

Call on hold:

	Call Log	Phone book	

Ringing:

Answer		Reject	

2) Hard keys

- vm
- forward/dnd
- hold/unhold
- redial
- recall
- transfer
- conf

3) Line keys

Not available.

4) VSC

- park/unpark
- transfer park

KX-UT123

1) Soft keys

Idle:

Settings	Call Log	Phone book	

Call:

Blind		Phone book	

Call on hold:

	Call Log	Phone book	

Ringing:

Answer		Reject	

2) Hard keys

- vm
- forward/dnd
- hold/unhold
- redial
- recall
- transfer
- conf

3) Line keys

Not available.

4) VSC

- park/unpark
- transfer park

KX-UT133

1) Soft keys

Idle:

Settings	Call Log	Phone book	

Call:

Blind		Phone book	

Call on hold:

	Call Log	Phone book	

Ringing:

Answer		Reject	

2) Hard keys

- vm
- forward/dnd
- hold/unhold
- redial
- recall
- transfer
- conf

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- unpark
- transfer park

KX-UT136

1) Soft keys

Idle:

Settings	Call Log	Phone book	

Call:

Blind		Phone book	

Call on hold:

	Call Log	Phone book	

Ringing:

Answer		Reject	

2) Hard keys

- vm
- forward/dnd
- hold/unhold
- redial
- recall
- transfer
- conf

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- park/unpark
- transfer park

KX-UT248

1) Soft keys

Idle:

Settings	Call Log	Phone book	

Call:

Blind		Phone book	

Call on hold:

	Call Log	Phone book	

Ringing:

Answer		Reject	

2) Hard keys

- vm
- forward/dnd
- hold/unhold
- redial
- recall
- transfer
- conf

3) Line keys

- BLF monitoring
- directed pickup

4) VSC

- park/unpark
- transfer park

15.1.13.4 Innovaphone

IP222

1) Soft keys

Idle:

Setup	All Calls	Home	Calls	My favorites	Phonebook

Call:

Hold	Transfer	Park	Cancel

Call on hold:

Resume	Transfer	Park	Cancel

Ringing:

Answer	Transfer	Silence	Reject

2) Hard keys

- hold
- redial

3) Line keys

- BLF monitoring

4) VSC

- unpark
- transfer park

IP232

1) Soft keys

Idle:

Setup	All Calls	Home	Calls	My favorites	Phonebook

Call:

Hold	Transfer	Park	Cancel

Call on hold:

Resume	Transfer	Park	Cancel

Ringing:

Answer	Transfer	Silence	Reject

2) Hard keys

- hold
- redial

3) Line keys

- BLF monitoring

4) VSC

- unpark
- transfer park

IP111

1) Soft keys

Idle:

Setup	All Calls	Home	Calls	My favorites	Phonebook

Call:

Hold	Transfer	Park	Cancel

Call on hold:

Resume	Transfer	Park	Cancel

Ringing:

Answer	Transfer	Silence	Reject

2) Hard keys

- hold
- redial

3) Line keys

- BLF monitoring

4) VSC

- unpark
- transfer park

IP240

1) Soft keys

Not available.

2) Hard keys

- hold
- redial
- conference
- dnd
- forward

3) Line keys

- BLF monitoring

4) VSC

- transfer park
- unpark

15.2 Sipwise sip:phone App (SIP client)

You can order two commercial Unified Communication Clients for full end-to-end integration of voice, video, chat and presence features. There are two applications available:

- the sip:phone Desktop Client for Microsoft Windows, Apple OSX, and Linux;

- the sip:phone Mobile App for iOS and Android.

Both clients are fully brandable to the customer’s corporate identity. The clients are not part of the standard delivery and need to be licensed separately. This handbook discusses the mobile client in details.

We continuously develop the mobile clients to provide new features, as they do not support the full range of features yet.

The sip:phone Mobile App is a mobile client for iOS and Android that supports voice calls via SIP, as well as presence and instant messaging via XMPP. The following sections describe the steps needed to integrate it into your sip:provider PRO.

15.2.1 Zero Config Launcher

Part of the mobile apps is a mechanism to sign up to the service via a 3rd party website, which is initiated on the login screen and rendered within the app. During the sign-up process, the 3rd party service is supposed to create a new account and subscriber in the sip:provider PRO (e.g. automatically via the API) and provide the end user with the access credentials.

The mobile apps come with a zero config mechanism to simplify the end-customer log in using these credentials (especially ruling out the need to manually enter them). It makes it possible to deliver the access credentials via a side channel (e.g. Email, SMS) packed into a URL. The user just clicks the URL, and it automatically launches the app with the correct credentials. The following picture shows the overall workflow.

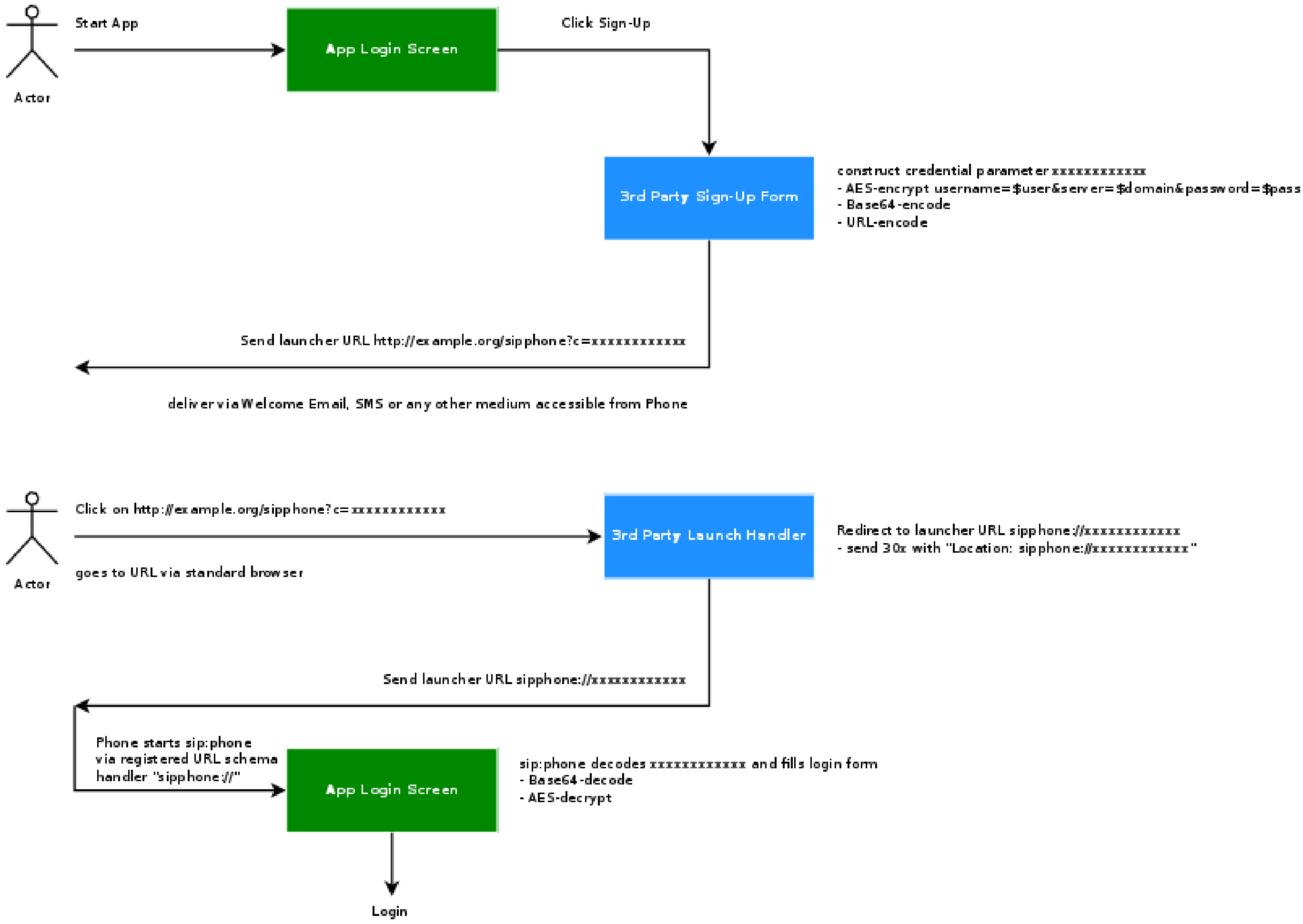


Figure 95: Provisioning Push Workflow

There are two components provided by a 3rd party system. One is the *3rd Party Sign-Up Form*, and the other is the *3rd Party Launch Handler*. The purpose of these components is to allow an end customer to open a link with the access credentials via the sip:phone app.

15.2.1.1 3rd Party Sign-Up Form

The 3rd Party Sign-Up Form is a website the app shows to the end user when he taps the sign-up link on the app *Login Screen*. There, the end customer usually provides his contact details like name, address, phone number and email address, etc. After validation, the website creates an account and a subscriber in the sip:provider PRO via the API.

After successfully creating the account and the subscriber, this site needs to construct a specially crafted URL, which is sent back to the end customer via a side channel. Ideally, this channel would be an SMS if you want to verify the end customer's mobile number, or an email if you want to check the email address.

The sip:phone app registers a URL schema handler for URLs starting with `sipphone://`. If you start such a link, the app performs a Base64 decoding of the string right after the `sipphone://` prefix and then decrypts the resulting binary string via AES using the keys defined during the branding step. The resulting string is supposed to be *username=$user&server=$domain&password=$password*.

Therefore, the *3rd Party Sign-Up Form* needs to construct this string using the credentials defined while creating the subscriber via the sip:provider PRO API, then encrypt it via AES, and finally perform a Base64 encoding of the result.

Note

Up until and including version mr4.5.3 of the sip:provider PRO, the SIP login credentials are used here. Future versions will connect to the REST interface of the sip:provider PRO using the web credentials first and fetch the SIP credentials along with other settings from there.

An example Perl code performs encoding of such a string. The AES key and initialization vector ($key and $iv) are the standard values of the sip:phone app and should work until you specified other values during the branding process.

```
#!/usr/bin/perl -w
use strict;
use Crypt::Rijndael;
use MIME::Base64;
use URI::Escape;

my $key = 'iBmTdavJ8joPW3HO';
my $iv = 'tww21lQe6cmywrp3';

my $plain = do { local $/; <> };
# pkcs#5 padding to 16 bytes blocksize
my $pad = 16 - (length $plain) % 16;
$plain .= pack('C', $pad) x $pad;

my $cipher = Crypt::Rijndael->new(
        $key,
        Crypt::Rijndael::MODE_CBC()
);
$cipher->set_iv($iv);
my $crypted = $cipher->encrypt($plain);
# store b64-encoded string and print to STDOUT
my $b64 = encode_base64($crypted, '');
print $b64, "\n";
# print to STDOUT using URL escaping also
print uri_escape($b64), "\n";
```

This snippet takes a string from STDIN, encrypts it via AES, encodes it via Base64 and sends the result to STDOUT. It also writes the second line with the same string, but this time, the URL is escaped. To test it, you would run it as follows on a shell, granted it's stored at `/path/to/encrypt.pl`.

```
echo -n 'username=testuser&server=example.org&password=testpass' \
  | /path/to/encrypt.pl
```

This command would result in the output strings `CI8VN8toaE40w8E4OH2rAuFj3Qev9QdLI/Wv/VaBCVK2yNkBZjxE9` `eafXkkrQfmYdeu01PquS5P40zhUq8Mfjg==` and `CI8VN8toaE40w8E4OH2rAuFj3Qev9QdLI%2FWv%2FVaBCVK2yNkBZjxE9eafXkkrQfmYdeu01PquS5P40zhUq8Mfjg%3D%3D`. The sip:phone can use the former string to automatically fill in the login form of the Login Screen if started via a Link like `sipphone://CI8VN8toaE40w8E4OH2rAuFj3Qev9QdLI/Wv/VaBCVK2yNkBZjxE9eafXkkrQfmYdeu01PquS5P40zhUq8Mfjg==`.

Here is the same code in PHP.

```
#!/usr/bin/php
<?php
$key = "iBmTdavJ8joPW3HO";
$iv = "tww21lQe6cmywrp3";

$clear = fgets(STDIN);
$cipher = fnEncrypt($clear, $key, $iv);

echo $cipher, "\n";
echo urlencode($cipher), "\n";

function fnEncrypt($clear, $key, $iv) {
        $pad = 16 - strlen($clear) % 16;
        $clear .= str_repeat(pack('C', $pad), $pad);
        return rtrim(base64_encode(mcrypt_encrypt(
                MCRYPT_RIJNDAEL_128, $key, $clear,
                MCRYPT_MODE_CBC, $iv)), "\0");
}
?>
```

Similar to the Perl code, you can call it like this:

```
echo -n 'username=testuser&server=example.org&password=testpass' \
  | /path/to/encrypt.php
```

However, a URL with the `sipphone://` schema is not displayed as a link in an SMS or an Email client and thus can not be clicked by the end customer, so you need to make a detour via a regular `http://` URL. To do so, you need a *3rd Party Launch Handler* to trick the phone to open such a link.

Therefore, that the *3rd Party Sign-Up Form* needs to return a link containing a URL pointing to the *3rd Party Launch Handler* and pass the URL escaped string gathered above to the client via an SMS or an Email. Since it is the regular `http://` link, it is clickable on the phone and can be launched from virtually any client (SMS, Email, etc.), which correctly renders an HTML link.

A possible SMS sent to the end customer (via the phone number entered in the sign-up from) could, therefore, look as follows (trying to stay below 140 chars).

```
http://example.org/p?c=CI8VN8toaE40w8E4OH2rAuFj3Qev9QdLI
%2FWv%2FVaBCVK2yNkBZjxE9eafXkkrQfmYdeu01PquS5P40zhUq8Mfjg%3D%3D to launch sipphone
```

An HTML Email could look like this:

```
Welcome to Example.org,
```

```
<a href="http://www.example.org/sipphone?c=CI8VN8toaE40w8E4OH2rAuFj3Qev9QdLI
%2FWv%2FVaBCVK2yNkBZjxE9eafXkkrQfmYdeu01PquS5P40zhUq8Mfjg%3D%3D">
click here
</a> to log in.
```

That way, you can do both: verify the contact details of the end customer, and send the end customer the login credentials in a secure manner.

15.2.1.2 3rd Party Launch Handler

The URL `http://www.example.org/sipphone` mentioned above can be any simple script, and its sole purpose is to send back a `301 Moved Permanently` or `302 Moved Temporarily` with a `Location:sipphone://xxxxxxxxxxxx` header to tell the phone to open this link via the sip:phone app. The *xxxxxxxxxxxx* is the plain (non-URL-escaped) string generated by the above script.

An example CGI script performing this task follows.

```perl
#!/usr/bin/perl -w
use strict;
use CGI;

my $q = CGI->new;
my $c = $q->param('c');
print CGI::redirect("sipphone://$c");
```

The script simply takes the URL parameter `c` from the URL `http://www.example.org/sipphone?c=CI8VN8toaE40w8E4OH2rAuFj3Qev9QdLI%2FWv%2FVaBCVK2yNkBZjxE9eafXkkrQfmYdeu01PquS5P40zhUq8Mfjg%3D%3D` crafted above and puts its content into a `Location` header using the `sipphone://` schema, and finally sends a `301 Moved Permanently` back to the phone.

The phone follows the redirect by opening the URL using the sip:phone app, which in turn decrypts the content and fills in the login form.

Note
Future versions of the sip:provider PRO will be shipped with this launch handler integrated into the system. Up until and including the version mr4.5.3, this script needs to be installed on any webserver manually.

15.2.2 Mobile Push Notification

The *mobile push* functionality provides the remote start of a mobile application on incoming calls via the Google GCM or the Apple APNS notification services. It enables you to offer your subscribers a modern and convenient service on mobile devices.

> **Caution**
> Although suspending an application on a phone and waking it up via the mobile push notification service extends battery life, the whole mobile push notification concept is the best effort framework provided by Apple and Google for iOS and Android respectively, and therefore does not guarantee 100% reliability.

15.2.2.1 Architecture

If the *mobile push* functionality is enabled and there are no devices registered for a subscriber, the call-flow looks as follows.

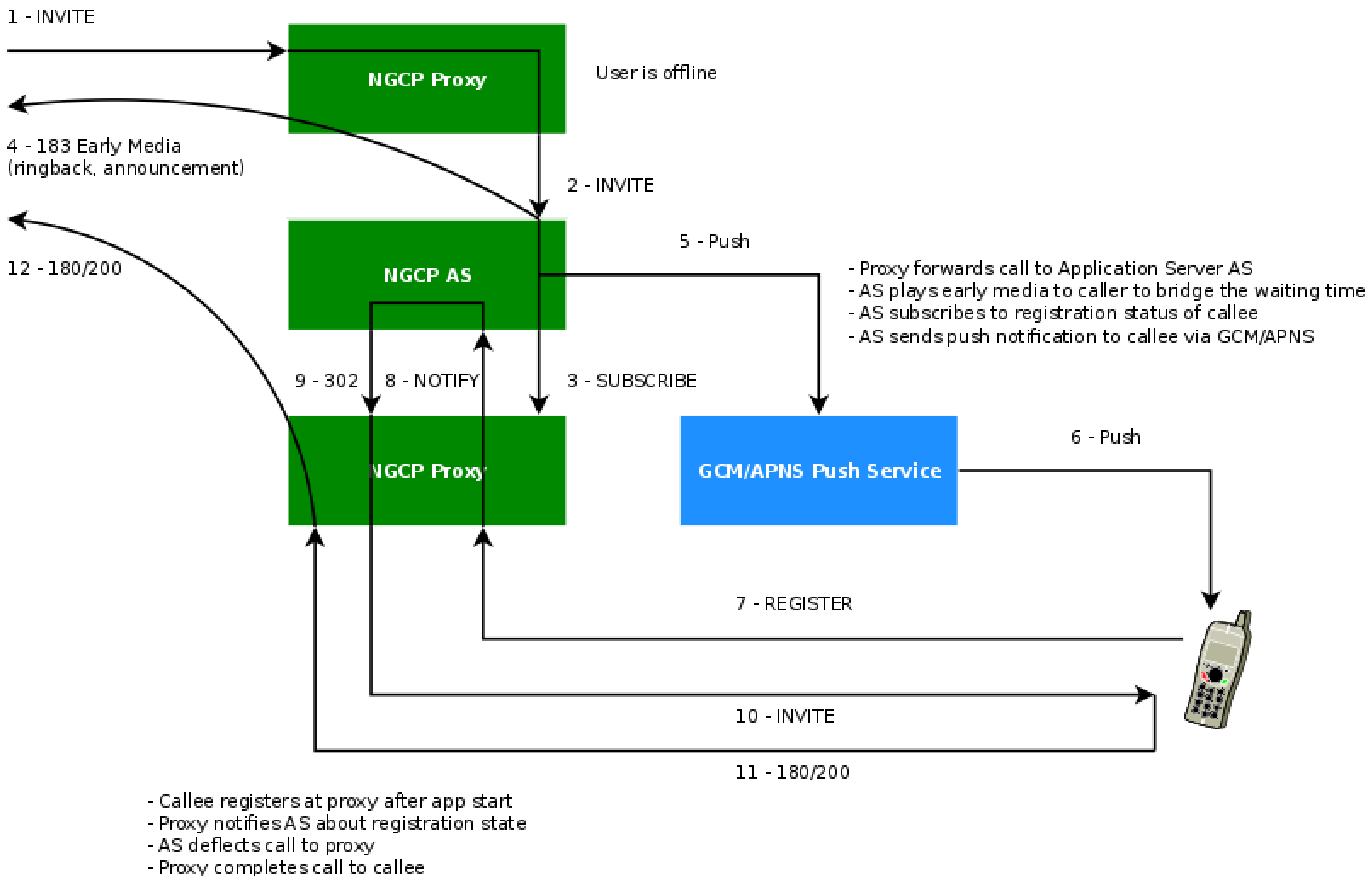


Figure 96: Mobile Push Workflow

1. The caller sends INVITE to proxy
2. The callee is offline, proxy forwards the call to AS (application server)
3. AS subscribes to the callee’s registration events on proxy
4. AS sends early media to the caller as a feedback, as the call initiation process might take a while
5. AS sends the push request to GCM/APNS service
6. GCM/APNS service delivers the push request to the callee

7. The callee accepts the push request and confirms the mobile application start (unattended on Android), then the mobile application registers to proxy
8. Proxy sends registration notification to AS
9. AS deflects the call back to proxy
10. Proxy sends INVITE to the callee
11. The callee accepts the call
12. The response is sent back to the caller. Hence, the call setup is completed

In the case of a time-out (no registration notification within a particular time), the application server rejects the call request with an error.

15.2.2.2 The Configuration Checklist

Follow this checklist to make sure you've completed all the steps. If you miss anything, the service may not work as expected.

Name	Description	Link
Obtain a trusted SSL certificate from a CA	Required for either application	Section 15.2.2.3
Create an Apple developer account and enable the push notification service	For iOS mobile application	Section 15.2.2.4
Obtain the Apple certificate for the app	For iOS mobile application	Section 15.2.2.5
Obtain the API key for the app from Google	For Android mobile application	Section 15.2.2.6
Provide the required information to developers	It is required to make beta builds and publish the apps	Section 15.2.2.7
Adjust the configuration	Adjust the config.yml file and apply the changes (usually performed by Sipwise)	Section 15.2.2.8
Recheck your DNS Zone configuration	Check that the DNS Zone is correctly configured	Section 15.2.2.9
Add DNS SRV records	Create specific DNS SRV records for SIP and XMPP services	Section 15.2.2.10
Check NTP configuration	Ensure that all your servers show exact time	Section 15.2.2.11
Enable Apple/Google Mobile Push in the Admin Panel	It can be enabled for a domain or separate subscribers	Section 15.2.2.12
Configure a mobile application	Check that subscribers can easily install and use your application	Section 15.2.2.13

15.2.2.3 Obtain the Trusted SSL Certificate

A *trusted* SSL certificate is required, and we suggest obtaining it before starting the configuration.

The mobile application uses respective iOS/Android libraries to establish a secure TLS connection with certain sip:provider PRO services, such as SIP/XMPP/pushd(https). A *signed* SSL certificate is required to guarantee the security of this connection.

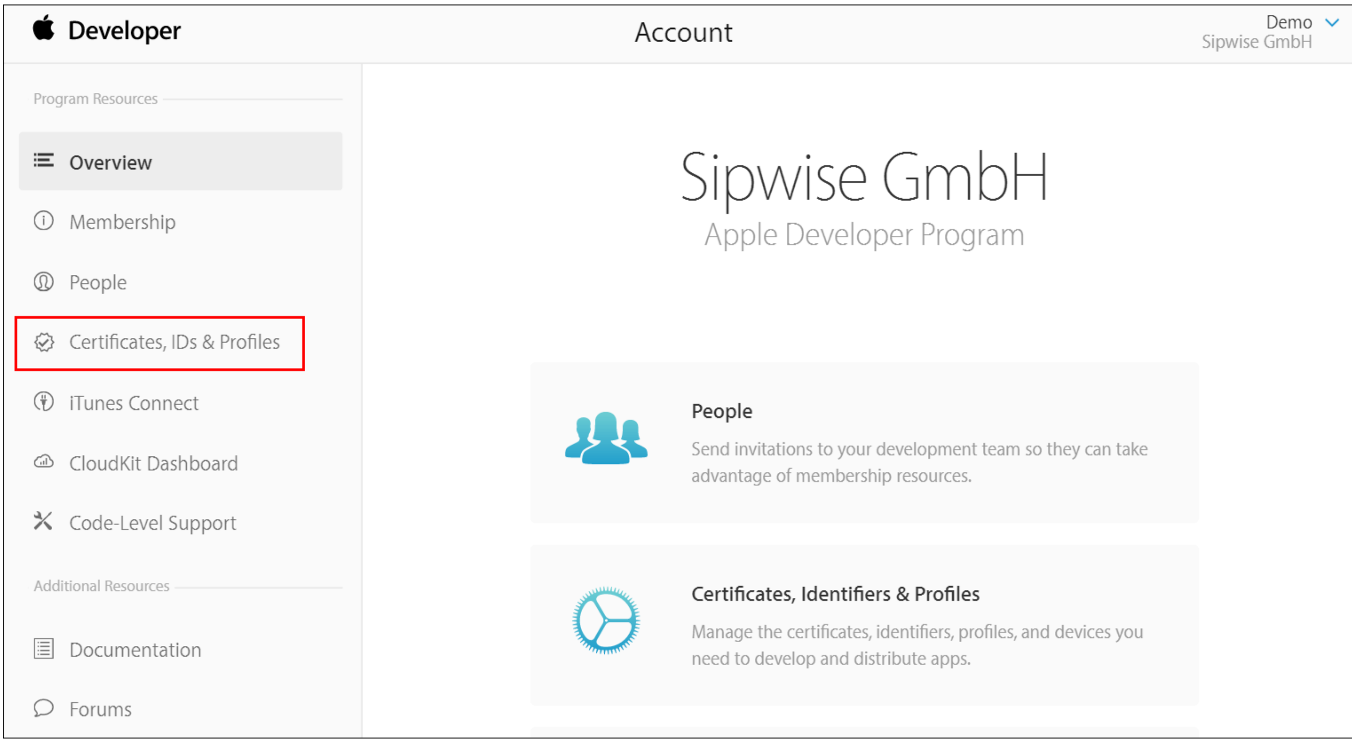
Any Certificate Authority (CA) such as Verisign and others can provide you with the required trusted SSL certificate (a certificate and the key files) which you will use in the configuration below.

15.2.2.4 Create an Apple Account and Enable the Push Notification Service

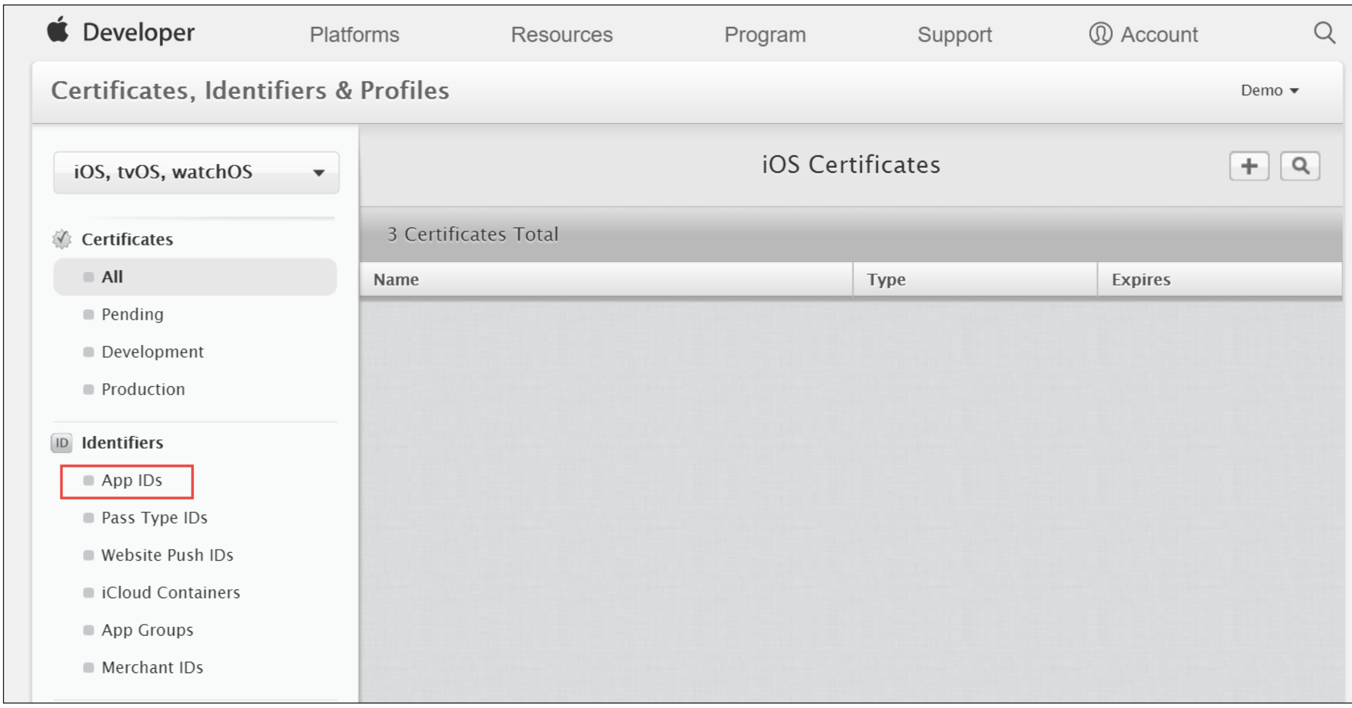
Below is a brief instruction on how to create an Apple account and enable the Push Notification Service in it. You may need to perform additional steps depending on your project.

Note
You may only create an Apple account (step 1 below) and enroll into the Apple Developer Program (step 2 below) and Sipwise developers will do the rest. Still, you can perform all the steps by yourself.

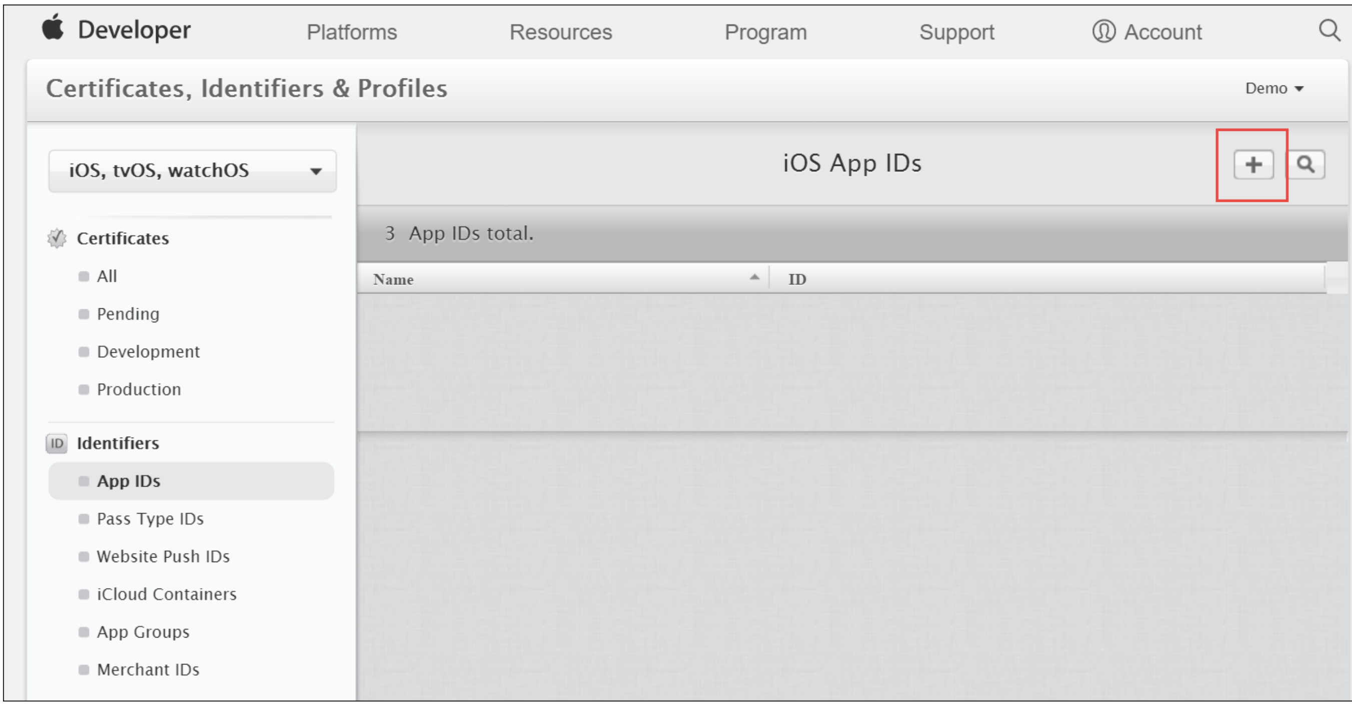
1. Create an Apple developer account to get the Apple ID for your company. For this, go to developer.apple.com/account
2. Enrol in the Apple Developer Program. It is required to configure push notifications as you will need a push notification certificate for your App ID, which requires the Apple Developer Program membership. Go to developer.apple.com/programs for more details.
3. Register an App ID:
   - Sign into developer.apple.com/account.



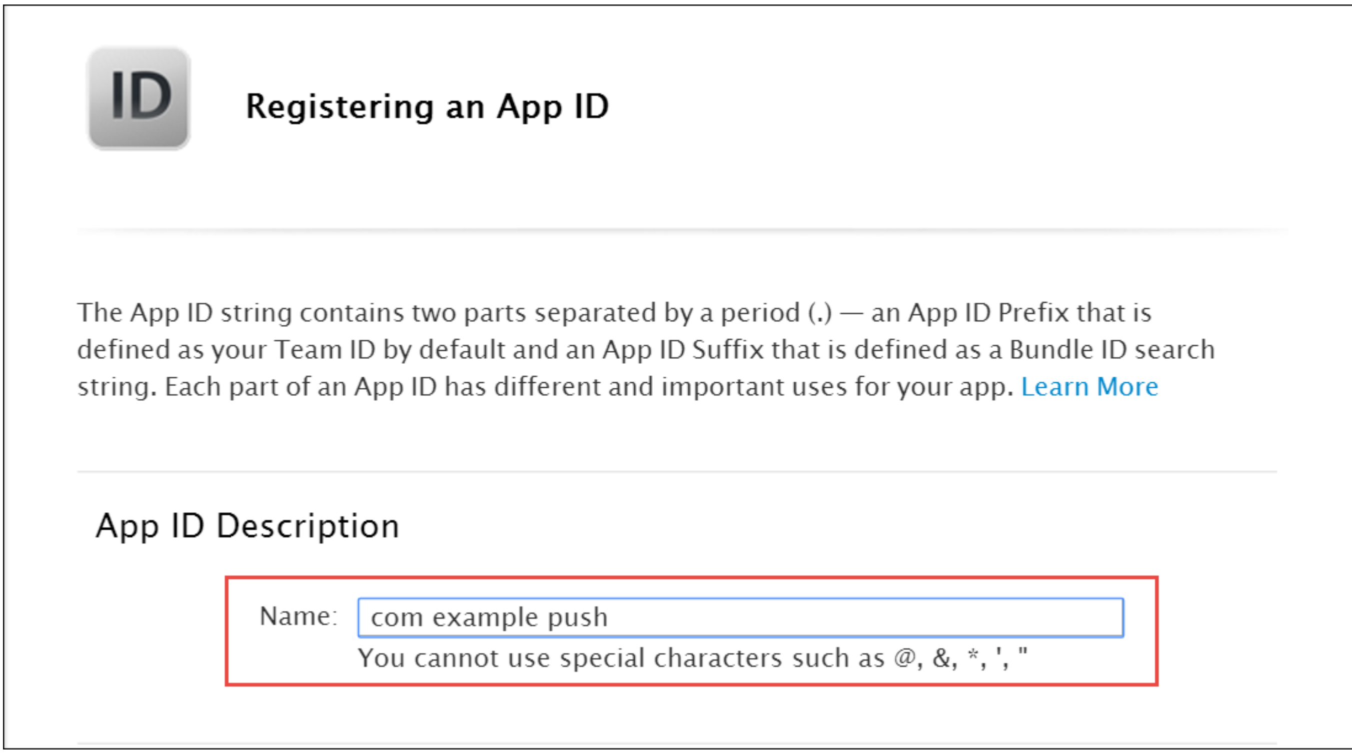
- Click *Certificates, IDs & Profiles*.



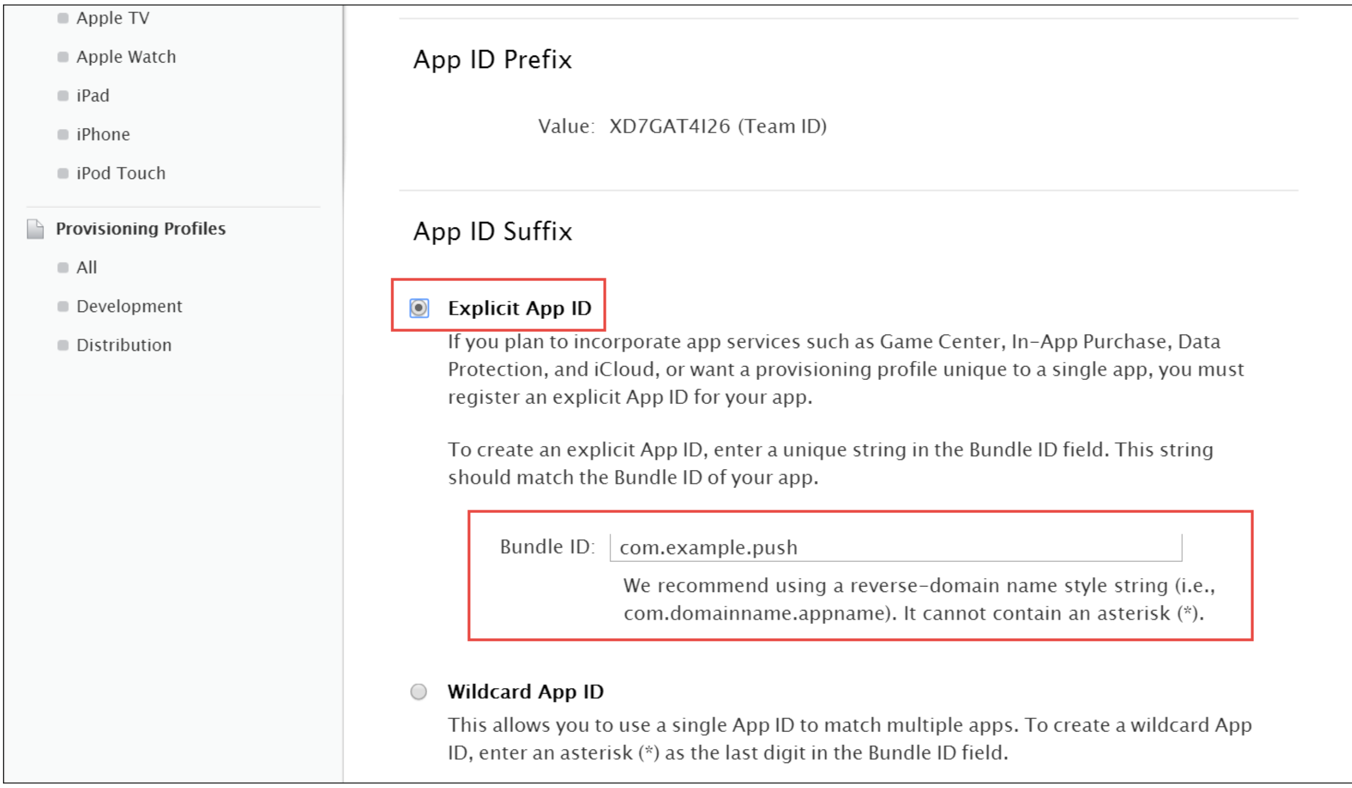
- Under *Identifiers*, select *App IDs*.



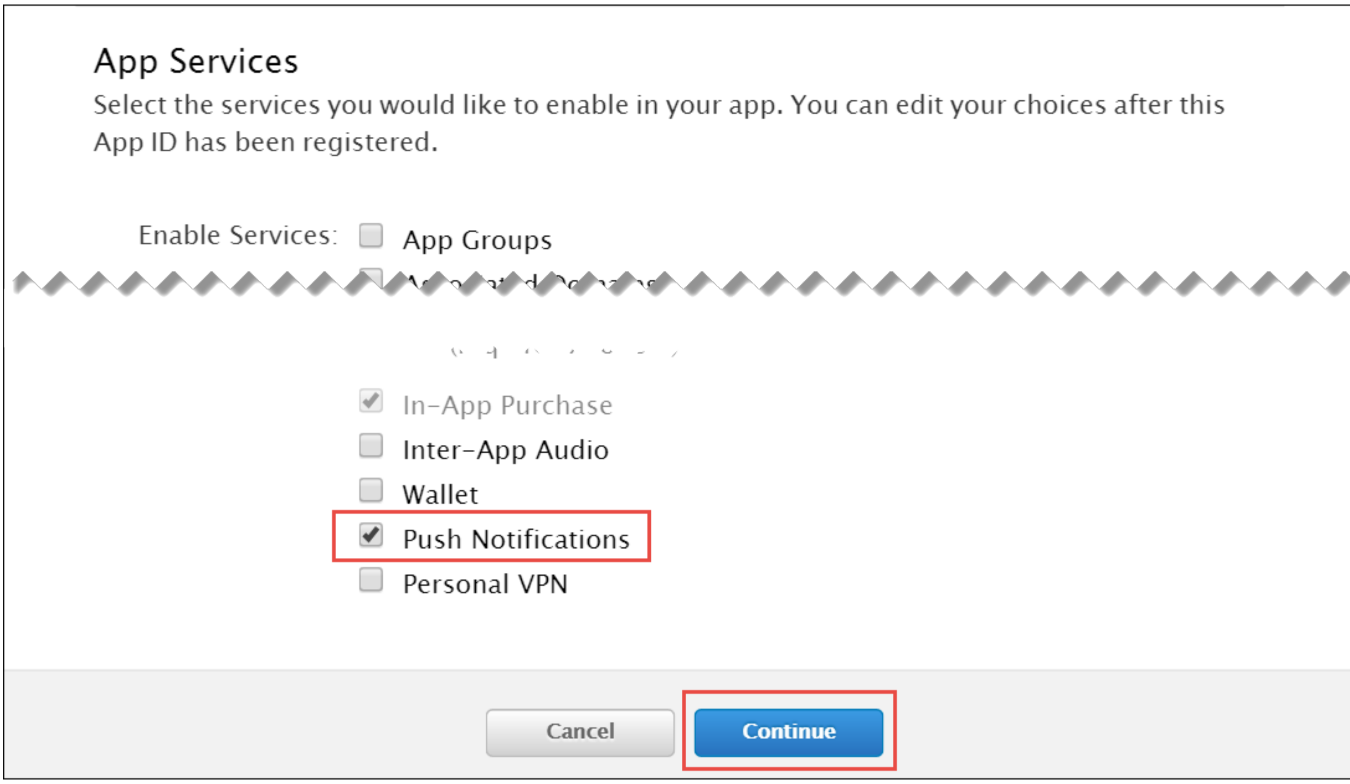
- Click the *Add* button (+) in the upper-right corner.



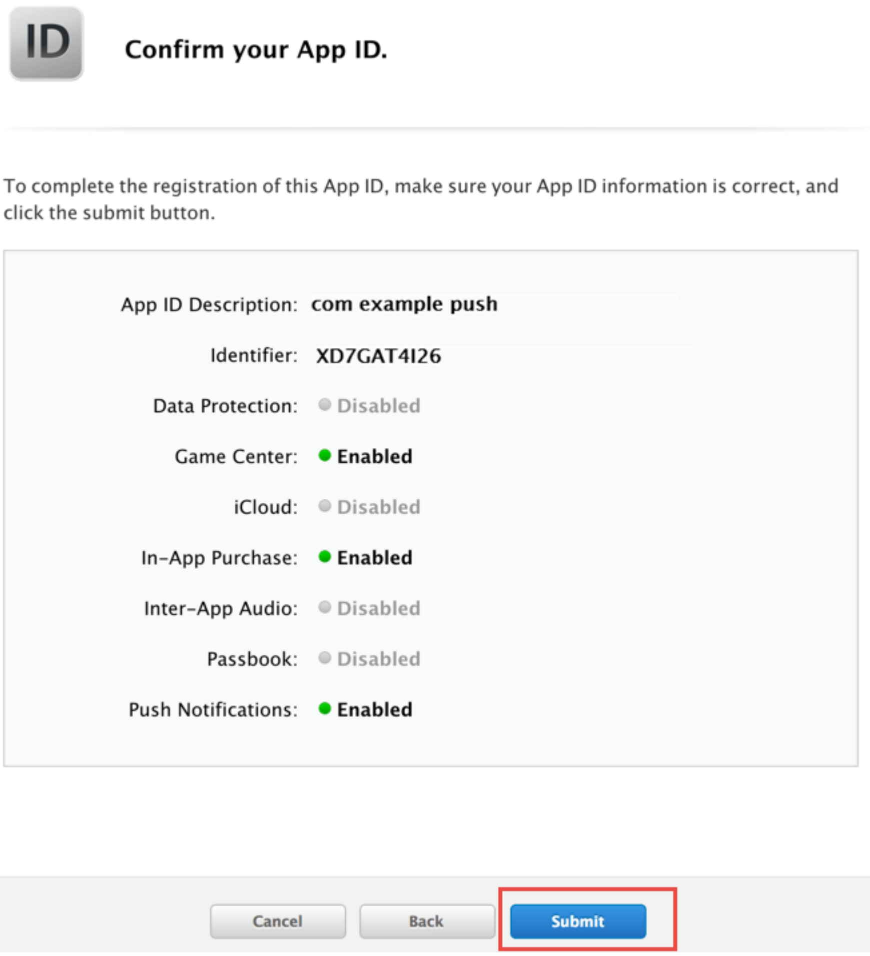
- Enter a name for the App ID in the *App ID Description* block. This helps you identify the App ID later.



- Select *Explicit App ID* and enter the app's bundle ID in the *Bundle ID* field. Note that an explicit App ID exactly matches the bundle ID of an app you are building — for example, com.example.push. An explicit App ID can *not* contain an asterisk (*).



- In the App Services section enable Push Notifications. Click *Continue* to submit the form

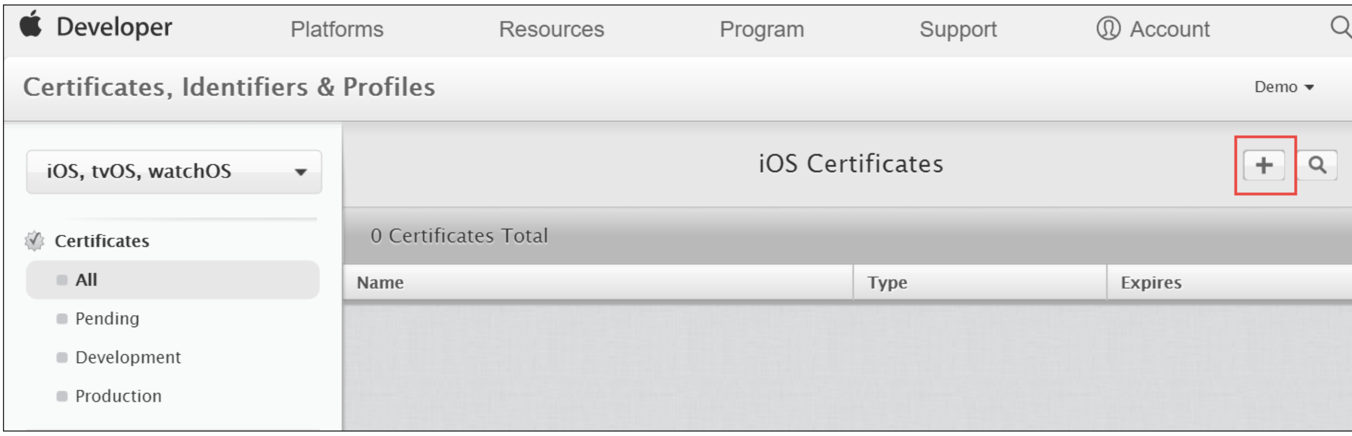


- Click *Submit* to create the App ID.

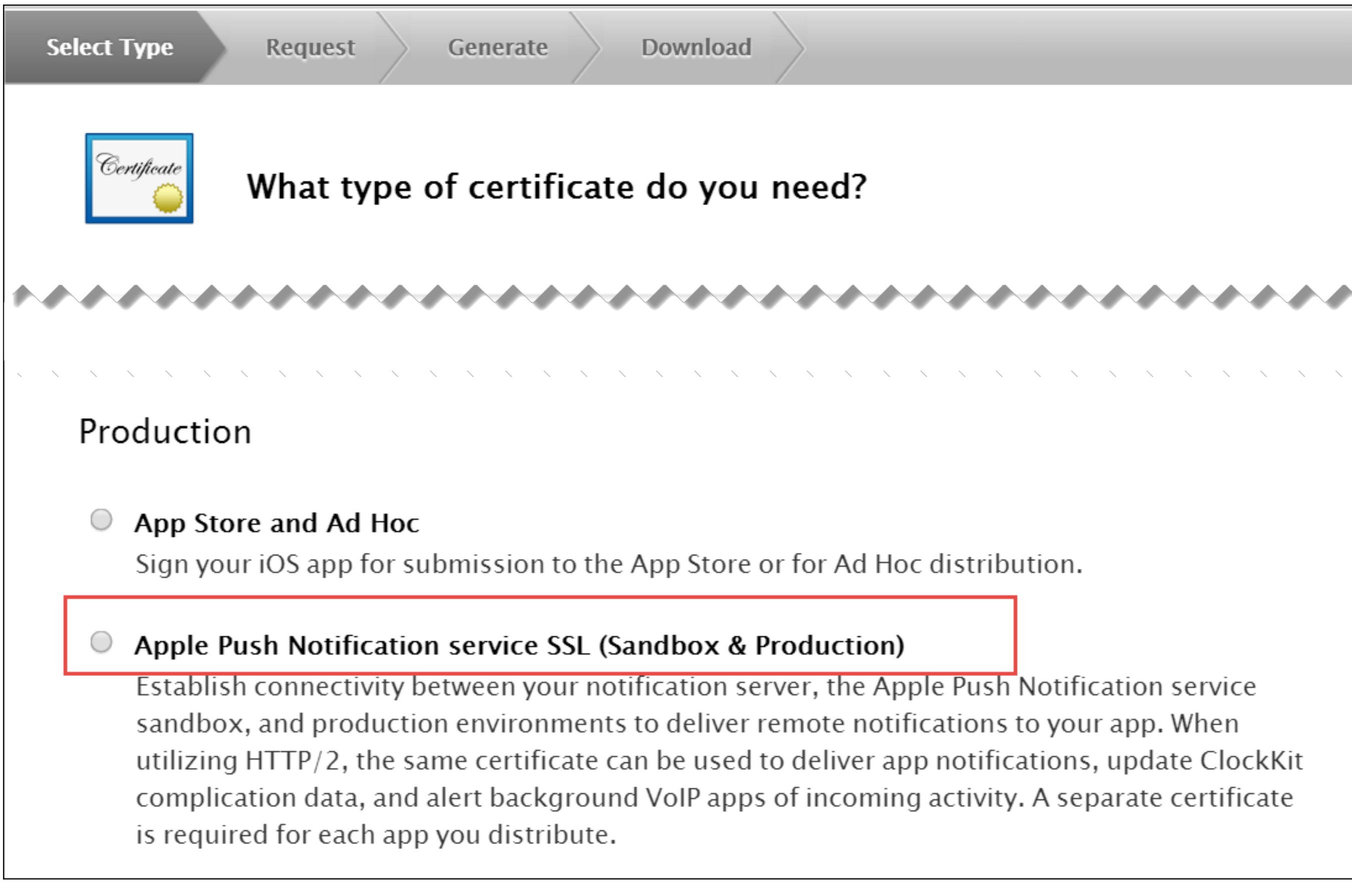
15.2.2.5 Obtain an Apple SSL Certificate and a Private Key

1. Create a CSR (Certificate Signing Request):

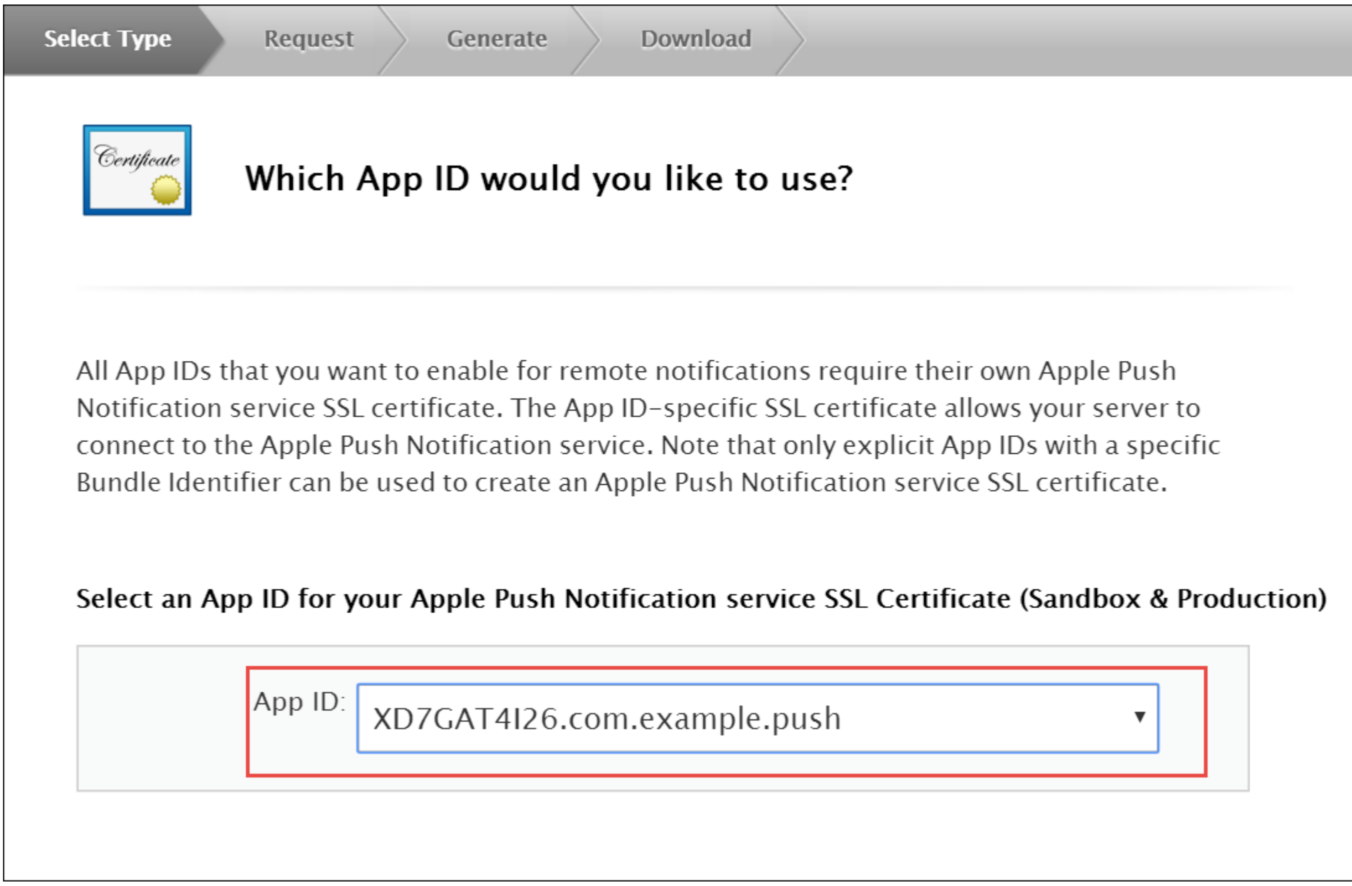
- Sign into developer.apple.com/account/ios/certificate.



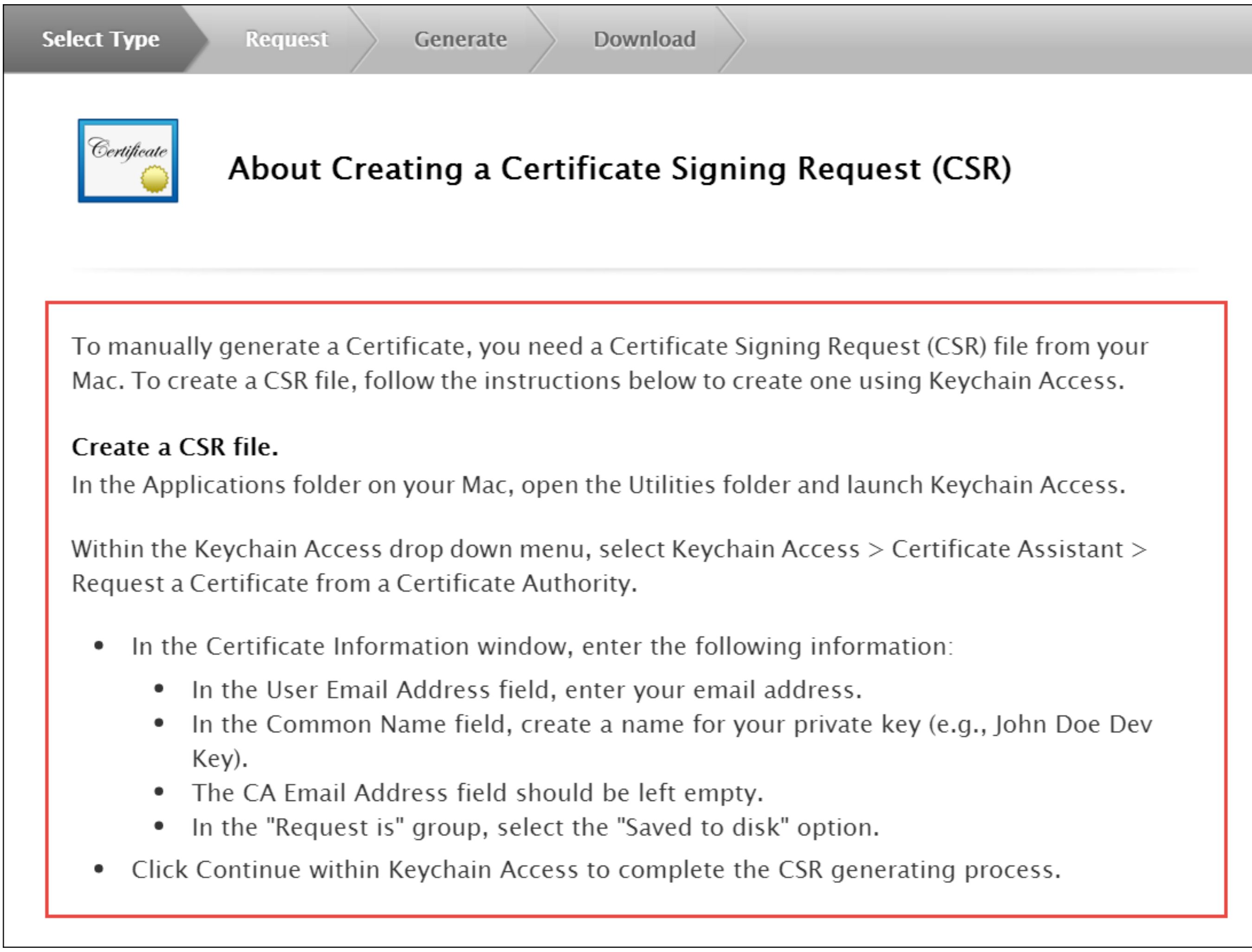
- Click the *Add* button (+) in the upper-right corner.



- Select *Apple Push Notification service SSL (Sandbox & Production)* as the certificate type and click *Continue*.



- Select your App ID and click *Continue*.



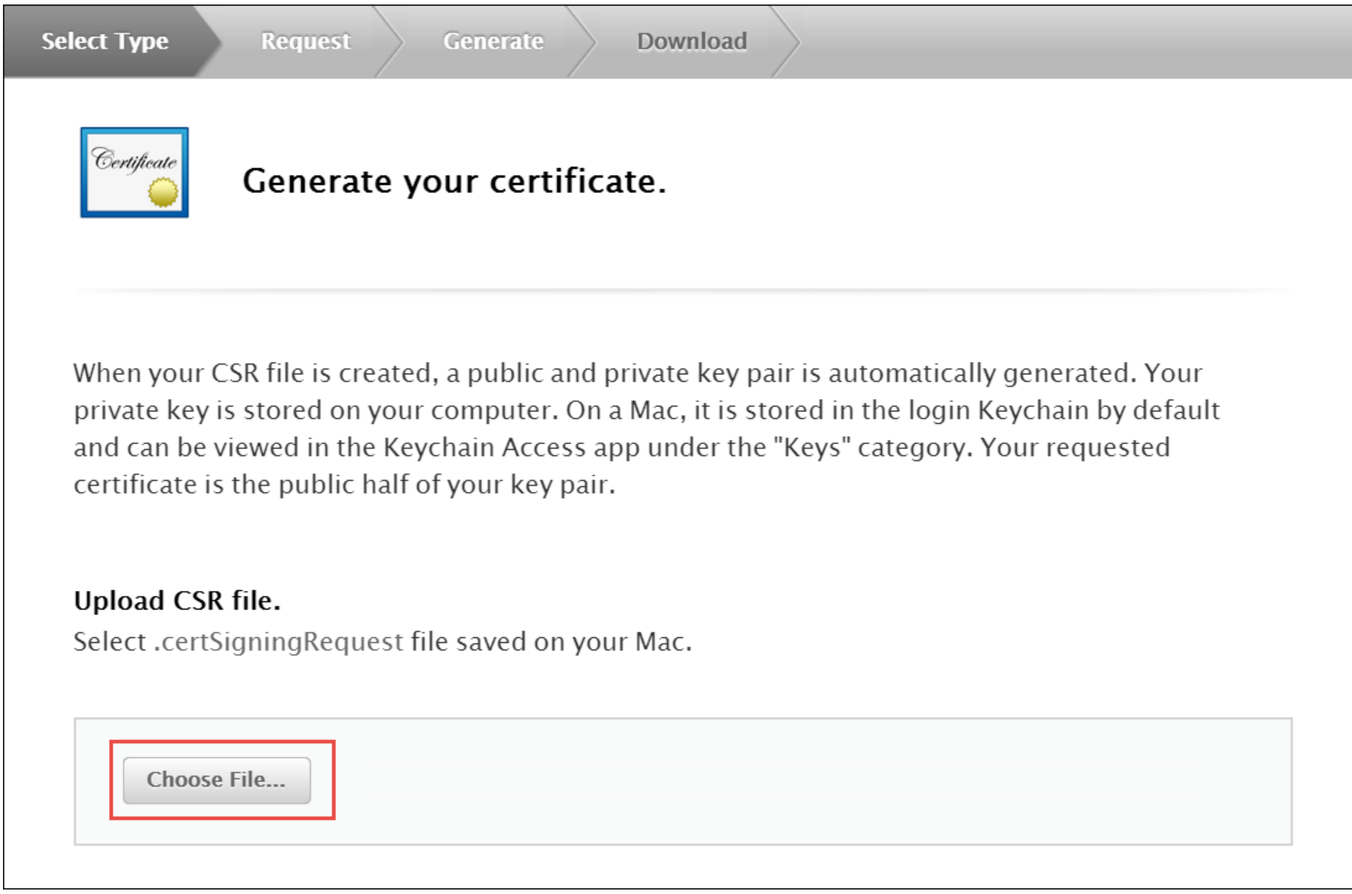
- Read the information about creating a CSR.
- Follow the instructions to create a CSR using Keychain Access in MAC.

Note

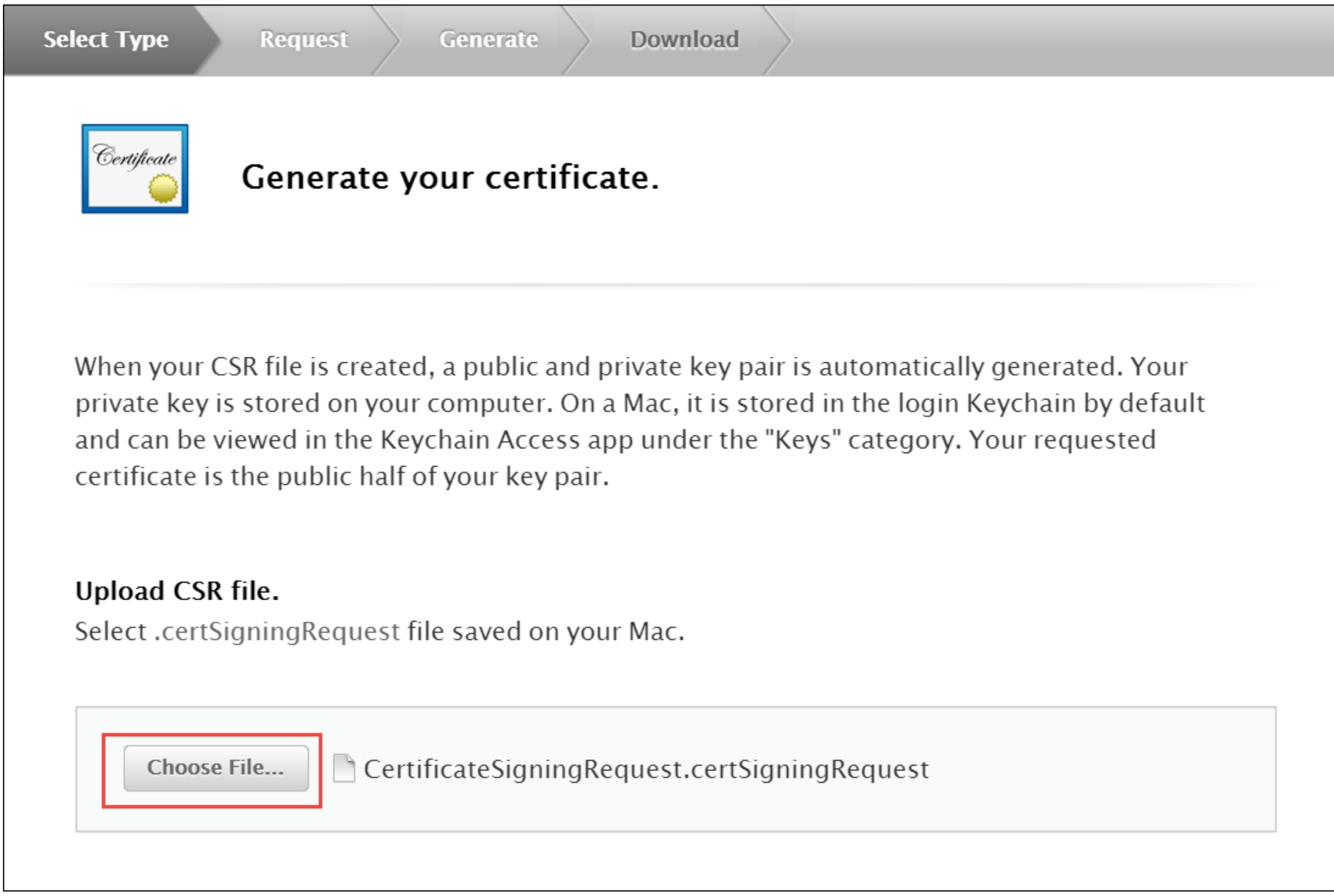
If you do not have access to a Mac, you can still create a CSR in Linux or Windows using OpenSSL, for example.

2. Get the Certificate and Private Key

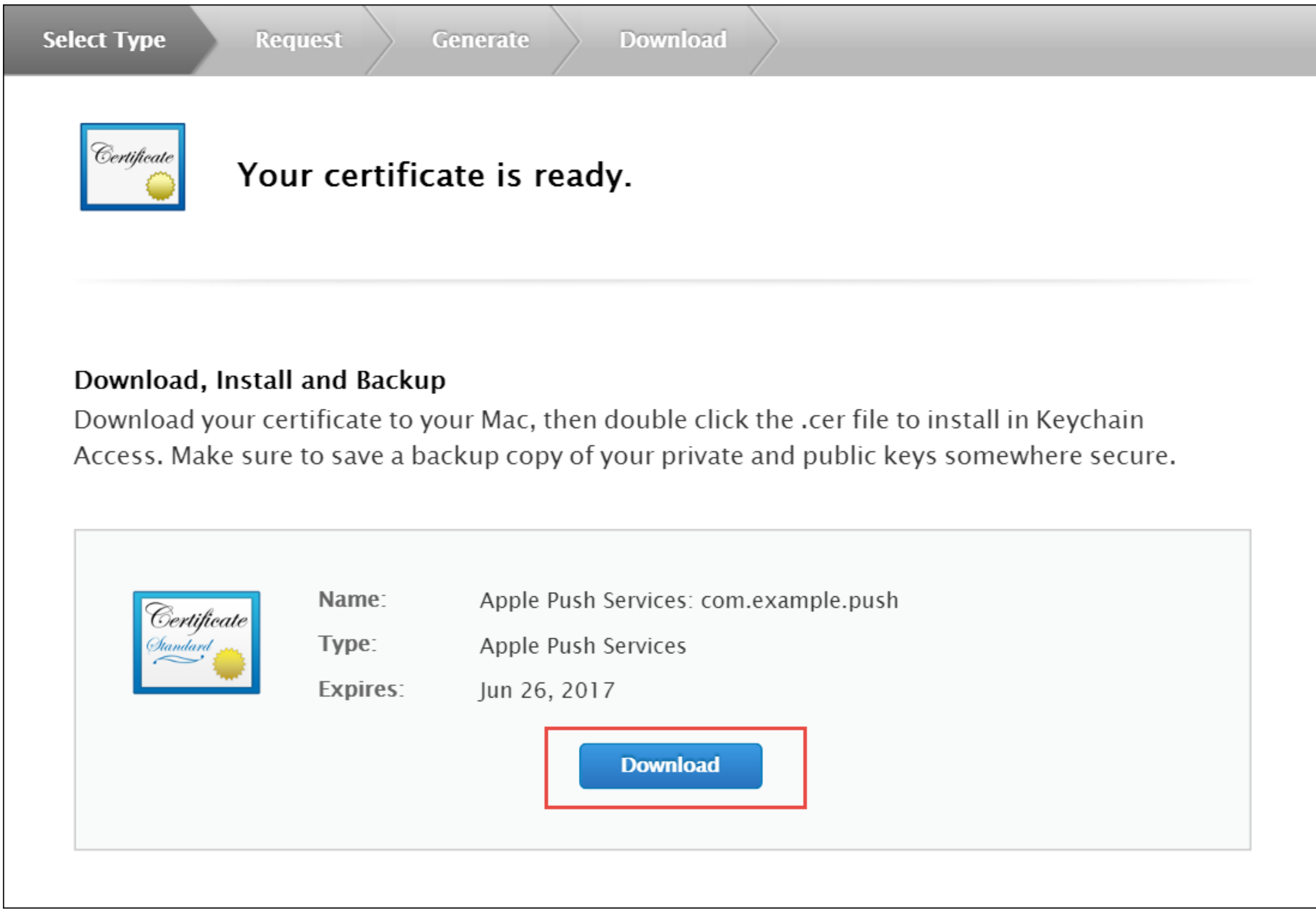
- When you have the CSR file return to the browser and click *Continue*.



- Click *Choose File...* in your browser.



- Select the CSR file you just created and saved and click *Continue*.



- Click *Download* to download the certificate (give it the **aps.cer** name).
- Open the downloaded certificate file (it should automatically be opened in Keychain Access, otherwise open it manually in Keychain Access).
- Find the certificate you just opened/imported in Keychain Access.
- Expand the certificate to show the Private Key.
- Select only the Private Key portion of the certificate, right-click on it and select *Export "Common Name"...* from the menu.
- Choose a location (e.g. Desktop) and filename to export the .p12 file to and click *Save*.
- **Optionally** pick a password for the .p12 file to protect its private key contents and click *OK*. (You will then need to enter your log-in password to permit the export).

3. Generate a PEM file from the p12 file:

   - Open up your terminal and run the following commands to create a PEM file from the p12 file (If you input a password for the p12 file, you will need to enter it here):

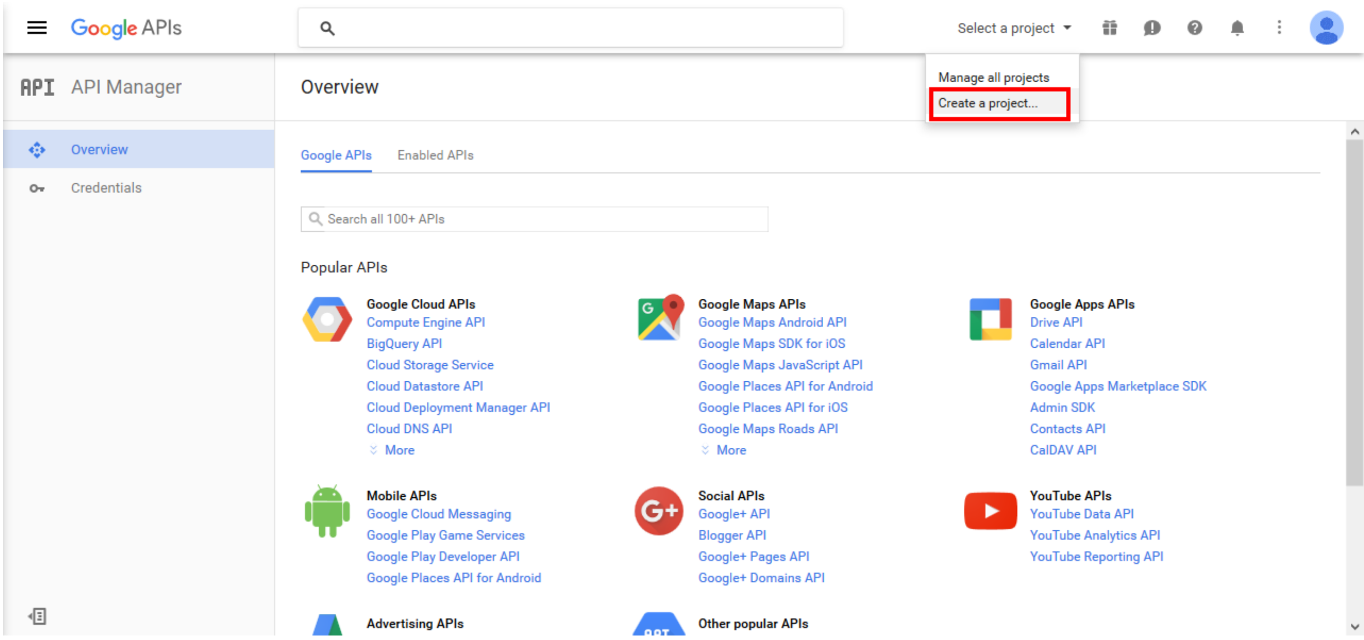
```
cd ~/Desktop
openssl x509 -in aps.cer -inform der -out PushChatCert.pem
openssl pkcs12 -in PushChatCert.p12 -out PushCertificate.pem -nodes -clcerts
openssl pkcs12 -nocerts -out PushChatKey.pem -in PushChatKey.p12
```

15.2.2.6 Obtain the API Key for the App from Google

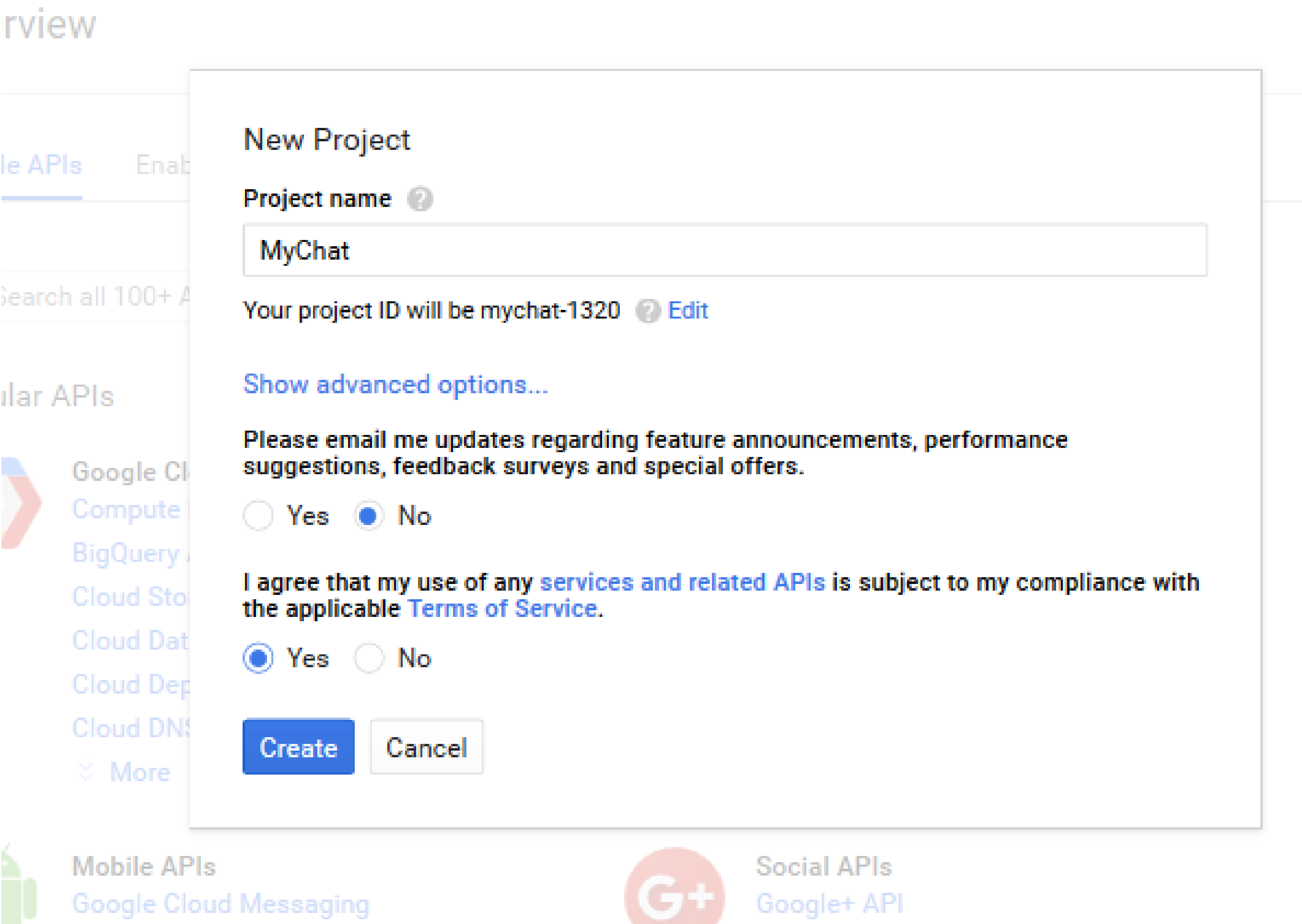
You can use Google Cloud Messaging (GCM) to send push notifications to your subscribers with Android-based mobile devices. Google Cloud Messaging is a free service that acts as an intermediary between the NGCP and devices of your subscribers. Google's Cloud Connection Server (CCS), a part of GCP, manages the persistent connections with mobile devices to deliver your push notifications.

While communicating with CCS, the NGCP identifies itself using an API key. To get it, follow the steps below.

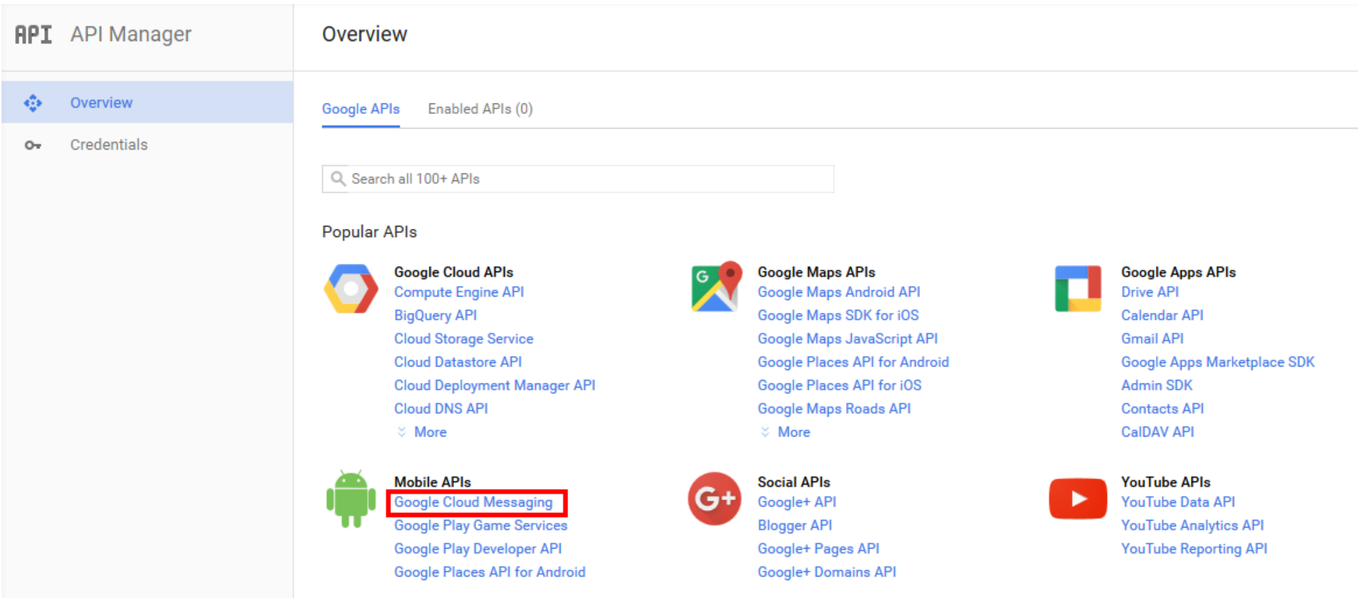
1. Create a new project in the Google APIs Console page. For this go to code.google.com/apis/console.



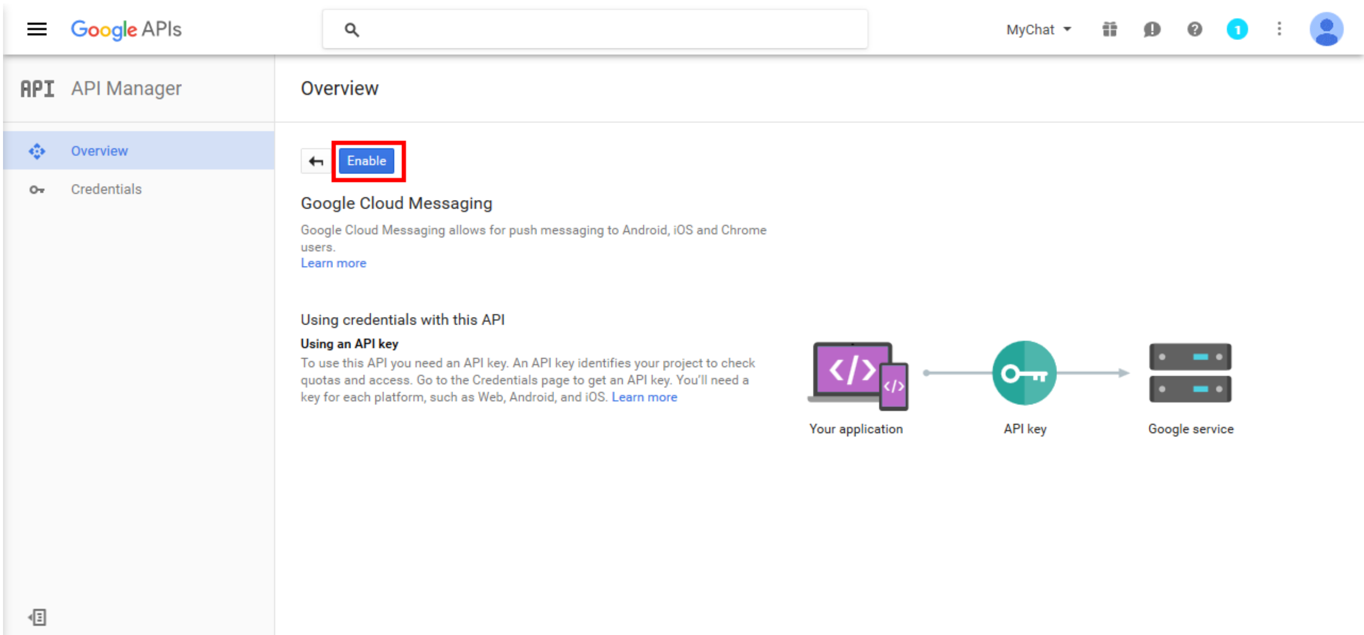
2. Click *Create a Project..*



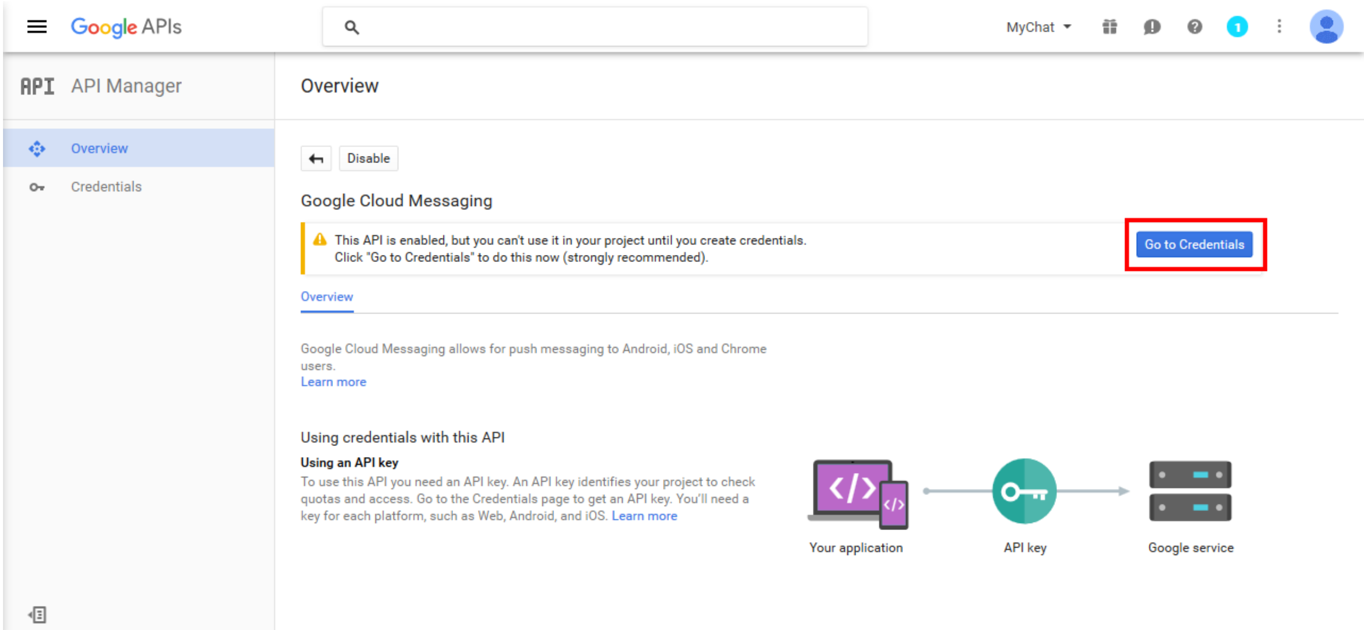
3. Input the project name, agree with the *Terms of Service* and click *Create*.



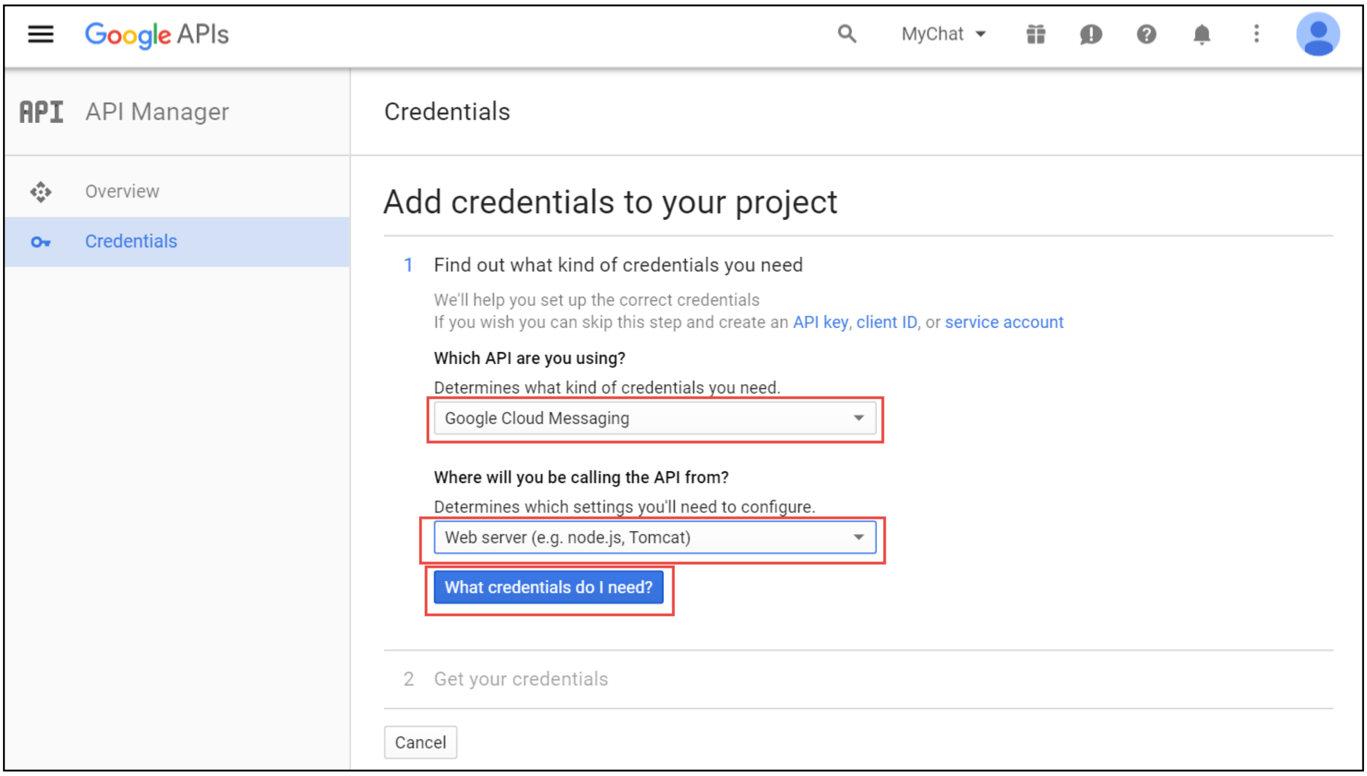
4. Click *Google Cloud Messaging* on the Overview page.



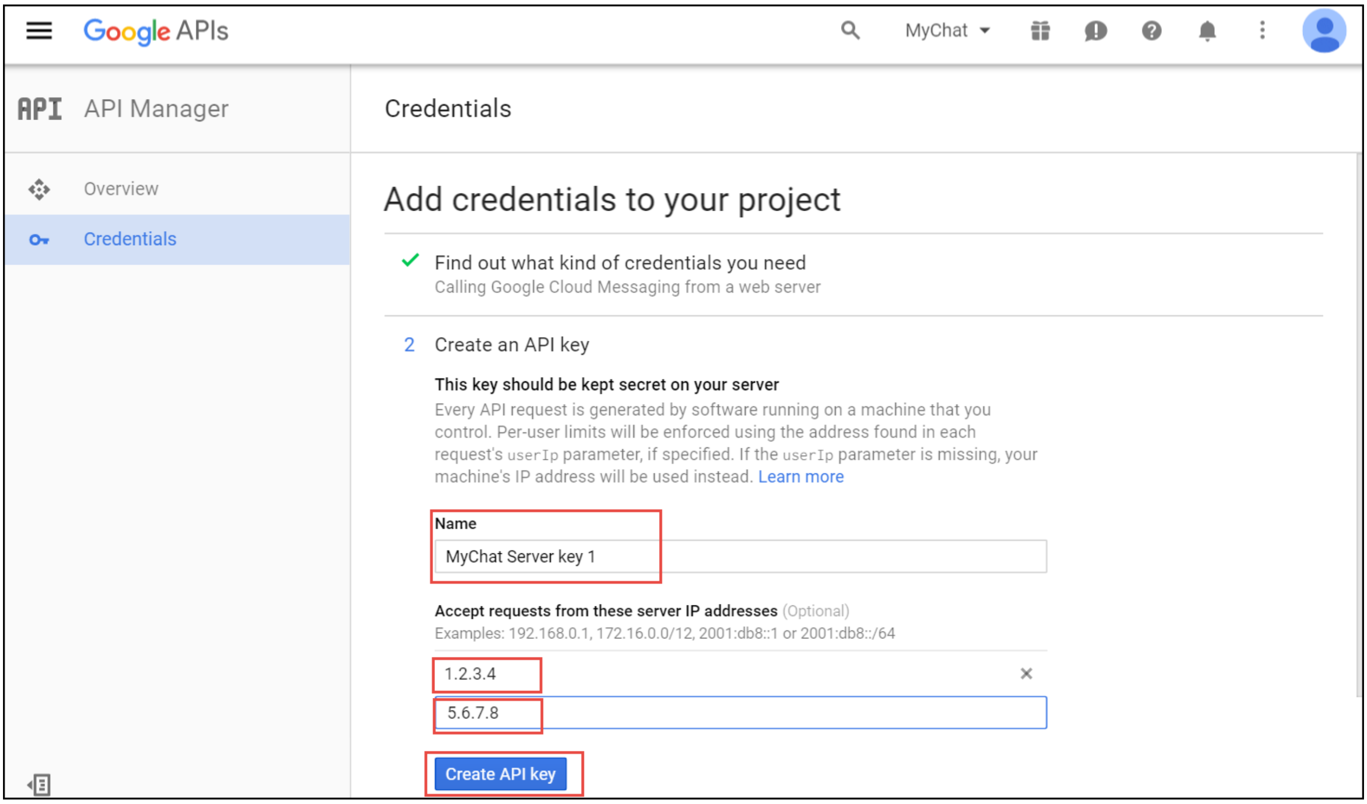
5. Click *Enable* for the Google Cloud Messaging.



6. Click *Go to Credentials*.



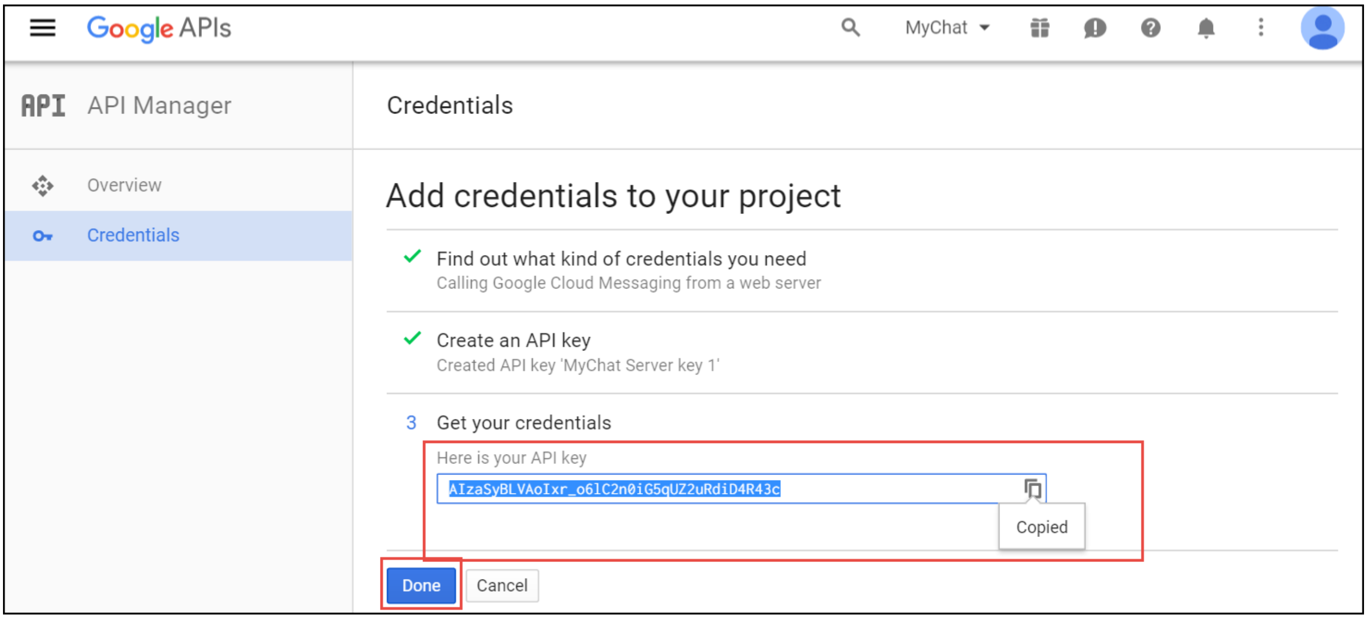
7. Select Google Cloud Messaging and Web Server from the corresponding lists and click *What credentials do I need*?



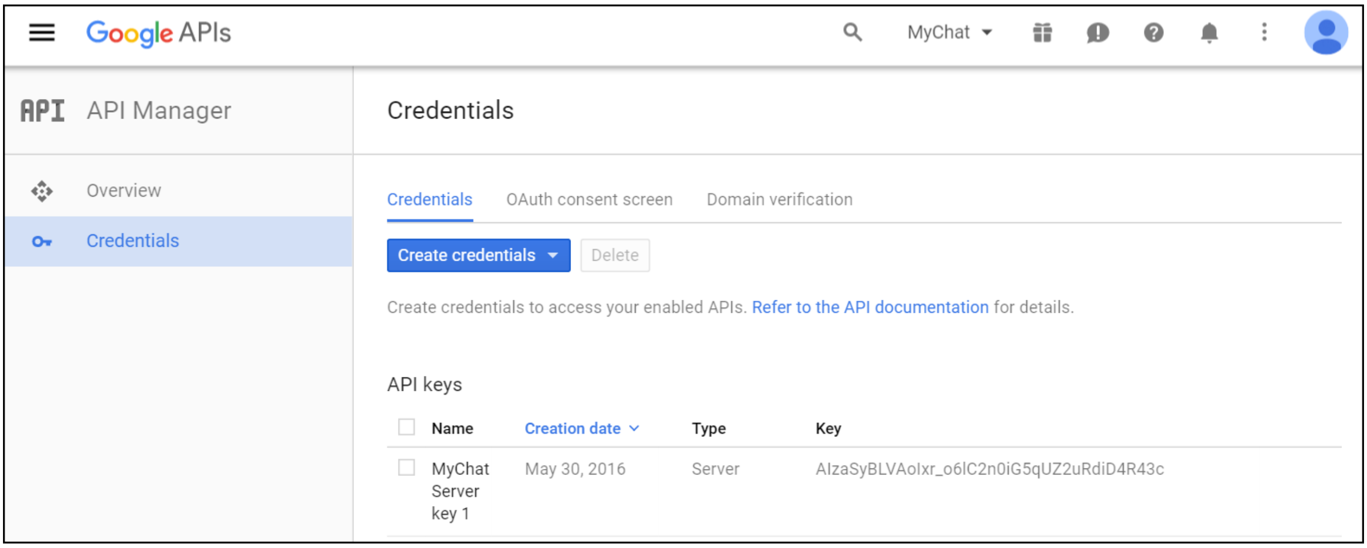
8. Adjust the API Key name and input the IP addresses of *all* your load balancers under *Accept requests from these server IP addresses*. Click *Create API key*.

Note

You may skip adding the IP addresses, otherwise list *ALL* your load balancers.



9. Copy your API key and click *Done*. Save the API key for future use.



15.2.2.7 Provide the Required Information to Developers

Please, provide Sipwise developers with the following files and information so that they can make beta builds and submit the application to the App Store:

- Access to your Apple developer account
- The trusted SSL certificate and its private key
- The Apple SSL certificate and its private key

For the Android application, provide the following:

- Access to your Google developer account
- Google application API key

15.2.2.8 Adjust the sip:provider PRO Configuration (Usually Performed by Sipwise)

1. Upload the Apple SSL certificate (**PushChatCert.pem**) and the private key (**PushChatKey.pem**) to */etc/ngcp-config/ssl/*
2. Upload the trusted SSL certificate (**CAsigned.crt**) and the private key (**CAsigned.key**) to */etc/ngcp-config/ssl/*
3. Specify the corresponding paths and names in the pushd section of the config.yml file:
   - apns: section (For iOS mobile application)
     - certificate: '/etc/ngcp-config/ssl/PushChatCert.pem'
     - enable: yes
     - key: '/etc/ngcp-config/ssl/PushChatKey.pem'
   - enable: yes
   - gcm: section (for Android mobile application)
     - enable: yes
     - key: 'google_server_api_key_here'
   - ssl: yes
   - sslcertfile: /etc/ngcp-config/ssl/CAsigned.crt
   - sslcertkeyfile: /etc/ngcp-config/ssl/CAsigned.key
     You can find an example of */etc/ngcp-config/config.yml* configuration in the config.yml overview section.
4. Apply your changes:

```
ngcpcfg apply 'enabled the backup feature.'
ngcpcfg push
```

15.2.2.9 Recheck Your DNS Zone Configuration

Check that your **NS** and **A** DNS records are correctly configured.

Let's consider the following example: * the load-balancers have the lb01a.example.com and the lb01b.example.com names * the shared name is lb01.example.com and the shared IP address is 1.1.1.1 * the service name is voipservice.example.com

The following DNS records must be present:

Server Name	Record type	IP Address
lb01a.example.com	A	1.2.3.4
lb01b.example.com	A	5.6.7.8
lb01.example.com	A	1.1.1.1
voipservice.example.com	A	1.1.1.1

15.2.2.10 Add SRV Records to DNS

Add at least one record for each service: **xmpp-server**, **xmpp-client**, **sips**.

A regular SRV record has the following form:

```
_service._proto.name. TTL class SRV priority weight port target
```

- service: the symbolic name of the service (xmpp-server, xmpp-client, sips).
- proto: the transport protocol of the desired service (TCP).
- name: the domain name (ending in a dot).
- TTL: standard DNS time to live field.
- class: the standard DNS class field (this is always IN).
- priority: the priority of the target host (lower value means more preferred).
- weight: a relative weight for records with the same priority (the higher the value, the more requests will be sent).
- port: the TCP or UDP port of the service.
- target: the canonical hostname of the machine providing the service (ending in a dot).

Here are examples of the SRV records:

```
_xmpp-server._tcp.voipservice.example.com. 18000 IN SRV 10 50 5269 voipservice.example.com.
_xmpp-client._tcp.voipservice.example.com. 18000 IN SRV 10 50 5222 voipservice.example.com.
_sips._tcp.voipservice.example.com. 18000 IN SRV 10 100 5061 voipservice.example.com.
```

You can always check whether the required SRV records are configured by executing the following commands:

```
dig SRV _xmpp-client._tcp.voipservice.example.net
dig SRV _xmpp-server._tcp.voipservice.example.net
dig SRV _sips._tcp.voipservice.example.net
```

15.2.2.11 Check NTP Configuration

We strongly suggest that the clocks of all the nodes within the platform are synchronized. To ensure this, check that the NTP service is correctly configured on all your sip:provider PRO servers and works reliably. Execute the following command for quick test of time synchronization:

```
ntpq -p
```

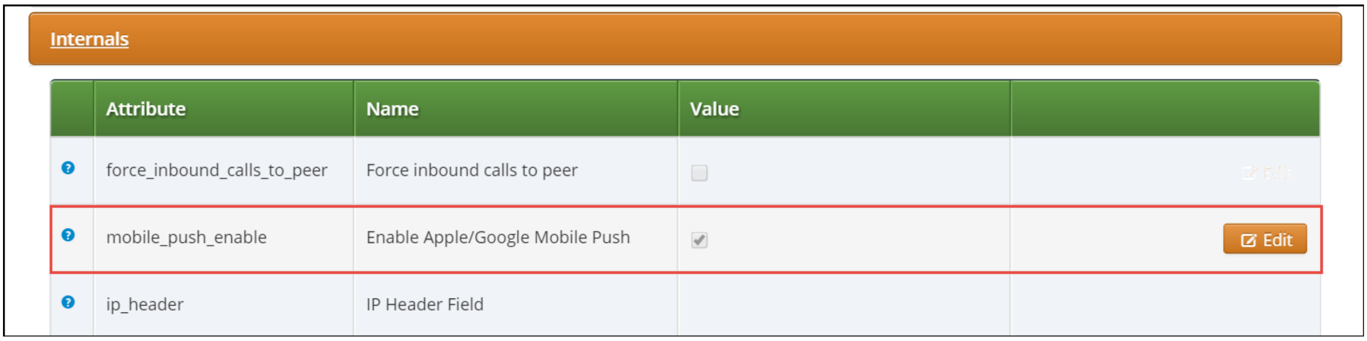
If the current node synchronizes with an NTP server, this server will be marked by the star (*) symbol.

15.2.2.12 Enable Apple/Google Mobile Push

It can be enabled for a domain or separate subscribers in the Admin Panel.

To enable the service for a domain:

1. Go to *Settings→Domains* and click on the *Preferences* button of the domain you want to enable Apple/Google Mobile Push for.
2. Go to the *Internals* group and enable the **mobile_push_enable** parameter.



15.2.2.13 Perform Tests

Perform tests when the application is available:

1. Download and install the application.
2. Open the application and input your registration username in the username@domain.name format and password.
3. Review the quality of application branding.
4. Make test calls.
5. Test the presence functionality.
6. Test the chat and group chat.
7. Test messaging.
8. Test the sharing functionality (e.g. pictures, video and voice messages and maps).
9. Check the application phone book integration with the phone's one

Make sure that the subscribers can start using your services in the easiest possible way.

15.3 Lawful Interception

15.3.1 Introduction

The Sipwise sip:provider PRO, as a communications platform carrying voice, fax and messaging data has to provide means for lawful interception of the content of communication by third party entities. Those Law Enforcement Agencies (LEAs) have to be able to connect to the Sipwise NGCP platform in a standardized way — ETSI, 3GPP and other organisations define the interface (and data exchange) between telecommunication operators and LEAs.

High level overview of lawful interception is shown in the following figure:

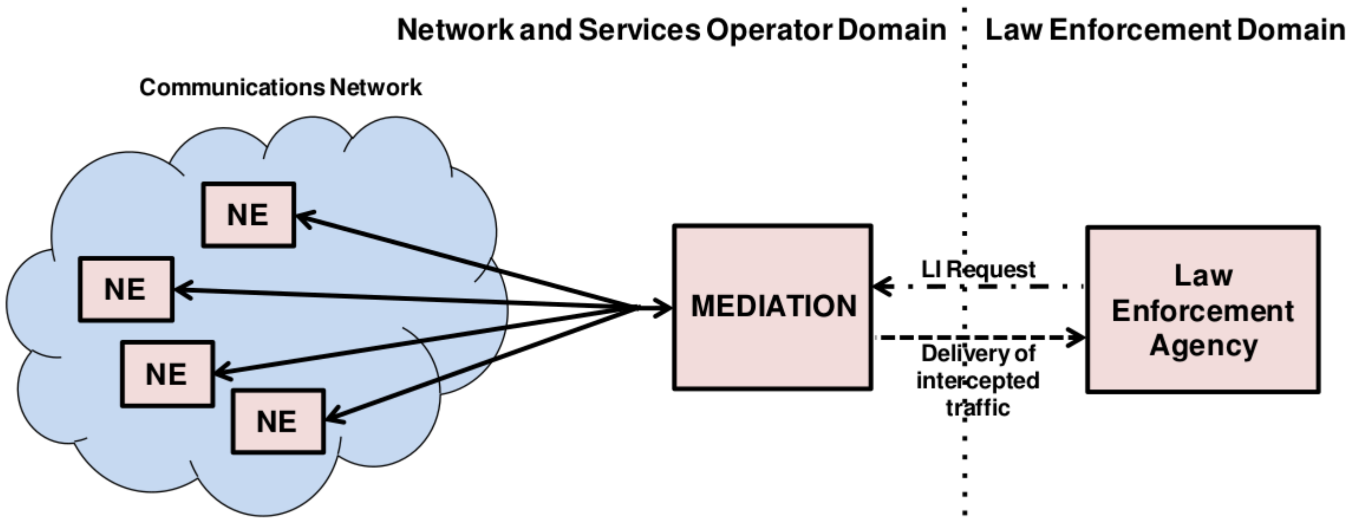


Figure 97: LI: High Level Overview

Main interfaces of lawful interception according to ETSI standard:

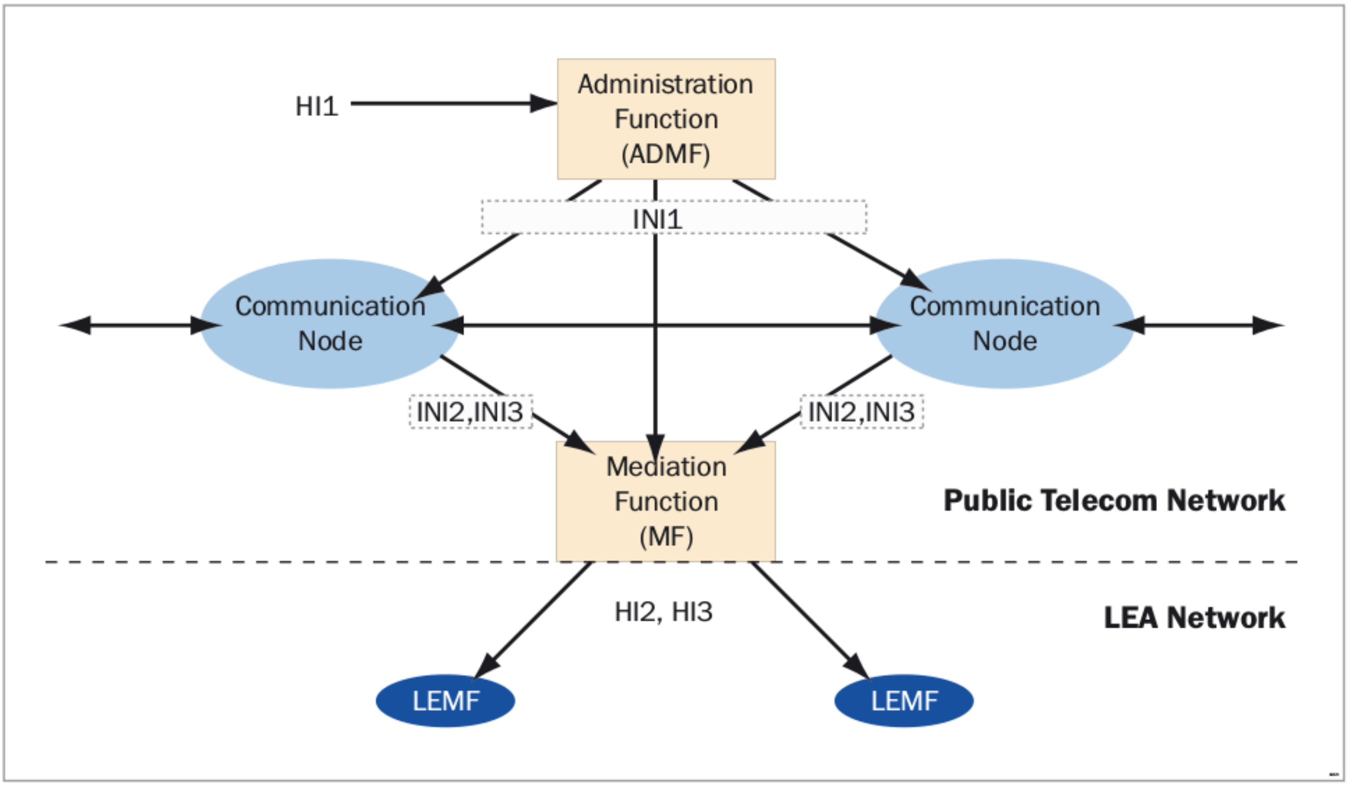


Figure 98: LI: ETSI Interfaces

15.3.1.1 Terms and Abbreviations

Content of Communication (CC)

Information exchanged between two or more users of a telecommunications service, excluding Intercept Related Information.

> **Note**
> This includes information which may, as part of some telecommunications service, be stored by one user for subsequent retrieval by another.

CC Internal Interception Function (CC-IIF)

The CC-IIF shall cause the CC, specified by the CCTF, via the CCCI to be duplicated and passed to the MF.

Content of Communication Control Interface (CCCI)

Carries controls information from the CCTF to the CC-IIF.

CC Trigger Function (CCTF)

The purpose of the CCTF is to determine the location of the CC-IIF device associated to the target CC traffic, and to control the CC-IIF via the CCCI interface.

Content of Communication Trigger Interface (CCTI)

Carries trigger information from the IRI-IIF to the CCTF.

Handover Interface (HI)
Physical and logical interface across which the interception measures are requested from an operator, and the results of interception are delivered from an operator to an LEMF.

Intercept Related Information (IRI)
Collection of information or data associated with telecommunication services involving the target identity, specifically call or service associated information or data (e.g. call identifier, unsuccessful call attempts) and location information.

Intercept Related Information Internal Interception Function (IRI-IIF)
The purpose of the IRI-IIF is to generate IRI information associated with sessions, calls, connections and any other information involving interception targets identified by Law Enforcement Agency (LEA) sessions.

Internal Network Interface (INI)
Network's internal interface between the Internal Intercepting Function and a mediation function.

Law Enforcement Agency (LEA)
Organization authorized, by a lawful authorization based on a national law, to request interception measures and to receive the results of telecommunications interceptions.

Law Enforcement Monitoring Facility (LEMF)
Law enforcement facility designated as the transmission destination for the results of interception relating to a particular interception subject.

Lawful Interception Administration Function (AF)
The AF ensures that an intercept request from a LEA for IRI or CC or both is provisioned for collection from the network, and subsequent delivery to the LEMF.

Lawful Interception Mediation Function (MF)
Mechanism which passes information between an access provider or network operator or service provider and a handover interface.

1. Firstly it receives information related to active intercepts from the IRI-IIF(s) and CC-IIF(s) within the service provider network.
2. Secondly correlates and formats that IRI and CC information in real time for delivery to the LEMF over the HI2 and HI3 handover Interfaces.

X1, X2 and X3 Interfaces
The 3GPP standard for Lawful Interception defines the handover interfaces with different names compared to the ETSI standard. The X*n* interface corresponds to the INI*n* interface and is functionally identical to the INI*n* interface.

15.3.2 Architecture and Configuration of LI Service

Sipwise sip:provider PRO platform implements the functions defined by LI requirements in a way that it relies on a third party provider for the Lawful Interception Mediation Function (MF).

Regarding other LI functions that are defined by ETSI / 3GPP standards there are 2 possible implementations:

1. Sipwise NGCP behaves as the Administration Function (AF) but the actual call data capturing is carried out by other SIP endpoints. In this case NGCP forwards the calls to be intercepted to its **SIP peers dedicated for LI service**. Within the scope of SIP peer based solution there are still 2 modes of operation:

   - *Call loopback to NGCP*: the LI peer receives the call, extracts IRI and CC data and then routes the call back to NGCP. NGCP handles the looped back call as if that was initiated from NGCP and sets up the second call leg to the destination.
   - *Call forwarded by peer* directly to destination: in this case NGCP will handle the call to LI peer as an ordinary second call leg to the destination.

2. Sipwise **NGCP itself provides** the required LI functions: AF and call data capturing; IRI and CC of intercepted calls are fowarded to the third party MF from NGCP.

This handbook will discuss the second setup in details in the following sections.

The below figure illustrates the logical connection of LI functions on Sipwise NGCP.

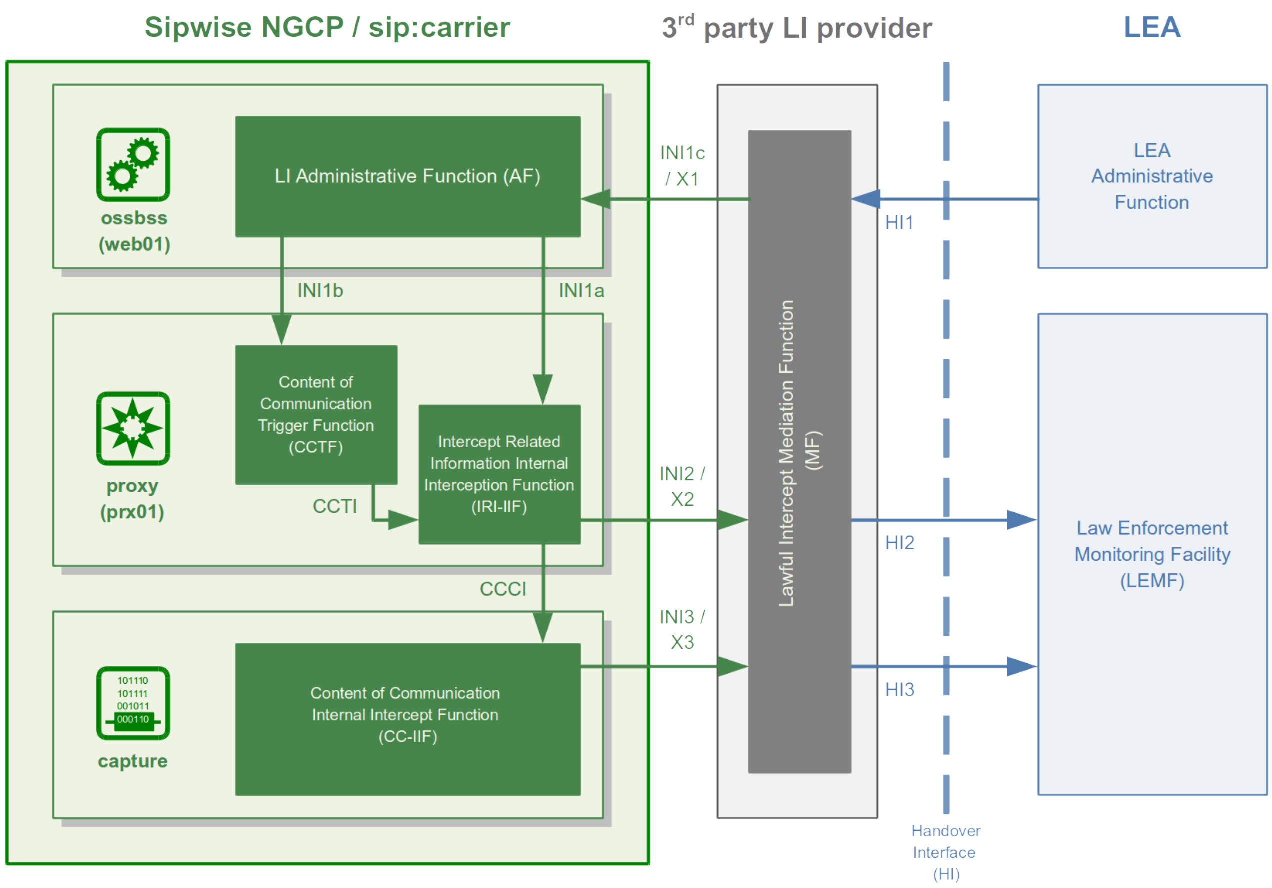


Figure 99: LI with 3rd Party Provider

15.3.2.1 Architecture Based on Captagent Module

Note

This kind of LI implementation will be phased out in future NGCP releases. A short description is kept here for reference, as NGCP still (as of version mr4.5.2) supports LI services with `captagent` module.

The `captagent` based implementation of LI functions on Sipwise NGCP includes the following components:

- **captagent**: a software module provided by a third party; its operation can be summarized as follows:
  1. the `captagent` process gets LI requests through an API
  2. the process listens for and analyses SIP (*INVITE*) messages; based on the message headers it decides whether the SIP session must be intercepted
  3. in case the session must be intercepted, `captagent` sends IRI through X2 interface to the MF element
  4. based on the SDP data, the process captures session media and forwards that through X3 interface to the MF element
- **third party MF**: Group2000's LIMA system playes the role of Mediation Function (MF) and interacts with `captagent` module, using X1, X2 and X3 interfaces.

15.3.2.2 Architecture Based on Voisniff-NG Module

Although the implementation of LI services with `captagent` is still available and configurable on sip:provider PRO, Sipwise suggests deploying a revised solution with its `voisniff-ng` software module. This newer implementation also relies on a 3rd party LI provider representing the LI Mediation Function (MF), where Sipwise currently (as of NGCP version mr4.5.2) cooperates with Group2000, Pine and Utimaco.

Sipwise NGCP components providing LI functions:

- **ngcp-panel:** this module is responsible for managing REST API for the whole NGCP in general
  - *runs on:* the active node (`sp1 /sp2`) on a sip:provider PRO platform
  - *LI functions:* AF; INI1 / X1 interface towards the MF
- **kamailio-proxy:** this module serves as a generic call control function on the NGCP
  - *runs on:* the active node (`sp1 /sp2`) on a sip:provider PRO platform
  - *LI functions:* CCTF and IRI-IIF; INI2 / X2 interface towards the MF
- **voisniff-ng:** this module is a generic element for capturing SIP and RTP traffic on the NGCP
  - *runs on:* the active node (`sp1 /sp2`) on a sip:provider PRO platform
  - *LI functions:* CC-IIF; INI3 / X3 interface towards the MF

Note

The `voisniff-ng` module is installed and activated by default on a Sipwise sip:provider PRO platform. It provides a possibility to get call statistics through Admin web interface, and is not readily configured for LI services.

Authentication and Confidentiality

It is required that the communication between the telecommunication operator's network element (that is: Sipwise NGCP) and the MF be authenticated and confidential, since the intercepted session related data and content of communication must not be disclosed to any 3rd party. For this purpose NGCP's LI service applies authentication and LI session data encryption based on public key cryptography mechanism (TLS).

Both Sipwise NGCP and the MF must authenticate themselves by certificates, for this reason the NGCP operator must ensure that valid certificates are deployed on the system. There is a need to contact the 3rd party LI provider, so that he can provide the necessary client certificates that NGCP will use to setup secured connection to the MF on X2 and X3 interfaces.

Similarly, the MF provider must contact the NGCP operator to offer him valid client certificates that the MF element will use to establish secured connection to the NGCP on X1 interface.

15.3.2.3 Configuration of LI Service

In order to enable LI services on sip:provider PRO the platform administrator has to explicitly enable lawful interception through the main configuration file (`config.yml`).

Here below is a sample configuration, which shows parameters of `intercept` and `voisniff` sections.

```
intercept:
  captagent:
    cin_max: '3000'
    cin_min: '0'
    country_code: '49'
    debug: '7'
    filter: 'port 5080'
    license: ''
    port: '18090'
    prefix_len: '3'
    schema: http
  enabled: no
  peer:
    acc: no
    inbound_prefix: LI_
    outbound_prefix: intercept_
  type: none

voisniff:
  admin_panel: yes
  daemon:
    bpf: 'port 5060 or 5062 or ip6 proto 44 or ip[6:2] & 0x1fff != 0'
```

```
    external_interfaces: 'eth0 eth2'
    filter:
      exclude:
        -
          active: '0'
          case_insensitive: '1'
          pattern: '\ncseq: *\d+ +(register|notify|options)'
      include: []
    internal_interfaces: lo
    li_x1x2x3:
      call_id:
        suffix:
          - _pbx-1
          - _b2b-1
      client_certificate: ''
      enabled: no
      local_name: sipwise
      x1:
        port: '18090'
    mysql_dump:
      enabled: yes
      num_threads: '4'
    mysql_dump_threads: '2'
    start: yes
    threads_per_interface: '2'
  partitions:
    increment: '700000'
    keep: '10'
```

Configuration Parameters

`intercept.enable`
: Set it to `yes` if you want to activate LI service. Default: `no`

`intercept.peer.acc`
: Calls to be intercepted may be forwarded to LI peers. The LI peer may forward the call to the original destination, without looping the call back to NGCP. Set this parameter to `yes` if you want to enable billing for such calls. Default: `no`

`intercept.peer.inbound_prefix`
: Calls to be intercepted may be forwarded to LI peers. This parameter specifies the prefix that is prepended to SIP usernames when the call is looped back to NGCP, in order to avoid sending the call again to any LI peer. Used by NGCP internally. Default: `LI_`

`intercept.peer.outbound_prefix`
: Calls to be intercepted may be forwarded to LI peers. This parameter specifies the prefix that is prepended to SIP usernames when the call is routed to an LI peer. It will be stripped off by rewrite rules of the peer, before sending the call effectively to the peer. Used by NGCP internally. Default: `intercept_`

intercept.type

The LI service provider module; allowed values are:

- `none`: LI service is not activated
- `peer`: LI service is activated and call data capturing is performed by SIP peers
- `captagent`: LI service is activated and call data capturing is performed by `captagent` module
- `voisniff`: LI service is activated and call data capturing is performed by `voisniff` module

Default: `none`

voisniff.admin_panel, voisniff.daemon.mysql_dump.*, voisniff.partitions.*

These parameters are not used in LI configuration, but only for call statistics which can be retrieved through the Admin web interface.

voisniff.daemon.bpf

This sets the basic packet filter applied by `voisniff-ng` module when capturing packets on network interfaces. Default: `"port 5060 or 5062 or ip6 proto 44 or ip[6:2] & 0x1fff !=0"`

> **Note**
>
> The default value basically allows capturing SIP traffic only. It is usually necessary to modify the parameter in order to capture both SIP and RTP traffic. An example of such a value: `"udp or ip6 proto 44 or ip[6:2] & 0x1fff !=0"`.

voisniff.daemon.external_interfaces

This is a list of network interfaces (typically VLAN IDs) where `voisniff-ng` should listen for and capture packets.

voisniff.daemon.filter.exclude

Additional filter to determine packets that need to be excluded from capturing.

voisniff.daemon.filter.include

Additional filter to determine packets that need to be included in capturing.

voisniff.daemon.internal_interfaces

A list of network interfaces which are considered only for internal communication between `voisniff-ng` and other NGCP components. Packets on these interfaces are not captured.

voisniff.daemon.li_x1x2x3.call_id

Pattern that determines which SIP Call-IDs should `voisniff-ng` listen for and store as IRI (Intercept Related Information).

voisniff.daemon.li_x1x2x3.client_certificate

The client certificate that NGCP uses to connect over TLS to a 3rd party LI provider.

voisniff.daemon.li_x1x2x3.enabled

Set it to `yes` to enable LI services via X1, X2 and X3 interfaces. Default: `no`

voisniff.daemon.li_x1x2x3.local_name

This parameter maps to the `header.source` field of the X2 protocol. It's an arbitrary string and can be used to identify the sending NGCP system. Default: `sipwise`

Note

As of NGCP version mr4.5.2, this is currently not used.

---

voisniff.daemon.li_x1x2x3.private_key

The private key used to encrypt data sent to a 3rd party LI provider.

voisniff.daemon.li_x1x2x3.x1.port

The port number on which `voisniff-ng` listens for incoming X1 messages. Default: `18090`

Note

You should leave the parameter set to the default value, unless there is a good reason to change it. The default value ensures backward compatibility with `captagent` LI module.

---

voisniff.daemon.start

Determines whether `voisniff` service must be started on the platform. Set it to `yes` if you'd like to activate `voisniff` that is needed for LI service too. Default: `yes`

voisniff.daemon.threads_per_interface

This is a performance tuning option and controls how many threads per enabled sniffing interface should be launched. Example: if it's set to 10 and 3 interfaces are enabled for sniffing, a total of 30 threads will be launched. Default: `2`

Caution

Do not set it to a high number, or simply leave it at its default value, unless there is a performance problem with `voisniff` service. Please keep in mind that a high number of threads might also decrease the overall system performance of NGCP!

15.3.3 X1, X2 and X3 Interface Specification

Short description of X*n* interfaces:

- The **X1** interface is used by an LI provider to create, modify, delete and list interceptions on the Sipwise NGCP. It is designed as RESTful HTTP interface using JSON (with JSON-HAL in responses from the NGCP) as content type to provision interceptions.
- The **X2** interface is a TLV based interface with JSON payload with a simple request/response mechanism over a secure TLS connection, used to pass intercepted signaling data towards an LI provider.
- The **X3** interface is also a TLV based interface with a binary payload encapsulating the intercepted RTP data.

15.3.3.1 X1 Interface

The resource used to work with interceptions is always `https://ngcp-ip:1443/api/interceptions/`

Authentication

Authentication and authorization on the NGCP API is performed via HTTP Basic Auth or SSL Client certificates.

- **HTTP Basic Auth**: With *cURL* use `--user username:password` option to specify your access credentials.

```
curl -i -X GET --user myuser:mypassword https://example.org:1443/api/interceptions/
```

  Additionally use the `--insecure` option if you are testing against a self-signed server certificate.

- **SSL Client Authentication**: You can generate and download client certificates for administrators and resellers via the NGCP Panel in the Administrators view.

  For the actual client authentication, you will need two files which you can download from the panel after creating the client certificates:

  1. The client certificate generated via the NGCP Panel. This is usually labelled NGCP-API-client-certificate-xxxxx.pem.
  2. The CA certificate used to sign the server certificate, in case it as been self-signed or the CA is not recognized by the client host environment.

  With *cURL* use `--cert /path/to/NGCPAPIclientcertificatexxxxx.pem` to specify the client certificate, and `--cacert /path/to/cacert.pem` to specify the CA certificate in case of a self-signed server certificate.

```
curl -i -X GET --cert /path/to/NGCPAPIclientcertificatexxxxx.pem \
--cacert /path/to/cacert.pem https://example.org:1443/api/interceptions/
```

  Additionally use the `--insecure` option if you are testing against a self-signed server certificate.

API Description

Collection Actions

Allowed methods for the collection as in METHOD `/api/interceptions/`

- OPTIONS
- POST
- GET
- HEAD

Item Actions

Allowed methods for a collection item as in METHOD `/api/interceptions/id`

- PATCH
- OPTIONS
- DELETE
- PUT
- GET
- HEAD

Properties

- `liid` (Number): The LI ID for this interception.
- `number` (String): The number to intercept.

- x2_host (String): The IP address of the X2 interface.
- x2_password (null, String): The password for authenticating on the X2 interface.
- x2_port (Number): The port of the X2 interface.
- x2_user (null, String): The username for authenticating on the X2 interface.
- x3_host (null, String): The IP address of the X3 interface.
- x3_port (null, Number): The port of the X3 interface.
- x3_required (null, Boolean): Whether to also intercept call content via X3 interface (false by default).

Query Parameters

- liid: Filter for interceptions of a specific interception ID
- number: Filter for interceptions of a specific number (in E.164 format)
- order_by: Order collection by a specific attribute. Possible values are: id, reseller_id, liid, number, cc_required, delivery_host, delivery_port, delivery_user, delivery_pass, modify_timestamp, create_timestamp, deleted, uuid, sip_username, sip_domain, cc_delivery_host, cc_delivery_port
- order_by_direction: Direction which the collection should be ordered by. Possible values are: asc (default), desc

API Examples

Get a specific interception

- Request:

```
curl - i -- insecure -- user administrator:administrator - X GET
https://localhost:1443/api/interceptions/528
```

- Response:

```
HTTP/1.1 200 OK
Server: nginx
Date: Tue, 01 Dec 2015 09:43:41 GMT
ContentType: application/hal+json; profile="http://purl.org/sipwise/ngcpapi/";
  charset=utf8
ContentLength: 634
Connection: keepalive
Link: </api/interceptions/>; rel=collection
Link: <http://purl.org/sipwise/ngcpapi/>; rel=profile
Link: </api/interceptions/528>; rel="item self"
SetCookie: ngcp_panel_session=35b56d921c36c1fc6edb8fcd0a86dd9af61ec62a; path=/;
  expires=Tue, 01 Dec 2015 10:43:41 GMT; HttpOnly
StrictTransportSecurity: maxage=15768000
  {
    "_links" : {
      "collection" : {
```

```
            "href" : "/api/interceptions/"
         },
         "curies" : {
            "href" : "http://purl.org/sipwise/ngcpapi/#rel{rel}",
            "name" : "ngcp",
            "templated" : true
         },
         "profile" : {
            "href" : "http://purl.org/sipwise/ngcpapi/"
         },
         "self" : {
            "href" : "/api/interceptions/528"
         }
      },
      "id" : 528,
      "liid" : 918273,
      "number" : "0014155550132",
      "x2_host" : "192.168.42.42",
      "x2_password" : null,
      "x2_port" : 3002,
      "x2_user" : null,
      "x3_host" : "192.168.42.42",
      "x3_port" : 3003,
      "x3_required" : true
   }
```

Get all interceptions for a number

- Request:

```
curl -i --insecure --user administrator:administrator -X GET \
https://localhost:1443/api/interceptions/?number=0014155550132
```

- Response:

```
HTTP/1.1 200 OK
Server: nginx
Date: Tue, 01 Dec 2015 09:47:36 GMT
ContentType: application/hal+json; profile="http://purl.org/sipwise/ngcpapi/";
  charset=utf8
ContentLength: 1283
Connection: keepalive
SetCookie: ngcp_panel_session=238550c5737058db619b183d925b5f9a61261cfe; path=/;
  expires=Tue, 01 Dec 2015 10:47:36 GMT; HttpOnly
StrictTransportSecurity: maxage=15768000
{
   "_embedded" : {
      "ngcp:interceptions" : {
```

```
         "_links" : {
            "collection" : {
               "href" : "/api/interceptions/"
            },
            "curies" : {
               "href" : "http://purl.org/sipwise/ngcpapi/#rel{rel}",
               "name" : "ngcp",
               "templated" : true
            },
            "profile" : {
               "href" : "http://purl.org/sipwise/ngcpapi/"
            },
            "self" : {
               "href" : "/api/interceptions/520"
            }
         },
         "id" : 520,
         "liid" : 1,
         "number" : "0014155550132",
         "x2_host" : "192.168.42.42",
         "x2_password" : null,
         "x2_port" : 3002,
         "x2_user" : null,
         "x3_host" : "192.168.42.42",
         "x3_port" : 3003,
         "x3_required" : true
      }
   },
   "_links" : {
      "curies" : {
         "href" : "http://purl.org/sipwise/ngcpapi/#rel{rel}",
         "name" : "ngcp",
         "templated" : true
      },
      "ngcp:interceptions" : {
         "href" : "/api/interceptions/520"
      },
      "profile" : {
         "href" : "http://purl.org/sipwise/ngcpapi/"
      },
      "self" : {
         "href" : "/api/interceptions/?page=1&rows=10"
      }
   },
   "total_count" : 1
}
```

Get all interceptions for all numbers

- Request:

```
curl -i --insecure --user administrator:administrator -X GET \
https://localhost:1443/api/interceptions/
```

- Response:


```
HTTP/1.1 200 OK
Server: nginx
Date: Tue, 01 Dec 2015 09:43:18 GMT
ContentType: application/hal+json; profile="http://purl.org/sipwise/ngcpapi/";
  charset=utf8
ContentLength: 2364
Connection: keepalive
SetCookie: ngcp_panel_session=68398eea5bdd3885ad0517e1f6d367ccc80111fa; path=/;
  expires=Tue, 01 Dec 2015 10:43:18 GMT; HttpOnly
StrictTransportSecurity: maxage=15768000
{
   "_embedded" : {
      "ngcp:interceptions" : [
         {
            "_links" : {
               "collection" : {
                  "href" : "/api/interceptions/"
               },
               "curies" : {
                  "href" : "http://purl.org/sipwise/ngcpapi/#rel{rel}",
                  "name" : "ngcp",
                  "templated" : true
               },
               "profile" : {
                  "href" : "http://purl.org/sipwise/ngcpapi/"
               },
               "self" : {
                  "href" : "/api/interceptions/520"
               }
            },
            "id" : 520,
            "liid" : 1,
            "number" : "0014155550132",
            "x2_host" : "192.168.42.42",
            "x2_password" : null,
            "x2_port" : 3002,
            "x2_user" : null,
            "x3_host" : "192.168.42.42",
            "x3_port" : 3003,
            "x3_required" : true
```

```
         },
         {
            "_links" : {
               "collection" : {
                  "href" : "/api/interceptions/"
               },
               "curies" : {
                  "href" : "http://purl.org/sipwise/ngcpapi/#rel{rel}",
                  "name" : "ngcp",
                  "templated" : true
               },
               "profile" : {
                  "href" : "http://purl.org/sipwise/ngcpapi/"
               },
               "self" : {
                  "href" : "/api/interceptions/528"
               }
            },
            "id" : 528,
            "liid" : 918273,
            "number" : "0014155550132",
            "x2_host" : "192.168.42.42",
            "x2_password" : null,
            "x2_port" : 3002,            "x2_user" : null,
            "x3_host" : "192.168.42.42",
            "x3_port" : 3003,
            "x3_required" : true
         }
      ]
   },
   "_links" : {
      "curies" : {
         "href" : "http://purl.org/sipwise/ngcpapi/#rel{rel}",
         "name" : "ngcp",
         "templated" : true
      },
      "ngcp:interceptions" : [
         {
            "href" : "/api/interceptions/520"
         },
         {
            "href" : "/api/interceptions/528"
         }
      ],
      "profile" : {
         "href" : "http://purl.org/sipwise/ngcpapi/"
      },
      "self" : {
```

```
         "href" : "/api/interceptions/?page=1&rows=10"
      }
   },
   "total_count" : 2
}
```

Get interception for specific LIID

- Request:

```
curl -i --insecure --user administrator:administrator  -X GET \
https://localhost:1443/api/interceptions/?liid=9876
```

- Response:

```
HTTP/1.1 200 OK
Server: nginx
Date: Tue, 01 Dec 2015 09:50:41 GMT
ContentType: application/hal+json; profile="http://purl.org/sipwise/ngcpapi/";
  charset=utf8
ContentLength: 1283
Connection: keepalive
SetCookie: ngcp_panel_session=23960dde6bb90f0c5c84575890194c53cce120ce; path=/;
  expires=Tue, 01  Dec 2015 10:50:40 GMT; HttpOnly
StrictTransportSecurity: maxage=15768000
{
   "_embedded" : {
      "ngcp:interceptions" : {
         "_links" : {
            "collection" : {
               "href" : "/api/interceptions/"
            },
            "curies" : {
               "href" : "http://purl.org/sipwise/ngcpapi/#rel{rel}",
               "name" : "ngcp",
               "templated" : true
            },
            "profile" : {
               "href" : "http://purl.org/sipwise/ngcpapi/"
            },
            "self" : {
               "href" : "/api/interceptions/520"
            }
         },
         "id" : 520,
         "liid" : 1,
         "number" : "0014155550132",
         "x2_host" : "192.168.42.42",
```

```
         "x2_password" : null,
         "x2_port" : 3002,
         "x2_user" : null,
         "x3_host" : "192.168.42.42",
         "x3_port" : 3003,
         "x3_required" : true
      }
   },
   "_links" : {
      "curies" : {
         "href" : "http://purl.org/sipwise/ngcpapi/#rel{rel}",
         "name" : "ngcp",
         "templated" : true
      },
      "ngcp:interceptions" : {
         "href" : "/api/interceptions/520"
      },
      "profile" : {
         "href" : "http://purl.org/sipwise/ngcpapi/"
      },
      "self" : {
         "href" : "/api/interceptions/?page=1&rows=10"
      }
   },
   "total_count" : 1
}
```


Create interception for a specific number

- Request:

```
curl -i --insecure --user administrator:administrator -X POST \
-H "ContentType: application/json" --data \
'{"liid":123, "number":"31032222203", "x2_host":"127.0.0.1", "x2_port":12345,
  "x3_required":true, "x3_host":"127.0.0.2", "x3_port":23456}' \
https://localhost:1443/api/interceptions/
```

- Response:

```
HTTP/1.1 201 Created
TransferEncoding: chunked
Connection: close
Location: /api/interceptions/528
SetCookie: ngcp_panel_session=e7817079d121fae4d86448b10e1fa21d0201c526; path=/;
  expires=Tue, 01 Dec 2015 10:43:18 GMT; HttpOnly
StrictTransportSecurity: maxage=15768000
```

The path to the newly created interception is found in the *Location* header of the response.

Update specific interception

- Request:

```
curl -i --insecure --user administrator:administrator  -X PUT \
-H "ContentType: application/json" -H 'Prefer: return=representation' --data \
'{"liid":918273, "number":"0014155550132", "x2_host":"192.168.42.42", "x2_port":5000,
  "x3_required":false}' \
https://localhost:1443/api/interceptions/123
```

- Response:

```
HTTP/1.1 200 OK
ContentType: application/hal+json; profile="http://purl.org/sipwise/ngcpapi/";
  charset=utf8
ContentLength: 621
Link: </api/interceptions/>; rel=collection
Link: <http://purl.org/sipwise/ngcpapi/>; rel=profile
Link: </api/interceptions/530>; rel=self
PreferenceApplied: return=representation
SetCookie: ngcp_panel_session=0b56e4a197b0e9f6e22a998e85473a0184770740; path=/;
  expires=Tue, 01  Dec 2015 10:56:17 GMT; HttpOnly
{
   "_links" : {
      "collection" : {
         "href" : "/api/interceptions/"
      },
      "curies" : {
         "href" : "http://purl.org/sipwise/ngcpapi/#rel{rel}",
         "name" : "ngcp",
         "templated" : true
      },
      "profile" : {
         "href" : "http://purl.org/sipwise/ngcpapi/"
      },
      "self" : {
         "href" : "/api/interceptions/530"
      }
   },
   "id" : 530,
   "liid" : 918273,
   "number" : "0014155550132",
   "x2_host" : "192.168.42.42",
   "x2_password" : null,
   "x2_port" : 5000,
   "x2_user" : null,
   "x3_host" : null,
   "x3_port" : null,
   "x3_required" : false
```

```
}
```

The *Prefer: return=representation* header forces the API to return the content, otherwise status *201* with no content is returned.

Update only certain items for a specific interception

- Request:

```
curl -i --insecure --user administrator:administrator -X PATCH \
-H "Content-Type: application/json-patch+json" -H 'Prefer: return=representation' \
  --data '[{"op":"replace", "path":"/x2_host", "value":"192.168.42.42"},{"op":"replace",
  "path":"/x2_port", "value":4000}]' \
https://localhost:1443/api/interceptions/530
```

- Response:

```
HTTP/1.1 200 OK
Server: nginx
Date: Tue, 01 Dec 2015 10:06:06 GMT
Content-Type: application/hal+json; profile="http://purl.org/sipwise/ngcp-api/";
  charset=utf-8
Content-Length: 620
Connection: close
Link: </api/interceptions/>; rel=collection
Link: <http://purl.org/sipwise/ngcp-api/>; rel=profile
Link: </api/interceptions/530>; rel=self
Preference-Applied: return=representation
Set-Cookie: ngcp_panel_session=0693129d63d543a85f96d464ff9a8f807cfc4d18; path=/;
  expires=Tue, 01 Dec 2015 11:06:06 GMT; HttpOnly
Strict-Transport-Security: max-age=15768000
{
   "_links" : {
      "collection" : {
         "href" : "/api/interceptions/"
      },
      "curies" : {
         "href" : "http://purl.org/sipwise/ngcp-api/#rel-{rel}",
         "name" : "ngcp",
         "templated" : true
      },
      "profile" : {
         "href" : "http://purl.org/sipwise/ngcp-api/"
      },
      "self" : {
         "href" : "/api/interceptions/530"
      }
   },
   "id" : 530,
```

```
    "liid" : 918273,
    "number" : "0014155550132",
    "x2_host" : "192.168.42.42",
    "x2_password" : null,
    "x2_port" : 4000,
    "x2_user" : null,
    "x3_host" : null,
    "x3_port" : null,
    "x3_required" : false
}
```


Delete specific interception

- Request:

```
curl -i --insecure  --user administrator:administrator  -X DELETE \
https://localhost:1443/api/interceptions/123
```

- Response:

```
HTTP/1.1 204 No Content
Server: nginx
Date: Tue, 01 Dec 2015 10:08:49 GMT
Connection: keepalive
SetCookie: ngcp_panel_session=570c66b66732629766f86b8ed9bd0d64902ae73e; path=/;
  expires=Tue, 01  Dec  2015 11:08:49 GMT; HttpOnly
XCatalyst: 5.90042
StrictTransportSecurity: maxage=15768000
```

15.3.3.2 X2 Interface

The communication via the X2 interface consists of request-response pairs.

Request

The request is formatted as: `X2/<bodylength>/<body>`

Body part has the following items:

Table 20: X2 Message Body Items

Element	Type	Length	Description
/x2/header/source	String	arbitrary length	identifier of Sipwise node which captured the data
/x2/header/destination	String	arbitrary length	identifier of LI mediation system

Table 20: (continued)

Element	Type	Length	Description
/x2/header/type	String	arbitrary length	always "sip" (but later potentially "xmpp" and others too)
/x2/header/version	PosInteger	arbitrary length	always "1"
/x2/header/timestamp	String	27 chars	format: YYYY-MM-DDThh:mm:ss.ffffffZ; timestamp in UTC when the X2 package is sent to mediation
/x2/body/dialogid	PosInteger	arbitrary length	globally increasing counter for each new communication dialog (e.g. call)
/x2/body/messageid	PosInteger	arbitrary length	increasing counter for each new x2 message within a dialog, starting from 0
/x2/body/timestamp	String	27 chars	format: YYYY-MM-DDThh:mm:ss.ffffffZ; timestamp in UTC when the package has been captured on the wire
/x2/body/interceptions			one or more elements containing the following information, one element per intercepted target:
/x2/body/interceptions/liid	PosInteger	arbitrary length	interception id ("liid") as set via X1 interface
/x2/body/interceptions/direction	String	arbitrary length	either "totarget" or "fromtarget" from the soft-switch perspective (if target is the called party, it is "totarget", if target is the calling party, it is "fromtarget").
/x2/body/data	Base64 encoded	arbitrary	content of full IP frame and up on the OSI layer; packets fragmented on the wire are provided in fully assembled format

Example of full message:

```
X2/418/
{
  "header": {
    "source": "prx01a.example.com",
    "destination": "x2destination.example.com",
    "type": "sip",
    "version": 1,
    "timestamp": "2015 03 11T09:18:04.729803Z"
  },
  "body": {
    "dialogid": 4,
    "messageid": 0,
    "timestamp": "2015 03 11T09:18:04.729123Z",
    "interceptions": [
      { "liid": 174, "direction": "fromtarget" },
      { "liid": 175, "direction": "totarget" }
```

```
    ],
    "data": "<base64 encoded ip,udp/tcp,sip frame>"
  }
}
```

Response

- Success: `X2-ACK/0/`
- Error: `X2-ERR/<length>/<error string>`

Keep-Alive Mechanism

A regular keep-alive mechanism with a default value of 10s is used on the connection if it is re-used across multiple messages.

- Request: `X2/0/`
- Response: `X2-ACK/0/`

15.3.3.3 X3 Interface

On the X3 interface TLV based packets are sent via secured (TLS) connection on a pre-established stream. X3 messages do not need to be acknowledged, except for keep-alive messages.

X3 Message Structure

Table 21: X3 Message Structure

Field	Length
Header	arbitrary
CCCID	4 bytes
MessageId	4 bytes
Timestamp	8 bytes
Payload	arbitrary

Header Details

Table 22: X3: Header Details

Field	Length	Content
type	2 bytes	always "X3"
delimiter	1 byte	always "/"
length	arbitrary	ASCII string

Table 22: (continued)

Field	Length	Content
delimiter	1 byte	always "/"

CCCID Details

`dialogid` (32 bit in network byte order, reset to 0 after 2^{32}-1)

The `dialogid` is referencing the `/x2/body/dialogid` field in order to correlate an X3 packet to an X2 call.

MessageId Details

`messageid` (32 bit in network byte order, reset to 0 after 2^{32}-1)

The `messageid` is a counter within a dialog sequencing the X3 packets sent from the NGCP. This counter is not correlated in any way with X2, rather than starting at 0 with the first RTP packet captured within a dialog.

Timestamp Details

- `seconds` (32 bit in network byte order)
- `fraction` (32 bit in network byte order)

The timestamp represents the Unix epoch starting from 1970-01-01.

Payload Details

Table 23: X3: Payload Details

Field	Length
original ip header	20 bytes for v4, 40 bytes for v6
original udp header	8 bytes
original rtp header	variable, 12-72 bytes
original rtp payload	arbitrary

Keep-Alive Mechanism

A regular keep-alive mechanism with a default value of 10s is used on the connection if it is re-used across multiple messages.

- Request: `X3/0/`
- Response: `X3-ACK/0/`

A Basic Call Flows

A.1 General Call Setup

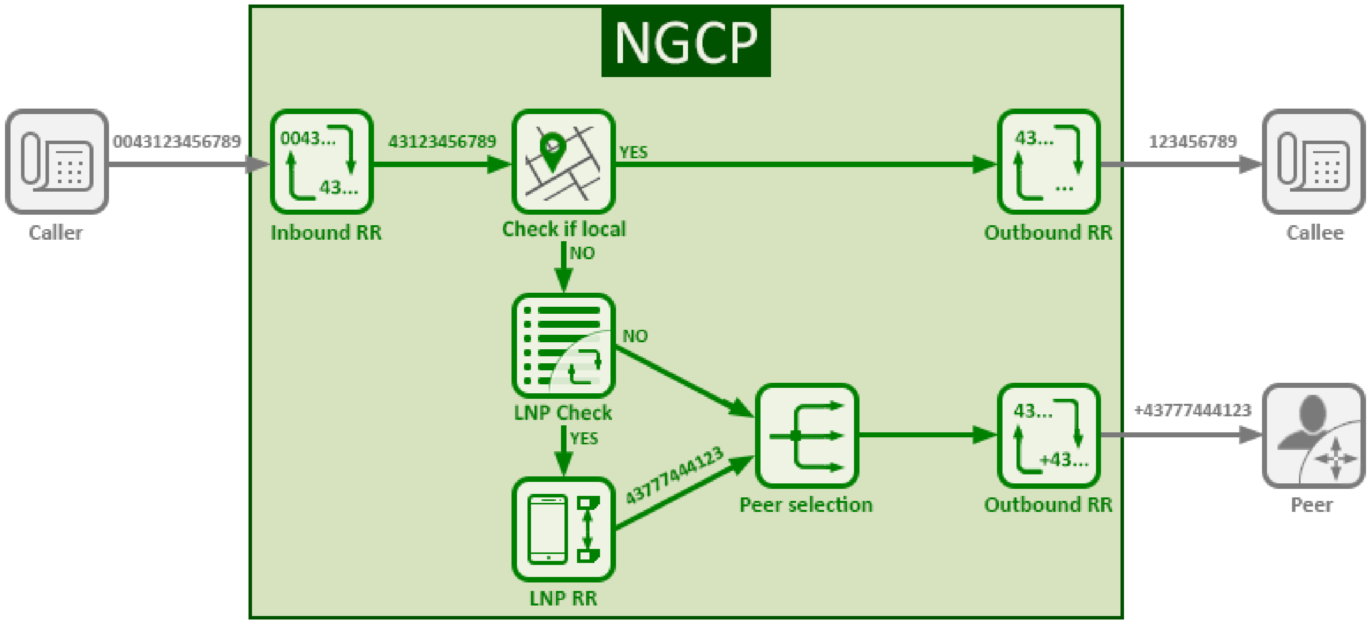


Figure 100: General Call Setup

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 5.4 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 4.5.2.1 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

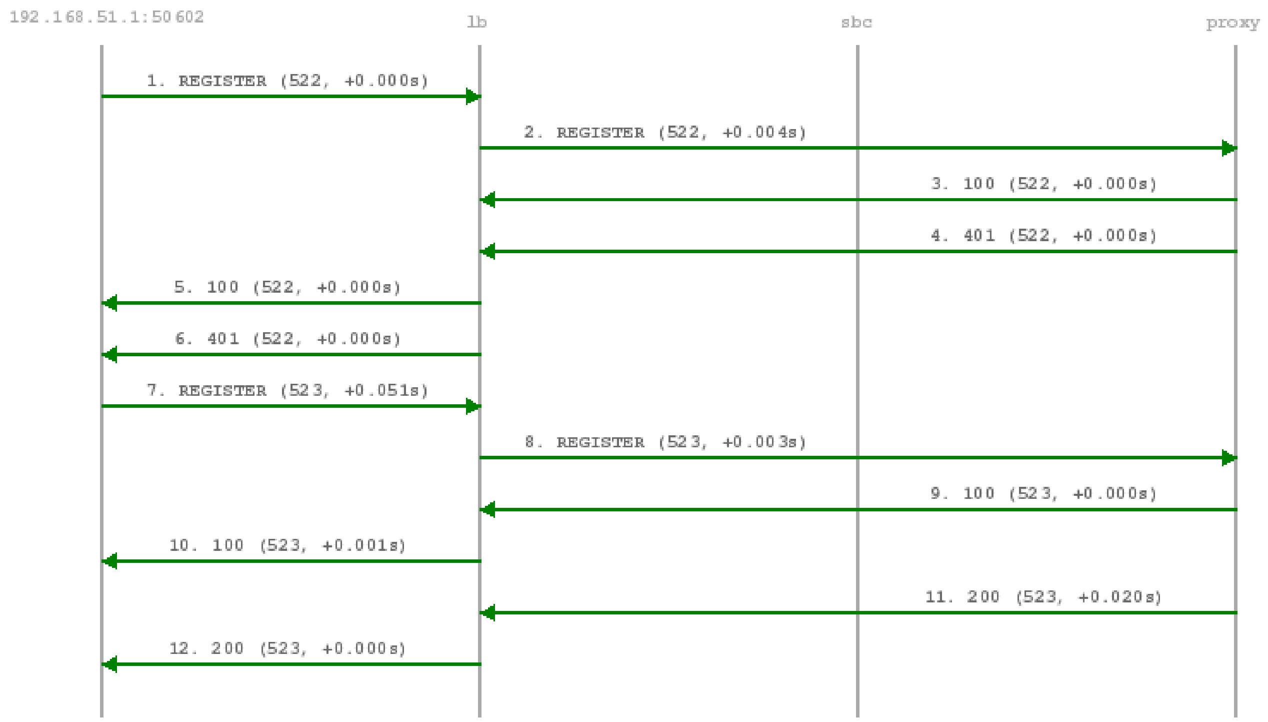


Figure 101: Registration Call-Flow

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).

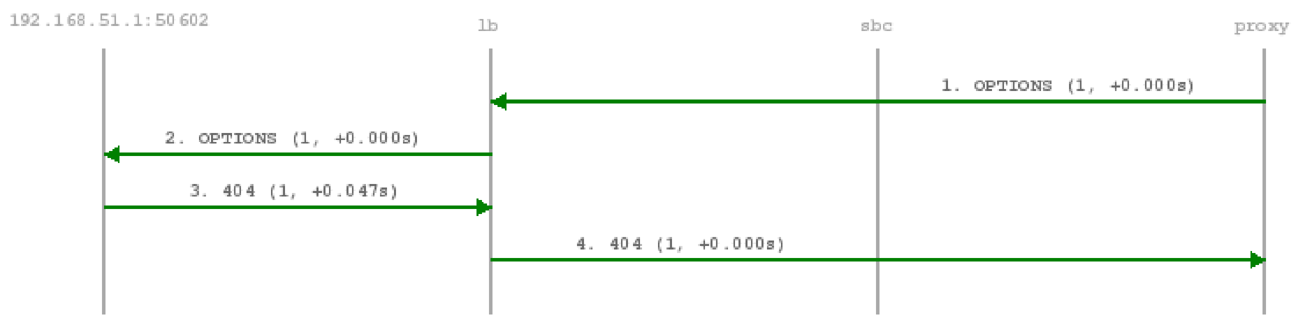


Figure 102: NAT-Ping Call-Flow

By default, a subscriber can register 5 contacts for an Address of Record (AoR, e.g. `sip:someuser@example.org`).

A.3 Basic Call

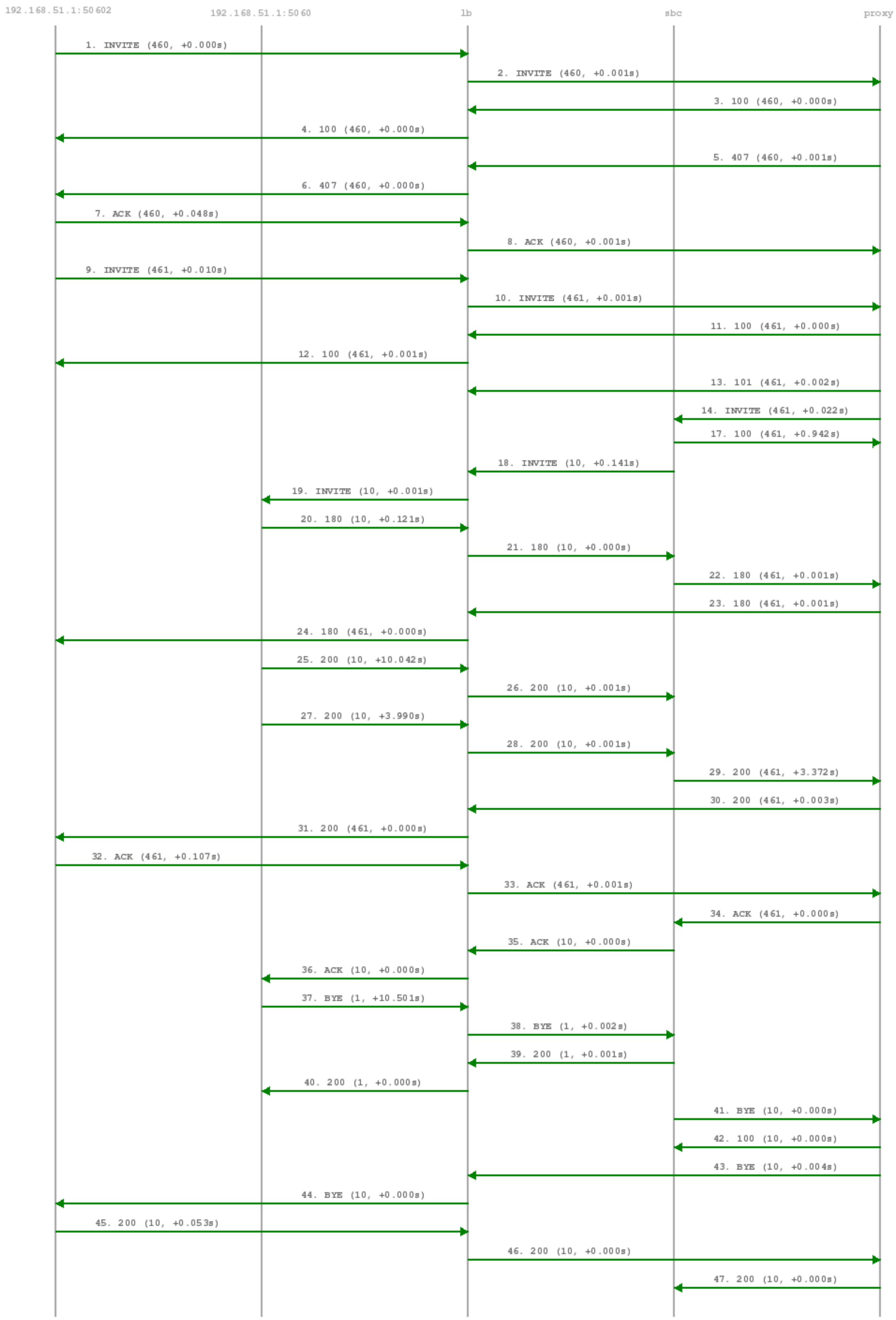


Figure 103: Basic Call Call-Flow

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. For calls made by subscribers on a prepaid plan, rating occurs at call runtime and is actually done by the B2BUA (which is necessary to properly support multiple parallel calls by the same subscriber). The final rating data is then passed on to *rate-o-mat* which will update the CDRs accordingly.

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

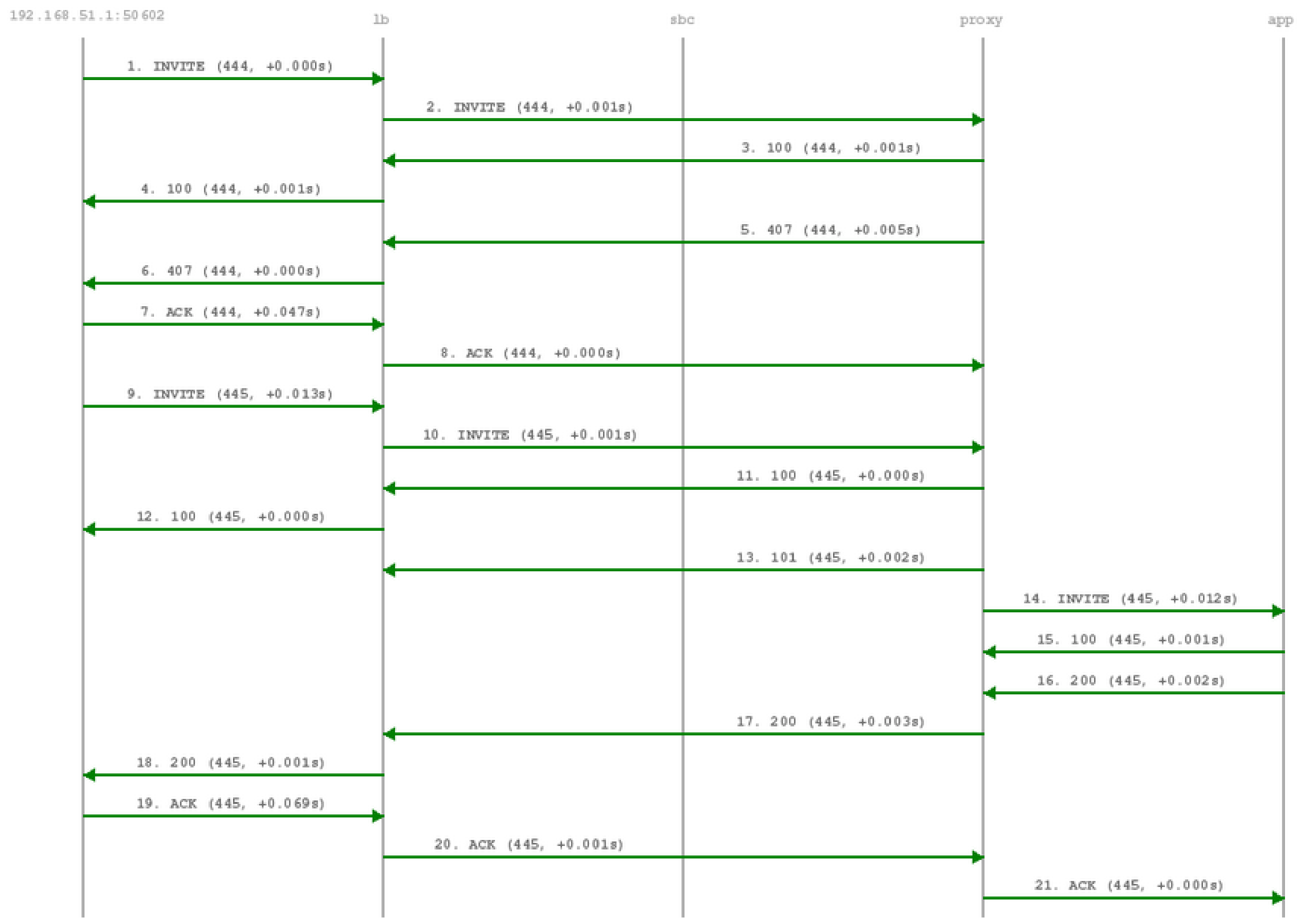


Figure 104: Voicebox Call-Flow

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 PRO 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 PRO.

```
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 backups to keep stored.
- backuptools.storage_dir: Storage directory of backups.
- 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:
  collcheck:
    cpuidle: 0.1
    dfused: 0.9
    eximmaxqueue: 15
    loadlong: 2
    loadmedium: 2
    loadshort: 3
    maxage: 600
    memused: 0.7
    siptimeout: 15
    swapfree: 0.5
  active_check_enable: 1
  asr_nsr_statistics: 1
```

```
exim_check_enable: 0
force: 0
kamailio_check_concurrent_calls_enable: 0
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
mpt_check_enable: 1
mysql_check_enable: 1
mysql_check_replication: 1
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
```

- 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 collectd.
- checktools.active_check_enable: Active node check plugin for collectd.
- checktools.asr_nsr_statistics: enable/Disable ASR/NSR 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: MySQL SNMP check plugin.

- checktools.mysql_check_replication: MySQL 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 (separated by comma , - ex. source: 127.0.0.1, 10.10.10.2, 10.10.10.3).

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 12.5 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
```

- general.adminmail: Email address used by monit to send notifications to.
- general.lang: Sets sounds language (e.g: *de* for German)

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:
    debug: 'no'
    extra_sockets: ~
    max_forwards: 70
    nattest_exception_ips:
      - 1.2.3.4
      - 5.6.7.8
    pkg_mem: 16
```

```
    port: 5060
    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
    shm_mem: 2012
    start: 'yes'
    strict_routing_safe: 'no'
    tcp_children: 8
    tcp_max_connections: 2048
    tls:
      enable: 'no'
      port: 5061
      sslcertfile: '/etc/kamailio/kamailio-selfsigned.pem'
      sslcertkeyfile: '/etc/kamailio/kamailio-selfsigned.key'
    udp_children: 8
    use_dns_cache: 'on'
  proxy:
    allow_info_method: 'no'
    allow_peer_relay: 'no'
    allow_refer_method: 'no'
    authenticate_bye: 'no'
    cf_depth_limit: 10
    children: 8
    debug: 'no'
    default_expires: 3600
    enum_suffix: e164.arpa.
    filter_100rel_from_supported: 'yes'
    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}
    foreign_domain_via_peer: 'no'
    ignore_auth_realm: 'no'
    keep_original_to: 'no'
```

```
  lnp:
    api:
      invalid_lnp_routing_codes:
        - ^EE00
        - ^DD00
      lnp_request_blacklist: []
      lnp_request_whitelist: []
      request_timeout: '1000'
    enabled: no
    type: api
  max_expires: 43200
  max_gw_lcr: 128
  max_registrations_per_subscriber: 5
  min_expires: 60
  nathelper_dbro: 'no'
  natping_interval: 30
  natping_processes: 7
  nonce_expire: 300
  pbx:
    hunt_display_indicator: '[h]'
  perform_peer_lcr: 0
  pkg_mem: 16
  port: 5062
  presence:
    enable: 'yes'
    max_expires: '3600'
    reginfo_domain: example.org
  proxy_lookup: 'no'
  set_ruri_to_peer_auth_realm: 'no'
  shm_mem: 2012
  start: 'yes'
  tcp_children: 4
  use_enum: 'no'
  usrloc_dbmode: 1
```

- kamailio.lb.debug: Enable intensive debug level.
- 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.nattest_exception_ips: List of IPs that don't need the NAT test.
- kamailio.lb.shm_mem: Shared memory used by Kamailio Load Balancer. The default value is auto generated by the system, depending on your system architecture.
- kamailio.lb.pkg_mem: PKG memory used by Kamailio Load Balancer. The default value is auto generated by the system, depending on your system architecture.

- 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.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.strict_routing_safe: Enable strict routing handle feature.
- 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.proxy.allow_info_method: Allow INFO method.
- kamailio.proxy.allow_peer_relay: Allow peer relay. Call coming from a peer that doesn't matcha 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.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.debug: Enable intensive debug level.
- kamailio.proxy.default_expires: Default expires value in seconds for REGISTER messages.
- kamailio.proxy.foreign_domain_via_peer: Enable calls to foreign domains via peers.
- kamailio.proxy.shm_mem: Shared memory used by Kamailio Proxy. The default value is auto generated by the system, depending on your system architecture.
- kamailio.proxy.pkg_mem: PKG memory used by Kamailio Proxy. The default value is auto generated by the system, depending on your system architecture.
- kamailio.proxy.enum_suffix: Sets ENUM suffix - don't forget . (dot).
- kamailio.proxy.filter_100rel_from_supported: Enable filtering of *100rel* from Supported header, to disable PRACK.
- 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.keep_original_to: Not used now.
- kamailio.proxy.lnp.enabled: Enable/disable LNP (local number portability) lookup during call setup
- 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.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.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.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.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.min_expires: Sets the minimum expires in seconds for registration.
- kamailio.proxy.natping_interval: Sets the NAT ping interval in seconds.
- kamailio.proxy.nathelper_dbro: Defaul is "no". This will be "yes" on CARRIER in order to activate the use of a read-only connection using LOCAL_URL
- kamailio.proxy.nonce_expire: Nonce expire time in seconds.
- kamailio.proxy.perform_peer_lcr: Enable/Disable Least Cost Routing based on peering fees.
- 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.set_ruri_to_peer_auth_realm: Set R-URI using peer auth realm
- kamailio.proxy.use_enum: Enable/Disable ENUM feature.

B.1.15 lnpd

The following section defines configuration of LNP daemon, that is used when LNP queries are served by external gateways → the so called LNP API mode.

```
lnpd:
  config:
    daemon:
      foreground: 'false'
      json-rpc:
        ports:
          - '8095'
      loglevel: '6'
      sip:
        port: '5095'
      threads: '4'
    instances:
      default:
        module: sigtran
        destination: 0.0.0.0
        from-domain: voip.example.com
        headers:
          - header: INAP-Service-Key
            value: '2'
        reply:
          tcap: raw-tcap
  enabled: no
```

- lnpd.enabled: Enable/disable LNP daemon
- lnpd.config: details are shown in Configuration of LNP daemon Section 5.4.2.3

B.1.16 mediator

The following is the mediator section:

```
mediator:
  interval: 10
```

- mediator.interval: Running interval of mediator.

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:
    certificate: '/etc/ngcp-config/ssl/PushChatCert.pem'
    enable: yes
    endpoint: gateway.push.apple.com
    feedback_endpoint: feedback.push.apple.com
    feedback_interval: 3600
    key: '/etc/ngcp-config/ssl/PushChatKey.pem'
    socket_timeout: 0
  enable: yes
  gcm:
```

```
    enable: yes
    key: 'google_api_key_here'
    priority:
      call: high
      groupchat: normal
      invite: normal
      message: normal
  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.certificate: Specify the Apple certificate for push notification https requests from the NGCP to an endpoint.
- pushd.apns.enable: Enable/Disable Apple push notification.
- pushd.apns.key: Specify the Apple key for push notification https requests from the NGCP to an endpoint.
- 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.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'
  maxport: 40000
  minport: 30000
  rtp_timeout: 21600
  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.maxport: Maximum port used by rtpengine for RTP traffic.
- rtpproxy.minport: Minimum port used by rtpengine for RTP traffic.
- rtpproxy.rtp_timeout: Maximum limit in seconds for a call (6h).
- rtpproxy.rtp_timeout_onhold: Maximum limit in seconds for an onhold (1h).

B.1.29 security

The following is the security section:

```
security:
  firewall:
    blacklist_networks_4: ~
    blacklist_networks_6: ~
    enable: 'yes'
    sipwise_support_access: 'no'
    whitelist_networks_4: ~
    whitelist_networks_6: ~
```

- security.firewall.enable: Enable/Disable security configuration for IPv6 and IPv6 (sysctl_ipv6.conf, sysctl_ipv4.conf).

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: Sets the maximum call duration for inter-domain calls.
- sems.sbc.outbound_timeout:: Sets the maximum call duration for outboud calls.
- 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 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.1.32 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.1.33 voisniff

The following is the voice sniffer section:

```
voisniff:
  admin_panel: 'no'
  daemon:
    bpf: 'port 5060 or 5062 or ip6 proto 44 or ip[6:2] & 0x1fff != 0'
    external_interfaces: 'eth0 eth1'
    filter:
      exclude:
        -
          active: 0
          case_insensitive: 1
          pattern: '\ncseq: *\d+ +(register|notify|options)'
      include: []
    internal_interfaces: lo
    mysql_dump_threads: 4
    start: 'no'
    threads_per_interface: 10
  partitions:
    increment: 700000
    keep: 10
```

- voisniff.admin_panel: Enable/Disable SIP STATS on Admin interface. Default is *no*.
- voisniff.deamon.external_interfaces: Define binding interfaces.
- voisniff.deamon.start: Change to *yes* if you want voisniff start at boot. Default is *no*.

B.1.34 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 PRO 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 10 section of the handbook.

Sample host configuration for sip:provider PRO

```
sp1:
  dbnode: '1'
  eth0:
    dns_nameservers:
      - 192.168.51.30
      - 192.168.51.31
    gateway: 192.168.22.1
    hwaddr: 06:1e:bc:e2:ec:fb
    ip: 10.0.2.15
    netmask: 255.255.255.0
    shared_ip: ~
    shared_v6ip: ~
    type:
      - web_ext
      - ssh_ext
      - web_int
  eth1:
```

```
    hwaddr: 6e:7f:3a:f9:db:1f
    ip: 192.168.255.251
    netmask: 255.255.255.248
    shared_ip:
      - 192.168.255.250
    shared_v6ip: ~
    type:
      - ha_int
      - ssh_ext
  eth2:
    ip: 10.15.20.107
    netmask: 255.255.255.0
    shared_ip:
      - 10.15.20.151
    type:
      - ssh_ext
      - web_ext
      - web_int
      - sip_ext
      - rtp_ext
      - mon_ext
  interfaces:
    - lo
    - eth0
    - eth1
    - eth2
  lo:
    advertised_ip: []
    cluster_sets:
      - default
    hwaddr: 00:00:00:00:00:00
    ip: 127.0.0.1
    netmask: 255.0.0.0
    shared_ip: []
    shared_v6ip: []
    type:
      - sip_int
      - web_ext
      - web_int
      - aux_ext
      - ssh_ext
      - api_int
    v6ip: '::1'
  peer: sp2
  role:
    - proxy
    - lb
    - mgmt
```

```
      - rtp
      - db
```

C NGCP-Faxserver Configuration

For an overview of Faxserver architecture and features, please see the Faxserver Section 5.8 chapter.

C.1 Faxserver Components

Starting from mr4.3 release there is a completely reworked fax server in a form of standalone daemon that uses Asterisk as its transmission component. No other component — such as `hylafax` or `iaxmodem` — is necessary to send and receive faxes on sip:provider PRO platform.

C.2 Enabling Faxserver

In order to configure functions of NGCP Faxserver one needs to update the main NGCP configuration file `/etc/ngcp-config/config.yml` with the correct fax options:

```
faxserver:
  enable: yes
  fail_attempts: '3'
  fail_retry_secs: '60'
  keep_failed_fax: yes
  keep_failed_fax_days: '60'
  keep_received_fax: yes
  keep_received_fax_days: '60'
  keep_sent_fax: yes
  keep_sent_fax_days: '60'
  mail_from: 'Sipwise NGCP FaxServer <voipfax@ngcp.sipwise.local>'
```

Parameters are:

- `enable`: must be `yes` to enable Faxserver
- `fail_...`: the number and timeout of fax sending retrials
- `keep_...`: fax retention definitions: enabling and length in days
- `mail_from`: the *From* header in the e-mail that is sent by Fax2Mail feature when a fax is received

C.3 Fax Templates Configuration

One needs to update `/etc/ngcp-config/templates/etc/ngcp-faxserver/faxserver.conf.tt2` if he wants to use custom content in the fax and e-mail templates that are used by Faxserver to generate the actual fax or e-mail. This may be done under the "User templates" section in the file.

Applying new Faxserver configuration

Once the above mentioned configuration files have been modified the new settings must be applied:

```
ngcpcfg apply 'Configured fax server'
ngcpcfg push
```

C.4 Fax Services Configuration per Subscriber

Fax services must be explicitly activated for subscribers before they can send or receive faxes. This activation and the custom settings may be set on the NGCP Web panel in the following way (as an administrator):

- Go to *Subscribers* and find the subscriber that you want to modify settings for
- Click on *Preferences* button
- Select *FaxFeatures*

In both sections *Fax2Mail and SendFax* and *Mail2Fax* there is a field: `Active`. This must be changed from `no` to `yes` if the particular fax service must be activated.

When fax services have been activated the user sees a summary of settings in *FaxFeatures* section on his Preferences page:

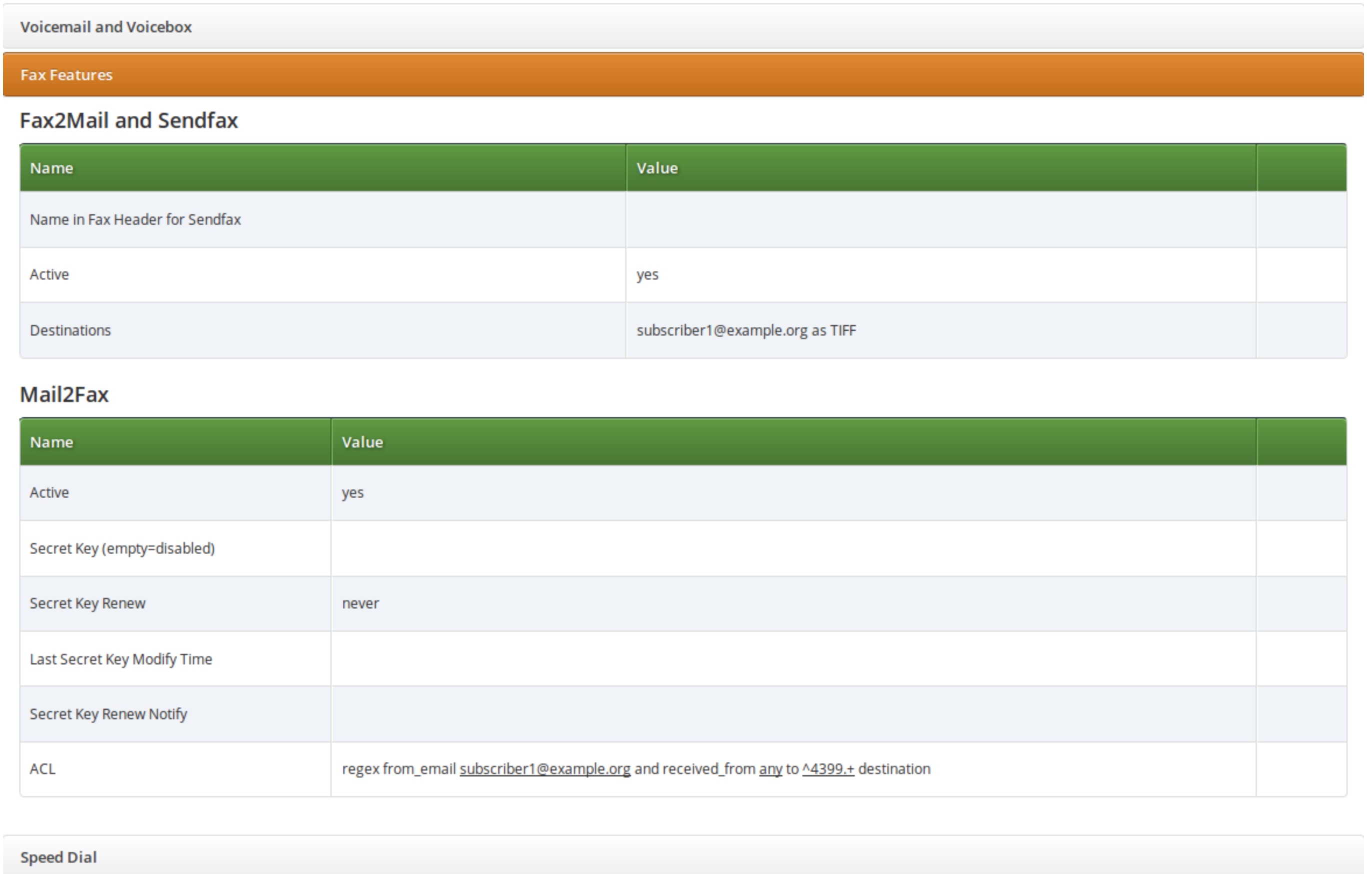


Figure 105: Fax Settings

Details of Fax2Mail, SendFax and Mail2Fax settings are described in subsequent paragraphs.

C.5 Fax2Mail and SendFax Settings

- `Name in Fax Header for SendFax`: optional field that contains the subscribers name on faxes sent from the Web panel directly
- `Destinations`: e-mail addresses and selections of notification items that define about which event and where an e-mail is sent; this is a list of such definitions

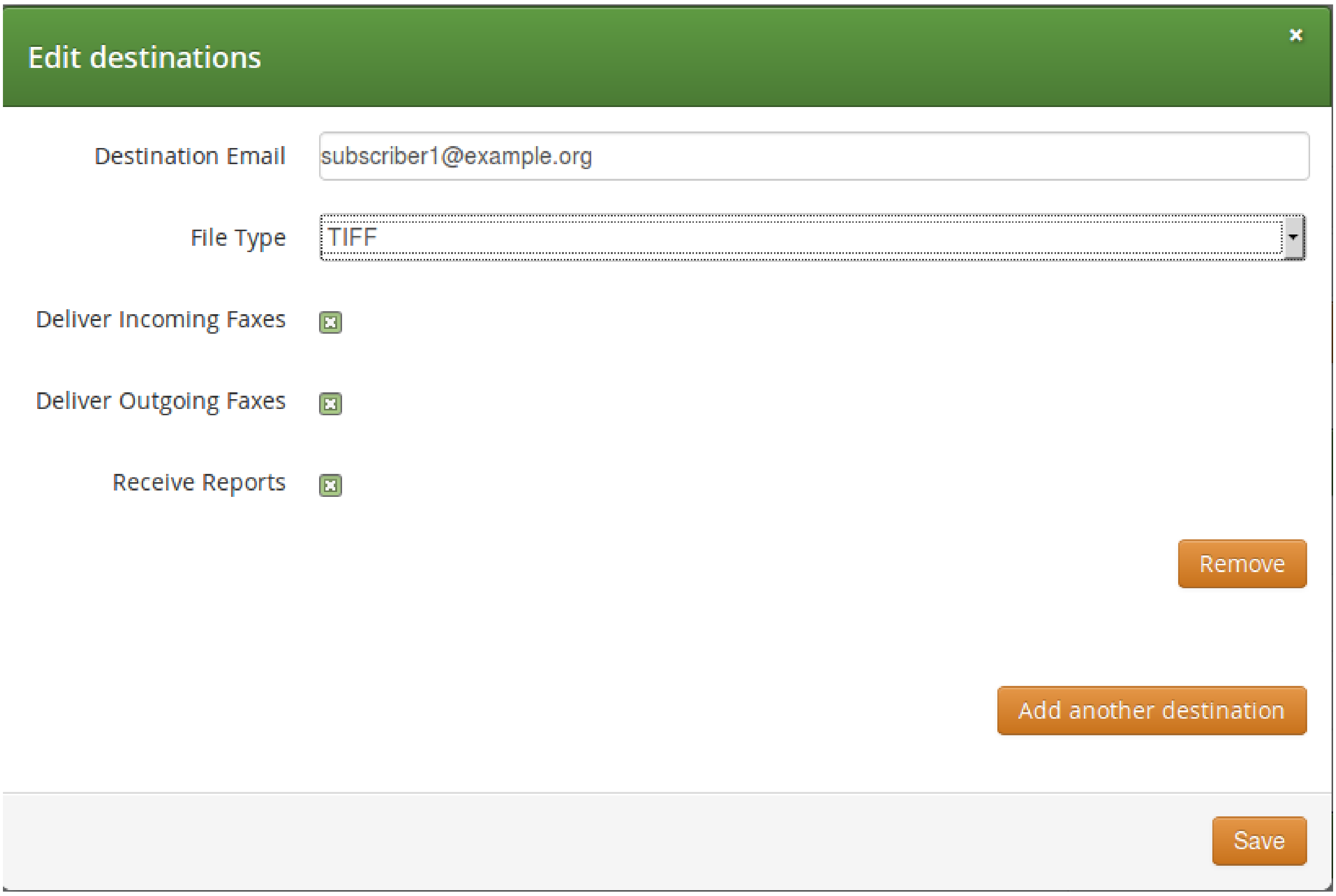


Figure 106: Fax2Mail Destination

The parameters for a destination are as follows:

- `Destination Email`: the e-mail address where the notification must be sent
- `File Type`: file format of faxes attached to e-mails
- `Deliver Incoming Faxes`: select this in order to receive incoming faxes in e-mail
- `Deliver Outgoing Faxes`: select this in order to receive a report about sent faxes
- `Receive Reports`: select this in order to receive reports about success / failure of fax transmissions

C.6 Mail2Fax Settings

A subscriber can restrict access to his Mail2Fax service with some methods, those can also be combined:

- using a *secret key* that is only known to him, and is inserted in every mail that he sends to NGCP to be forwarded as fax
- using an *access control list (ACL)* that determines from which endpoint and for which destination a mail-to-fax is accepted by NGCP platform
- `Secret Key`: the secret key used to validate the sender of an e-mail; not used if left empty
- `Secret Key Renew`: secret key renewal period; NGCP platform will enforce renewal of the secret key when the defined time has elapsed
- `Last Secret Key Modify Time`: information about the last secret key modification time
- `Secret Key Renew Notify`: an e-mail address where the notification about secret key modification is sent
- `ACL`: access control list, see the details below; this is a list of access control rules

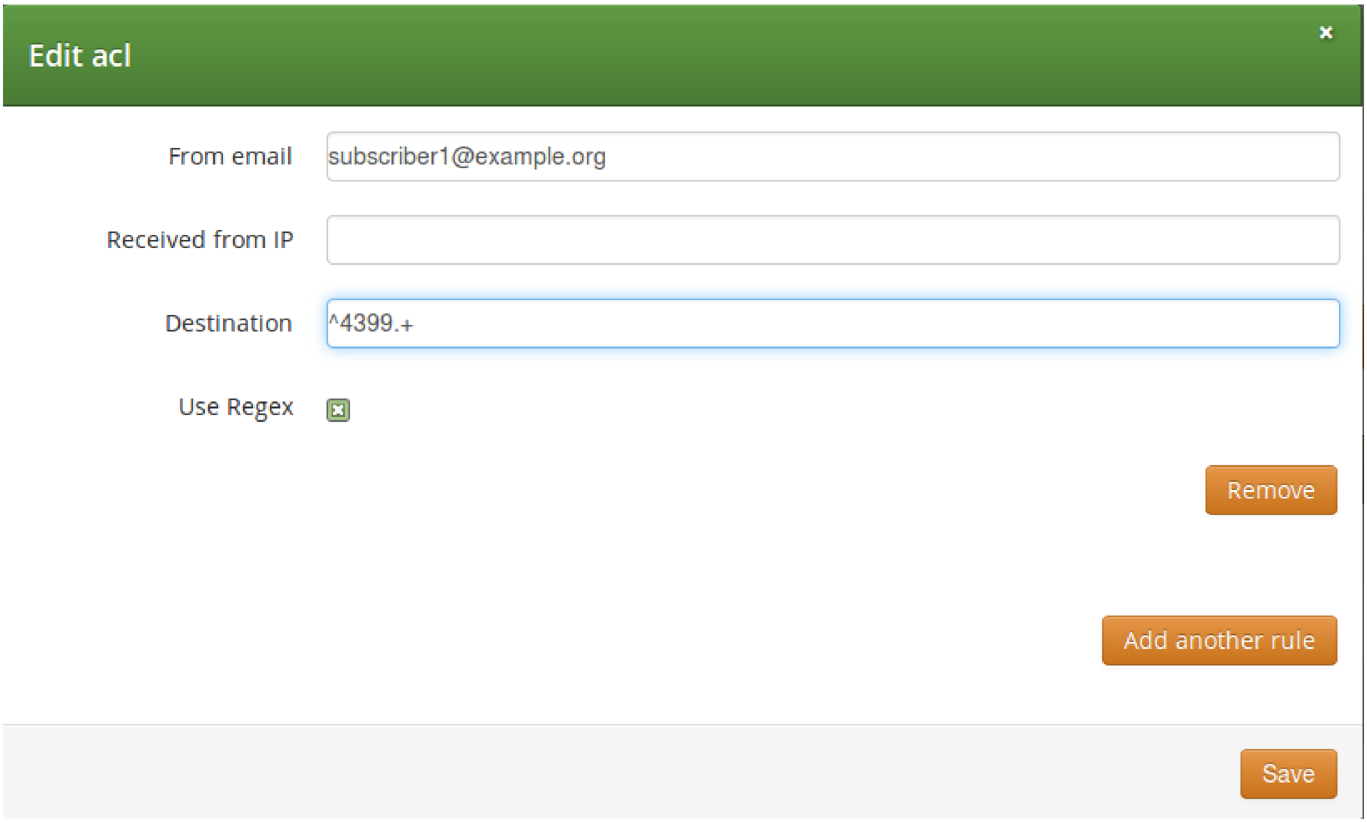


Figure 107: Mail2Fax Access Control List

The parameters for access control rules:

- `From email`: this sender is allowed to use Mail2Fax service

- `Received from IP`: this IP address or host name must be present in From e-mail header
- `Destination`: either a complete phone number in E.164 format, or a regular expression ("Use Regex" checkbox must be ticked) that may define a range of numbers. Examples: "4313334445" as a single number; "ˆ4399.+" as a regular expression: all destinations starting with "4399"

Caution

When neither *Secret Key*, nor *ACL* is defined then Mail2Fax service will deny accepting any e-mail for sending faxes!

C.7 Sending Fax from Web Panel

A subscriber can log in to his *Customer Self Care* website and send faxes directly from there. In order to do this, one needs to do the following:

- Go to *Settings* → *Web Fax* page

  Tip

  The list of received faxes is also available here.

- Press *Send Fax* button to start entering data, such as recipient and content for the fax being sent:

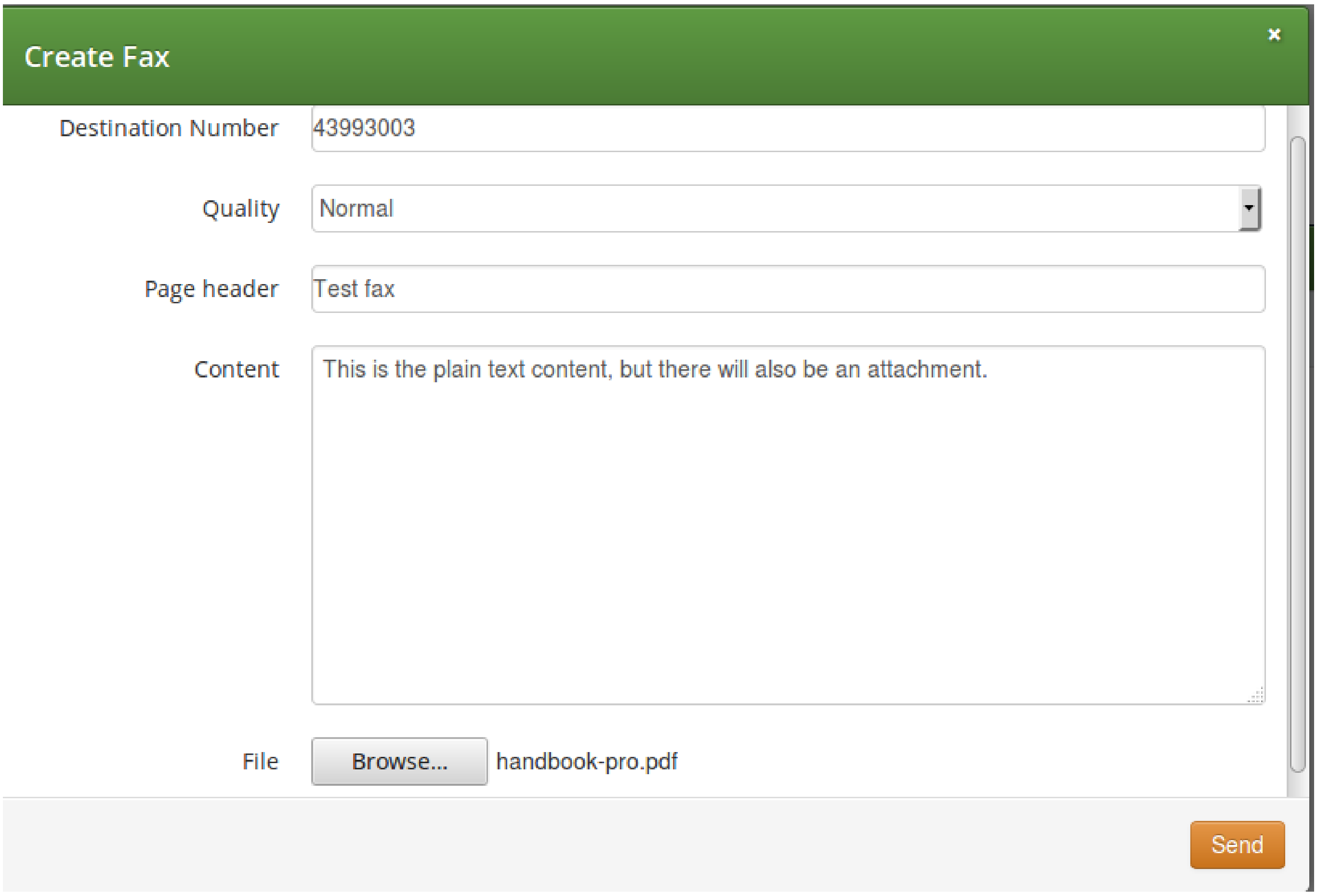


Figure 108: Sending Fax from Web Panel

Both plain text message and attached files can be sent in the fax. First page(s) will contain the plain text message and the content of attached files will follow that.

C.8 Faxserver Mail2Fax Configuration

Using NGCP Faxserver's Mail2Fax service requires the configuration of sip:provider PRO's local mail server that is *Exim*. It has to be configured in a way that it can receive mails from outside of the server, because *Exim* by default listens only on the local interfaces for incoming mails.

Exim Configuration

The NGCP platform administrator must reconfigure *Exim* in order to enable receiving e-mails for fax sending:

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

PLEASE NOTE: When entering configuration data the following points must be kept in mind:

- operation mode has to be set to "`mail sent by smarthost;no local mail`"

- "`mail2fax.example.org`" must be added to accepted domains, where "example.org" is the domain name of the NGCP platform operator

DNS Configuration

It is necessary to add a subdomain starting as `mail2fax.` to the list of domain names. That is where the faxes will be sent by users to trigger Mail2Fax service.

Tip

Alternatively, edit `/etc/ngcp-config/templates/etc/exim4/conf.d/router/999_mail2fax.tt2` file and adjust it to your personal preferences. Although this is not recommended and should only be done by Sipwise support engineers.

C.9 Sending Fax Using E-mail Clients

When sending an e-mail that should be converted to a fax, there are some points to keep in mind so that Faxserver properly processes the e-mail.

- ***To* header**:
  - must contain the subscriber's number who is sending the fax, as the username part of the mail address
  - must contain the specific domain starting with `mail2fax.`
- ***Subject* header**: must contain the fax destination number
- ***Body*** should consist of plain text data
- Adding **attachments** is possible, but only plain text and PDF formats are supported

Secret Key

In order to use the "secret key" access control feature, it should be either put in the first row of the e-mail body followed by an empty line, or included as a plain text attachment. Once it has been validated, it will be removed from the email.

Important

Either add the secret key to the body, or attach it. Never do both as only one will be recognized and removed, leaving the other one to be sent as part of the fax.

Mail Example

Provided there is a subscriber on sip:provider PRO platform with the 43130111 number, the destination fax is 43130222 and the secret key is "MySecretKey":

```
From: User Name <username@example.org>
To: 43130111@mail2fax.example.org
```

```
Subject: 43130222
- - - - - - - - - - - - - - - - - - -
MySecretKey

This is a test fax.

Cheers
```

C.10 Managing Faxes via the REST API

It is possible to send and receive faxes and configure fax settings using the built-in REST API interface.

In subsequent sections you can find examples of using the API for sending, receiving faxes and changing fax settings.

C.10.1 Configuring Fax Settings

C.10.1.1 Retrieving Fax Settings

The following example retrieves the fax settings for the subscriber with ID 3.

```
Method: GET
Content-Type: application/hal+json

https://127.0.0.1:1443/api/faxserversettings/3
```

The output format is as follows (only the relevant output data is shown):

```
  "active" : true,
   "destinations" : [
      {
         "destination" : "user@company.com",
         "filetype" : "PDF14",
         "incoming" : true,
         "outgoing" : true,
         "status" : true
      }
   ],
   "name" : null,
   "password" : null
```

C.10.1.2 Updating Fax Settings

The following example updates a specific parameter. Namely, it deactivates the fax feature for the subscriber with ID 3.

```
Method: PATCH
Content-Type: application/json-patch+json
```

```
https://127.0.0.1:1443/api/faxserversettings/3

--data-binary '[ { "op" : "replace", "path" : "/active", "value" : 0 } ]'
```

C.10.2 Sending a Fax

The following request sends a PDF file located at `/tmp/test_fax.pdf` as fax to 431110002 from the subscriber with ID 3.

```
Method: POST
Content-Type: multipart/form-data

https://127.0.0.1:1443/api/faxes/

--form 'json={"destination" : "431110002", "subscriber_id" : 3}' --form 'faxfile=@/tmp/ ↩
    test_fax.pdf'
```

C.10.3 Receiving a Fax

All received faxes are stored on the server and can be retrieved on demand. You can retrieve a stored fax by following these steps:

1. Firstly, obtain the internal ID of the fax:

   ```
   Method: GET
   Content-Type: application/json

   https://127.0.0.1:1443/api/faxes/3
   ```

   This request returns the list of stored faxes for the subscriber with ID 3. One of the available faxes is returned like this:


   ```
      "callee" : "431110002",
      "caller" : "431110001",
      "direction" : "out",
      "duration" : "0",
      "filename" : "d9799276-b7d9-454f-98c3-714edf7e3072.tif",
      "id" : 5,
      "pages" : "1",
      "quality" : "8031x7700",
      "reason" : "Normal Clearing / SIP 200 OK [1/3]",
      "signal_rate" : "14400",
      "status" : "SUCCESS",
      "subscriber_id" : 1,
      "time" : "2016-07-30 09:49:59"
   ```


2. Now, to retrieve the fax with ID 5, use the following request:

```
Method: GET
Content-Type: application/hal+json

https://127.0.0.1:1443/api/faxerecordings/5
```

By default, the fax is in the TIFF format. It is also possible to request it in a different format. To retrieve the same fax in PDF14, use the following request:

`https://127.0.0.1:1443/api/faxerecordings/5?format=pdf14`

C.10.4 Configuring Mail2Fax Settings

The configuration of Mail2Fax settings via the REST API is similar to the fax settings configuration.

C.10.4.1 Retrieving Mail2Fax Configuration

To get the Mail2Fax configuration for the subscriber with ID 3, use the following request:

```
Method: GET
Content-Type: application/hal+json

https://127.0.0.1:1443/api/mailtofaxsettings/3
```

The output format is as follows (only the relevant output data is shown):

```
   "acl" : [],
   "active" : false,
   "secret_key" : "secretkeypassword",
   "secret_key_renew" : "daily",
   "secret_renew_notify" : [
      {
         "destination" : "user1@company.com"
      }
   ]
```

C.10.4.2 Updating Mail2Fax Configuration

The following set of requests changes the Mail2Fax configuration with new secret key settings.

- Secret key value:

```
Method: PATCH
Content-Type: application/json-patch+json
```

```
https://127.0.0.1:1443/api/faxserversettings/3

--data-binary '[ { "op" : "replace", "path" : "/secret_key", "value" : " ↩
    newsecretkeypassword" } ]'
```

- Secret key renewal interval:

```
Method: PATCH
Content-Type: application/json-patch+json

--data-binary '[ { "op" : "replace", "path" : "/secret_key_renew", "value" : "monthly" }  ↩
    ]'
```

- List of email addresses that receive the automatic secret key update notifications:

```
Method: PATCH
Content-Type: application/json-patch+json

--data-binary '[ { "op" : "replace", "path" : "/secret_renew_notify", "value" : [ { " ↩
    destination": "user2@company.com" }, { "destination": "user3@company.com" } ] } ]'
```

C.10.5 Using Advanced Faxserver and Mail2Fax Settings via the REST API

On the NGCP REST API documentation web page you can find the complete list of available Faxserver and Mail2Fax configuration parameters: https://<ngcp_ip_address>:1443/api

Important
The information on the web page is relevant for your platform version and may change in next releases.

After visiting the API documentation main page, you can find the following entries related to Faxserver operations:

- Faxes (https://<ngcp_ip_address>:1443/api/#faxes)
- FaxRecordings (https://<ngcp_ip_address>:1443/api/#faxrecordings)
- FaxserverSettings (https://<ngcp_ip_address>:1443/api/#faxserversettings)

C.11 Troubleshooting

The following log file may be used to check Faxserver functionality: `/var/log/ngcp/faxserver.log`

C.11.1 Session ID (SID)

Faxserver stores basic information about each processed fax in a session file. The most important element within this set of data is the *Session ID* (SID) that uniquely identifies a fax throughout its lifetime.

Session ID is a long hexadecimal string (a kind of UUID) that can be read from the above mentioned Faxserver logfile, and which itself is used also as the filename in files that belong to a specific sent / received fax. An example:

```
root@sp1:~# cat /var/spool/ngcp/faxserver/failed/1e480167-5de6-4cc2-948b-de58d1a0bb8c.err

created: 2016-09-06 04:41:32
caller: 111111111
callee: 222222222
file: 1e480167-5de6-4cc2-948b-de58d1a0bb8c.tif
sid: 1e480167-5de6-4cc2-948b-de58d1a0bb8c
dir: out
attempts: 0
fail_attempts: 3
fail_retry_secs: 60
quality: normal
status: FAILED
error: Internal error
modified: 2016-09-06 17:41:30

root@sp1:~#
```

The data element `sid` is the session ID. Other important elements are:

- `caller` and `callee`: these are probably searched for when trying to figure out what happened to a specific fax transmission, if you don't know the SID
- `dir`: direction of fax transmission: *in'coming or 'out'going or 'mtf* for mail-to-fax
- `status`: shows success or failure
- `error`: the error cause in case of failed faxes

C.11.2 Fax Storage Location

Faxserver stores all of its processed faxes at the path: `/var/spool/ngcp/faxserver/...` Within that directory the most relevant subdirectories are `failed` and `completed` that store the SID file and the fax itself in TIFF format of those faxes that failed or were successful, respectively.

D RTC:engine

D.1 Overview

WebRTC is an open project providing browsers and mobile applications with Real-Time Communications (RTC) capabilities. The RTC:engine protocol is a light weight messaging and signaling protocol for WebSocket clients. Technically it is a WebSocket sub protocol. It consists of JSON messages that are used to initiate and control call dialogs, send chat messages, join and control conferences and share files. It is similar to well known signaling protocols like SIP, but much simpler. It does not care about the underlying network protocols, like SIP does.

D.2 RTC:engine enabling

The RTC:engine is not activated by default and needs a few steps to setup.

D.2.1 Enabling services via CLI

First you have to enable it first on your server via CLI. Connect with SSH on your server, open **/etc/ngcp-config/config.yml** with your editor of choice and change the following properties:

```
fileshare:
  enable: yes

rtcengine:
  conference:
    relay:
      app_id: bormuth
      url: http://xms.sipwise.com:81
    call:
      relay:
      app_id: bormuth
      url: http://xms.sipwise.com:81
  enable: yes
  expose_provisioning_api: yes

www_admin:
  http_csc:
  servername: '$IP_OF_VM'
```

Save the config.yml file and run *$* ***ngcpcfg apply*** **enable rtcengine**. After the script ran, check the status of all services via *$* ***monit summary*** or *$* ***monit status***.

D.2.2 Enabling via Panel for resellers and subscribers

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.

D.2.3 Create RTC:engine session

D.2.3.1 Create sessions

Request:

```
curl -i -X POST --insecure --user SUBSCRIBER_ID:SUBSCRIBER_PW -H 'Content-Type: application ↩
    /json' --data-binary '{}' https://IP_OF_VM/api/rtcsessions/
```

Response Header:

```
Location: /api/rtcsessions/7
```

D.2.3.2 Receive sessions

Request:

```
curl -i -X GET --insecure --user SUBSCRIBER_ID:SUBSCRIBER_PW -H 'Content-Type: application/ ↩
    json' https://IP_OF_VM/api/rtcsessions/{ID_FROM_LAST_REQUEST_HEADER}
```

Response Header:

```
{
  ...
  "rtc_app_name" : "default_default_app",
  "rtc_browser_token" : "22fz8e51-ad6e-481e-a389-15c58c3fe5ac",
  "rtc_network_tag" : "",
  "subscriber_id" : "263"
}
```

Tip
Use **rtc_browser_token** in your cdk.Client.

D.3 RTC:engine protocol details

D.3.1 Terminology

D.3.1.1 Connector

There are two kinds of connectors. The front and the back connectors. The only front connector is the BrowserConnector. It has access to all WebSocket connections and is responsible for delivering RCT:engine protocol messages to the WebSocket clients, and for forwarding messages from the WebSocket clients to the router.

Currently there are four back connectors (SipConnector, XmppConnector, WebrtcConnector, ConferenceConnector). Every back connector implements a certain communication use case.

D.3.1.2 Router

The router is very simple stateless message broker, that is responsible for delivering the messages to the right connector. To decide where to send the message, the router takes a look at the recipient address (to) and forwards the message to the specified connector.

D.3.1.3 User

D.3.1.4 App

An app is a scope for a certain RTC:engine integration. Every user can have multiple apps. And an app contains sessions.

D.3.1.5 Network

A network is a user wide configuration, that maps a custom network name (tag) to a certain back connector. Additionally it can also store network specific configurations. And any account that is related to a certain network, will merge its custom configs with the network configs, and send its messages to the specified connector.

D.3.1.6 Session

D.3.1.7 Account

An account represents the credentials for a specific network. Usually it consists of an identifier like a SIP uri (sip:user@domain.tld) and an access token or rather a password.

D.3.1.8 Browser SDK

The Browser SDK is an abstraction layer on top of the RTC:engine protocol. It is served as bundled javascript library, and provides convenient components and methods for all use cases.

D.3.2 Messages

A typical message created by the browser sdk contains the following fields:

```
{
  "method": "module.action",
  "from": "connector:id",
  "to": "connector:id",
  "session": "session",
  "body": {
    ...
  }
}
```

D.3.2.1 Fields

D.3.2.2 method

It is separated in two parts. The first part is the module. It is a delegation key to separate concerns in the code. The second part is the action, which represents a specific method in a module.

D.3.2.3 from

It represents the current sender of a message. For example the user creates a new call via the browser sdk, the message would look like this:

```
{
  "method": "call.start",
  "from": "",
  "to": "webrtc:b2bua1",
  "session": "session1",
  "body": {
    ...
  }
}
```

The content of the field is completely irrelevant, because the BrowserConnector will overwrite this field. The reason is to avoid user manipulation.

```
{
  "method": "call.start",
  "from": "browser:ws1",
  "to": "webrtc:b2bua1",
  "session": "session1",
  "body": {
    ...
  }
}
```

D.3.2.4 to

In general this field represents the recipient of a message. The recipients address consists of two parts. First part is the prefix that targets the connector. Second part is the identifier of the recipient.

D.3.2.5 session

If you provisioned with the RTCEngine, you get a session and its token property. The browser SDK adds this token to every message.

D.3.2.6 body

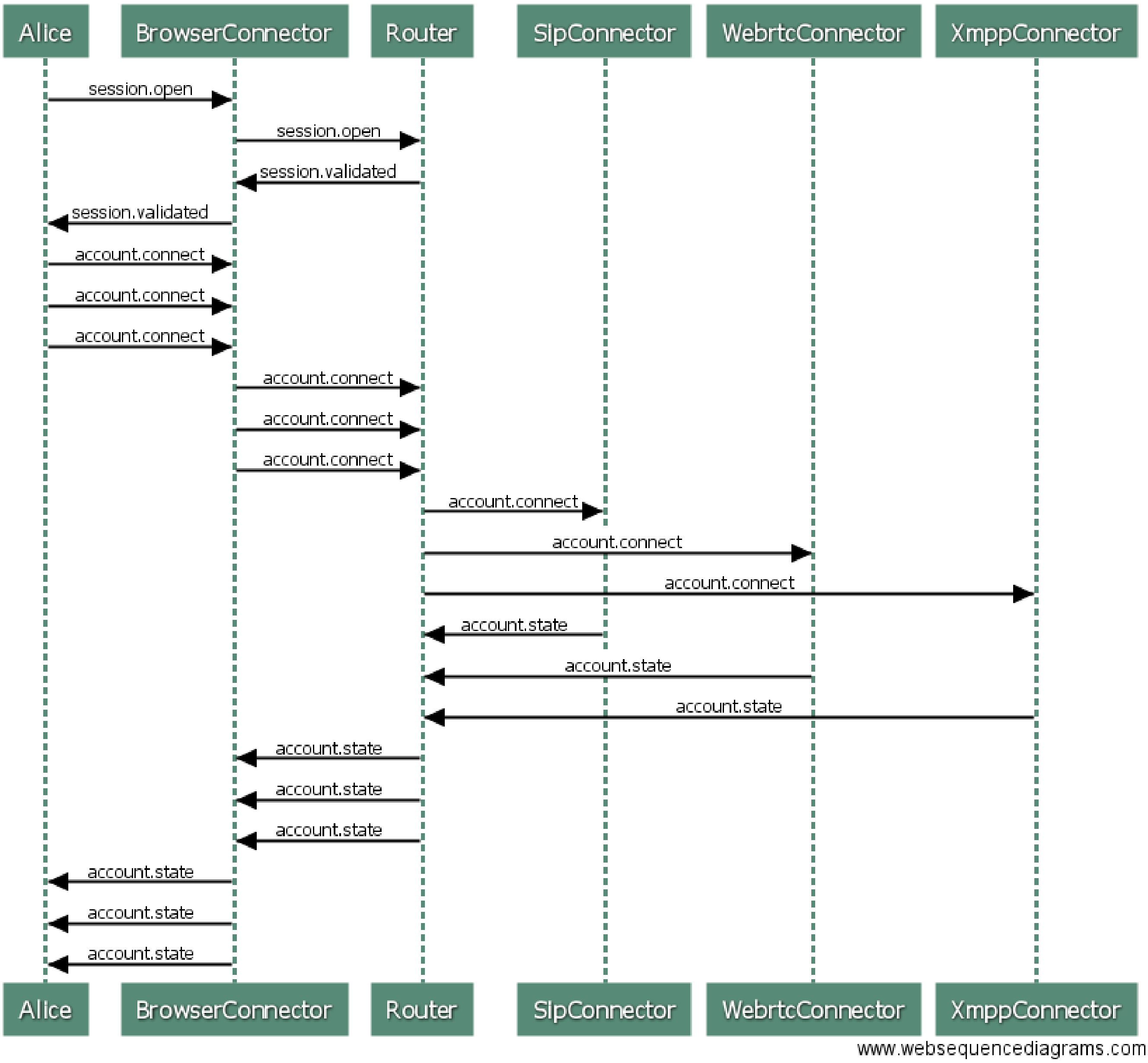
The body contains the payload of the message. Every message type has its own body schema.

D.3.3 Account

Mainly an account consists of credentials (identifier, accessToken), that are needed to authenticate against the related network. Its lifecycle is bound to the lifecycle of the related session.

After RTC:engine received session.open, it responds a session.validated message. This message contains all provisioned accounts in its property "body.accounts".

D.3.3.1 Flow



D.3.3.2 Messages

D.3.3.3 account.connect

RTC:engine needs one message per account. The message should contain the id of the account. The id is the object key in the accounts object from the [session.validated](../session/index.md) message.

```
{
  "from": "",
  "to": "...:...",
  "method": "account.connect",
  "session": "...",
  "body": {
    "id": "..."
  }
}
```

D.3.3.4 account.state

This message gives state information about the authentication and registration process of the related network and the corresponding connector. For example, if the related connector is the SipConnector, it creates a new SIP B2BUA in background, and notify the browser if any state change happens.

```
{
  "from": "...:...",
  "to": "browser:...",
  "method": "account.state",
  "session": "...",
  "body": {
    "id": "...",
    "reason": "...",
    "state": "..."
  }
}
```

D.3.3.5 State reasons

- OK
- CONNECTING
- DISCONNECTING
- SERVICE_UNAVAILABLE
- SERVICE_ERROR
- BAD_CONFIGURATION

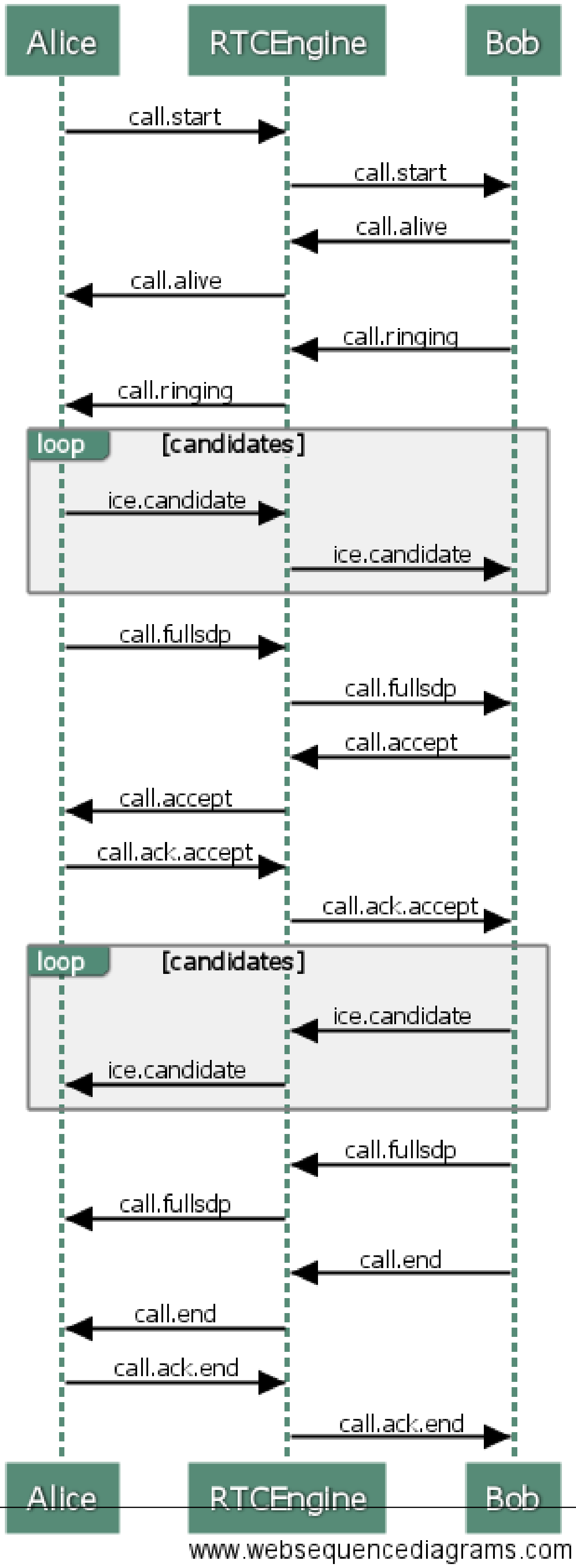
- WRONG_CREDENTIALS
- CONNECTOR_UNAVAILABLE
- CONNECTOR_BUSY
- CONNECTOR_ERROR
- ACCOUNT_NOT_FOUND

D.3.3.6 States

- CONNECTED
- DISCONNECTED

D.3.4 Call

D.3.4.1 Flow



D.3.4.2 call.start

The caller sends this message to the RTC:engine to initiate a new call dialog.

```
{
  "from": "local",
  "to": ["...:..."],
  "method": "call.start",
  "session": "...",
  "body": {
    "id": "...",
    "gcid": "...",
    "account": "..."
    "replace": true|false,
    "trickle": true|false,
    "target": "...",
    "sdp": "..."
  }
}
```

D.3.4.3 Body properties

D.3.4.4 id

The id is a UUID version 4 that identifies the call dialog in the system. But caller and callee never have the same.

D.3.4.5 gcid

Whereas the gcid is a system wide and end-to-end consistent call identifier. It is necessary to track the entire call dialog.

D.3.4.6 account

It contains the callers account id. [(See accounts)](../account/index.md)

D.3.4.7 replace

This property is not used yet. It should support a call handover scenario.

D.3.4.8 trickle

If is set to true, the callee expects ice candidates, before the full sdp delivered by the caller, to accelerate the negotiation process.

D.3.4.9 target

It's the URI (sip:user@domain.tld) of the callee.

D.3.4.10 sdp

The sdp property contains a very early state of the browsers media machine. It contains no ice candidates so far.

D.3.4.11 call.alive

After the callee received the "call.start" message, it responds with a "call.alive" to the RTC:engine, immediately.

```
{
  "from": "...",
  "to": "...",
  "method": "call.alive",
  "session": "...",
  "body": {
    "id": "...",
    "gcid": "..."
  }
}
```

D.3.4.12 call.ringing

After the callee received the "call.start" message, it responds with a "call.ringing" to the RTC:engine, immediately.

```
{
  "from": "...",
  "to": "...",
  "method": "call.ringing",
  "session": "...",
  "body": {
    "id": "...",
    "gcid": "...",
    "account": null
  }
}
```

D.3.4.13 call.accept

The callee sends this message after accepting the call explicitly.

```
{
  "from": "...",
  "to": "...",
```

```
  "method": "call.accept",
  "session": "...",
  "body": {
    "id": "...",
    "gcid": "...",
    "account": null,
    "trickle": true|false,
    "sdp": "..."
  }
}
```

D.3.4.14 call.ack.accept

Caller sends this message after it received the "call.accept" message from the callee.

```
{
  "from": "...",
  "to": "...",
  "method": "call.ack.accept",
  "session": "...",
  "body": {
    "id": "...",
    "gcid": "..."
  }
}
```

D.3.4.15 call.candidate

Both, caller and callee send ice candidates immediately after initiating respectively accepting the call.

```
{
  "from": "...",
  "to": "...",
  "method": "call.candidate",
  "session": "...",
  "body": {
    "id": "...",
    "gcid": "...",
    "candidate": {
      "payload": "...",
      "type": "WEBRTC_LEGACY"
    }
  }
}
```

D.3.4.16 call.fullsdp

Both, caller and callee send this message after the ice gathering finished and all candidates are available.


```
{
  "from": "...",
  "to": "...",
  "method": "call.fullsdp",
  "session": "...",
  "body": {
    "id": "...",
    "gcid": "...",
    "sdp": "..."
  }
}
```


D.3.4.17 call.change. . . .

All messages, that begin with "call.change", are important for renegotiation and glare handling.

D.3.4.18 call.change.lock.reset

D.3.4.19 call.change.lock

D.3.4.20 call.change.lock.ok

D.3.4.21 call.change.offer

D.3.4.22 call.change.answer

D.3.4.23 call.dtmf

Only works if the connector of the related account supports DTMF messages.


```
{
  "from": "...",
  "to": "...",
  "method": "call.dtmf",
  "session": "...",
  "body": {
    "id": "...",
    "gcid": "...",
    "dtmf": "...",
    "account": null
  }
}
```

D.3.4.24 call.end

Both, caller and callee can send this message. It forces the counter part to end and destroy the call.

```
{
    "from": "...",
    "to": "...",
    "method": "call.end",
    "session": "...",
    "body": {
        "id": "...",
        "gcid": "...",
        "reason": "..."
    }
}
```

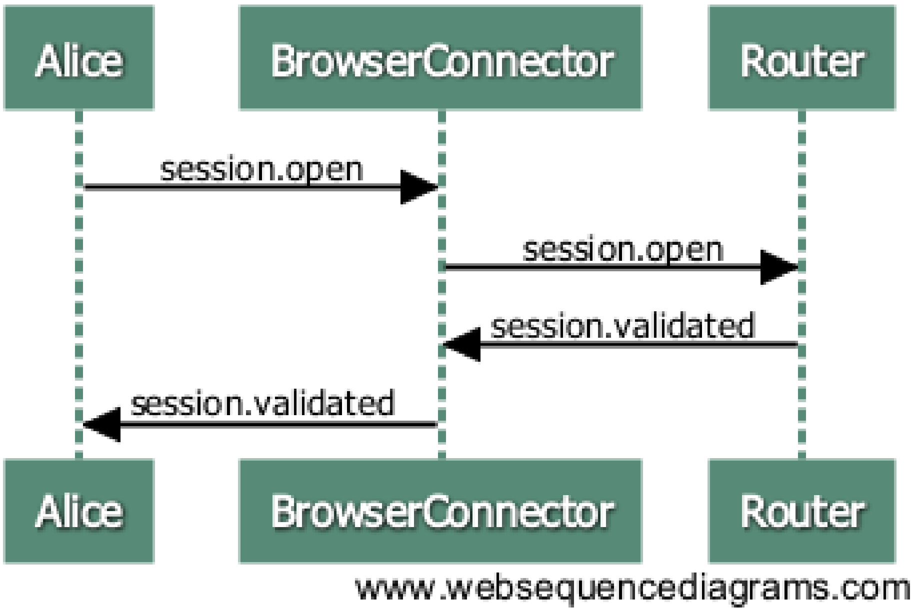
D.3.4.25 call.ack.end

The counter part, that receives the "call.end" message, sends the "call.ack.end" message.

```
{
  "from": "...",
  "to": "...",
  "method": "call.ack.end",
  "session": "...",
  "body": {
    "id": "...",
    "gcid": "...",
    "account": null
  }
}
```

D.3.5 Session

D.3.5.1 Flow



D.3.5.2 Messages

D.3.5.3 session.open

```
{
  "method": "session.open",
  "from": "",
  "to": "",
  "session": "session1",
  "body": {
    "credentials": {
      "userSession": "session1"
    }
  }
}
```

D.3.5.4 session.validated

This message is the response to **session.open**. If the session property is a valid session, you get a response where the result property is true. In addition you get the account information to connect to the networks.

```
{
  "method": "session.validated",
  "from": "core",
```

```
  "to": "browser:ws1",
  "session": "session1"
  "body": {
    "result": true,
    "accounts": {
      "account1": {
        "identifier": "sip:account1@foo.bar"
        "target": "sip-connector:b2bua-account1",
        "network": {
          "tag": "sip-network"
        }
      }
    }
  },
}
```

If something went wrong, result is set to false and an error reason appears.

```
{
  "method": "session.validated",
  "from": "core",
  "to": "browser:ws1",
  "session": "session1"
    "body": {
      "result": false,
      "reason": {
        "type": "invalidToken",
        "message": "Your token is not a valid user session token!"
      }
    }
  },
}
```

D.3.5.5 Reason types

- invalidToken
- tokenExpired
- missingCredentials

E NGCP Internals

This chapter documents internals of the sip:provider PRO that should not be usually needed, but might be helpful to understand the overall system.

E.1 Pending reboot marker

The sip:provider PRO 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.

E.2 Redis id constants

The list of current sip:provider PRO Redis DB IDs:

Service	Redis DB N:	central	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	SNMP agent monitoring data
proxy	redis_db:	10	-	mr4.3+	MT#16079	SIP Loop detection

E.2.1 Redis monitoring keys

The redis monitoring database contains a cache of several current monitoring values. These values are stored in namespaced hashes:

node:<nodename>	Cluster node information.
fsys:<nodename>:<fsysname>	Mounted filesystems information.
proc:<nodename>:<procname>	Monitored processes information.
mysql:<nodename>	MySQL database information.

To access all *fsys* and *proc* hashes there are two sets that list them:

fsys-list:<nodename>	Set of mounted filesystems.
proc-list:<nodenam>	Set of monitored processes.

The *node* hashes contain the following keys:

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).
num_cpus	Total number of CPUs on cluster node.

The *fsys* hashes contain the following keys:

name	The mounted filesystem name (such as */*).
size	The filesystem total size in bytes.
used	The filesystem used size in bytes.

The *proc* hashes contain the following keys:

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* hashes contain the following keys:

last_io_errno	Last IO error number.
last_io_error	Last IO error description.
last_sql_errno	Last SQL error number.
last_sql_error	Last SQL error description.
seconds_behind_master	Delay in seconds since last db replication.
slave_io_running	Status of slave IO thread.
slave_sql_running	Status of slave SQL thread.

E.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*.