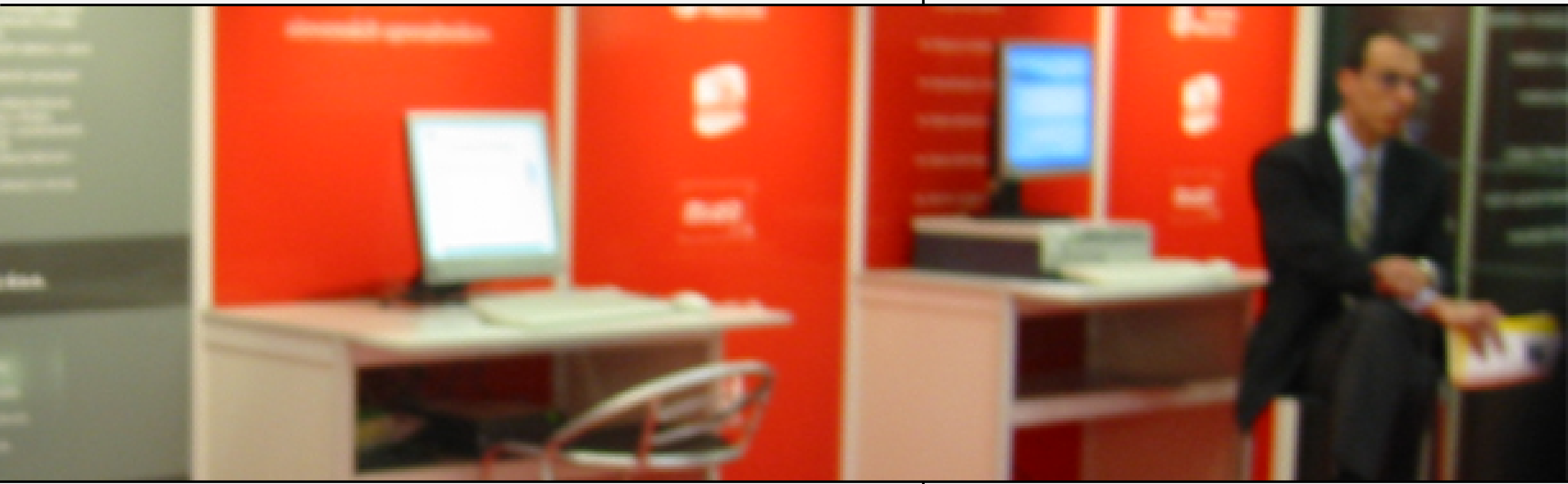
PORTA ® ONE

Porta SIP ®



User Guide

www.portaone.com

Copyright notice & disclaimers

PortaSIP User Guide
V.1.2.1, November 2003

Please address your comments and suggestions to: Sales Department, PortaOne, Inc., Suite 400, 2963 Glen Drive, Coquitlam, BC, V3B 2P7, Canada

Table of Contents

Preface

This document provides a general overview of PortaSIP, a platform for the delivery of enhanced business and residential communications services.

Where to Get the Latest Version of this Guide

The hard copy of this guide is updated at major releases only, and does not always contain the latest material on enhancements occurring in-between minor releases. The online copy of this guide is always up-to-date, integrating the latest changes to the product. You can access the latest copy of this guide at:
http://www.portaone.com/solutions/portasip/docs

Conventions

This publication uses the following conventions:
- Commands and keywords are given in **boldface**
- Terminal sessions, console screens, or system file names are displayed in `fixed width font`

Caution indicates that the described action might result in program malfunction or data loss.
NOTE: Notes contain helpful suggestions about or references to materials not contained in this manual.
Timesaver means that you can save time by performing the action described in the paragraph.

Tips provide information that might help you solve a problem

Hardware and Software Requirements

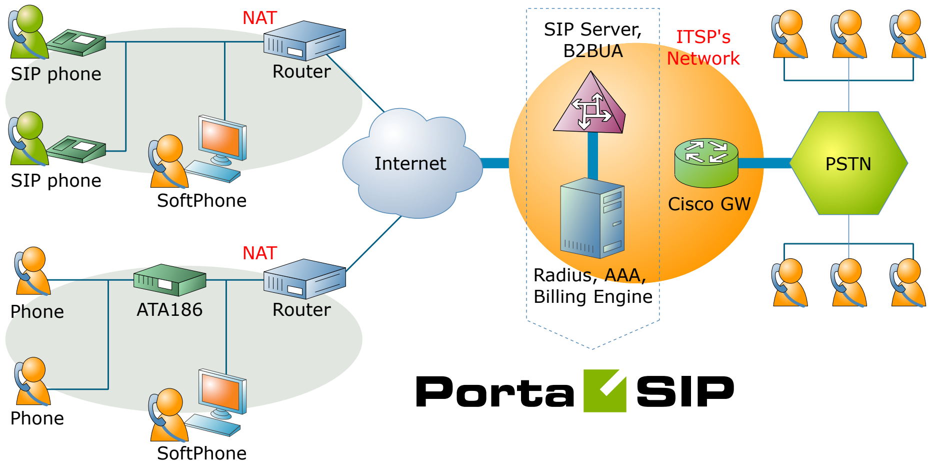
Client System Recommendations

- OS: Windows 95-XP, UNIX or Mac OS
- Browser: Internet Explorer 5 or higher, Netscape 6.2 or higher supporting DOM and with JavaScript enabled.
- Spreadsheet processor (MS Excel)
- Display Settings:
    - Min Screen Resolution: 1024 x 768
    - Color Palette: 16 bit color (minimum)

NOTE: To view downloaded CDR files in Windows please proceed as follows: My Computer -> Control Panel -> Regional Settings -> Number -> List Separator type "," to match the PortaBilling default list separator.

1. System Concepts

PortaSIP's Role in Your VoIP Network

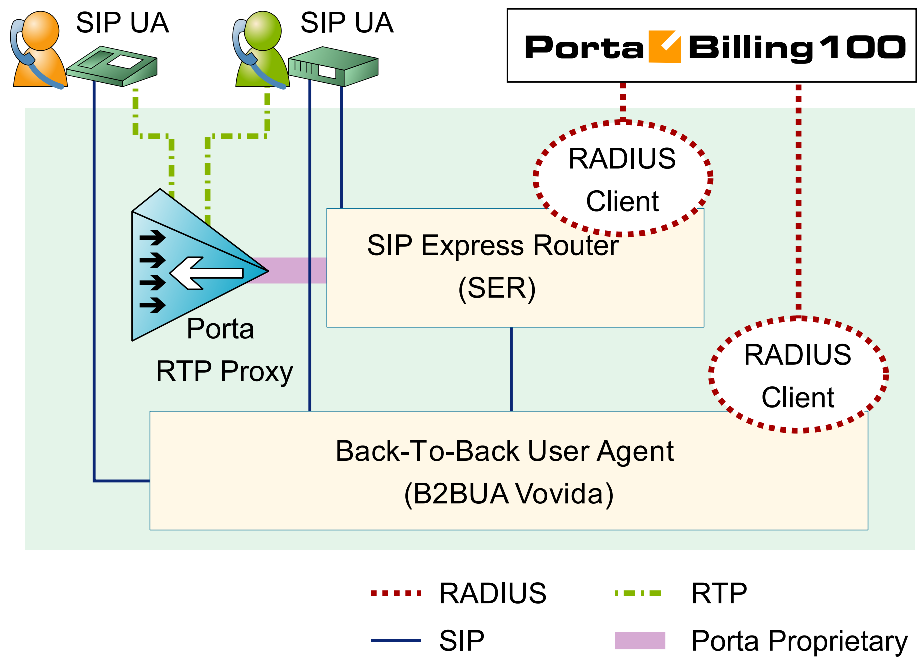


PortaSIP is a call control software package enabling service providers to build scalable, reliable VoIP networks. Based on the Session Initiation Protocol (SIP), PortaSIP provides a full array of call routing capabilities to maximize performance for both small and large packet voice networks.

PortaSIP allows IP Telephony Service Providers to deliver communication services at unusually low initial and operating costs that cannot be matched by yesterday's circuit-switched and narrowband service provider PSTN networks.

In addition to conventional IP-telephony services, PortaSIP provides a solution to the NAT traversal problem and enhances ITSP network management capabilities.

PortaSIP Components



PortaSIP consists of the following main components:

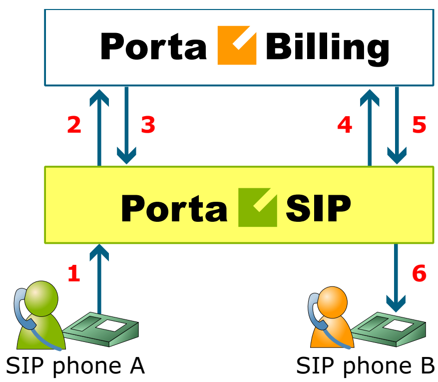
- SIP Proxy Server (SIP Express Router on the diagram): The SIP Proxy Server performs a number of functions, such registering SIP telephones, dealing with NAT issues, etc.
- Back-To-Back User Agent (B2BUA): The B2BUA SIP-based logical entity can receive and process INVITE messages as a SIP User Agent Server (UAS). It also acts as a SIP User Agent Client (UAC), determining how the request should be answered and how to initiate outbound calls. Unlike a SIP proxy server, the B2BUA maintains the complete call state. Integrating B2BUA with PortaSIP ensures that every call made between endpoints (off-net, on-net, etc.) is authorized, authenticated and billed. The system is also able to provide "Debit" billing (i.e. to disconnect a call if the account balance falls below zero).
- RTP Proxy: The RTP Proxy is an optional component used to ensure a proper media stream flow from one SIP telephone to another when one or both of them are behind a NAT firewall.
- Media Server: The Media Server is used to play a number of short voice prompts to an SIP user when an error occurs, such as zero balance, invalid password, and so on.

Call Process / Supported Services

SIP UA ←→ SIP UA

An example: a customer purchases our VoIP services, and two of his employees, A and B, are assigned SIP phone numbers 12027810003 and 12027810009, respectively. For convenience, the administrator creates two abbreviated dialing rules: 120 for 12027810003 and 121 for 12027810009.

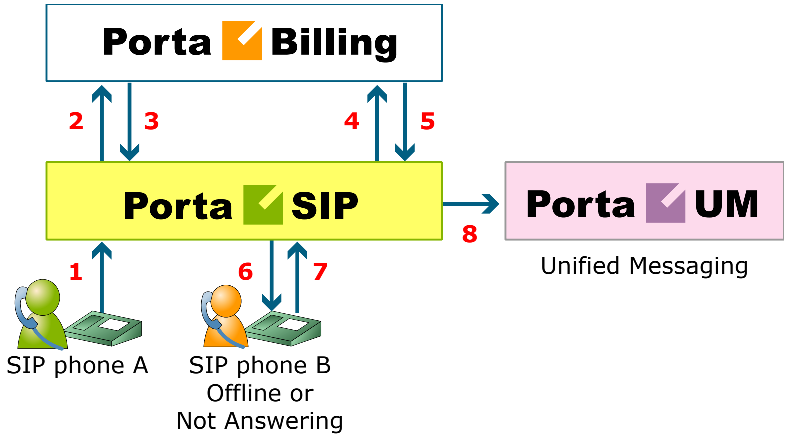
When the called party is online



This is the simplest case:

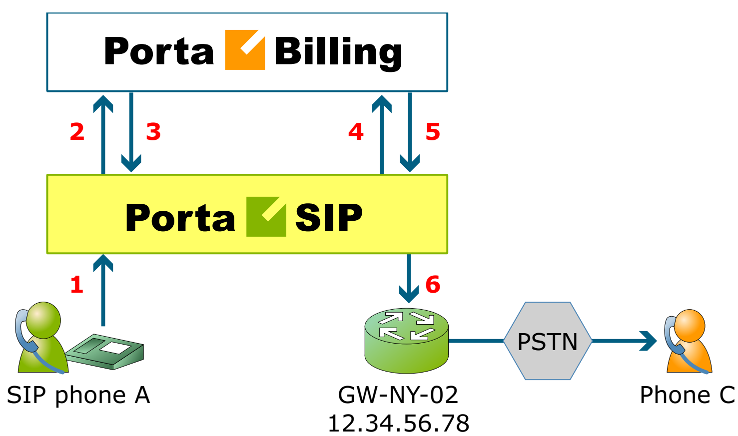
- User A dials B's number (121). His SIP user agent sends an INVITE request to the SIP server (1).
- The SIP server performs a number lookup in the billing (2). For example, if user A dials 121, the billing will inform the SIP server that the actual number is 12027810009, and that this number belongs to B (3).
- The SIP server checks its registration database to find the actual contact address of the SIP user agent with that number.
- The SIP server transfers control to the B2BUA. The B2BUA sends an INVITE request to the billing (4).
- The billing engine checks if A is actually allowed to call that number and what is the maximum call duration (5).
- The SIP server sends INVITE to the SIP user agent for user B (6).
- Depending on the configuration, the SIP server may let A and B's user agents talk directly to each other, or else route the call through the RTP proxy.
- When the call is finished, the SIP server sends accounting information to the billing.

The called party is not online



- User A dials 121 in an attempt to reach B. His SIP user agent sends an INVITE request to the SIP server (1).
- The SIP server performs a number lookup in the billing (2). For example, if user A dials 121, the billing will inform the SIP server that the actual number is 12027810009, and that this number belongs to B (3).
- The SIP server checks its registration database, but finds that this account is not online at the moment. If B has unified messaging services enabled, the call will be redirected to a voice mail system, and A can leave a message for him (8). The same thing would happen if B were online, but not answering his phone (6), (7).
- In any other case, the call will fail.

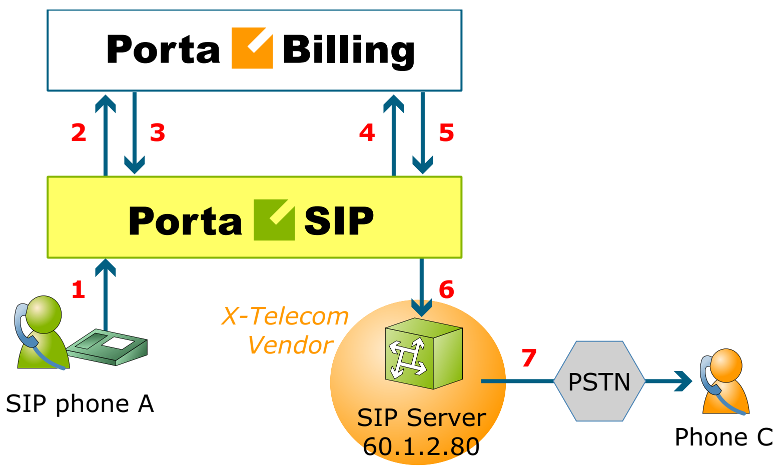
SIP UA → PSTN



- User A attempts to call his co-worker, C. C has not been assigned an SIP phone yet, thus he only has a normal PSTN phone number, 12023001234. A's SIP user agent sends an INVITE request to the SIP server (1).

- The SIP server performs a number lookup in the billing (2). It finds that this number does not belong to any SIP accounts, i.e. it does not exist in the customer's VoIP network, and therefore must be routed to the proper vendor for termination (3).
- The SIP server transfers control to the B2BUA. The B2BUA sends an INVITE request to the billing, asking for routing information (4).
- The billing engine checks if A is actually allowed to call this number and what is the maximum call duration. After that, it checks which vendors are able to terminate the call to 12023001234 and produces a list of possible routes according to routing preferences (5).
- The SIP server tries to send a call to all routes returned by the billing sequentially, until either a connection is made or the list of routes is exhausted (6).
- When the call is finished, the SIP server sends accounting information to the billing.

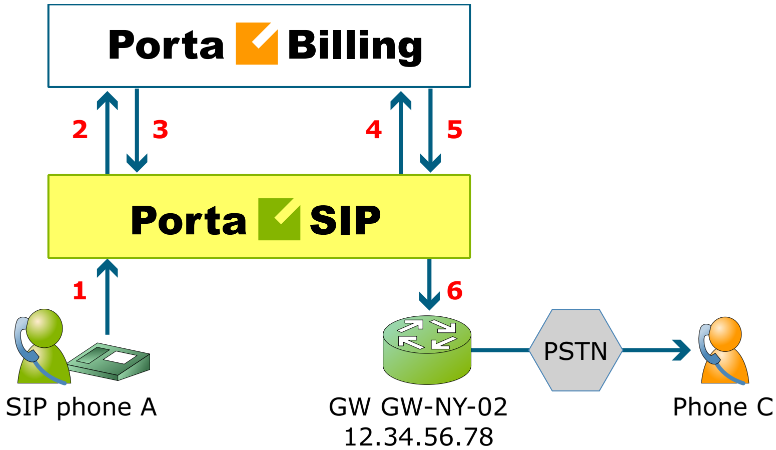
Terminating SIP calls to a vendor using VoIP



- An example: we are able to terminate calls to the US and Canada to a vendor, X-Telecom. This would then be described as a **VoIP to vendor** connection in the billing, with the remote address being the address of the vendor's SIP server (or SIP-enabled gateway).
- The billing engine returns the IP address of the vendor's SIP server in the route information, with login/password optional. The PortaSIP server sends an INVITE request to that address (providing the proper credentials), and then proceeds in basically the same way as if it were communicating directly with C's SIP user agent.
- After the call is established, the B2BUA starts the call timer, disconnecting the call once the maximum call duration is exceeded.

- After the call is completed, the B2BUA sends accounting information for the call to the billing.

Terminating SIP calls to a vendor using telephony

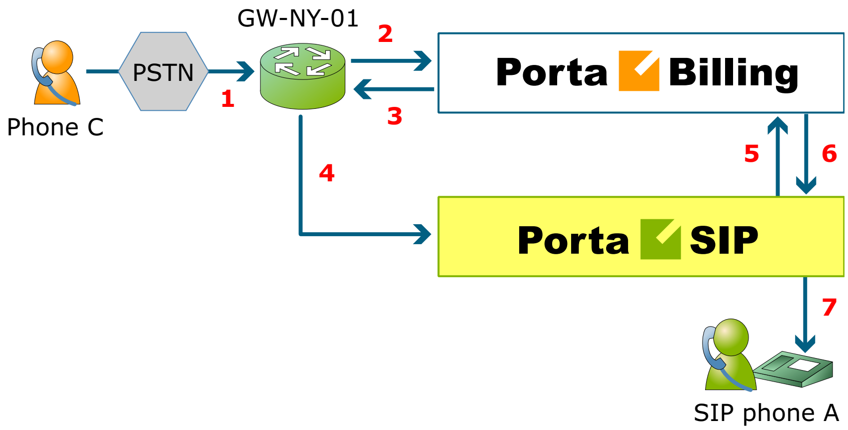


- Let's assume that T1 is connected to Qwest on our gateway **GW-NY-02** in New York, where we are able to terminate calls to the US. This connection would be described as a **PSTN to vendor** connection. The PortaSIP server obtains the address of the GW-NY-02 gateway in the route information.
- The B2BUA sends an INVITE to the remote gateway (GW-NY-02).
- GW-NY-02 performs authentication on the incoming call via the remote IP address. Even if the call was actually originated by A (a dynamic IP address), but the INVITE request to GW-NY-02 arrived from the PortaSIP server, the PortaSIP's IP address will be authenticated. Since PortaSIP is defined as our node, authentication will be successful.

NOTE: Remote IP authentication on the gateway is not required in this case, but is highly recommended. Otherwise, someone else might try to send calls directly to the gateway, bypassing authentication and making such calls for free.

- The call will be routed to the PSTN on the gateway.
- After the call is established, the B2BUA starts the call timer, disconnecting the call once the maximum call duration is exceeded.
- After the call is completed, the B2BUA sends accounting information for the two VoIP call legs to the billing. The gateway will also send accounting information about the answer/VoIP and originate/Telephony call legs. The billing engine will combine this information, since accounting from the SIP server allows us to identify who made the call, while accounting from the gateway carries other useful information – for example, through which telephony port the call was terminated.

PSTN->SIP



This is another important aspect of SIP telephony. Your subscribers not only want to make outgoing calls, they also want other people to be able to call them on their SIP, regardless of where they are at the moment. In order to do so, you will need to obtain a range of phone numbers from your telecom operator, and make sure that calls made to these numbers on the PSTN network are routed to your gateway via the telephony interface.

- C wishes to call A. He thus dials A's phone number (since C is in the US, he dials it using the North American format, 2027810003).
- This call is routed through the telecom network to gateway GW-NY-01. When the incoming call arrives on the gateway (1), it starts a special TCL application to handle this call. This application does several things:
    - Converts the phone number to the E.164 format, so that 2027810003 becomes 12027810003.
    - Performs authorization in the billing (2) – whether A is allowed to receive incoming telephony calls from GW-NY-01, and, if you charge for incoming calls, what is the maximum call time allowed, based on A's current balance (3). One important point is that authorization should happen without a password check, since the application does not know the valid password for the SIP account.
    - Starts outgoing call to 12027810003.
    - Starts the timer once the call is established, disconnecting the call when the maximum call duration is exceeded.
    - The gateway is configured such that it knows that calls to 1202781…. numbers should be sent to the PortaSIP server, thus it sends an INVITE to PortaSIP (4).

NOTE: The gateway cannot make this call "on behalf" of A, since even if we know A's account ID, we do not know A's password; therefore, such a call will be rejected. In addition, Cisco gateways currently do not support INVITE with authorization.

- PortaSIP receives the INVITE, but without authorization information. So the PortaSIP server performs authentication based on the IP address (**5**), (**6**). Since this call is made from our trusted node – gateway GW-NY-01 – the call is authorized.
- PortaSIP checks if the SIP user agent of the dialed number (12027810003) is registered at the time. If yes, a call setup request is sent (**7**).
- If the dialed number belongs to an SIP account with unified messaging services enabled, but this account is not online at the moment or does not answer, the call will be redirected to a voice mail system.
- After the call is completed, the B2BUA sends accounting information for the two VoIP call legs to the billing. The gateway will also send accounting information about the answer/Telephony and originate/VoIP call legs. The billing engine will combine this information, since accounting from the SIP server allows us to recognize that the call was terminated directly to the SIP user agent, and not to a vendor, while accounting from the gateway will contain information as to which account should be billed for this call.

Understanding SIP Call Routing

When a call goes through the PortaSIP server, the SIP server may:

- Direct the call to one of the registered SIP clients, if the called number belongs to the registered agent.
- Optionally, direct the call to a voice mailbox, if the called number belongs to an account in PortaBilling, but this account is not currently registered to the SIP server, i.e. is off-line (PortaUM is required in this case).
- Route the call to one of the gateways for termination, according to the routing rules specified in PortaBilling.

Routing of SIP On-net Calls

The SIP server automatically maintains information about all currently registered SIP user agents, so it is able to determine that a call should be sent directly to the SIP user agent. In addition, the billing engine informs the SIP server, in response to a LOOKUP request, as to whether the dialed number is actually a valid SIP account. In this case, for instance, a call can be redirected to the unified messaging service if this account is not available online at the moment.

Routing of SIP Off-net Calls

You can have different vendors for terminating off-net calls. For example, calls to the US can be terminated either to AT&T, via a T1 connected to your gateway in New York, or by sending the call to a remote gateway from Qwest. You need a tool allowing you to manage routing policies for the different destinations. This tool is extensions routing for tariffs. Tariffs define the termination costs for each connection to a vendor, while extensions routing simply adds a few more fields to the rates in a given tariff. This allows you to easily manage both termination costs and routing from a single location on the web interface. The routing principle is simple:

- The SIP server asks PortaBilling for routing destinations for a number.
- PortaBilling checks every tariff with routing extensions associated with connection to the vendor for rates matching this phone number.
- A list of possible termination addresses will be produced (this will include remote IP addresses for the VoIP connections and IP addresses of your own nodes with telephony connections).

- This list will be sorted according to the routing preference, with entries after the first huntstop being ignored.
- A list of these IP addresses (with optional login and password for SIP authentication) will be returned to the SIP server.

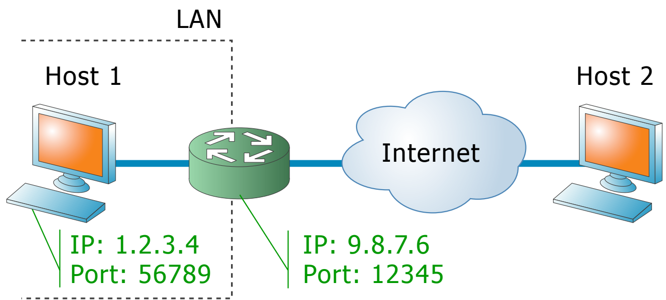
NAT Traversal Guidelines

NAT Overview

The purpose of NAT (Network Address Translation) is to allow multiple hosts on a private LAN not directly reachable from a WAN to send information to and receive it from hosts on the WAN. This is done with the help of the NAT server, which is connected to the WAN by one interface with a public IP address, and to the LAN by another interface with a private address. This document describes issues connected with the implementation of NAT and its implications for the operation of PortaSIP, with an overview of some fundamental NAT concepts.

The NAT server acts as a router for hosts on the LAN. When an IP packet addressed to a host on the WAN comes from a host on the LAN, the NAT server replaces the private IP address in the packet with the public IP address of its WAN interface and sends the packet on to its destination. The NAT server also performs in-memory mapping between the public WAN address the packet was sent to and the private LAN address it was received from, so that when the reply comes, it can carry out a reverse translation (i.e. replace the public destination address of the packet with the private one and forward it to the destination on the LAN).

Since the NAT server can potentially map multiple private addresses into a single public one, it is possible that a TCP or UDP packet originally sent from, for example, port A of the host on the private LAN will then, after being processed in the translation, be sent from a completely different port B of the NAT's WAN interface. The following figure illustrates this: here "HOST 1" is a host on a private network with private IP address `1.2.3.4`; "NAT" is the NAT server connected to the WAN via an interface with public IP address `9.8.7.6`; and "HOST 2" is the host on the WAN with which "HOST 1" communicates.



A problem relating to the SIP User Agent (UA) arises when the UA is situated behind a NAT server. When establishing a multimedia session, the NAT server sends UDP information indicating which port it should use to send a media stream to the remote UA. Since there is a NAT server between them, the actual UDP port to which the remote UA should send its RTP stream may differ from the port reported by the UA on a private LAN (`12345` vs. `56789` in the figure above) and there is no reliable way for such a UA to discover this mapping.

However, as was noted above, the packets may not have an altered post-translation port in all cases. If the ports are equal, a multimedia session will be established without difficulty. Unfortunately, there are no formal rules that can be applied to ensure correct operation, but there are some factors which influence mapping. The following are the major factors:

- How the NAT server is implemented internally. Most NAT servers try to preserve the original source port when forwarding, if possible. This is not strictly required, however, and therefore some of them will just use a random source port for outgoing connections.

- Whether or not another session has already been established through the NAT from a different host on the LAN with the same source port. In this case, the NAT server is likely to allocate a random port for sending out packets to the WAN. Please note that the term "already established" is somewhat vague in this context. The NAT server has no way to tell when a UDP session is finished, so generally it uses an inactivity timer, removing the mapping when that timer expires. Again, the actual length of the timeout period is implementation-specific and may vary from vendor to vendor, or even from one version by the same vendor to another.

NAT and SIP

There are two parts to a SIP-based phone call. The first is the signaling (that is, the protocol messages that set up the phone call) and the second is the actual media stream (i.e. the RTP packets that travel directly between the end devices, for example, between client and gateway).

SIP Signaling

SIP signaling can traverse NAT in a fairly straightforward way, since there is usually one proxy. The first hop from NAT receives the SIP messages from the client (via the NAT), and then returns messages to the same location. The proxy needs to return SIP packets to the same port it received them from, i.e. to the `IP:port` that the packets were sent from (not to any standard SIP port, e.g. 5060). SIP has tags which tell the proxy to do this. The "received" tag tells the proxy to return a packet to a specific IP and the "rport" tag contains the port to return it to. Note that SIP signaling should be able to traverse any type of NAT as long as the proxy returns SIP messages to the NAT from the same source port it received the initial message from. The initial SIP message, sent to the proxy `IP:port`, initiates mapping on the NAT, and the proxy returns packets to the NAT from that same `IP:port`. This is enabled in any NAT scenario.

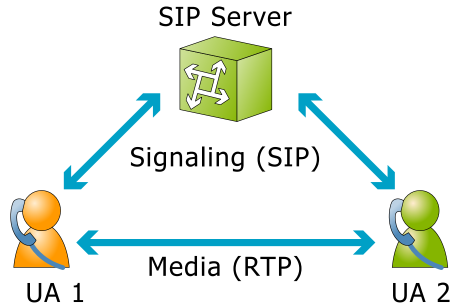
Registering a client which is behind a NAT requires either a registrar that can save the `IP:port` in its registration information, based on the port and IP that it identifies as the source of the SIP message, or a client that is aware of its external mapped address and port and can insert them into the contact information as the `IP:port` for receiving SIP messages. You should be careful to use a registration interval shorter than the keep-alive time for NAT mapping.

RTP – Media Stream

An RTP that must traverse a NAT cannot be managed as easily as SIP signaling. In the case of RTP, the SIP message body contains the information that the endpoints need in order to communicate directly with each other. This information is contained in the SDP message. The endpoint clients fill in this information according to what they know about themselves. A client sitting behind a NAT knows only its internal `IP:port`, and this is what it enters in the SDP body of the outgoing SIP message. When the destination endpoint wishes to begin sending packets to the originating endpoint, it will use the received SDP information containing the internal `IP:port` of the originating endpoint, and so the packets will never arrive.

Understanding the SIP Server's Role in NAT Traversal

Below is a simplified scheme of a typical SIP call:



It must be understood that SIP signaling messages between two endpoints always pass through a proxy server, while media streams usually flow from one endpoint to another directly. Since the SIP Server is located on a public network, it can identify the real IP addresses of both parties and correct them in the SIP message, if necessary, before sending this message further. Also, the SIP Server can identify the real source ports from which SIP messages arrive, and correct these as well. This allows SIP signaling to flow freely even if one or both UAs participating in a call are on private networks behind NATs.

Unfortunately, due to the fact that an RTP media stream uses a different UDP port, flowing not through the SIP server but directly from one UA to another, there is no such simple and universal NAT traversal solution. There are 3 ways of dealing with this problem:

1. Insert an RTP proxy integrated with the SIP Server into the RTP path. The RTP proxy can then perform the same trick for the media stream as the SIP Server does for signaling: identify the real source IP address/UDP port for each party and use these addresses/ports as targets for RTP, rather than using the private addresses/ports indicated by the UAs. This method helps in all cases where properly configured UAs supporting symmetric media are used. However, it adds another hop in media propagation, thus increasing audio delay and possibly decreasing quality due to greater packet loss.

2. Assume that the NAT will not change the UDP port when resending an RTP stream from its WAN interface, in which case the SIP Server can correct the IP address for the RTP stream in SIP messages. This method is quite unreliable; in some cases it works, while in others it fails.

3. Use “smart” UAs or NAT routers, or a combination of both, which are able to figure out the correct WAN IP address/port for the media by themselves. There are several technologies available for this purpose, such as STUN, UPnP and so on. A detailed description of them lies beyond the scope of this document, but may easily be found on the Internet.

NAT Call Scenarios and Setup Guidelines

In the context of NAT traversal, there are three distinct SIP call scenarios, each of which should be handled differently. These scenarios differ in that, in cases 1 and 3, the media stream will always pass through one or more NATs, as the endpoints cannot communicate with each other directly, while in case 2 it is possible to arrange things so that a media stream flows directly from one endpoint to another through a LAN. These scenarios are as follows:

1. A call is made from/to a UA under the NAT from/to another UA on the WAN or under a different NAT. There are three main approaches to this scenario:
   - Enable an RTP proxy integrated into the PortaSIP. With proper UAs, this allows correct NAT traversal for the media in all cases.
   - Assume that clients' NAT will not change their UDP port when resending an RTP stream from the WAN interface, in which case the SIP Server will correct the IP address for the RTP stream in SIP messages. This method will not work in all cases, so the best you can do is compile a list of specific NAT servers (AKA routers) which always try to preserve the original source port number when forwarding traffic from the external WAN interface (see Appendix A for a list of some that we have already tested). Since this information is usually not easily available in the documentation, you should either inquire from the manufacturer or use a trial-and-error approach. Also, if possible, try to configure a different RTP port in each UA sold to clients, which will allow them to make calls from/to several devices under the same NAT from/to UAs outside NAT. Quite large UDP port space should be allocated for general use (more than 60,000 distinct ports); there should be no problem in doing this, even for large-scale VoIP providers. When selecting the allowed UDP ports, care should be taken to exclude ports used by any popular UDP-based services (such as ICQ).
   - Use "smart" UAs or NAT routers, or a combination of both, which are able to figure out the correct WAN IP address/port for the media by themselves. There are several technologies available for this purpose, such as STUN, UPnP and so on. Also, some modern routers available on the market are able to rewrite SIP messages on the fly, in such a way that they contain the correct `IP:port` of the NAT's WAN interface, thus solving the problem completely. It might be feasible to compile a list of such models and recommend them to new clients who are planning to acquire a router.

2. A call is made from/to a UA under the NAT from/to another UA under the same NAT. This scenario is likely to be encountered in a corporate environment, where a company may decide to use VoIP technologies to extend or replace its existing telephony infrastructure. In such cases, employees located in the same private network should be able to call each other from their IP phones. PortaSIP already handles such situations correctly out of the box, by allowing the RTP flow from one UA to another via the LAN, so that no additional setup/configuration is necessary.

3. A call is made from/to a UA under the NAT from/to a Cisco GW with support for SIP COMEDIA extensions. This scenario is probably the most common one. For example, a home user installs a router to share one DSL/Cable connection among several machines, and then also installs a SIP phone for making cheap long-distance calls. In this case, you need to configure your Cisco GW as per Appendix B in order to ensure proper NAT traversal.

In Appendixes A through C you will find a list of tested routers, as well as a typical configuration for Cisco IOS software and Cisco ATA 186 telephones that has been adapted for optimal NAT traversal performance.

2. Setting Up SIP Services

Please refer to the *PortaBilling Administrator Interface* PDF file:
(**http://www.portaone.com/resources/PB-100_Administrator_Interface.pdf**)
for detailed instructions on how to navigate and operate the web interface, as well as detailed explanations for particular fields.

Initial Configuration of PortaBilling

The following steps are normally performed only once, after the system is installed. Proceed as follows:

Visit **Company Info** on the main menu. Enter information about your company and set up your base currency. Naturally, this does not limit your operations to this currency only. However, on cost/revenue reports and the like different currencies will be converted to the one you specify here.
NOTE: Once you set up a base currency it cannot be changed. If you make a mistake, you will have to start with a new PortaBilling environment.
From the main menu, choose **Users** and create login entries for users who will be working with the system. It is not recommended that the default PortaBilling root user (pb-root) be used for any operations other than initial set-up. Make sure you are able to login as the newly-created user, and change the password for the pb-root user.
NOTE: It is possible that you will require assistance from PortaBilling support personnel in the future. In order to provide such assistance, they will need access to the web interface. Therefore, when submitting a problem report, please either provide them with a new password for the **pb-root** user, or create a special user with root permissions for them.
If you plan to do billing in multiple currencies, define these in the **Currencies** section and specify exchange rates in **Exchange Rates.**

Create Destinations

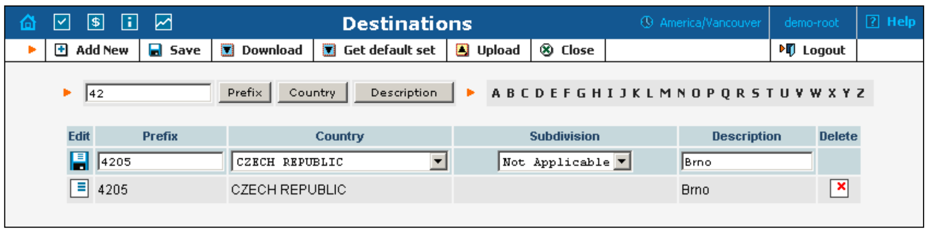
This step is only required if you have not previously defined the necessary destinations. There are two ways to insert a new destination into the system:

- One-by-one, using the **Add new** functionality on the web interface.
- A bulk update, by uploading destinations from a file.

NOTE: **PortaBilling supplies a file with a set default destination, which you can download and then upload to the server. However, it is possible that your business requires different types of prefixes, so please check the data in the file before uploading.**

Creating destinations "one-by-one":

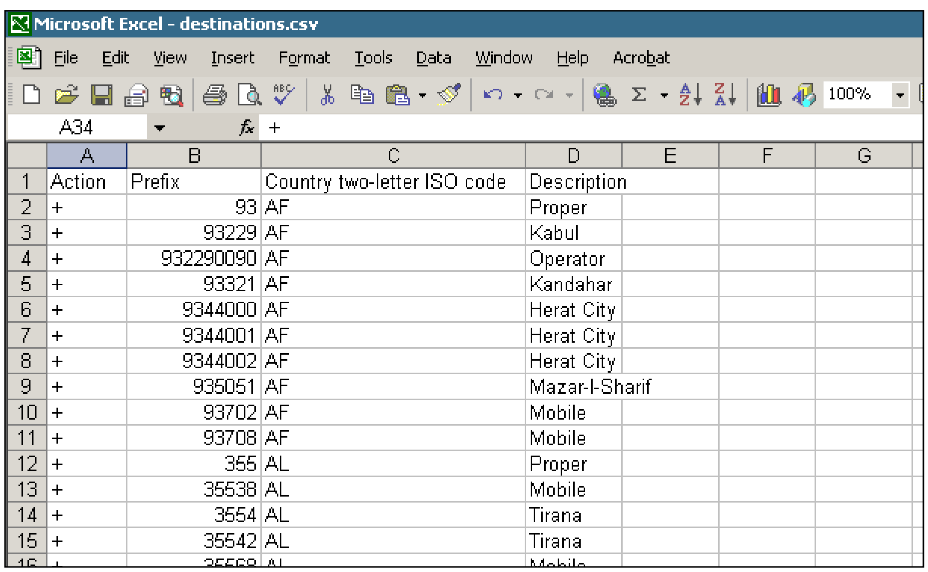
1. In the Management section of Admin-Index, choose **Destinations**.
2. Click on the **+ Add New** button.
3. Fill in the required information. This includes the phone prefix and country name. The country subdivision is optional. You can use the **Description** column to store extra information about the destination (for example, if it is a mobile or fixed number).



4. Click **Save**.
5. Repeat these steps for any additional destinations you would like to add.

Uploading a set of destinations from a file:

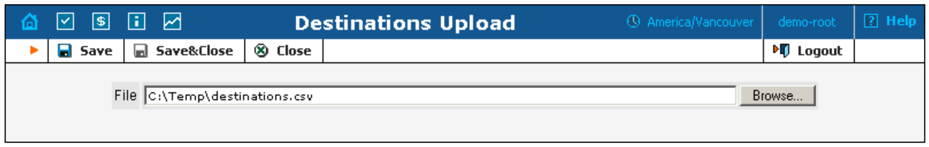
1. In the Management section of Admin-Index, choose **Destinations**.
2. Click on **Default set** to download a set of destinations as a CSV (Comma-Separated Values) file.



	A	B	C	D	E	F	G
1	Action	Prefix	Country two-letter ISO code	Description			
2	+	93	AF	Proper			
3	+	93229	AF	Kabul			
4	+	932290090	AF	Operator			
5	+	93321	AF	Kandahar			
6	+	9344000	AF	Herat City			
7	+	9344001	AF	Herat City			
8	+	9344002	AF	Herat City			
9	+	935051	AF	Mazar-I-Sharif			
10	+	93702	AF	Mobile			
11	+	93708	AF	Mobile			
12	+	355	AL	Proper			
13	+	35538	AL	Mobile			
14	+	3554	AL	Tirana			
15	+	35542	AL	Tirana			

3. Open this file in Microsoft Excel or any other suitable program. Edit the data if necessary.

4. Save the file and close it in Excel.
5. Switch back to the PortaBilling web interface, and click **Upload** on the Destinations screen.



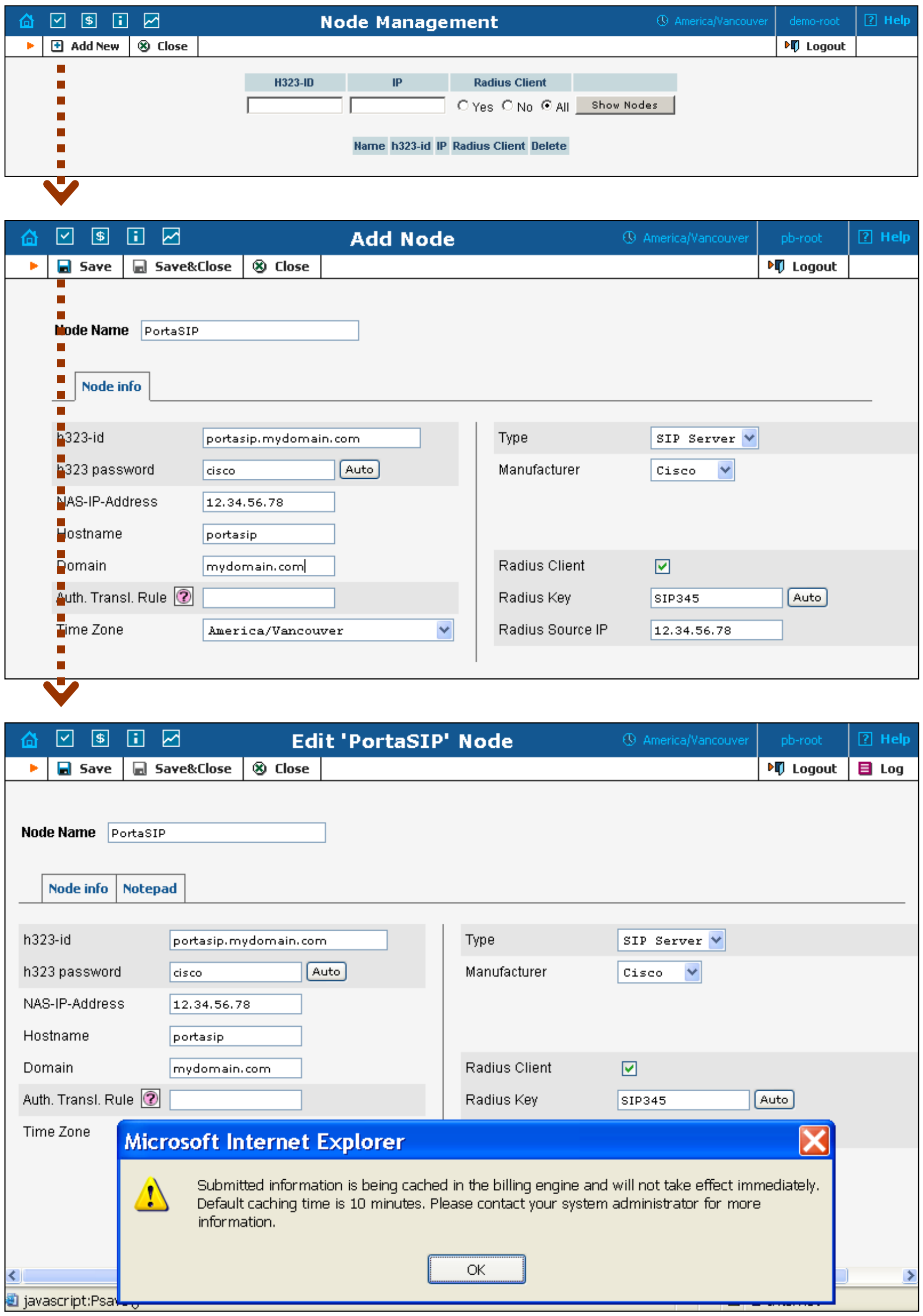
6. Type in the filename for the file you have edited, or click on the **Browse…** button and select the file.
7. Click **Save&Close.**

Destinations for SIP phones

In order to receive an incoming call, an SIP user agent must be configured with a phone number. Normally, you will obtain a range of phone numbers from your local telecom, and you will be able to assign these to your customers. For example, you will be assigned range 12027810000 – 12027819999. It is, therefore, a good idea to create a special destination **1202781**. This prefix will cover all of your SIP phones, and thus its actual purpose is to set up your pricing or routing.
Even if you have not obtained an official phone prefix, it is highly recommended not to assign IDs to your SIP user agents at random. Choose a non-existing prefix, e.g. **099**, and create it as the destination with **N/A** country and the description **SIP phones**. Then use SIP IDs such as **09900001, 0990002, … 0990999, …**

Create Nodes

Now you have to enter your SIP server (and, optionally, other gateways) as nodes. PortaBilling requires some key information about your network equipment, such as the IP address, h323-id, Radius shared secret, and so on.



1. In the management section of the Admin-Index page, choose **Nodes.**
2. In the Node management window, click the **Add new** icon.

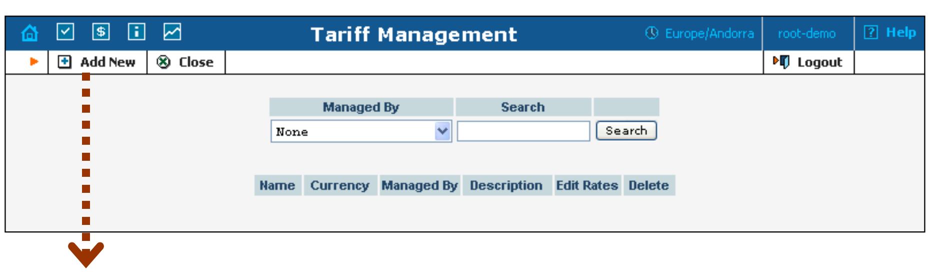
3. Fill in the New Node form:
   - **Node name** – a short descriptive name for your SIP server (this will be used in the select menus).
   - **H323-ID** (recommended: `hostname.domainname`)
   - **H323 Password** – if you plan to send calls from your SIP server to your Cisco gateways, where the default Cisco remote IP authentication script will be used, enter **`cisco`** here.
   - **NAS-IP-Address** – the IP address of the SIP server.
   - **Auth. Translation rule** – if you plan to use E.164 numbering for your SIP phones (highly recommended), you can just leave this empty.
   - **Type** – H.323 node type; select **SIP Server**.
   - **Manufacturer** – select **Cisco** (even if the PortaSIP is not manufactured by Cisco, it uses a Cisco-compatible means of communicating with the billing).
   - **Radius Client** – check this, since PortaSIP will need to communicate with the billing.
   - **Radius Key** – enter the radius shared secret here; this must be the same **key** which you entered during the PortaSIP installation.
   - **Radius Source IP** – see the **Node ID, NAS IP address and Radius source IP** section in **PortaBilling100 User Guide** for more information. Unless your PortaSIP server uses multiple network interfaces, the value here should be the same as the NAS-IP-Address.
4. Click **Save&Close**.
5. Repeat steps 2-4 for any additional gateways you may have. Use **VOIP-GW** as the node type.

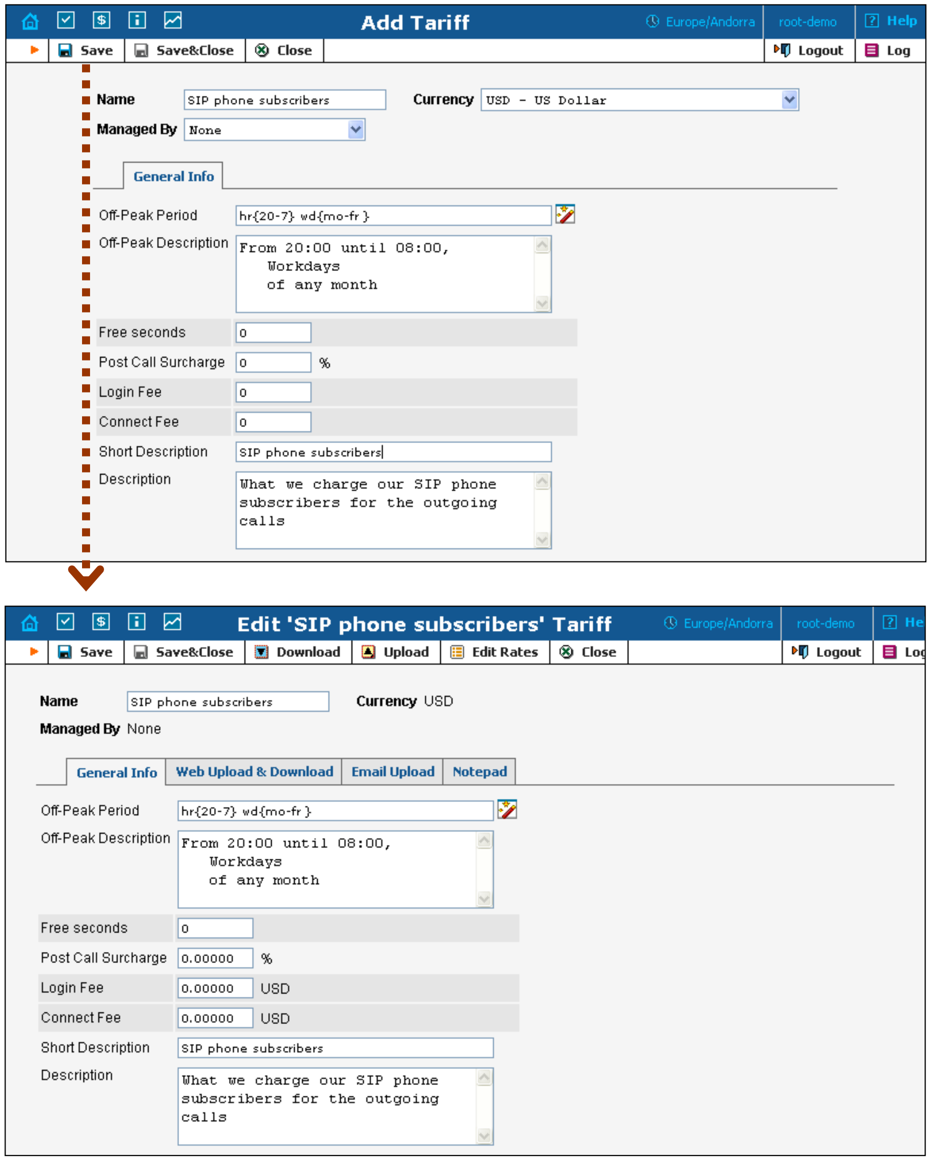
NOTE: There is some propagation delay between the database and the Radius server configuration file; however, it is no more than 15 minutes.

Create Tariff

The tariff is a single price list for calling services or for your termination costs. A tariff combines:

- conditions which are applicable for every call regardless of the called destination;
- per destination rates.





1. In the Management section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, choose **Add New**.
3. Fill in the **New Tariff** form:
   - **Name** – a short name for the tariff object; this is the name you will then see in the select menus.
   - **Managed by** – If this is a tariff that describes your vendor's termination costs, choose **None – Routing Ext.** here, as this tariff will be used not only to calculate termination costs, but also for routing SIP calls. If you want this tariff to be used for your wholesale customer accounts, so that the customer himself can edit rates in this tariff , choose a customer name from the menu. Otherwise, if this is the tariff for a retail customer's accounts, choose **None**.

- **Off-peak Period** – Defines the off-peak period. Click on the Off-peak period wizard icon ( ) to summon the wizard, which will help you construct the correct period definition. Click **Help** for more information on period format definition. If you do not differentiate between peak and off-peak rates, just leave this field empty.
- **Off-Peak Description** – a description of the off-peak period, automatically filled in by the off-peak period wizard; thus you do not have to fill in this field.
- **Currency** – Indicates in which currency pricing information is defined. All pricing information for a single tariff must be defined in the same currency.

NOTE: The currency for the tariff may be chosen only once, and cannot be changed later.

- **Free seconds** – The number of free seconds allowed for each call. In order to claim free seconds, the length of the call must be at least one billing unit (first interval; see the 'Enter Rates' section above).
- **Post Call Surcharge** – percentage of the amount charged for the call.
- **Login Fee** – amount to be charged immediately after the first user authentication (i.e. after the user enters his PIN).
- **Connect Fee** – amount to be charged for each connected call (call with a non-zero duration).
- **Short Description** – A short tariff description will be shown in the rate lookup on the admin interface and the self-care pages for your accounts and customers. For example, you can have a tariff called **EasyCall-tollfree**, and the short description will provide more information such as **Easy Call – when using a toll-free number.** This field is mandatory.
- **Description** – an extended tariff description.

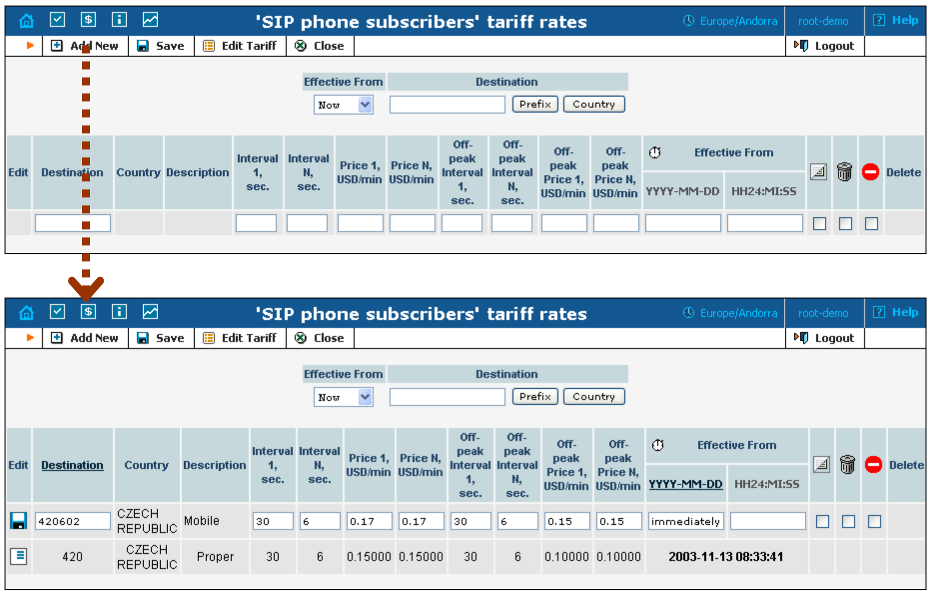
4. Click **Save**.
5. Repeat steps 1-4 until you have entered all of the tariffs. You will need at least two tariffs – one which you will use to charge your customers, and another which describes your termination costs. Make sure you choose **None – Routing Ext.** in the **Managed by** select menu when creating tariffs for your vendors.

Enter Rates

Rates are per-destination prices. Please refer to the **System Concepts** chapter for more details on billing parameters.

Managing rates online

Managing rates online is very convenient for maintaining existing rate tables, as well as for reference purposes. For new price lists or for major updates, an offline method is better.



1. On the Tariff Management page you will see a list of the available tariffs. Click the **Edit Rates** icon next to the name of the tariff. When you are in Tariff Management for a particular tariff, click on **Edit Rates** in the toolbar.
2. On the **Edit Rates** screen, click **Add New**.
3. Fill in the required information:
   - **Destination** – A destination prefix may be entered directly, e.g. **47** for Norway, or you can access the destinations directory by clicking the **Destination** link (in the column header). Here you can find the desired prefix by country name.

NOTE: The phone prefix you are trying to create a rate for must already exist in Destinations.

   - **Interval 1** – first billing unit in seconds.
   - **Interval N** – next billing unit in seconds.
   - **Price 1** – per minute price for first interval.
   - **Price N** – per minute price for next interval.
   - **Off-peak Interval 1** – first billing unit in seconds for off -peak time.
   - **Off-peak Interval N** – next billing unit in seconds for off-peak time.
   - **Off-peak Price 1** – per-minute price for first interval for off-peak time.

- **Off-peak Price N** – per-minute price for next interval for off-peak time.

NOTE: Off-peak fields appear only if an **off-peak period** has been defined for the tariff.

- **Effective from** – If you want this rate to take effect sometime in the future, you can either type in a date manually, or use the calendar (click on the DD-MM-YYYY link).

NOTE: When using the calendar, you can specify that the date you are entering is in a different time zone than your present one. PortaBilling will then automatically adjust the time.

- The **Hidden**, **Forbidden** or **Discontinued** flags are optional.

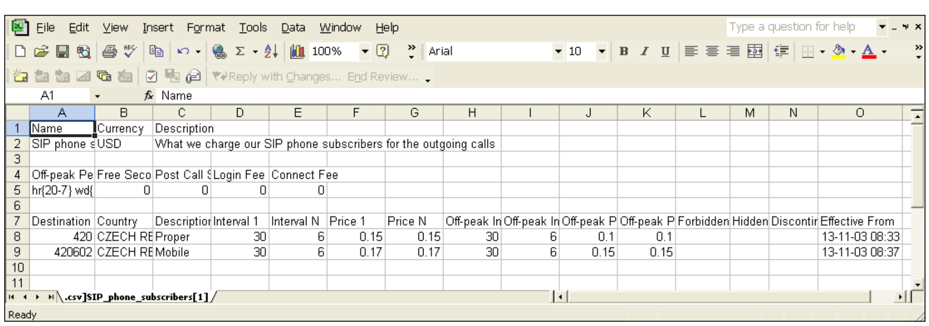
4. Click the **Save** button in the toolbar, or the icon on the left side of the row.
5. Repeat these steps if you need to enter more rates.

Managing rates offline

NOTE: Templates are available in PortaBilling, a powerful tool for uploading rates from custom format data files. However, in this particular example we assume that you will enter data using the PortaBilling default format.

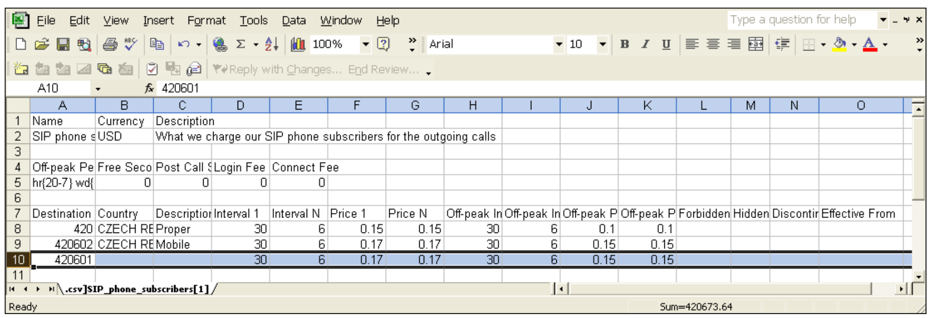
The rates table may be prepared using a spreadsheet processor (i.e. Microsoft Excel) and easily imported into PortaBilling. This is very convenient if you are going to make many changes. For example, you might increase all prices by 10%.

1. If you are not in Tariff Management for your tariff, go to the main menu, click on **Tariffs**, and then click on the tariff name.
2. In the Edit Tariff window, move the mouse over the **Download** button and hold it there until a popup menu appears. Choose the **Now** menu item and click on it. This will download the current set of rates (empty), but will also provide you with an overview of the file structure.
3. You will see the **File download** dialog and be prompted to choose whether to save the file or open it from the current location. We recommend that you save the file into the folder you will be using in the future to store tariff data files, then open it in Excel.
4. Now you should see something similar to the screenshot below:



	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O
1	Name	Currency	Description												
2	SIP phone s	USD	What we charge our SIP phone subscribers for the outgoing calls												
3															
4	Off-peak Pe	Free Seco	Post Call S	Login Fee	Connect Fee										
5	hr{20-7} wd{	0	0	0	0										
6															
7	Destination	Country	Descriptior	Interval 1	Interval N	Price 1	Price N	Off-peak In	Off-peak In	Off-peak P	Off-peak P	Forbidden	Hidden	Discontir	Effective From
8	420	CZECH RE	Proper	30	6	0.15	0.15	30	6	0.1	0.1				13-11-03 08:33
9	420602	CZECH RE	Mobile	30	6	0.17	0.17	30	6	0.15	0.15				13-11-03 08:37
10															
11															

5. Edit the file by adding more rows with rate data, so that it resembles the screenshot below.
6. Note that the **Country** and **Description** columns are only for reference, and are ignored during import. Also, when using the default template you must fill in data in the Off-peak columns even if your tariff does not have an off-peak period (use the clipboard to easily copy the values from the 4 peak columns).
7. Also note that you may only use those phone prefixes which you already have defined as destinations (see the **Create destinations** step above).
8. Make sure that you clear the values in the **Effective from** column (which would mean that the new rates are effective immediately), or enter a future date there. Otherwise, if you retain past dates, these rates will fail to upload.



9. Save the file in Excel. You will probably get a warning from Excel that your file "*may contain features that are not compatible with CSV (Comma delimited)*". Ignore this, and choose **Yes** to retain the CSV format.
10. Close the file in Excel. If you performed step 6, then disregard the message "*Do you want to save the changes you made*", since this arises only because your format is not the default Excel XLS format.
11. Go back to the PortaBilling web interface, and then go to the **Edit Tariff** screen.
12. Click on the **Upload** button.
13. Either enter the name of your file manually, or click **Browse…** and choose the file.
14. Click **Save&Close**. You should return to the **Edit Tariff** screen, where a message will tell you about the status of the import. Also, you will receive an email confirmation of the tariff upload. If any operations have failed, you will receive whatever data was not uploaded as an attachment, so you can try to import it later.

You can verify your work using the **Edit Rates** feature. After you have done so, go to the **Main menu** (by clicking on the **Home** icon).

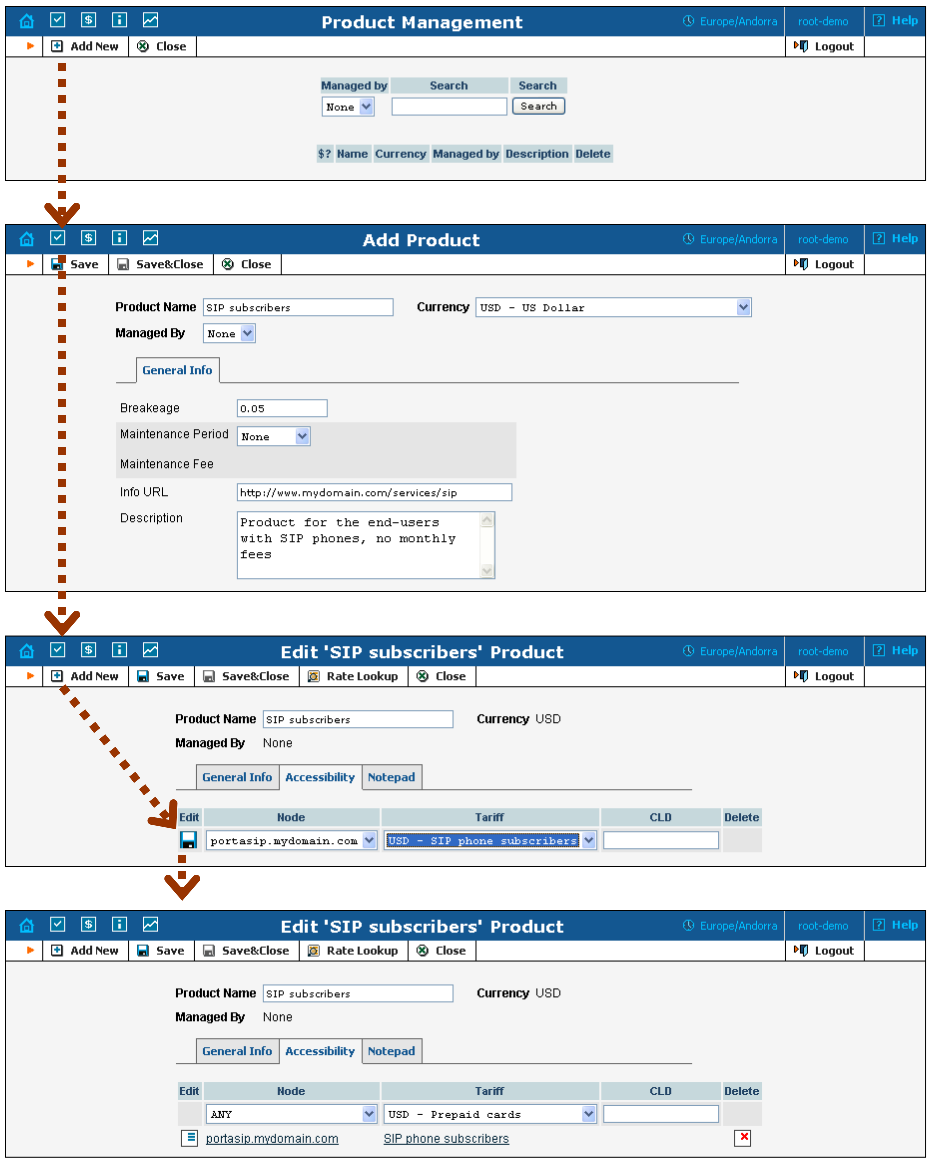
Create All Required Tariffs

Repeat the **Create Tariff** and **Enter Rates** steps, after which you will create:

- A tariff for each account billing scheme. For example, if you plan to charge your customers more when they access a toll-free line instead of a local one, you will need two tariffs, i.e. "Normal" and "Using Toll-free line".
- Create a tariff with the termination costs for each termination partner you have.
- If you have wholesale customers, create the tariffs you will use to charge each of them. Do not create tariffs which will be applied to subscribers of your wholesale customers yet. Create customers first, then return to this step. Make sure that, when creating these subscriber tariffs, you choose the corresponding wholesale customer in the **Managed By** menu.

Create Product

Accounts for accessing your SIP services will be issued for a specific product. Products are a powerful feature that define different ways to bill an account. Product definition is always done in two steps: product definition and creation of an accessibility list.



In the Management section of the Admin-Index page, choose **Products**.

1. On the Product Management page, click the **Add new** icon.
2. Fill in the "Add product" form:
   - **Product name** – product object name.
   - **Currency** – product currency; only tariffs which have the same currency will be permitted in the accessibility list.

- **Managed by** – If you want this product to be used for your wholesale customer's accounts, so the customer himself can change the parameters of this tariff and create new accounts using this product, choose a customer name from the menu. Otherwise, choose **None** here.
- **Breakage** – The left-over balance which is considered "useless" (for statistical purposes). Accounts with a balance below the breakage will be counted as *depleted*.
- **Maintenance period** – The surcharge application interval, which will be reflected in call history as a separate line each time it is charged at the end of a specified period.
- **Maintenance fee** – the surcharge amount.
- **Info URL** – If you have an external server with a description of product features, enter the URL here. Your customers will be able go here from their self-care page.
- **Description** – your comments about the intended use of this product.

3. Click **Save**.
4. Click on the **Accessibility** tab to edit this product's accessibility.

Enter Node and Tariff into the product's accessibility list

The Accessibility List has two functions: it defines permitted access points (nodes and access numbers) and specifies which tariff should be used for billing in each of these points.

1. When the Accessibility tab is selected, click on the **Add new** icon.
2. In the accessibility entry window, select the node where your IVR is running, and choose the appropriate tariff.
3. You can also use an access number for accessibility in the **CLD** field. For example, if you have a node with two access numbers, local (12345) and toll-free (1800 12345), you can set up the product's accessibility in such a way that, if a customer calls via a toll-free line, he will be billed using a different tariff (one which includes surcharges). The CLD field only makes sense when a call originates from your customer in a public telephony network. Therefore, just leave it empty for the SIP service.

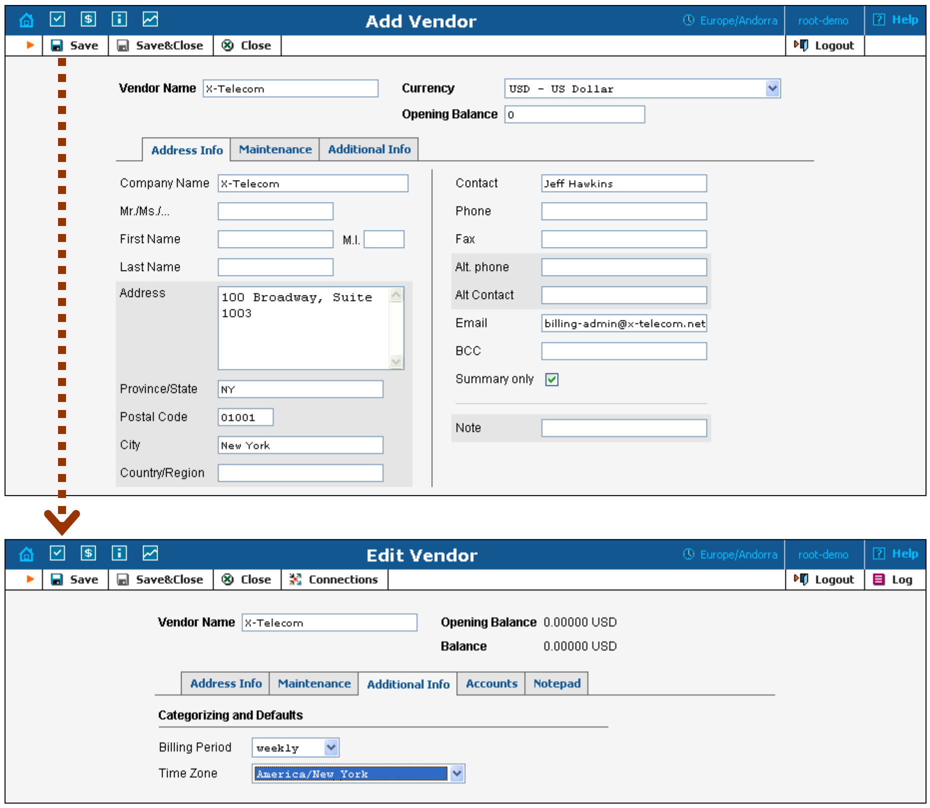
NOTE: The CLD feature requires modification of the application scripts on the Cisco platform; see the **Help** item on the Accessibility screen for more info. The CLD feature is not available for the Quintum platform.

4. Click **Save** to save this accessibility entry.
5. Repeat steps 1-4 if you want to define more accessibilities. Make sure that you have a row in Accessibility containing the PortaSIP server and the tariff you want to use for outgoing SIP calls.

Create Vendors

This step is only required if you have not entered information about your vendors into the system before. Vendors are your termination partners or providers of incoming toll-free lines.

1. In the Management section of the admin interface, choose **Vendors**.
2. On the Vendor Management page, choose **Add New**.



3. Fill in the **New Vendor** form. Please note that there are two tabs available on the screen. The most important fields are:

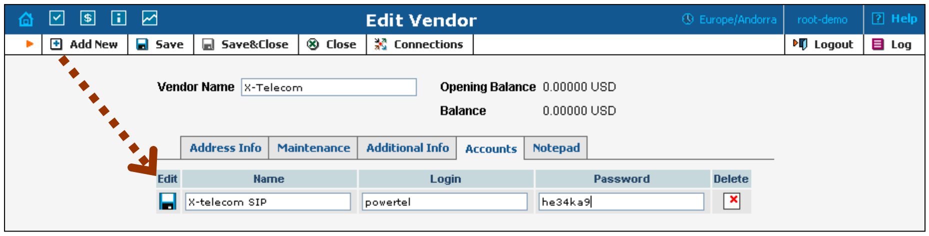
Main form (top):

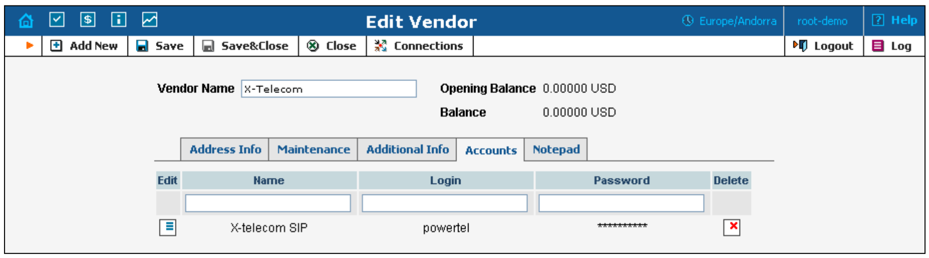
- **Vendor name** – short name for the Vendor object; this will be used on the web interface.
- **Currency** – the currency in which this vendor charges you.
- **Opening balance** – starting balance for the vendor; the default is zero.

Additional info:

- **Billing period** – split period for vendor statistics.
- **Time zone** – the time zone that the vendor uses.

4. Click **Save**.
5. If you plan to terminate your calls to the vendor's SIP server, typically he would provide you with a username/password which will authorize you to send calls to his server. Enter this information as **Vendor account**.



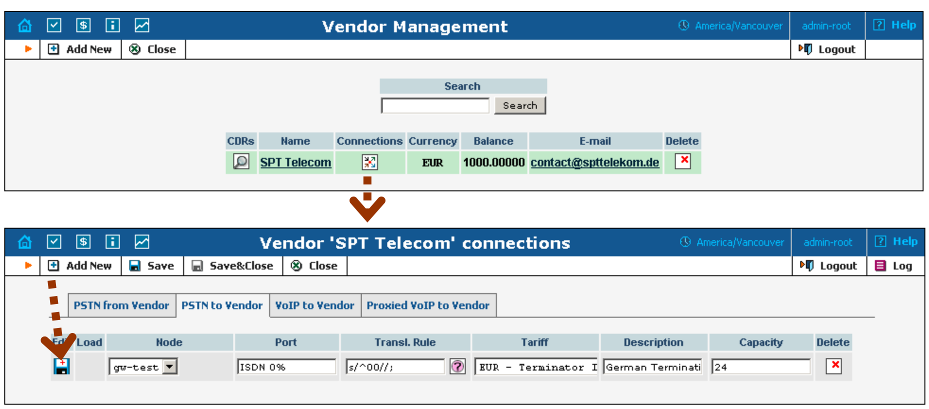


6. Click **Close** in order to return to the **Vendors** admin page.
7. Repeat steps 2-6 to add all of your vendors.

Define Connections

Connections are points at which calls leave or enter a network and are directed to or from vendors, whereby costing occurs.

1. In the Management section of the admin interface, choose **Vendors**.
2. Click on the **Connections** icon next to the vendor name.



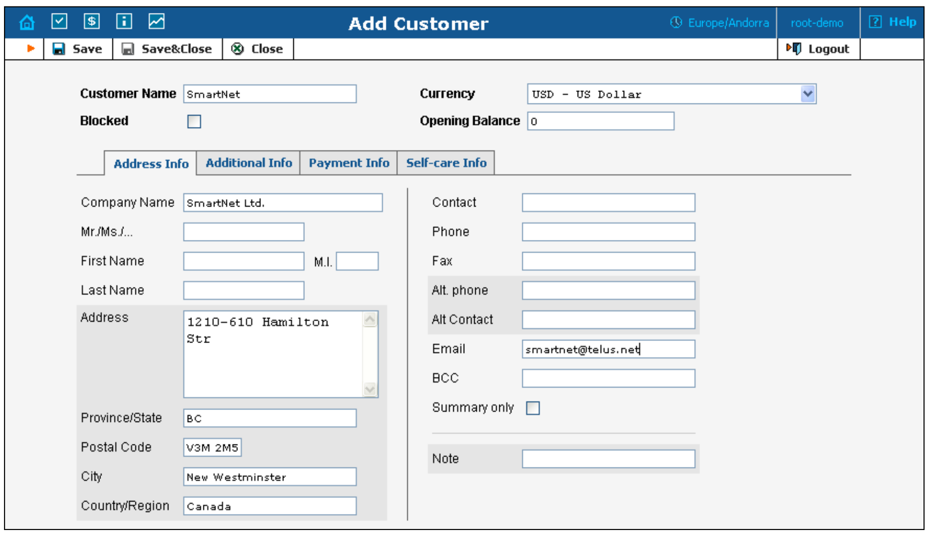
3. Choose the type of connection (**PSTN to Vendor**, **VoIP to Vendor**, etc.) by clicking on the corresponding tab.
4. Press **Add New** to add a new connection.
5. Fill in the connection information. If you send traffic to a vendor via telephony, choose the node and enter the optional port pattern. If you send traffic via VoIP, enter the remote IP address (address of the vendor's gateway or SIP server). Choose the tariff which defines your termination costs for this connection/vendor. **Description** and **Capacity** are mandatory for all connection types. For VoIP

connections where you have been assigned a login name and password, choose the corresponding vendor account.

6. Click **Save**.
7. Repeat steps 3-5 to add more connections to the same vendor, then click **Close** to exit to the **Vendor Management** screen.
8. Repeat steps 2-7 to add connections for other vendors.

Create a Customer

A customer is an account owner. The customer's contact information is used to distribute generated accounts data and account usage information. Even if your company owns and distributes all of its pre-paid cards, you will need at least one customer object for your company.



1. In the Management section of the Admin-Index page, choose **Customers**.
2. On the Customer Management page, choose **Add New**.
3. Fill in the **New Customer** form. Please note that there are several tabs with extra information available on the screen. The most important fields are:

Main form (top):

- **Name** – short name for the customer object; this will be used on the web interface.
- **Currency** – the currency in which this customer will be billed.
- **Opening balance** – a starting balance for the customer; the default is zero.

Address info tab:

- **Email** – An email address for the distribution of accounting information. After the billing period is over, a list of CDRs and other statistics will be sent to this address
- **Bcc** – Blind carbon copy in email; may be used for debug and archiving purposes.
- **Summary only** – Distributes summary only, and does not attach a details file; might be useful when the amount of calls is very large.

Additional info tab:

- **Type** – specify wholesale or retail customer.
- **Billing period** – The frequency of accounting information distribution. Available billing periods:
  - **Daily** – one day, midnight to midnight, sent on the next day;
  - **Weekly** – [Mon-Sun] inclusive, sent on Monday;
  - **Bi-weekly** – [1-15] inclusive, sent on the 16th, and [16-last day] inclusive, sent on the 1st;
  - **Monthly –** [1-last day] inclusive, sent on the 1st of the following month.

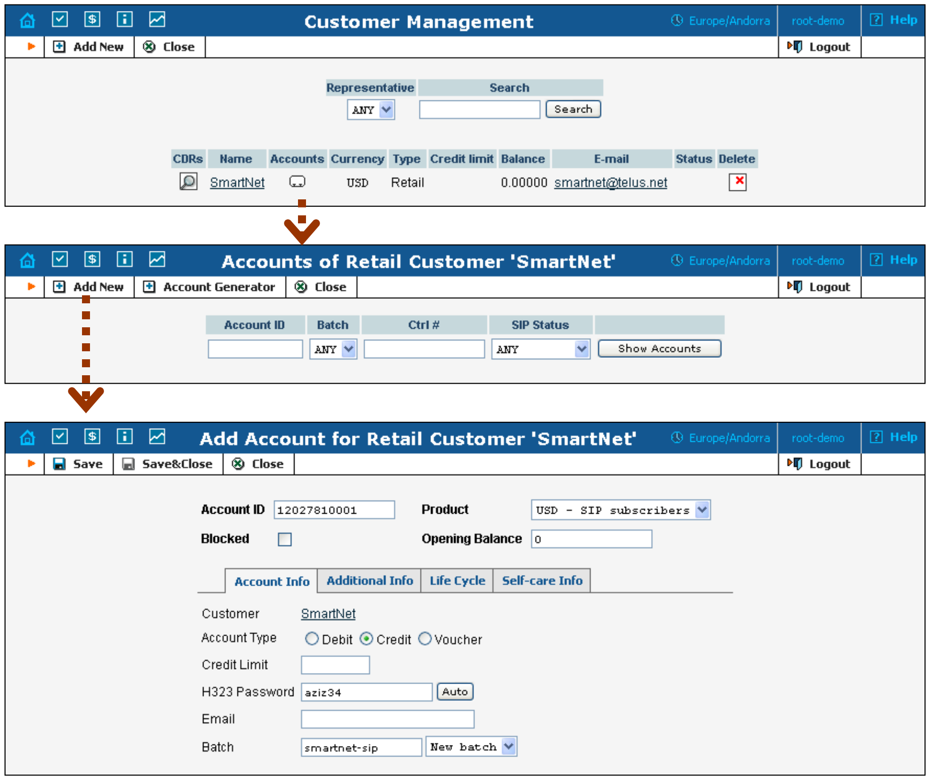
Payment info tab:

- **Credit limit** – if left empty, then there is no credit limit.

4. Click **Save&Close**.

Create Accounts

1. Go to the **Customers** screen (the one containing the list of customers). It should resemble the screenshot below.



2. Next to the customer name, click on the icon (the one in the **Accounts** column) to go to the account management for that customer.
3. Click on **Account New**.
4. Fill in the "Add account" form:
   - **Account ID** – SIP ID, i.e. the phone number which will be used to login to the SIP server and receive incoming calls.
   - **Product** – choose the product which you would like your account to have.
   - **Blocked** – you may create your account as blocked, although this is rarely done with SIP service accounts.
   - **Opening balance** – the initial balance on the account.

Account Info tab:

- **Account type** – account type; select credit for post-paid and debit for prepaid service.
- **Credit limit** – For a credit account, specify the credit limit. If you leave this field empty, it means there is no credit limit for this account (but a customer credit limit may still apply).

- **H323 password** – This password is used for SIP services as well. The account ID and this password will be used to authenticate SIP server login.
- **Email** – Enter the account owner's email address here. If he ever forgets his password for the web self-care pages, he will be able to reset it, and a new password will be sent to this email address. You can also just leave this field empty.
- **Batch** – A batch is a management unit for accounts. The batch name is alphanumeric. You can type a new name here, or use an existing name in order to generate more accounts for the same batch.

Additional Info tab:

- **Preferred language** – This is a custom attribute, which is transferred to the IVR. Leave English here if you are unsure whether your IVR supports this function.
- **Time zone** - When an account owner (pre-paid card user) accesses web self-care pages to see a list of his calls, we can show the time in the time zone most appropriate for him.
- **Redirect number** – redirect number (discussed in the **Advanced features** section); leave this empty.
- **UM enabled** – check the box if this account has unified messaging (e.g. voicemail) services enabled.

Life Cycle tab:

- **Activation date** – account activation date.
- **Expiration date** – account expiration date.
- **Life time** – Relative expiration date: account will expire on "first usage date" + "life time" days. If you do not want to use this feature, leave the field blank.

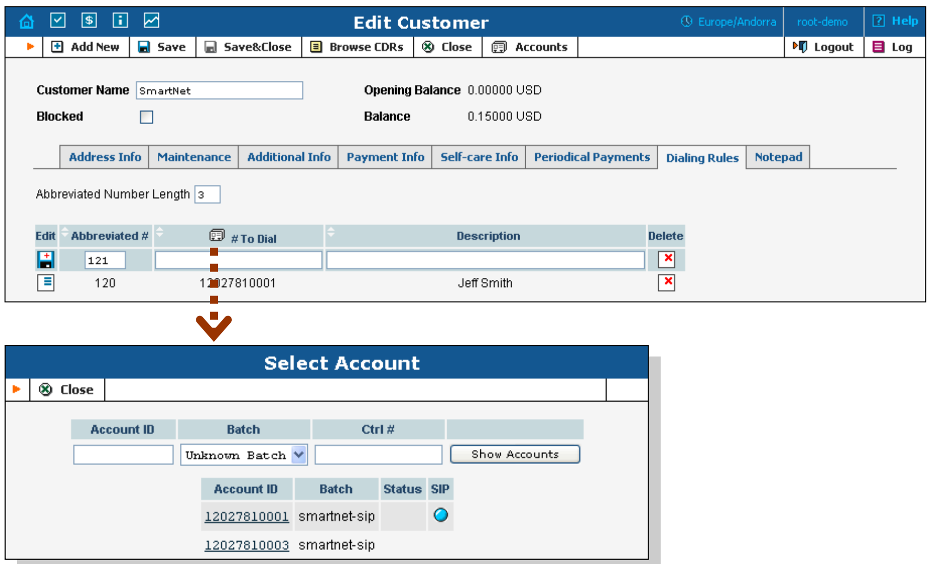
Self-care Info Cycle tab:

- **Login** – Account login to web self-care pages. Can be the same as account ID.
- **Password** – password for the web self-care pages.

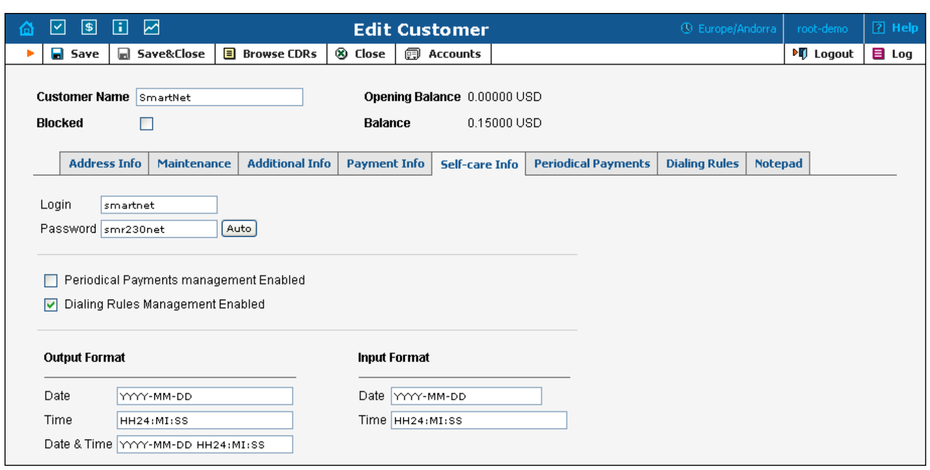
5. After clicking **Save&Close**, you will see a confirmation screen saying that the new account has been created.

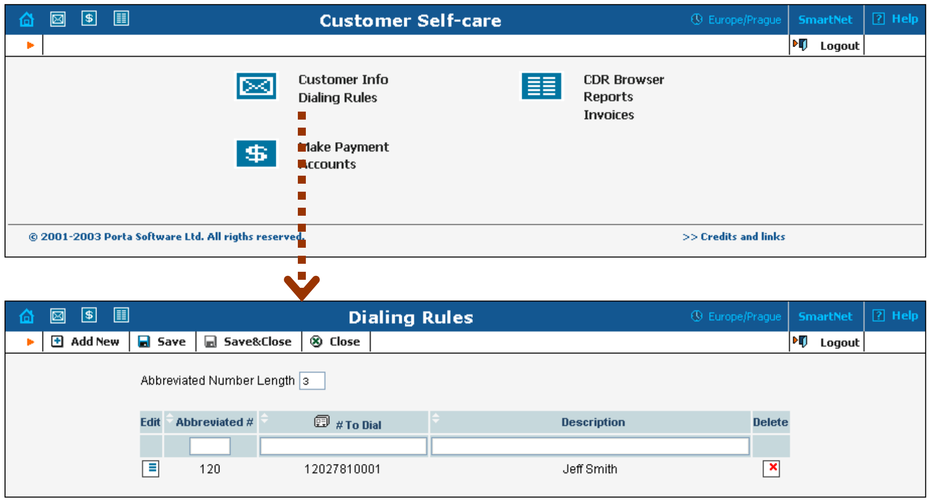
Set Up Abbreviated Dialing for the Customer (optional)

If your customer has multiple SIP accounts, and plans to make calls between them, it would be very inconvenient to dial a complete E.164 number each time. You may create abbreviated dialing rules, so that from any SIP phone using this customer's account it will suffice to dial, for example, 120 to reach Jeff Smith.



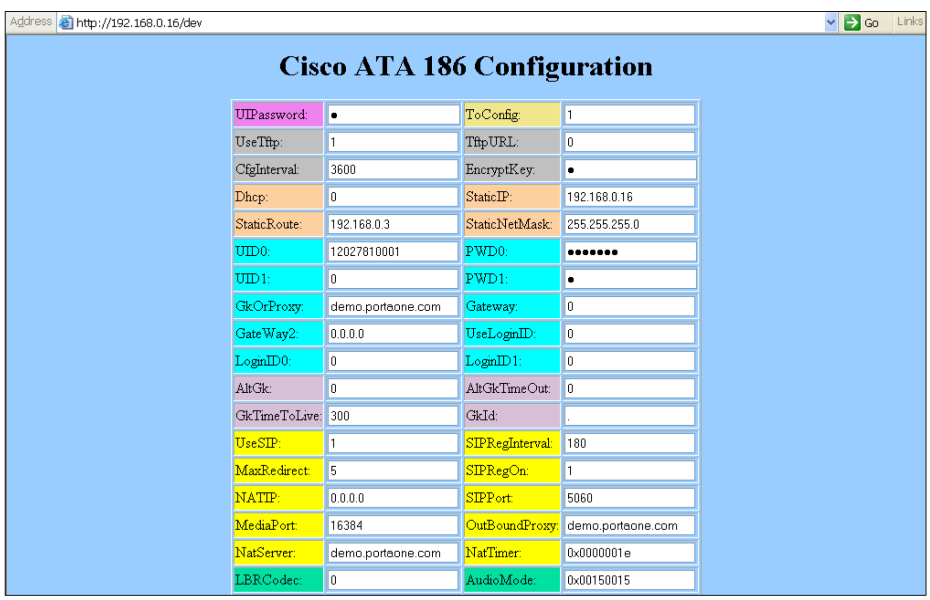
This is much better than programming every phone used in the organization. In addition, the customer himself can manage these dialing rules on his self-care pages, if you allow him.





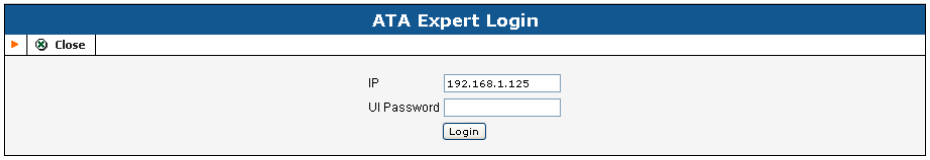
Configure Cisco ATA Using ATA Expert (optional)

Cisco ATA could be configured from the web interface accessible at **http://<ata-IP-address>/dev**. However, this web interface is designed to be used by experts, and parameter values must be entered in the protocol-specific format (e.g. `0x00150015`). You may find more information at: **http://www.cisco.com/en/US/products/hw/gatecont/ps514/products_configuration_example09186a00800c3a50.shtml**
However, this complicated way of entering the parameters makes it virtually impossible for end-users to employ.



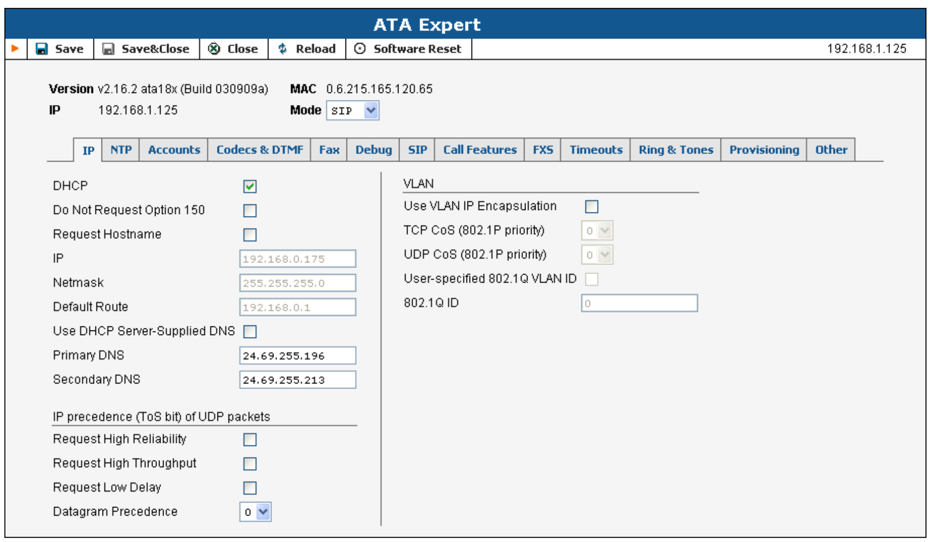
Fortunately, PortaBilling provides a safe and user-friendly way to configure your Cisco ATA from the web interface via **ATA Expert**:

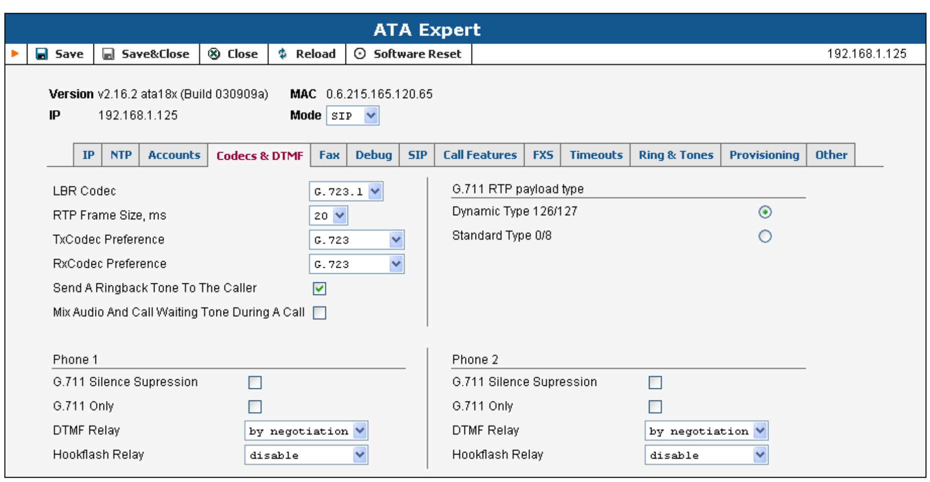
1. Login to the PortaBilling admin interface, and go to **http://<your-porta-billing-domain>/ATAExpert/expert.html**
2. Type in your Cisco ATA IP address, as well as the administrator's password if you have set up one.



NOTE: The PortaBilling ATA Expert needs to communicate directly with the Cisco ATA. So make sure that the ATA is connected to the network and configured with an IP address. This IP address must be either a public IP address (accessible from anywhere on the Internet) or a private IP address (e.g. 192.168.xxx.xxx) which is accessible from the PortaBilling web server.

3. You can browse current configuration parameters on the expert screen.

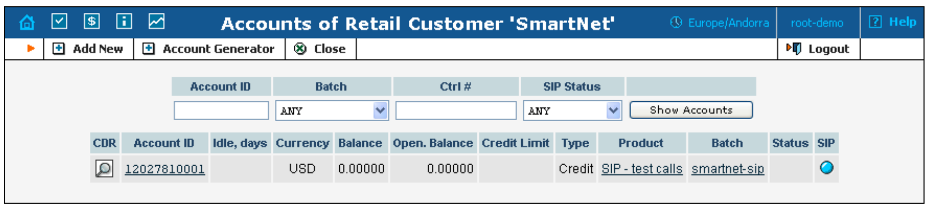




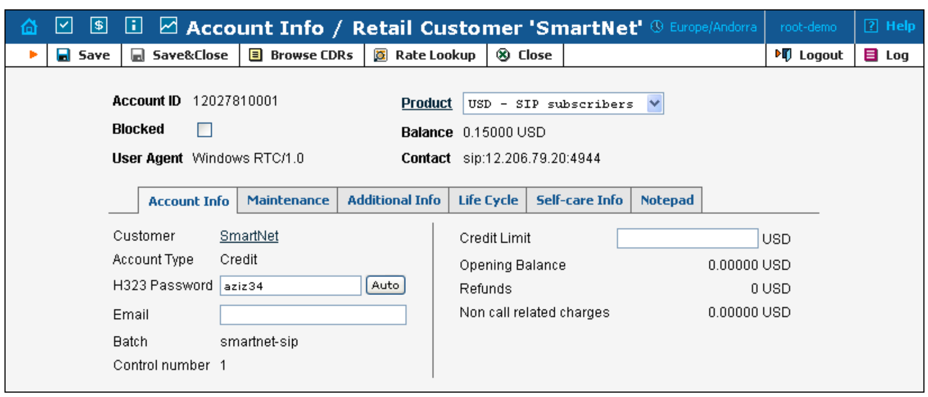
4. Press **Save** to save the new configuration to the ATA.

Testing the Whole System

1. Make sure the PortaBilling radius and PortaSIP servers are running.
2. Configure your SIP user agent with the account ID and password. (See appendixes for configuration guidelines for some SIP UAs). Then have your SIP user agent login to the SIP server.
3. Check that the account is logged into the SIP server:
   - Go to the account list screen and see if the SIP indicator button (a blue circle) is on for this account.



   - Go to the account info page for this account, and check that the **User Agent** and **Contact** fields contain some values. These fields will show the account's current registration information.



4. Try to make a call using one of the accounts
5. Browse the SIP server log file (`/var/log/sip.log` on the SIP server host). In the real system, each line in the log will be prefixed with the current time and call-id. In the sample debug output below, this has been replaced with *** for greater clarity.

```
…
Registration request received
***/ser: processing REGISTER received from 12.206.79.20
***/ser: sending AAA request for 12027810001, method REGISTER
***/ser: AAA request accepted
Registration successful
…
No account is trying to make an outgoing call
***/ser: processing INVITE received from 12.206.79.20
***/ser: no auth info or auth failure, sending challenge
```

```
SIP user agent is forced to use secure authorization
***/ser: processing INVITE received from 12.206.79.20
***/ser: sending AAA request for 12027810001, method LOOKUP
***/ser: AAA request accepted
***/ser: got complete number 380442447182, rewriting URI
LOOKUP in billing successful, proceeding with call
***/b2bua: session received from 195.70.151.35:5060, creating
new answering call leg
***/b2bua: setting UID (12027810001), realm (195.70.151.35),
nonce (3fb6c99aad9906b93ad99618fe2f4cb8cac090fb), method
(INVITE), uri (sip:380442447182@195.70.151.35), algorithm
(MD5), response (a53293d9f49068259c34240593d3c4b0) from
Authorization header
***/b2bua: requesting billing-assisted routing
***/b2bua: sending AAA request
***/b2bua: authorization accepted
***/b2bua: call duration unlimited
***/b2bua: got route: sip:380442447182@23.45.170.16;
auth=powertel:h324ka9;user=phone
Call is being sent to the remote SIP server
***/b2bua: placing outgoing session to
sip:380442447182@23.45.170.16;user=phone, h323-conf-id=2484F919
5CC2BBF9 49EBF817 8D847E7E
***/b2bua: outgoing session started
Call is connected
***/b2bua: outgoing session ended successfully
***/b2bua: sending Acct Stop (Answering)
***/b2bua: sending Acct Stop (Originate)
***/b2bua: session duration is 10
…
```

6. Browse information in the PortaBilling log file (type "`less /var/log/porta-billing.log`")

```
Here is an example of a registration request:
Thu 13 23:58:29: Processing request:
NAS-IP-Address       = '195.70.151.35'
User-Name            = '12027810001'
User-Name contains the SIP account ID, so normally it is
identical to the ANI number (Calling-Station-Id below)
Calling-Station-Id   = '12027810001'
Service-Type         = '15'
call-id              = '1093122365@192.168.0.16'
Digest-Attributes    = 'User-Name = "12027810001"'
Digest-Attributes    = 'Realm = "195.70.151.35"'
Digest-Attributes    = 'Nonce =
"3fb48c51eb1a02bdc027dc528c789f54a8a0e194"'
Digest-Attributes    = 'URI = "sip:195.70.151.35"'
Digest-Attributes    = 'Method = "REGISTER"'
Digest-Attributes    = 'Algorithm = "MD5"'
Digest-Response      = '1dcfd64301fa495c4d0a9bf9c8533b6d'
You cannot see a clear text password because a secure digest-
based method is used to verify the password.
NAS-Port-Id          = '5060'
Thu 13 23:58:29: h323-conf-id=00000000 00000000 00000000
00000000/22, call-id=1093122365@192.168.0.16/22
Thu 13 23:58:29: Pure SIP call, use
'1093122365@192.168.0.16/22' as h323-conf-id
Thu 13 23:58:29: This is GK auth or SIP register, authenticate
only
Thu 13 23:58:29: PrepareNexecute 'AccountAuth'
Thu 13 23:58:29: Found account&customer
Thu 13 23:58:29: Account 12027810001 is not logged in yet
Thu 13 23:58:29: Authentication acknowledge response
```

```
Cisco-AVPair         = h323-ivr-in=Tariff: SIP subscribers
h323-return-code     = 0
h323-preferred-lang = en
h323-currency        = USD
h323-billing-model   = 0
```

Authentication was successful, so the SIP user agent is allowed to register.

Next, the LOOKUP request arrives. We need to validate what would be the actual phone number to dial (for example, if the customer has a private numbering plan, a short abbreviated number can be converted into a complete one).

```
Thu 13 23:58:53: Processing request:
NAS-IP-Address      = '195.70.151.35'
User-Name           = '12027810001'
Called-Station-Id   = '380442447182'
Calling-Station-Id = '12027810001'
Service-Type        = '15'
call-id             = '3252619179@192.168.0.16'
Digest-Attributes   = 'User-Name = "12027810001"'
Digest-Attributes   = 'Realm = "195.70.151.35"'
Digest-Attributes   = 'Nonce =
"3fb48c692f4fbbd1bf786242a79a67db93a30f08"'
Digest-Attributes   = 'URI = "sip:380442447182@195.70.151.35"'
Digest-Attributes   = 'Method = "LOOKUP"'
Digest-Attributes   = 'Algorithm = "MD5"'
Digest-Response     = 'a79d53fb2faa251c2f1e6bd9421bf3b6'
NAS-Port-Id         = '5060'
Thu 13 23:58:53: h323-conf-id=00000000 00000000 00000000
00000000/22, call-id=3252619179@192.168.0.16/22
Thu 13 23:58:53: Pure SIP call, use
'3252619179@192.168.0.16/22' as h323-conf-id
Thu 13 23:58:53: PrepareNexecute 'AccountAuth'
Thu 13 23:58:53: Found account&customer
Thu 13 23:58:53: Account 12027810001 is not logged in yet
Thu 13 23:58:53: Using peak rate, since no off-peak is defined
Thu 13 23:58:53: PrepareNexecute custom SQL
Thu 13 23:58:53: Authentication acknowledge response
Cisco-AVPair         = h323-ivr-in=Tariff:SIP subscribers
Cisco-AVPair         = h323-ivr-in=
PortaBilling_CompleteNumber:380442447182
h323-return-code     = 13
h323-preferred-lang = en
h323-currency        = USD
h323-billing-model   = 0
```

Lookup was successful, and the actual number to be dialed (380442447182) and maximum call duration (unlimited) are returned.
Now, after the SER has obtained the correct number to dial and verified that the customer is allowed to make this call, it transfers control to the B2BUA, which issues another request (INVITE). One of the goals of the INVITE request is to obtain call routes according to the configuration in billing:

```
Thu 13 23:58:56: Processing request:
NAS-IP-Address      = '195.70.151.35'
User-Name           = '12027810001'
Called-Station-Id   = '380442447182'
Calling-Station-Id = '12027810001'
h323-conf-id        = '2484F919 5CC2BBF9 49EBF817 8D847E7E'
call-id             = '3252619179@192.168.0.16'
Digest-Attributes   = 'Realm = "195.70.151.35"'
Digest-Attributes   = 'Nonce =
"3fb48c692f4fbbd1bf786242a79a67db93a30f08"'
Digest-Attributes   = 'Method = "INVITE"'
```

```
Digest-Attributes  = 'URI = "sip:380442447182@195.70.151.35"'
Digest-Attributes  = 'Algorithm = "MD5"'
Digest-Attributes  = 'User-Name = "12027810001"'
Digest-Response    = 'a79d53fb2faa251c2f1e6bd9421bf3b6'
h323-remote-address= '195.70.151.35'
h323-session-protocol = 'sipv2'
h323-ivr-out       = 'PortaBilling_Routing:YES'
Thu 13 23:58:56: h323-conf-id=2484F919 5CC2BBF9 49EBF817
8D847E7E/22, call-id=3252619179@192.168.0.16/22
Thu 13 23:58:56: H323/SIP call, use h323-conf-id, but remember
call-id
Thu 13 23:58:56: PrepareNexecute 'AccountAuth'
Thu 13 23:58:56: Found account&customer
Thu 13 23:58:56: Account 12027810001 is not logged in yet
Thu 13 23:58:56: Registering account '12027810001' (120715) as
logged in
Thu 13 23:58:56: PrepareNexecute 'UpdateAccountFirstUsage'
Thu 13 23:58:56: Using peak rate, since no off-peak is defined
Thu 13 23:58:56: Result routes: [ {
            'login' => 'powertel',
            'ip' => '23.45.170.16',
            'description' => 'X-Telecom',
            'password' => 'he34ka9'
          } ];

Thu 13 23:58:56: Authentication acknowledge response
Cisco-AVPair        = h323-ivr-in=Tariff:SIP subscribers
Cisco-AVPair        = h323-ivr-in =
PortaBilling_Routing:380442447182@23.45.170.16;auth=powertel:he
34ka9
h323-return-code    = 13
h323-preferred-lang = en
h323-currency       = USD
h323-billing-model  = 0
```

The billing engine confirms that the call is authorized, and provides for the B2BUA destination route.

…

After the call is completed, we receive accounting records:

```
Thu 13 23:59:20: Processing request:
NAS-IP-Address      = '195.70.151.35'
User-Name           = '12027810001'
Called-Station-Id   = '380442447182'
Calling-Station-Id  = '12027810001'
Acct-Status-Type    = 'Stop'
h323-call-origin    = 'answer'
h323-call-type      = 'VoIP'
h323-setup-time     = '07:58:56 GMT Fri Nov 14 2003'
h323-connect-time   = '07:59:10 GMT Fri Nov 14 2003'
h323-disconnect-time= '07:59:20 GMT Fri Nov 14 2003'
h323-disconnect-cause= '10'
h323-voice-quality  = '0'
h323-conf-id        = '2484F919 5CC2BBF9 49EBF817 8D847E7E'
call-id             = '3252619179@192.168.0.16'
Acct-Session-Id     = '3252619179@192.168.0.16'
Acct-Session-Time   = '10'
Acct-Delay-Time     = '14'
h323-remote-address = '195.70.151.35'
h323-session-protocol = 'sipv2'
h323-ivr-out        = 'PortaBilling_Session:unlock'
Acct-Terminate-Cause= 'User-Request'
Exec-Program-Log    = 'porta-billing.pl'
Thu 13 23:59:20: h323-conf-id=2484F919 5CC2BBF9 49EBF817
8D847E7E/22, call-id=3252619179@192.168.0.16/22
Thu 13 23:59:20: Found a call in cache with such id
```

```
Thu 13 23:59:20: Copying account and customer information into
the current request
Thu 13 23:59:20: Unlocking sessions for this call
Thu 13 23:59:20: Scheduling 12027810001 for logout, call
lifetime reduced to 15
Thu 13 23:59:20: Logging out account '12027810001' from
2484F919 5CC2BBF9 49EBF817 8D847E7E
Thu 13 23:59:20: Looking up vendor/connection
Thu 13 23:59:20: PrepareNexecute 'GetAllConnections'
Thu 13 23:59:20: Trying to match connection for call
Thu 13 23:59:20: Looking for a connection VoIP/answer
Thu 13 23:59:20: Answer VoIP (h323-proxy or just junk),
matching by the node IP address '195.70.151.35'
Thu 13 23:59:20: Connection to vendor not found
Thu 13 23:59:20: Connection to vendor not found - on-net call
leg
Thu 13 23:59:20: Accounting response
```

This is the call leg from the SIP user agent to the SIP server - it is still considered part of our network, so the call is not billed yet.

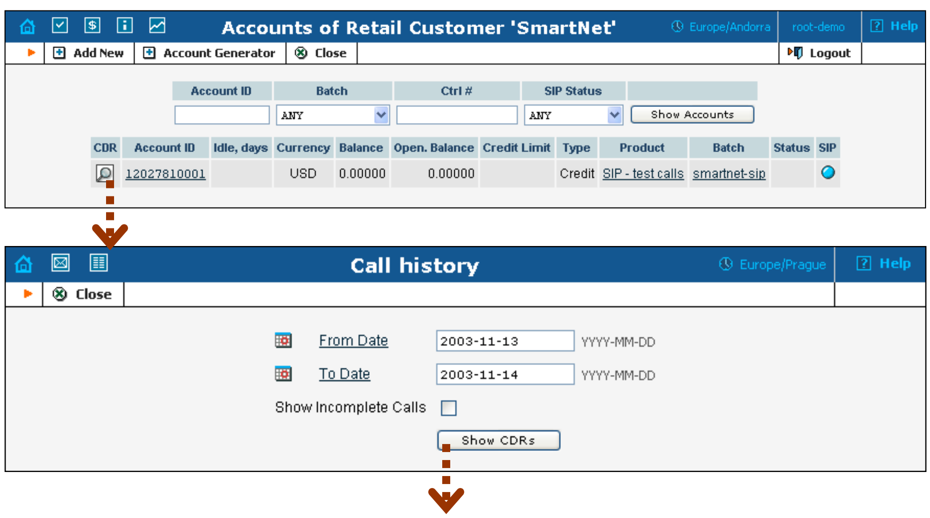
And here is the accounting for the second call leg:

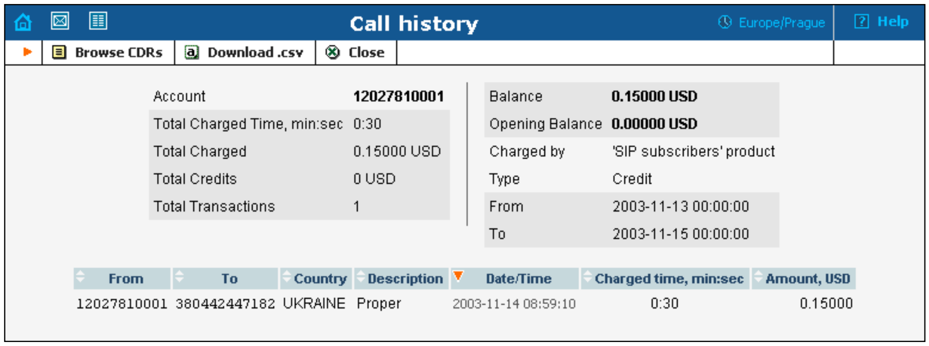
```
Thu 13 23:59:20: Processing request:
NAS-IP-Address       = '195.70.151.35'
User-Name            = '12027810001'
Called-Station-Id    = '380442447182'
Calling-Station-Id   = '12027810001'
Acct-Status-Type     = 'Stop'
h323-call-origin     = 'originate'
h323-call-type       = 'VoIP'
h323-setup-time      = '07:58:57 GMT Fri Nov 14 2003'
h323-connect-time    = '07:59:10 GMT Fri Nov 14 2003'
h323-disconnect-time= '07:59:20 GMT Fri Nov 14 2003'
h323-disconnect-cause= '10'
h323-voice-quality   = '0'
h323-conf-id         = '2484F919 5CC2BBF9 49EBF817 8D847E7E'
call-id              = '3252619179@192.168.0.16'
Acct-Session-Id      = '3252619179@192.168.0.16'
Acct-Session-Time    = '10'
Acct-Delay-Time      = '13'
h323-remote-address = '23.45.170.16'
h323-session-protocol= 'sipv2'
Acct-Terminate-Cause= 'User-Request'
Exec-Program-Log     = 'porta-billing.pl'
Thu 13 23:59:20: h323-conf-id=2484F919 5CC2BBF9 49EBF817
8D847E7E/22, call-id=3252619179@192.168.0.16/22
Thu 13 23:59:20: Found a call in cache with such id
Thu 13 23:59:20: Copying account and customer information into
the current request
Thu 13 23:59:20: End of the outgoing call for account - waiting
for another outgoing call or hang up, lifetime 60 sec
Thu 13 23:59:20: Looking up vendor/connection
Thu 13 23:59:20: Trying to match connection for call
Thu 13 23:59:20: Looking for a connection VoIP/originate
Thu 13 23:59:20: Outgoing VoIP, matching by the remote IP
address '23.45.170.16' (env 22)
Thu 13 23:59:20: Found connection 22 'X-Telecom' to vendor 'X-
Telecom'
Thu 13 23:59:20: Found vendor/connection
Thu 13 23:59:20: Charging call
Thu 13 23:59:20: Calculating account charge
Thu 13 23:59:20: Using peak rate, since no off-peak is defined
Thu 13 23:59:20: PrepareNexecute custom SQL
```

```
Thu 13 23:59:20: Call to '380442447182' with duration 10
seconds will be charged for 30 seconds at a cost of 0.15000
(30*0.3)
Thu 13 23:59:20: Calculating vendor charge
Thu 13 23:59:20: Using peak rate, since no off-peak is defined
Thu 13 23:59:20: PrepareNexecute custom SQL
Thu 13 23:59:20: Call to '380442447182' with duration 10
seconds will be charged for 10 seconds at a cost of 0.04167
(1*0.25+9*1*0.25)
Thu 13 23:59:20: Charging account for the call
Thu 13 23:59:20: Charging credit account 12027810001 0.15000
Thu 13 23:59:20: PrepareNexecute 'UpdateAccountBalance'
Thu 13 23:59:20: Inserting CDR
Thu 13 23:59:20: PrepareNexecute 'InsertAccountCDR'
Thu 13 23:59:20: Charging account owner for the call
Thu 13 23:59:20: Charging customer 30 'SmartNet' 0.15000
Thu 13 23:59:20: PrepareNexecute 'UpdateCustomerBalance'
Thu 13 23:59:20: Charging vendor for the call
Thu 13 23:59:20: Charging vendor 20 'X-Telecom' 0.04167
Thu 13 23:59:20: Inserting CDR
Thu 13 23:59:20: PrepareNexecute 'InsertVendorCDR'
Thu 13 23:59:20: PrepareNexecute 'UpdateVendorBalance'
Thu 13 23:59:20: Accounting response
The call was completed and billed successfully.
```

Verify Call History for the Account

To view the CDR of an account, go to Customers, select the customer owning the account, and click on the Accounts icon; or, alternatively, select the Account Info link from the Main Menu. You can also go to the account self-care page (accessible via the **Accounts** menu item in the **Home** popup menu).



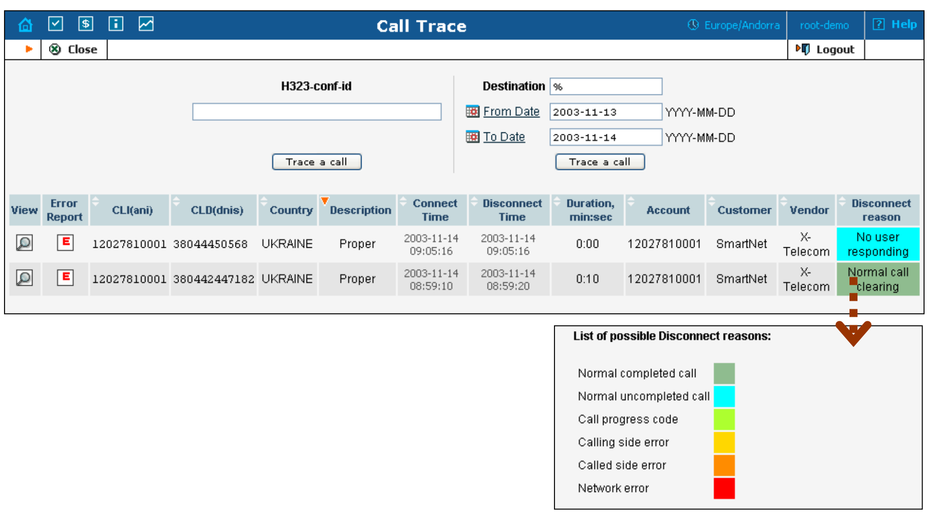


Login with the account's web access login and password. After that, you will be able to see the account's self care menu.

Choose the date range for which you want to see a list of calls, and press **Show CDR**. In the results table you will see call charges and other fees, such as maintenance fees or refunds (if any). The report can be also downloaded by clicking the **Download .csv** icon.

Check the Call History

If you want to see a list of all calls going through the system, or perhaps only ones for a particular destination, use the **Trace Call** function.



- In the Helpdesk section of Admin-Index choose **Trace a Call**.
- Fill in the check phone number form:
  - **h323-conf-id** – if you need to trace a specific call, enter h323-conf-id here; otherwise this leave empty.
  - **Destination** – the phone number you are looking for or a destination pattern (first digits and a percent sign, for example **380%**).
  - **From, To Date –** the date range.
- Click **Trace a Call.**

The advantage of this method: you may view all the call attempts, including unsuccessful calls, with disconnect reasons displayed. Also, you can see the billing history for a call.

For administrator convenience, accounts' CDRs may also be accessed from the Account Management window by clicking the **CDR** icon for an account.

3. How to ...

... configure my Cisco gateway to accept incoming SIP calls and terminate them to a telephony network?

Configuration of the Cisco gateway for SIP is not much more difficult than H323. First of all, make sure that the rest of your system is configured properly – that the gateway can place the outgoing calls, and is able to communicate with the billing using RADIUS.

Codecs

First of all, make sure you have set up a list of codecs which are supported by your SIP agents on your GW. Your actual configuration might differ, but here is a good example which should work in most cases:

```
voice class codec 1
 codec preference 1 g723r63
 codec preference 2 g729r8
 codec preference 3 g729br8
 codec preference 4 g723r53
 codec preference 7 g726r16
 codec preference 8 g726r24
 codec preference 9 g726r32
 codec preference 10 g711alaw
 codec preference 11 g711ulaw
 codec preference 12 g723ar53
 codec preference 13 g723ar63
```

SIP agent

Now enable the SIP agent functionality on your gateway. Also enable it on gateways where NAT symmetric traversal is supported, as this will facilitate calls from SIP agents behind the firewall.

```
sip-ua
 nat symmetric check-media-src
```

NOTE: Cisco GWs are currently unable to log in to the SIP server using the REGISTER method.

Dial-peers

Finally, create an SIP-enabled incoming dial-peer:

```
dial-peer voice 100 voip
 incoming called-number .T
 voice-class codec 1
 session protocol sipv2
 dtmf-relay rtp-nte
!
```

Note that this gateway provides no authentication of incoming SIP calls, so that potentially anyone could route calls to you from their SIP server. This is why the recommended configuration is as follows:

```
call application voice remote_ip flash:app_remote_authenticate.tcl

dial-peer voice 100 voip
 incoming called-number .T
 voice-class codec 1
 session protocol sipv2
 dtmf-relay rtp-nte
 application remote_ip
!
```

Thus, every incoming call will be authenticated by the IP address of the remote peer. Since signaling for the SIP call comes from the SIP server, this would be the address of the SIP server. This means that calls coming from your own SIP server will be authenticated by billing, since your SIP server is entered in the system as a trusted node.

... configure my Cisco gateway to send outgoing calls using SIP?

Configuration of the Cisco gateway for SIP is not much more difficult than H323. First of all, make sure that the rest of your system is configured properly – that the gateway can place the outgoing calls, and is able to communicate with the billing using RADIUS.

SIP server parameters

Specify general parameters of the SIP server, such as hostname. You can also refer to the SIP server by its IP address; however, this method will require reconfiguration of each individual gateway if you change the IP address of your SIP server.

```
sip-ua
 aaa username proxy-auth
 sip-server dns:<hostname-of-your-SIP-server>
```

NOTE: Cisco GWs are currently unable to register to SIP servers using the REGISTER method, or to perform proper authorization of an outgoing call using the INVITE method. Therefore, remote IP address authorization is performed by PortaSIP when it detects an incoming call from the Cisco gateway. In order for this authorization to be successful, the gateway should be registered among the PortaBilling nodes.

Dial-peers

Now you can create an SIP-enabled outgoing dial-peer:

```
dial-peer voice 200 voip
 destination pattern .T
 session protocol sipv2
```

```
 session target sip-server
!
```

You probably will need an application on the incoming telephony dial-peer to properly authenticate and authorize incoming calls.

... support incoming H323 and SIP calls on the same gateway?

This configuration is supported, as Cisco GW can handle both H323 and SIP calls at the same time. However, please note that Cisco matches an incoming dial-peer by the incoming called number, not by the protocol. Thus, the dial-peer shown below will match both incoming SIP and H323 calls, even if it gives the session protocol sipv2:

```
dial-peer voice 101 voip
 description *** Incoming SIP calls
 incoming called-number .
 voice-class codec 1
 session protocol sipv2
 dtmf-relay rtp-nte
 fax protocol cisco
```

... configure my Cisco ATA186 to work with PortaSIP?

Perform the initial network configuration of the ATA using the built-in IVR. After your ATA is assigned an IP address, you can go to the web configuration screen at **http://<your-ATA-IP-address>/dev**.
Consult **APPENDIX C. Clients' Cisco ATA 186 Configuration for PortaSIP**. For other options not listed in the table below, the default manufacturer value is assumed.

... provide services and bill a customer who has a SIP-enabled gateway but no authorization capability (e.g. Cisco AS5350)?

PortaSIP is able to authenticate incoming calls using the IP address of the remote side. This method ensures that PortaSIP will accept calls from your own gateways, but it can also be used to bill traffic from your customers. You just need to create an account for your customer with an account ID identical to the IP address of his gateway. Authentication and billing will be done in the same way as IP-based billing using H323.

... make all SIP calls to a certain prefix NNN go to my gateway XXX?

Normally it is only possible to use the REGISTER command for user-agents, i.e. for devices which represent a single physical phone. An SIP user agent cannot register with the SIP server and report: "I am going to receive all calls for prefix NNN". (Cisco 5300 supports the REGISTER command, but this only works for numbers assigned to FXS ports or IP phones). Therefore, if you have a gateway with E1/T1 connected to it and wish to route certain prefixes there for termination, you must define the routing in the billing. To do this, proceed as follows:

- Create a new tariff with the "Routing Ext".
- When you enter rates into this tariff, two new columns will appear: **Preference** and **Huntstop**. Enter the desired routing preference. (The higher the number, the more desirable this route is. 0 means no route at all.) Turn the huntstop on if you do not wish to use any routes with a lower priority.
- Create a **PSTN to vendor** connection to the vendor, specify the gateway which will handle termination as your **Node**, and select the tariff you have created as the termination tariff.
- Make sure that your gateway is actually configured to accept incoming VoIP calls and send them to telephony for the destinations you plan to terminate.

... create an application to handle PSTN->SIP calls?

You can create this application yourself according to the functionality description in this guide. A PSTN2SIP application may be purchased from **http://store.portaone.com**.

... configure SIP phone X made by vendor Y?

Obviously, we cannot provide a sample configuration for every possible SIP phone model. Please check the documentation shipped with your device. Essentially, however, you need to configure the following settings:

- **IP address of the SIP proxy** - IP address or hostname of the PortaSIP server.
- **CID** (Caller Identification).
- **Login** and **password** – account ID and password of the corresponding account in PortaBilling.
- **Preferred audio codec** – depends on your network characteristics; should be compatible with the codec used by other components (e.g. VoIP gateways used for PSTN termination).

In the case of PortaSIP, both the login name and CID should be set to the same value. Set the preferred audio codec to G.723 if your phone supports this. Likewise, enable in-band alerting if your phone supports it, as this will help in situations when the phone is behind a NAT.

… bill SIP-to-SIP calls?

By default, calls from one SIP account to another are treated as on-net ones, and are therefore not billed. However, if you want to bill your customers for such calls, you can do the following:

- Add the appropriate rate to the tariff associated with the accounts to be charged. For example, if you have SIP accounts with the prefix **078,** then you should add the appropriate rate for destination **078** to the tariff used to charge for outgoing calls.
- Create a special tariff with rates corresponding to the prefixes allocated for your SIP accounts (**078** in the example above). This will be the tariff used to calculate your termination expenses. Since you do not pay anything for such termination, you can enter zero prices for all of the rates.
- Create a new vendor with a descriptive name, for example, "Direct termination to SIP phones". Add a **VoIP to Vendor** connection to that vendor with the tariff created in the previous step and enter **sip-ua** in the **Remote IP** field.

So now, if a call is made from one SIP phone to another, the originating party will be charged according to the rates you have entered in the customer's tariff. This call will be counted as terminated to the vendor **Direct termination to SIP phones**, with zero termination cost – but it will still be recorded in the database, so you can easily view statistics for all SIP-SIP calls.

… bill incoming calls from PSTN to SIP using a special rate?

In order to properly bill a SIP account for such calls, do the following:

- Install a PSTN2SIP application on your Cisco gateway which handles incoming PSTN calls.
- Create an appropriate tariff with the desired rates. For example, if your SIP customer has account **12021234567** and you want to charge him for incoming calls from PSTN to that number, there should be a rate with a prefix matching this number, for example, **1202**.
- In the product associated with this account, add an accessibility entry with this PSTN-SIP gateway as the node and the tariff created in the previous step.

Now calls originating from a SIP phone to 1202 numbers will be charged using the tariff associated in the product's accessibility with the PortaSIP node. Calls terminated from the PSTN to the SIP phone will be charged using a different tariff, one associated with the PSTN gateway.

4. Troubleshooting/ FAQ

Common Problems

No or one-way audio during SIP Phone – SIP Phone calls

This problem usually means that one or both phones are behind a NAT firewall. Unfortunately, unless the RTP Proxy is turned on or certain "smart" SIP phones/NAT routers are used, there is no way to guarantee proper performance in such cases (see Nat Traversal section for details).

One-way audio during SIP Phone – Cisco gateway calls

This problem can occur if the Cisco GW is not configured properly. Please check that the GW contains the following in its IOS configuration:

```
sip-ua
 nat symmetric check-media-src
```

I have problems when trying to use SIP phone X made by vendor Y with PortaSIP

Unfortunately, not all of the many SIP phones available on the market today fully comply with the SIP standard, especially low-end products. We use Cisco ATA 186 as a reference phone, and the Cisco ATA – PortaSIP combination has been thoroughly tested.
If you are unable to get your third-party vendor SIP phone working properly, follow the instructions below:

- Make sure the phone has been configured properly, with such parameters as account ID, password, SIP server address, etc. Consult the product documentation regarding other configuration settings.
- Check the PortaSIP and PortaBilling logs to ensure that there is not a problem with the account you are trying to use (for example, an expired or blocked account).
- Connect the Cisco ATA to the same network as your SIP phone. If possible, disconnect the SIP phone and use the same IP address for the Cisco ATA as was previously used by the third-party SIP phone. Configure the Cisco ATA with the same account as was used on your third-party SIP phone.
- Try to make test calls from the Cisco ATA.
- If you have followed the preceding steps and the problem disappears, then this means your third-party vendor SIP phone is not working according to the standard. Contact the vendor of the SIP phone, and describe the problem.
- If this problem with the Cisco ATA persists, contact support@portaone.com. Provide a full description of the

problem, the ID of the account being used for testing, and the relevant parts of the sip.log and porta-billing.log

FAQ

Why can't my debit account initiate 3-way calling using the features of a SIP phone such as Cisco ATA 186?

Since 3-way calling requires 2 simultaneous outgoing SIP sessions from one SIP telephone, debit accounts will be unable to use it, as the first session will lock the account and not allow the second one to go through. Therefore, if you want to enable your clients to use such services, create a credit account for them instead.

Does PortaSIP support conferencing?

No. Full-scale SIP conferencing requires a separate software or hardware solution. However, you can make use of the features available in some SIP phones, such as Cisco ATA 186, to allow your clients to set up simple, so-called chain conferences. For more information, please refer to the documentation for each specific SIP phone.

Can you assist me in integrating SIP device X (gateway, media server, conference server, etc.) made by vendor Y with PortaSIP?

Yes, we can; however, you will have to purchase an additional consulting contract. Generally speaking, there should be no compatibility problems between PortaSIP and any standards-compliant SIP device. However, for obvious reasons we only provide detailed setup instructions for the Cisco AS5300 gateway.

Can I use PortaSIP with a billing system other than PortaBilling100?

Yes, this is possible. PortaSIP uses the standard Radius protocol to communicate with the billing engine, and its AAA behavior was purposely made very similar to that of Cisco IOS. So it should work with any billing system that supports Radius and can bill Cisco gateways. However, advanced services, such as billing-assisted routing, abbreviated dialing, PortaUM integration, and so on, require support from the billing engine. Detailed specifications of the protocol used to exchange information between PortaBilling100 and PortaSIP are available upon request.

Can PortaSIP be installed on a different Unix-like operating system (Linux, Solaris etc) than FreeBSD?

Yes, this is possible. PortaSIP is easily portable to any modern Unix-like operating system. However, the base price of the product only covers PortaSIP/FreeBSD installation to a clean server using the CD provided by Porta. Once you have purchased the product, we can provide you with the full source code used to build PortaSIP, so that you can compile and install it yourself to an operating system of your choice. Alternatively, we can do this for you; however, you will be additionally charged for the time required for this task at our standard hourly rate. Please also note that a support contract for a non-FreeBSD PortaSIP installation will be more expensive, and must be negotiated on a case-by-case basis. Please contact sales@portaone.com for details.

I want to terminate my SIP customers to a vendor that only supports H.323 traffic – what should I do?

To do this you need to use a SIP->H.323 protocol converter. Either purchase a dedicated solution, available from a number of vendors, or use one of your existing Cisco AS53XX gateways by looping one or more pairs of E1/T1 ports on it to allow SIP->ISDN->H323 call flow.

Please note that, in the latter approach, one ongoing session will consume 1 timeslot in each looped E1/T1 (2 total), as well as 2 DSPs. For example, if you have two E1 interfaces connected back-to-back, the maximum number of simultaneous SIP sessions that you will be able to terminate to your H.323 provider will be 30, and each such session will use 2 DSPs. In Appendix D you will find information on how to set up such a back-to-back connection physically and configure it in Cisco IOS.

I have connected the Cisco AS53XX gateway to PSTN in order to send calls from PSTN to my SIP accounts and terminate calls from my SIP accounts to PSTN. How many simultaneous sessions will it be able to handle?

A rule of thumb is that each SIP->PSTN call or PSTN->SIP call will use up one DSP and one timeslot in E1/T1 interface. Therefore, if you have connected your gateway to PSTN using, for example, two E1 ports, and are using both of those ports for SIP<->PSTN, the maximum number of simultaneous calls you will be able to handle will be 60, provided that you have enough free DSPs in the system.

I have problems with the audio quality of SIP calls, what can I do?

First of all, please make sure that both the user agents and SIP<->PSTN gateway are configured for use of the same low-bitrate codec, such as G.723.
In Appendix E, there are details on how to configure Cisco IOS and Cisco ATA 186; for other SIP phones or gateways, check the

documentation supplied with the device. If you are sure that the codec used for SIP calls is a low-bitrate one (for example, by inspecting the gateway logs), but the quality is still suboptimal, you need to determine where packet loss is occurring in the media path. To do this, you can use standard network tools such as ping, traceroute and the like. Keep in mind that for SIP UA<->PSTN calls the RTP audio stream flows directly between SIP UA and PSTN GW, while for SIP UA<->SIP UA calls the RTP path depends on whether or not an RTP proxy is enabled. If an RTP proxy is not enabled, the RTP flows directly from one SIP UA to another. Otherwise, each RTP packet sent by one UA goes first to the machine running PortaSIP and is then resent from that machine to another SIP UA.

Configuration

There are several separate configuration files for PortaSIP components:

SIP Proxy Server

The main configuration file is called ser.cfg, and is located in the `/usr/local/etc/ser` directory. This file defines the rules for processing incoming SIP messages by the proxy's core. You can read more about its syntax here: **http://www.iptel.org/ser/doc/seruser/seruser.html**.

NOTE: Please note that this file is quite complex and is not intended for modification by the user. Therefore, you must be careful if making changes to it, always keeping a backup of the current version.

There are another two configuration files in the `/usr/local/etc/ser` directory: `servers` and `radiusclient.conf`. These files define the parameters of the Radius stack used by the Proxy to communicate with the billing engine.

The following parameters can be adjusted in `radiusclient.conf`:

- `authserver` – IP address of the Radius server, usually an address of the master PortaBilling100 host;
- `radius_timeout` – the length of time to wait for a reply from the Radius server;
- `radius_retries` – the number of times to resend a request before giving up on it;
- `bindaddr` – if PortaSIP is running on a machine with multiple IP addresses, set this parameter to the IP address from which you want Radius requests to be sent.

The `servers` file contains the Radius Key for the Radius server as specified in the `radiusclient.conf`.

Back-to-Back User Agent

The configuration file for the B2BUA is called `b2bConfig.xml` and is located in `/usr/local/etc`. There are several parameters that can be adjusted here:

- `SIP/Local/Address`: the IP address which the B2BUA should use on a machine with multiple IP addresses;
- `SIP/Proxy_Server/Address`: the IP address on which the SIP Proxy is running;
- `SIP/Proxy_Server/Port`: the UDP port used by the SIP Proxy;
- `RADIUS/Billing_Server/Address`: the IP address or hostname of the Radius server, usually an address of the master PortaBilling100 host;
- `RADIUS/Billing_Server/Authentication_Port`: the UDP port to which authentication requests are to be sent;
- `RADIUS/Billing_Server/Accounting_Port`: the UDP port to which accounting requests are to be sent;
- `RADIUS/Billing_Server/Password`: Radius Key.

Starting/Stopping PortaSIP Services

If you need to stop all PortaSIP services, then execute the following command:

```
$ sudo /usr/local/erc/rc.d/sip.sh stop
```

This will properly terminate all components. To start PortaSIP, use the following command:

```
$ sudo /usr/local/erc/rc.d/sip.sh start
```

NOTE: Please always make sure that you have stopped services as described above before trying to start them again, since trying to start services when they are already running may render the service inoperable.

5. Appendices

APPENDIX A. Tested Routers and NAT Software

Commodity routers and NAT software bundled with popular operating systems which attempt to preserve the RTP source port:

1. Linksys BEFSX41
2. Belkin F5D5230-4
3. `natd` bundled with FreeBSD 4.x and 5.x operating systems
4. `iptables` bundled with Linux kernel 2.4.x

Commodity routers and NAT software bundled with popular operating systems which do not attempt to preserve the RTP source port:

1. Internet connection sharing software bundled with the Windows XP operating system
2. Netgear RP614

APPENDIX B. Cisco GW Setup for PortaSIP (COMEDIA)

```
sip-ua
  nat symmetric check-media-src
```

APPENDIX C. Clients' Cisco ATA 186 Configuration for PortaSIP

UID0	[CLIENT'S ACCOUNT ID (PHONE NUMBER) 1]
PWD0	[CLIENT'S PASSWORD FOR ACCOUNT ID 1]
UID1	[CLIENT'S ACCOUNT ID (PHONE NUMBER) 2]
PWD1	[CLIENT'S PASSWORD FOR ACCOUNT ID 2]
GkOrProxy	[IP ADDRESS OF SERVER RUNNING PORTASIP]
Gateway	0.0.0.0
GateWay2	0.0.0.0
UseLoginID	0
LoginID0	0
LoginID1	0
AltGK	0.0.0.0
AltGKTimeOut	0
GkTimeToLive	300

GkId	.
UseSIP	1
SIPRegInterval	180
MaxRedirect	5
SIPRegOn	1
NATIP	0.0.0.0
SIPPort	5060
MediaPort	[DIFFERENT FOR EACH CLIENT AS DESCRIBED IN THE SETUP GUIDELINES]
OutBoundProxy	0.0.0.0
NatServer	[IP ADDRESS OF SERVER RUNNING PORTASIP]
NatTimer	0x1e
LBRCodec	0
AudioMode	0x00150015
RxCodec	0
TxCodec	0
NumTxFrames	1
CallFeatures	0xffffffff
PaidFeatures	0xffffffff
CallerIdMethod	0xc0019e60
FeatureTimer	0
Polarity	0
ConnectMode	0xe0400
AuthMethod	0
TimeZone	[SEE CISCO ATA 186 DOCUMENTATION FOR ENTERING CORRECT VALUE]
NTPIP	192.43.244.18
AltNTPIP	131.188.3.222
DNS1IP	0.0.0.0
DNS2IP	0.0.0.0
UDPTOS	0xb8
SigTimer	0x64
OpFlags	0x62
VLANSettings	0x2b
NPrintf	0.0.0.0
TraceFlags	0

The manufacturer's default values are assumed for all options not listed here.

APPENDIX D. Setting up back-to-back T1/E1 connection

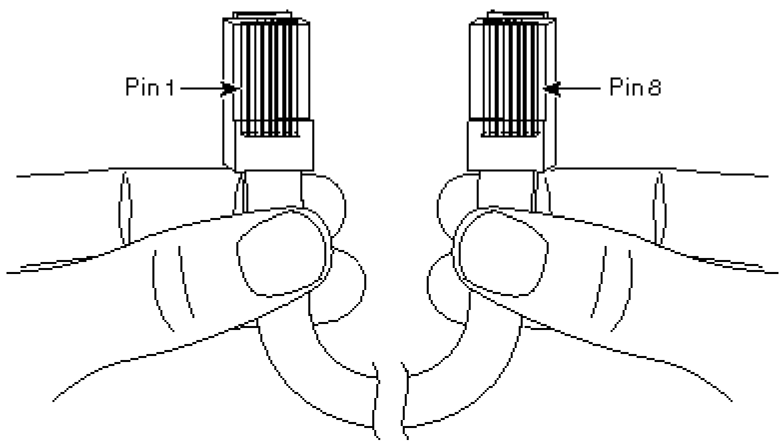
Hardware Setup

In order to make one or more back-to-back connections, you will need to construct one or more RJ-48C cross-over cables using the following table:

T1/E1 CSU/DSU Cross-Over Pinout

From RJ 48C Pin	To RJ 48C Pin
1	4
2	5
4	1
5	2

Make sure you count the RJ-48C pins as shown in the illustration below:



Alternatively, you can order ready-made ones. You can find a number of vendors producing such cables by searching for "RJ-48C cross-over cable" on www.google.com.

Once the cable is ready, plug it into the designated pair of T1/E1 ports in your Cisco AS5300 gateway.

Software Configuration

You also have to configure the T1/E1 interfaces. The sample configuration below is for T1; adjust the time slots for E1:

```
isdn switch-type primary-5ess
!
controller T1 0
framing sf
clock source line primary
linecode ami
pri-group timeslots 1-24
!
controller T1 1
framing sf
clock source line secondary 1
linecode ami
pri-group timeslots 1-24
!
controller T1 2
framing sf
linecode ami
pri-group timeslots 1-24
!
controller T1 3
framing sf
linecode ami
pri-group timeslots 1-24
!
interface Serial0:23
no ip address
isdn switch-type primary-5ess
isdn protocol-emulate network
no cdp enable
!
interface Serial1:23
no ip address
isdn switch-type primary-5ess
no cdp enable
!
interface Serial2:23
no ip address
isdn switch-type primary-5ess
isdn protocol-emulate network
no cdp enable
!
interface Serial3:23
no ip address
isdn switch-type primary-5ess
no cdp enable
```