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Preface

This document provides PortaBilling100 users with the most common examples and guidelines for setting up a VoIP network. The last section of the document answers the most frequent questions users ask after running PortaBilling100 for the first time.

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 between minor releases. The online copy of this guide is always up to date, and integrates the latest changes to the product. You can access the latest copy of this guide at **www.portaone.com/solutions/billing/docs**

Conventions

This publication uses the following conventions:

- Commands and keywords are in **boldface**
- Terminal sessions, console screens and system file names are displayed in `fixed width font`

Caution means 'reader beware'. You are capable of doing something that might result in a program malfunction or loss of data.

NOTE: Means 'reader take note'. Notes contain helpful suggestions 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 are information that might help you to solve a problem.

Hardware and Software Requirements

Server System Recommendations

- One UNIX Server.
- A minimum of 40 GB of available disk space, this space is required for storing various log files
- A processor running at 2.4 GHz or greater. Additional processor speed is needed for networks with a high call volume.
- At least 512MB of RAM (RDRAM or DDR), 1 GB recommended.
- At least one USB port.

For information about whether particular hardware is supported by FreeBSD from the JumpStart Installation CD, consult the related document on the FreeBSD website:
http://www.freebsd.org/doc/en_US.ISO8859-1/books/faq/hardware.html

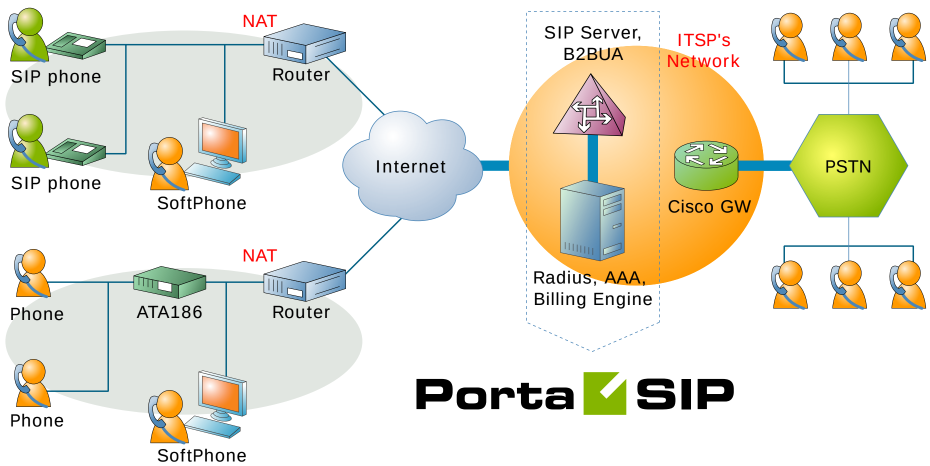
Client System Recommendations

- OS: Windows 95-XP, UNIX or Mac OS
- Browser: Internet Explorer 6.0 or higher, Netscape 7.1 / Mozilla 1.6 or higher supporting DOM and with JavaScript enabled.
- Spreadsheet processor (MS Excel)
- Display settings:
  - Minimum screen resolution: 1024 x 768
  - Color palette: 16 bit color (minimum)

NOTE: To view downloaded CSV (Comma-Separated Values) files in Windows, please do the following to match PortaBilling's default list separator: My Computer -> Control Panel -> Regional Settings -> Number -> List Separator type ",".

1. System Concepts

PortaSIP's Role in Your VoIP Network

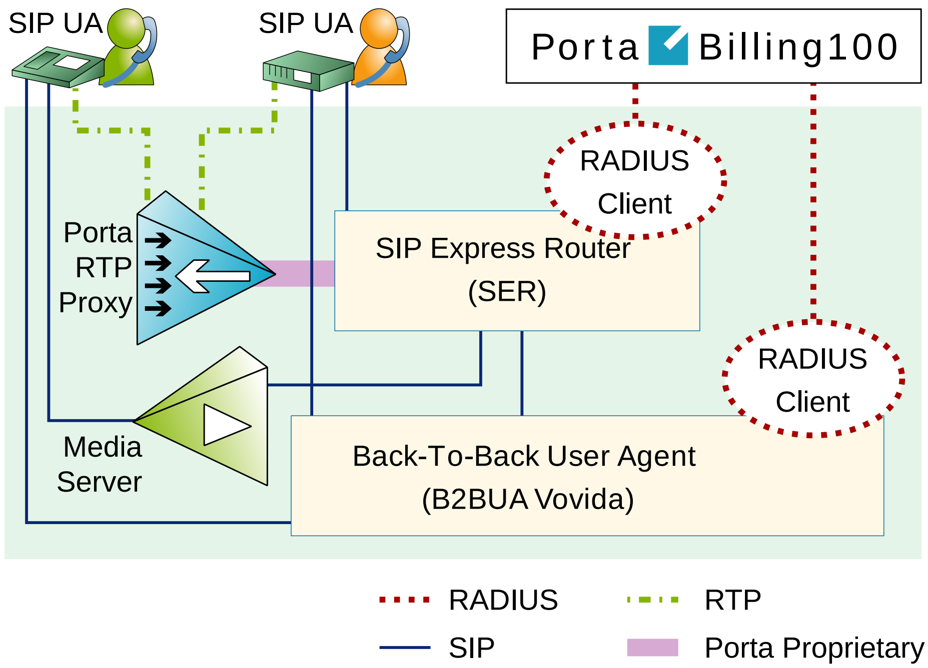


PortaSIP is a call control software package enabling service providers to build scalable, reliable VoIP networks. Based on the Session Initiation Protocol (SIP), PortaSIP provides a full array of call routing capabilities to maximize performance for both small and large packet voice networks.

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:

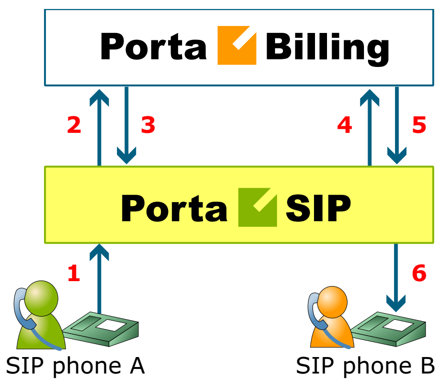
- SIP Proxy Server (SIP Express Router on the diagram): The SIP Proxy Server performs a number of functions, such registering SIP telephones, dealing with NAT issues, etc.
- Back-To-Back User Agent (B2BUA): The B2BUA SIP-based logical entity can receive and process INVITE messages as a SIP User Agent 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.

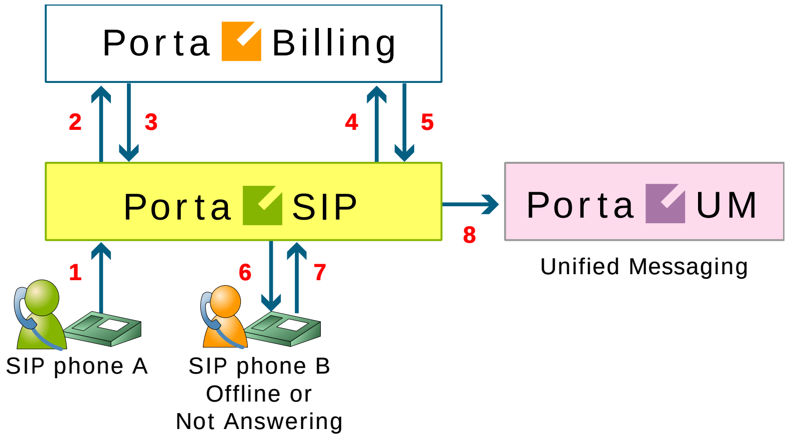
When the called party is online



This is the simplest case:

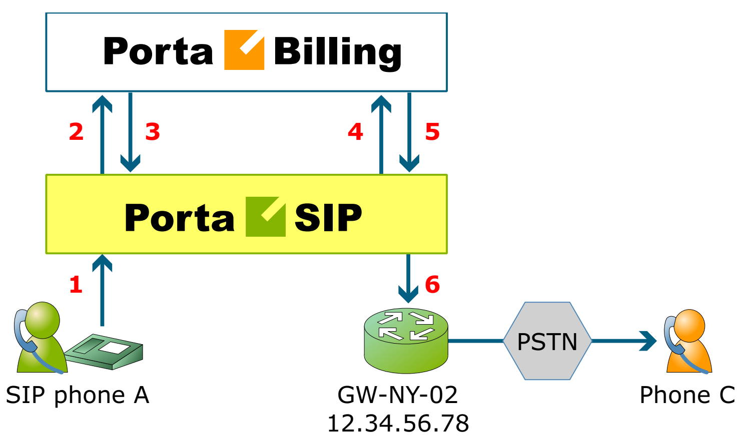
- User A dials B's number (121). His SIP user agent sends an INVITE request to the SIP server (1).
- The SIP server performs a number lookup in the billing (2). For example, if user A dials 121, the billing will inform the SIP server that the actual number is 12027810009, and that this number belongs 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.

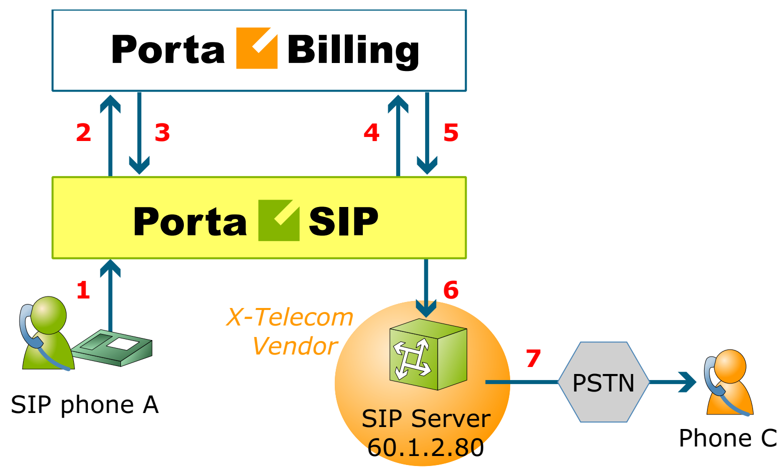
SIP UA -> PSTN



- 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.

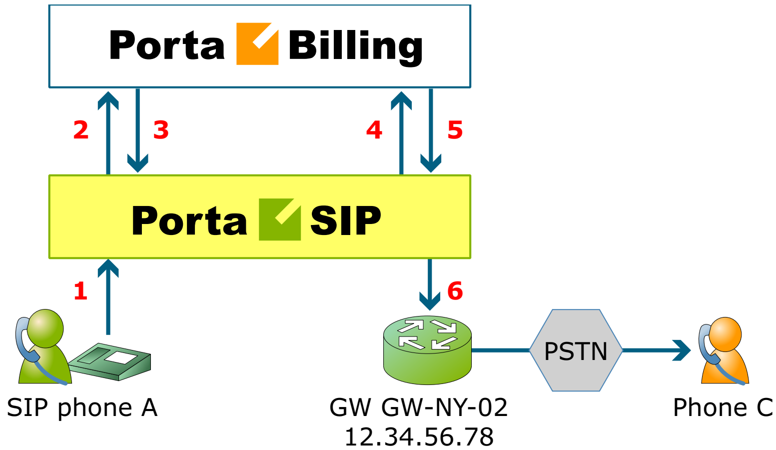
Terminating SIP calls to a vendor using VoIP



- 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



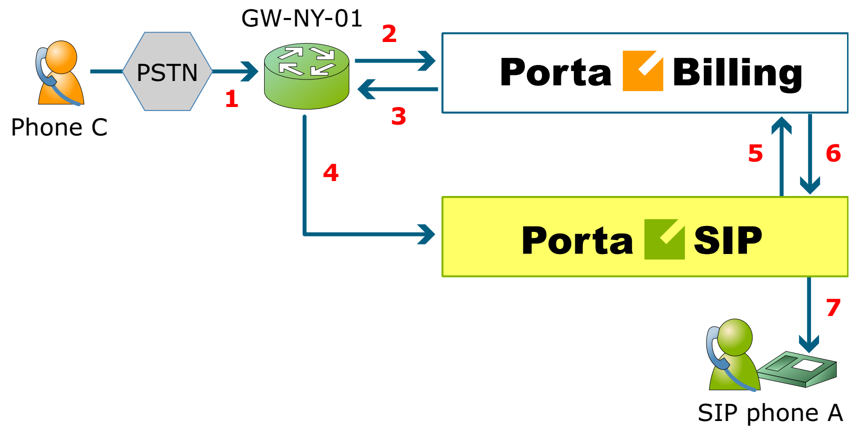
- Let's assume that T1 is connected to Qwest on our gateway **GW-NY-02** in New York, where we are able to terminate calls to the US. This connection would be described as a **PSTN to vendor** connection. The PortaSIP server obtains the address of the GW-NY-02 gateway in the route information.
- The 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.

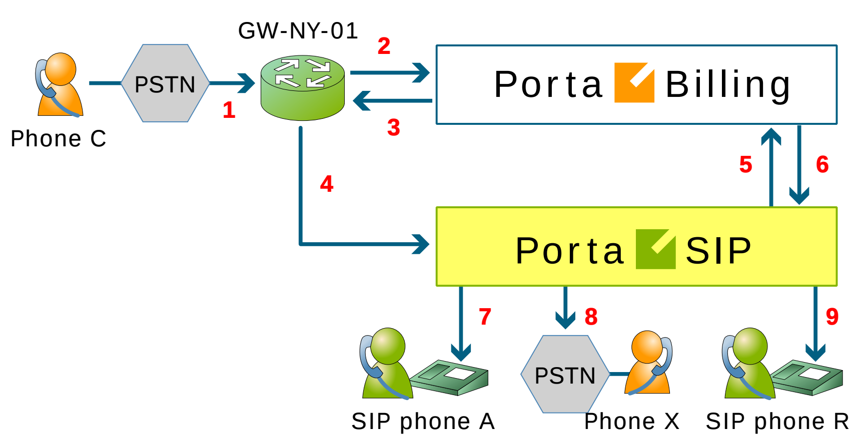
- C wishes to call A. He thus dials A's phone number (since C is in the US, he dials it using the North American format, 2027810003).
- This call is routed through the telecom network to gateway GW-NY-01. When the incoming call arrives on the gateway (1), it starts a special TCL application to handle this call. This application does several things:
  - Converts the phone number to the E.164 format, so that 2027810003 become 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.

Follow-me services

Due to the volatile nature of VoIP networks, the customer may wish to use standard PSTN calls as a backup. He can define a list of follow-me numbers (for each of which a period of validity can be defined, e.g., he wants to receive calls to his mobile phone only from 8am to 9pm). When a call arrives on his original SIP account, the SIP server can try the alternative numbers until the call is answered.



- C wishes to call A. So he dials A's phone number (since C is in the US, he dials it using the North American format, 2027810003).
- The call is routed through the telecom network to gateway GW-NY-01. When the incoming call arrives at the gateway (**1**), it is processed there in exactly the same way as a normal PSTN->SIP call: the number is transformed, the call is authorized in the billing (**2**), and the timer starts to measure the maximum call time allowed, based on A's current balance (**3**).
- The call is sent to PortaSIP (**4**).
- PortaSIP receives the INVITE, but without authorization information. So the PortaSIP server performs authorization in the billing based on the IP address, and also requests billing-assisted routing (**5**).
- PortaBilling recognizes that the destination is an account with follow-me services enabled, and produces a special list of routes:
  - If the follow-me mode chosen is "When unavailable", then a direct route to the account's SIP UA is included as the first route in the list, with a default timeout.
  - A list of follow-me numbers is produced. If the current time falls outside the specified period for a certain number, it is removed from the list.
  - If the follow-me order is "Random", then the list of phone numbers is shuffled.
  - The maximum call duration is calculated for each follow-me number, based on the balance and rates for the **called** account (A).
  - The resulting list of routes is produced and sent back to PortaSIP (**6**).
- PortaSIP tries the first route (**7**); if the call is not connected within the timeout interval, it moves to the next route (**8**), then to the next one (**9**), until either the call is put through or no more routes are left.
- If such a call was completed to follow-me number R, two CDRs will appear i90
- n the system: one for the call C->A (charged per the incoming rates for A) and the other for C->R (charged per the outgoing rates for A).
- If the call did not originate in the PSTN network, but rather from user B's SIP UA, two CDRs will likewise be generated. B will be charged for call B->A, while A will be charged for call B->R.

Note that the follow-me service is not recursive. Thus A can forward calls from his SIP phone to B's SIP phone, but the call will only go directly to B's SIP phone. If B has set up forwarding to his mobile phone, this forwarding will not be used, in order to prevent excessive routing and routing loops.

Service announcements via the media server

A customer might be unable to make a call not only due to network problems, but also for various administrative reasons, for example, if his account is blocked or he does not have enough money on his account. If the end user can be informed of such administrative problems, instead of just being given a busy signal, this will greatly simplify troubleshooting. Here is what would happen in the event that, for instance, an account which is blocked attempts to make a call:

- The customer tries to make a call. SIP proxy receives the INVITE request and sends an authorization request (LOOKUP) to the billing.
- PortaBilling determines that this account is blocked. An authorization reject is returned to the SIP server. In addition to the h323-return-code, a special attribute is sent back to the SIP server. This attribute contains a description of the type of error – in this case, "user_denied".
- The SIP server receives the authorization reject from the billing. However, instead of just dropping the call, it redirects the call to the media server, including the error message as a parameter.
- The media server establishes a connection with the SIP UA. It locates a voice prompt file based on the error type and plays it to the user. After this the call is disconnected.

The media server and prompt files are located on the SIP server. So as to avoid dynamic codec conversion, there are three files for each prompt (.pcm, .723 and .729). These files are located in `/usr/local/share/asterisk/sounds`, and you can change them according to your needs. Here is a list of the currently supported error types:

- **account_expired** – the account is no longer active (expired as per the expiration date or life time)
- **cld_blocked** – there was an attempt to call a destination which is not in the tariff, or is marked as forbidden
- **credit_disconnect** – a call is disconnected because the maximum credit time is over
- **in_use** – currently not available, since this error condition occurs in B2BUA during INVITE, and redirection to the media server is done by the SIP proxy during LOOKUP
- **insufficient_balance** – there are not enough funds to make a call to the given destination
- **invalid_account** – incorrect account ID, or account is not permitted to use SIP services
- **user_denied** – the account is blocked
- **wrong_passwd** – an incorrect password has been provided

Understanding SIP Call Routing

When the PortaSIP server has to establish an outgoing call, it must find out where the call is being sent to. To do this, it will ask billing for a list of possible routes. In this case the routing configuration is in one central location, and billing can use information about termination costs to choose the best route (least-cost 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 the voice mailbox (PortaUM required) if the called number belongs to an account in PortaBilling, but this account is not currently registered to the SIP server (is off-line).
- 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 whether a call should be sent directly to a SIP user agent.

Routing of off-net calls

You can have different vendors for terminating off-net calls. For example, you can terminate calls to the US either to AT&T, via a T1 connected to your gateway in New York, or to a remote gateway from Qwest.

Rate routing parameters

Ordinarily, tariffs define the termination costs for each connection to a vendor. If you create a tariff with the **Routing** type, a few more fields will be added to rates in that tariff:

- **Route category** – you can split this into categories such as “Premium”, “Cheap”, etc. and use these categories in routing plans
- **Preference** – routing priority (0-10), higher values mean higher priority, 0 means do not use this rate for routing at all
- **Huntstop** – signals that no routes with a lower preference should be considered

This allows you to easily manage both termination costs and routing from a single location on the web interface. Thus, when such a routing tariff is associated with a connection, you can send calls for termination to all prefixes for which rates exist in the tariff.

Multiple routes

It is dangerous to have only one termination partner: if it is down, your customers will not be able make any more calls. Normally, you will try to find several vendors and enter their rates into the system. Each connection to a vendor (with routing tariff) will produce one possible route, and PortaBilling will arrange them according to cost or your other preferences.

Routing plans

Routing preferences in the rate allow you to specify that, for example, you would rather send a call to MCI than to T-Systems. However, this decision is "global", and so will apply to all calls made in your system. But what if you would like to use MCI first for customer A, while T-Systems should be the first route for customer B, and customer C should be routed to MCI only?
This can be accomplished using routing plans. A routing plan defines the routes for which categories are available, as well as in which order they should be arranged. For instance, in the example above MCI may be assigned as the "Normal" route category and T-Systems as the "Premium" category. After that, three routing plans will be created:

- **Quality** - includes first Premium and then Normal routing categories
- **Ordinary** - includes first Normal and then Premium routing categories
- **Cost-efficient** – includes only Normal routing category

So, depending on which routing plan is assigned to the current customer, the system will offer a different set of routes.

Routing algorithm

The routing principle is simple:

- The SIP server (or MVTS, or some other entity) asks PortaBilling for routing destinations for a given number.
- PortaBilling checks every tariff with routing extensions associated with a vendor connection for rates matching this phone number. In each tariff the best-matching rate is chosen; this rate will define the routing parameters.
- A list of possible termination addresses will be produced (this will include the remote IP addresses for VoIP connections and IP addresses of your own nodes with telephony connections).

- This list will be sorted according to routing plan, routing preference and cost; entries after the first huntstop will be ignored.
- A list of these IP addresses (with optional login and password for SIP authentication) will be returned to the SIP server. (To avoid extremely long delays, only a certain number of routes from the beginning of the list are returned; the default is 15, but this can be changed in `porta-billing.conf`).

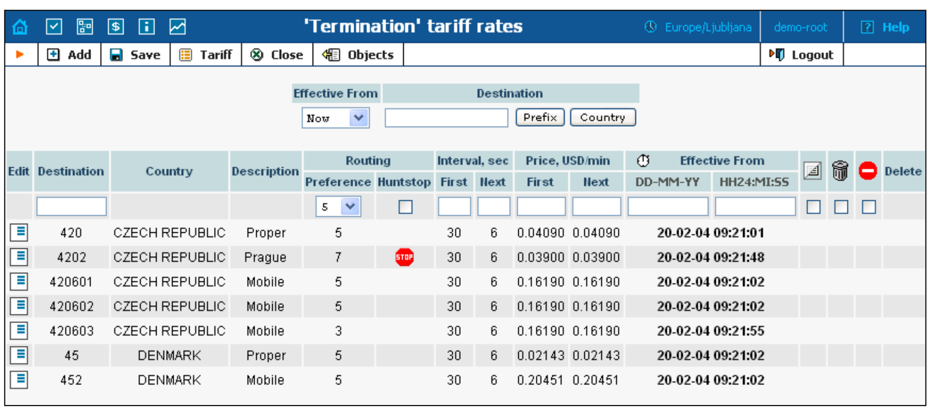
Route sorting

How exactly does PortaBilling100 arrange multiple available routes?

1. By route category. Only route categories which are included in the routing plan will be used, following the order given in the routing plan.
2. Then, routes within the same route category are arranged according to preference.
3. For routes with the same preference, the system can arrange them according to cost (a comparison is made on the **Price_Next** rate parameter) so that cheaper routes will be among the first ones, or in random fashion.

Does PortaSwitch support LCR?

Yes, we support LCR – and much more besides. In fact, "just LCR" is the simplest type of routing PortaSwitch handles. If you decide not to use routing plans (one default plan for everyone) or routing preferences (same preference for all vendors), then the routing decision will be affected solely by the vendor's cost.



Edit	Destination	Country	Description	Routing Preference	Routing Huntstop	Interval, sec First	Interval, sec Next	Price, USD/min First	Price, USD/min Next	Effective From DD-MM-YY	Effective From HH24:MI:SS	Delete
	420	CZECH REPUBLIC	Proper	5		30	6	0.04090	0.04090	20-02-04	09:21:01	
	4202	CZECH REPUBLIC	Prague	7	STOP	30	6	0.03900	0.03900	20-02-04	09:21:48	
	420601	CZECH REPUBLIC	Mobile	5		30	6	0.16190	0.16190	20-02-04	09:21:02	
	420602	CZECH REPUBLIC	Mobile	5		30	6	0.16190	0.16190	20-02-04	09:21:02	
	420603	CZECH REPUBLIC	Mobile	3		30	6	0.16190	0.16190	20-02-04	09:21:55	
	45	DENMARK	Proper	5		30	6	0.02143	0.02143	20-02-04	09:21:02	
	452	DENMARK	Mobile	5		30	6	0.20451	0.20451	20-02-04	09:21:02	

Example

If we have:

1. "Standard" routing plan, which includes the routing categories "Default", "Cheap", "Expensive" (in that order)
2. Six vendors (A, B, C, D, E, F) with the following rates (prefix, route category, preference, price):

a. 4202, Cheap, 7, 0.04
b. 420, Default, 5, 0.06
c. 420, Cheap, 6, 0.03
d. 420, Cheap, 6, 0.025
e. 420, Expensive, 5, 0.11
f. 4202, Premium, 5, 0.09

then when a customer with this routing plan makes a call to 42021234567, the system will arrange the possible routes as follows:

Vendor	Parameters	Comment
B	Default, 5, 0.06	The "Default" route category is first in the route plan
A	Cheap, 7, 0.04	This vendor has the highest preference in the "Cheap" category
D	Cheap, 6, 0.025	This vendor has the same preference as vendor C, but a cheaper per-minute rate
C	Cheap, 6, 0.03	
E	Expensive, 5, 0.11	This is the only vendor in the last route category

(vendor F was not included in the routing, because his route category is not in the customer's routing plan).

Number translation

There are many different phone number formats, some used by your customers, others by your vendors. How to deal with all of them without making mistakes? PortaBilling offers a powerful tool called **translation rules** for converting phone numbers, with several different types depending on customers' needs.

Your network numbering plan

The key to avoiding problems with number formats is to choose a certain number format as the standard for your network and make sure that calls travel on your network only in this format. The ideal candidate for such a format is E.164 (of course it is highly recommended that you use this same format in billing as well). When a call arrives from your customer (with a phone number from a customer-specific number plan), PortaSwitch will convert the number into your network format. It will then travel on your network until it is sent to a vendor for termination. Just before this happens, it can be converted into the vendor-specific format.

Customer-based translation rules

Very often your customer will have his own numbering format, for example, dialing 00 for international numbers, while dialing just the phone number for local calls. Customer-based translation rules allow you to convert a number from a format specific to this particular customer. There is a special dialing rules wizard available to make such configuration easier, so that customers can even do this themselves. Customer-based translation rules have two applications:

- When a number is submitted for authorization, these rules will be applied, with the resulting number used to search rates. Thus, if your customer dials 0042021234567, you can convert it to 42021234567 and find the correct rates for the 420 prefix.
- This number will be returned to the node which requested it.

Connection-based outgoing translation rules

If your vendor requires a special number format (e.g. tech-prefix), it is possible to apply this rule to convert the number. When billing returns a list of routes, the phone numbers for routes for this connection will be converted. This only works for a routing model in which the VoIP node (e.g. PortaSIP) requests billing for routing information. If your gateway uses dial-peers or an external gatekeeper for routing, then you must configure number translation there.

Connection-based translation rules

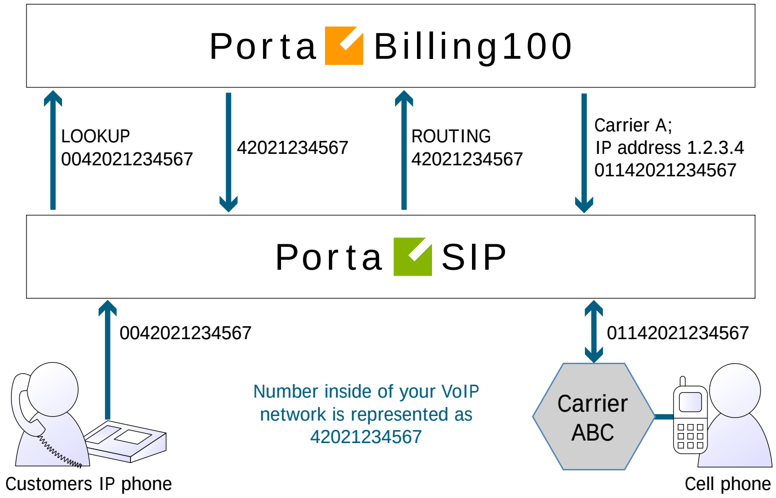
When the call has been terminated to the vendor in a vendor-specific format, it will be reported to billing in this same format (e.g. 7834#42021234567). Now it is necessary to convert this number to the proper format for billing (4202134567), which may be done using connection translation rules. These rules will be applied to all calls which go through a given connection (even those routed there using dial-peers or other external tools)

Node-based translation rules

These serve the purpose of converting a number from a custom format used by the customer into billing's internal format during authorization, depending on the gateway. For example, on a gateway in Prague, Czech Republic, there may be the translation rule "strip leading 00", while on a gateway in Moscow, Russia, the rule will be "strip leading 810 or replace leading 8 with 7".
Since customer-based translation rules were introduced, node-based translation has become obsolete. Therefore, a node-based translation rule is applied only if there is no customer-based translation rule defined for a given customer.

Number translation on your network

Below is an illustration of how different translation rules are applied during a call.



1. The customer dials a phone number on his SIP phone. He enters the number in the same format he uses on his conventional phone, i.e. 0042021234567.
2. The number is delivered to the PortaSIP server and translated using the **customer's dialing rule**, which states that the international dialing prefix for this customer is 00. So the number becomes 42021234567 (E.164 format). This number is used to search for the customer's rate for this destination.
3. PortaSIP then requests routing for this number. Carrier ABC is defined for terminating calls to the Czech Republic in PortaBilling. However, this carrier requires the number to be in US dialing format, so the international number must be prefixed by 011. An **outgoing translation rule** s/^/011/; to carrier ABC has been defined, and is now applied to the phone number, with the result 01142021234567. Note that there may be several carriers who can terminate this call, each with its own numbering format. In such a case there will be several alternative routes with different phone numbers.
4. PortaSIP attempts to establish a connection to remote gateway 1.2.3.4 using phone number 01142021234567.
5. After the call is completed, PortaSIP sends an accounting request to PortaBilling, stating that a call to remote gateway 1.2.3.4 has just been completed. PortaBilling finds a connection to vendor ABC with remote IP address 1.2.3.4, and applies the **translation rule** s/^011//; for this connection in order to convert the number from the vendor-specific format into your billing format. Thus 011 is removed from 01142021234567, and the number becomes 42021234567. PortaBilling searches for the vendor and customer rates for this number and produces the CDRs.

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.

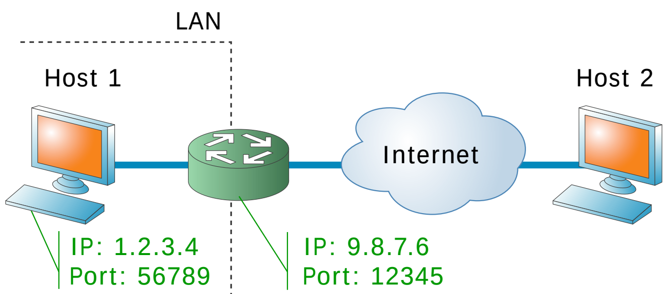
NAT Traversal Guidelines

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

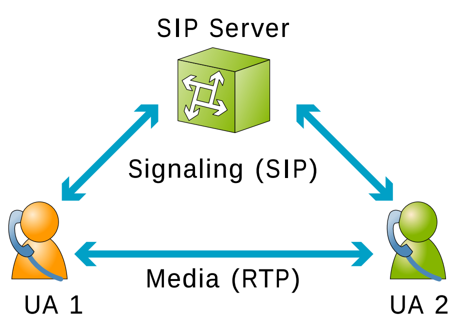
IP that it identifies as the source of the SIP message, or a client that is aware of its external mapped address and port and can insert them into the contact information as the `IP:port` for receiving SIP messages. You should be careful to use a registration interval shorter than the keep-alive time 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: www.portaone.com/resources/documentation/ for detailed instructions on how to navigate and operate the web interface, as well as detailed explanations for particular fields.

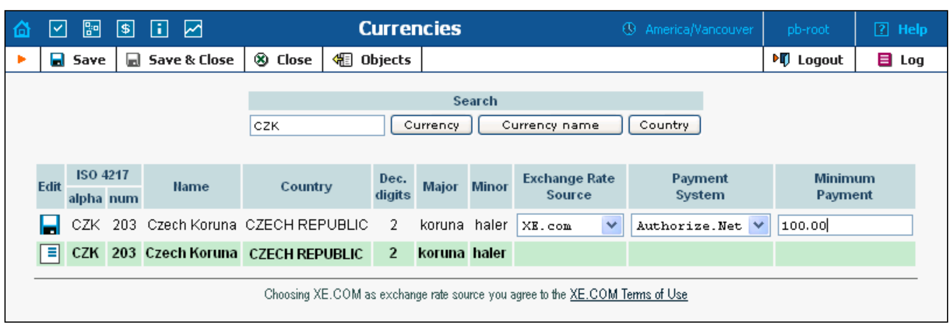
Initial Configuration of PortaBilling

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

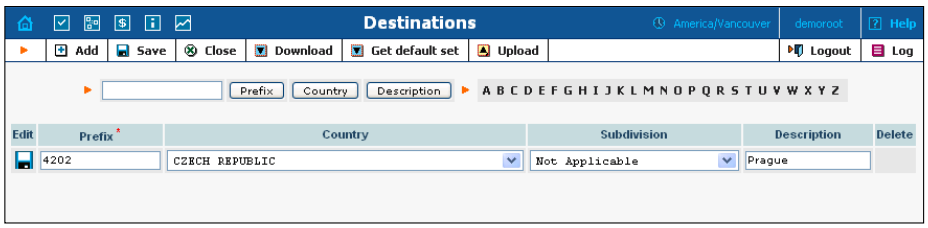
This step is only required if you have not previously defined the necessary destinations. There are two ways to insert a new destination into the system:

- One-by-one, using the **Add** 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”:

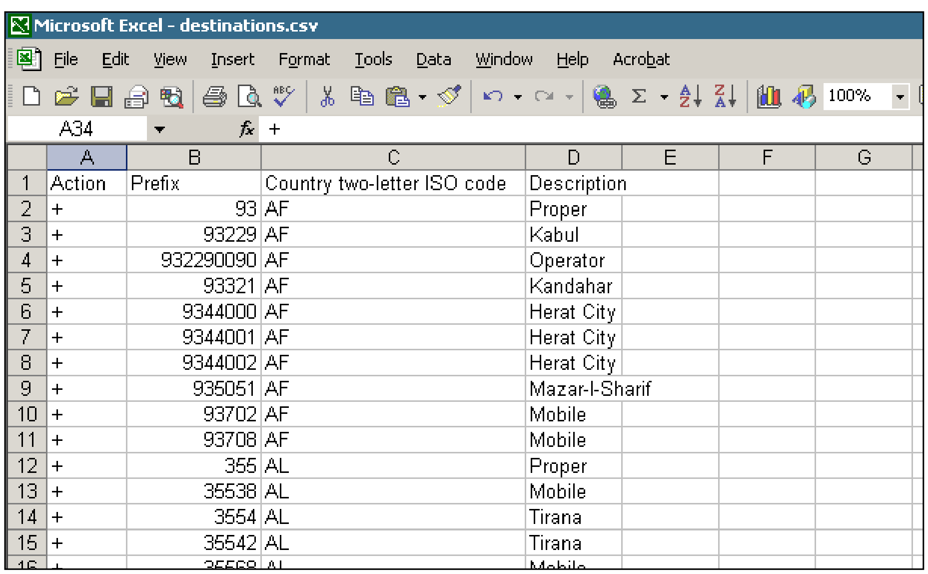
1. In the Management section of Admin-Index, choose **Destinations**.
2. Click on the **Add** button.
3. Fill in the required information. This includes the phone prefix and country name. The country subdivision is optional. You can use the **Description** column to store extra information about the 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.

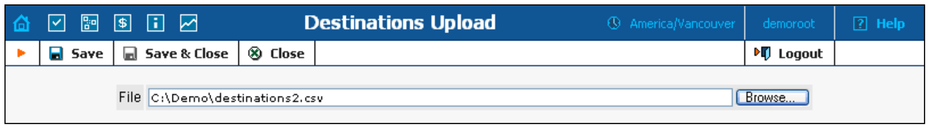
Uploading a set of destinations from a file:

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
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.



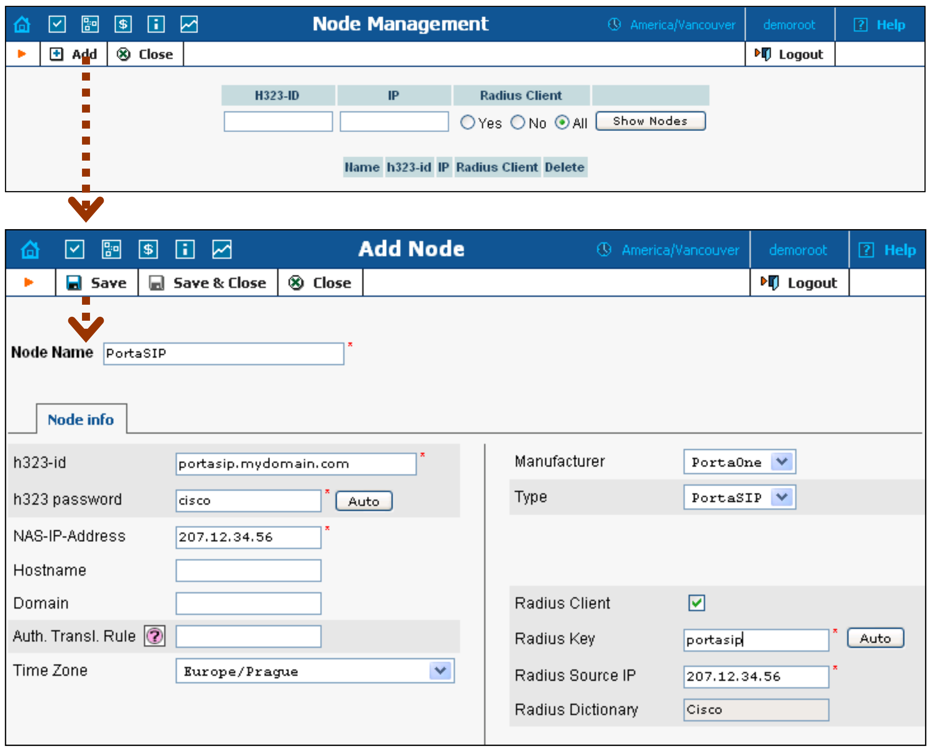
6. Type in the filename for the file you have edited, or click on the **Browse…** button and select the file.
7. Click **Save&Close.**

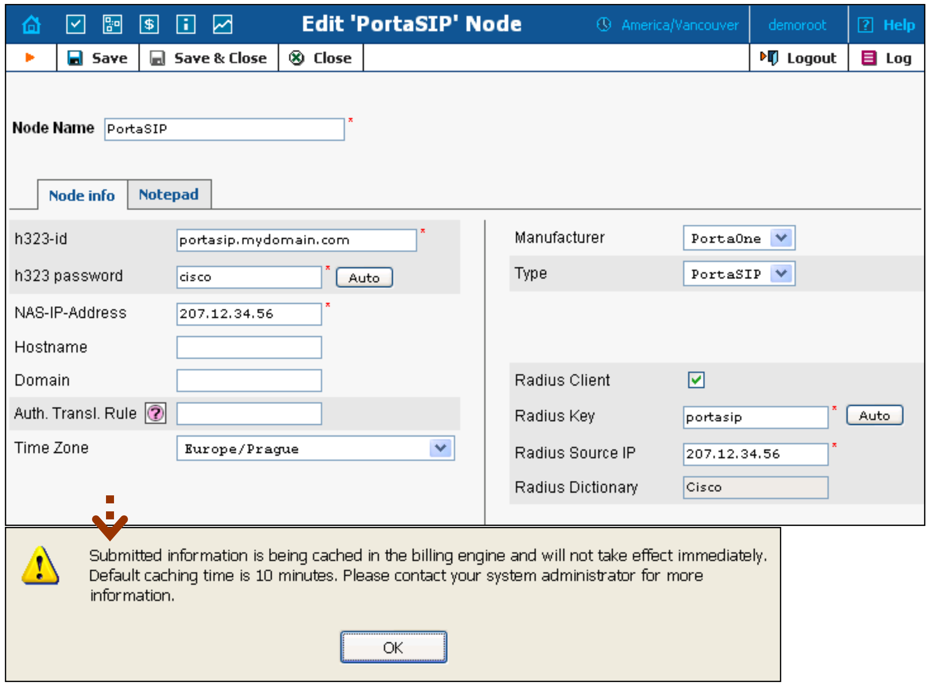
Destinations for SIP phones

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** 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. And if some of your customers wishes to use his own numbering – you set up customer based translation rule (dialing rules) for each of them individually.
   - **Manufacturer** – select **PortaOne**.
   - **Type** – VoIP node type; select **PortaSIP**.
   - **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**.

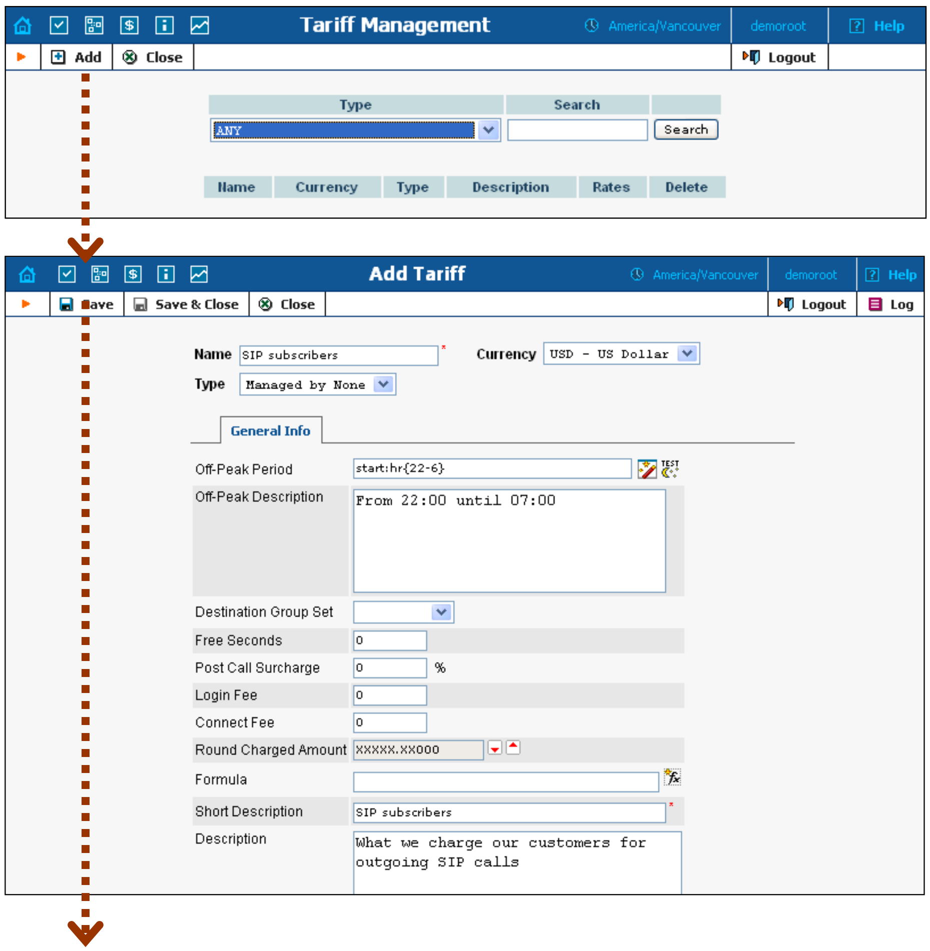
5. Repeat steps 2-4 for any additional gateways you may have. Use **VOIP-GW** as the node type.

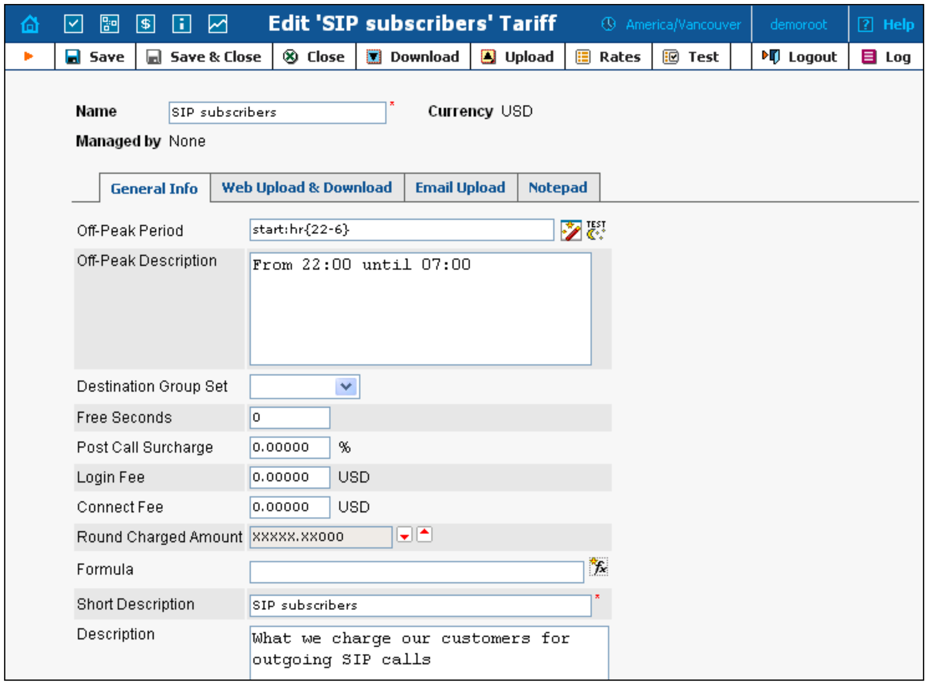
NOTE: There is some propagation delay between the database and the Radius server configuration file; however, it is no more than 15 minutes.

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**.
3. Fill in the **Add Tariff** form:
   - **Name** – a short name for the tariff object; this is the name you will then see in the select menus.
   - **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.

   - **Type** – If this is a tariff that describes your vendor's termination costs, choose **Routing** here, as this tariff will be used not only to calculate termination costs, but also for routing SIP calls. if you plan for this tariff to be used for your reseller's accounts, so that the reseller himself can edit rates in this tariff, choose "Managed by NNN", where NNN is the reseller's name. Otherwise, if this is the tariff for a retail customer's accounts, choose **Managed by 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.
- **Destination group set** – if you wish to enter rates in the tariff not for every individual prefix, but for a whole group of prefixes at once, you should create a destination group set and destination groups beforehand. Leave this select menu empty for now.
- **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).
- **Round charged amount** – Instead of calculating CDRs with a 5-decimal-place precision, round up CDR amount values (e.g. to cents, so that 1.16730 becomes 1.17). Set the rounding pattern to XXXX.XX000 (as shown on the picture) so every call will be rounded to the equal cent amount.
- **Formula** – Default rating formula, which will be applied to every rate created in the tariff. If you leave this empty, the "old-style" rating will be used.
- **Short Description** – a short tariff description. This will be shown in the rate lookup on the admin interface and the self-care pages for your accounts and customers.
- **Description** – an extended tariff description.

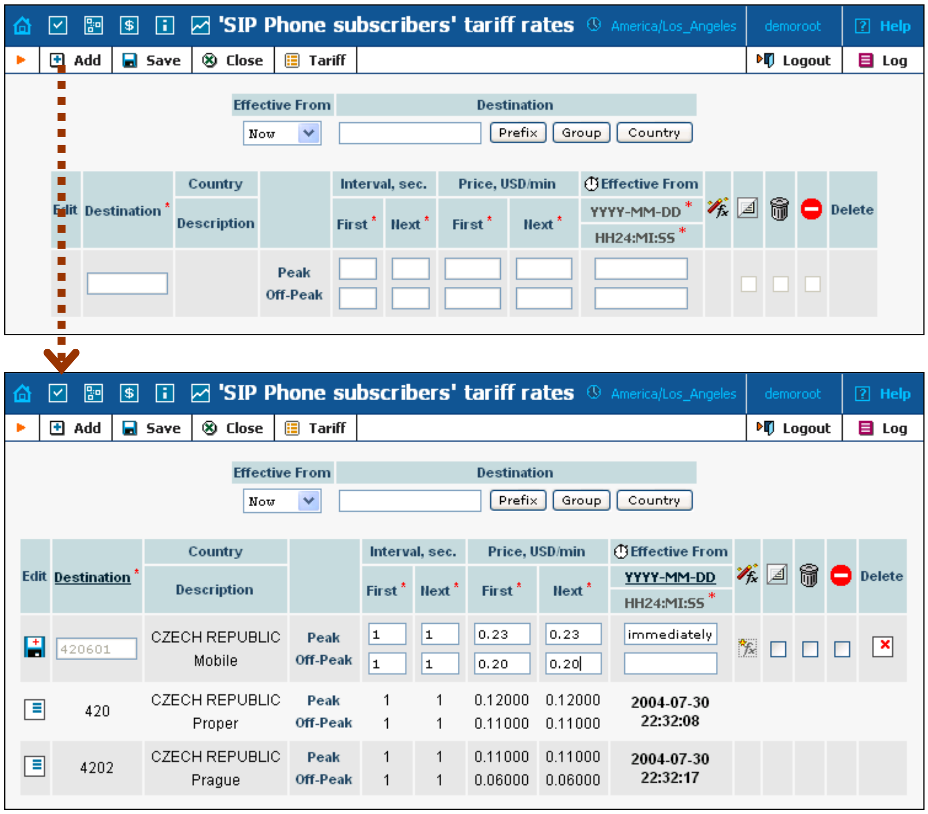
4. Click **Save**.
5. Repeat steps 1-4 until you have entered all of the tariffs. You will need at least two tariffs – one, which you will use to charge your customers, and another, which describes your termination costs. Make sure you choose **Routing** in the **Type** select menu when creating tariffs 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 **Rates** icon next to the name of the tariff. When you are in Tariff Management for a particular tariff, click on **Rates** in the toolbar.
2. On the **Edit Rates** screen, click **Add**.
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 First** – first billing unit in seconds
   - **Interval Next** – next billing unit in seconds
   - **Price First** – per minute price for first interval
   - **Price Next** – per minute price for next interval
   - **Off-peak Interval First**– first billing unit in seconds for off - peak time
   - **Off-peak Interval Next** – next billing unit in seconds for off-peak time
   - **Off-peak Price First** – per minute price for first interval for off-peak time

- o **Off-peak Price Next** – 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.

- o **Formula** – launches the wizard for creating a custom rating formula
- o **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.

- o 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.

Tariffs with routing extensions

These tariffs are created for your vendors. In addition to the billing parameters described above, you can also specify your routing preferences.

Pic_64

- o **Route category** - you can split your available routes into several categories, such as "Cheap", "Very good", etc., then create routing plans for your customers. Use the Default route category for now.
- o **Preference** - routing priority for the specific destination. 10 is the highest priority, 0 is the lowest (i.e. do not use destination for routing at all). For now, you can just set all of your vendor rates at preference 5, and the system will organize available routes according to cost (LCR).
- o **Huntstop** – do not try any routes with a lower preference.

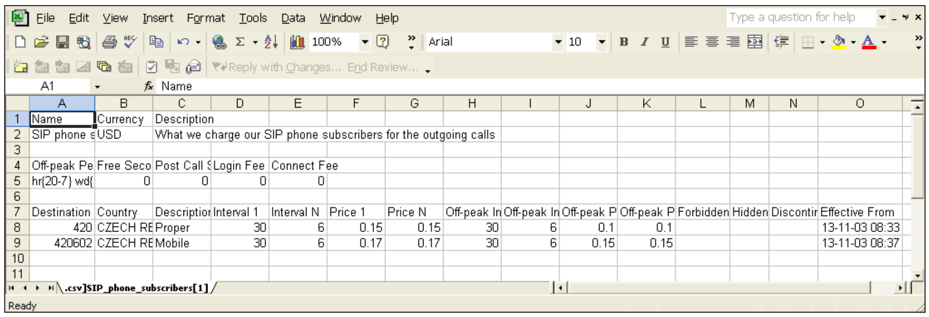
Managing rates offline

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.

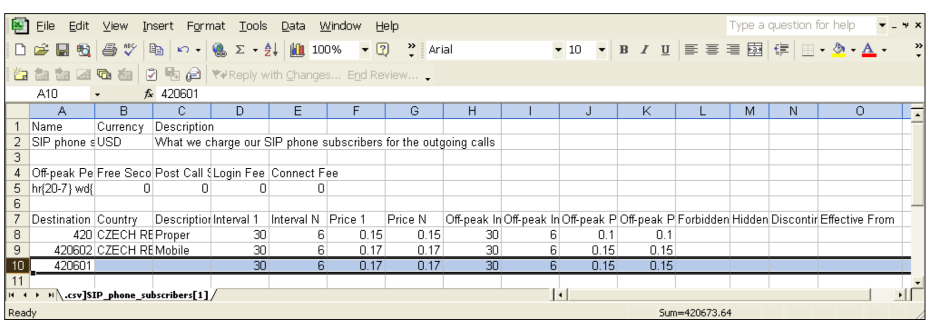
The rates table may be prepared using a spreadsheet processor (i.e. Microsoft Excel) and easily imported into PortaBilling. This is very convenient if you are going to make many changes. For example, you might increase all prices by 10%.

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:



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 **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 **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).

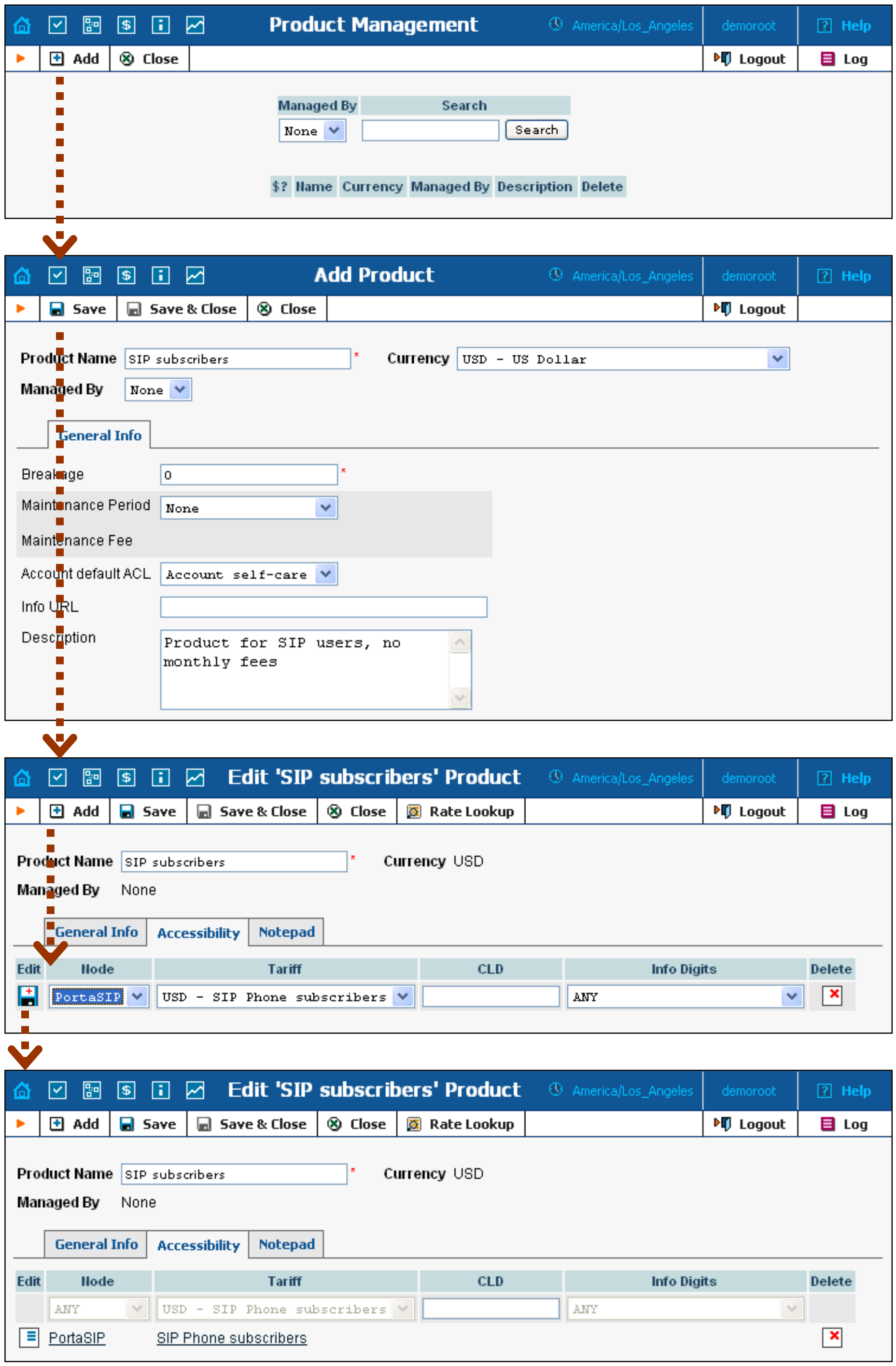
Create All Required Tariffs

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, these tariffs will also include your routing preferences.
- 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 and then return to this step. Make sure that, when creating these subscriber tariffs, you choose the *Managed by NNN* in the **Type** menu, where *NNN* is the name of the corresponding wholesale customer.

Create Product

Accounts for accessing your SIP services will be issued for a specific product. Products are a powerful feature that defines 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** 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 reseller's accounts, so the reseller 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

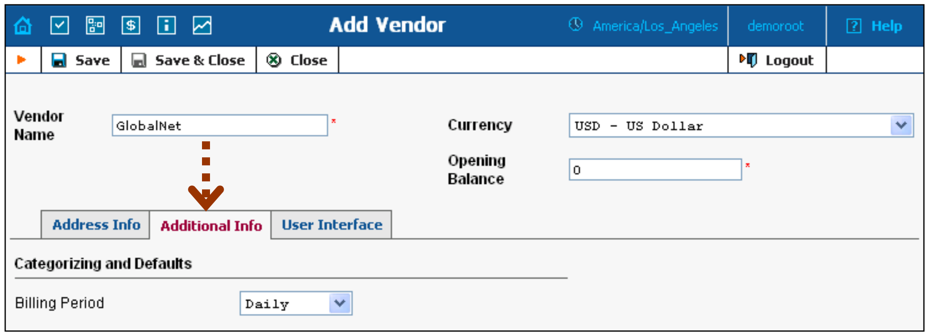
The Accessibility List has two functions: it defines permitted access points (nodes and access numbers) and specifies which tariff should be used for billing in each of these points.

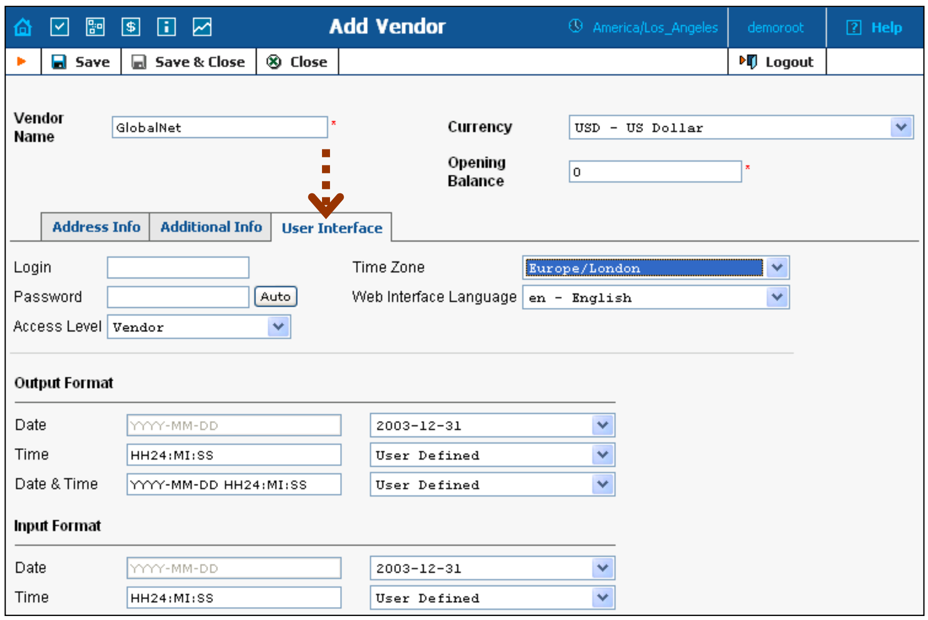
1. When the Accessibility tab is selected, click on the **Add** icon.
2. In the accessibility entry window, select the node where your IVR is running, and choose the appropriate tariff.
3. The **CLD** or **Info-digits** fields only make sense when a call originates from your customer in a public telephony network. Therefore, just leave it empty for the SIP service.
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**.





3. Fill in the **Add 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.

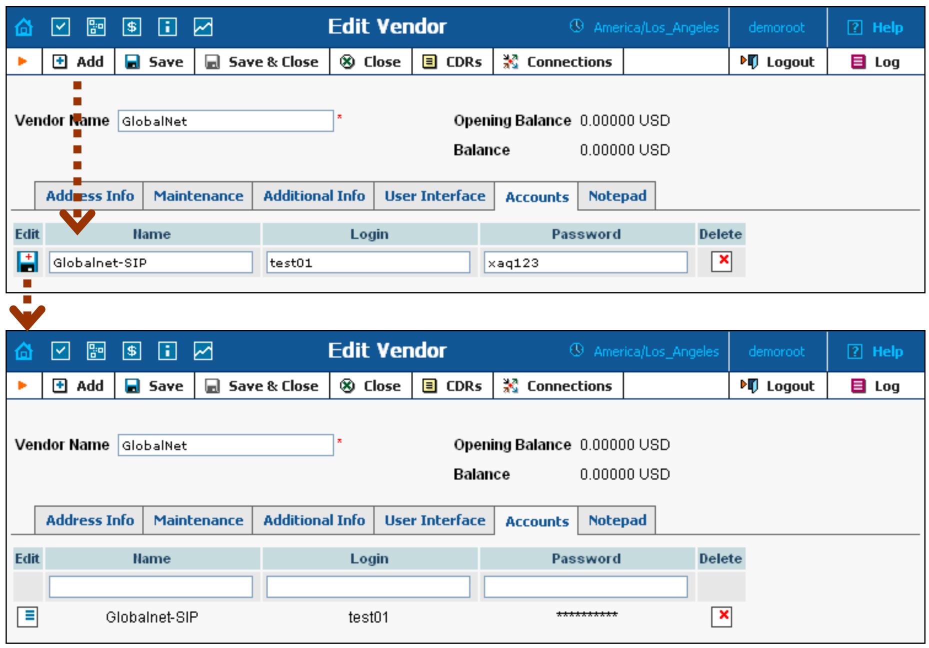
Additional info:

- **Billing period** – split period for vendor statistics.

User Interface:

- **Time zone** – time zone, which vendor uses for his billing period. Statistics will be split between the periods in this time zone.

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**.

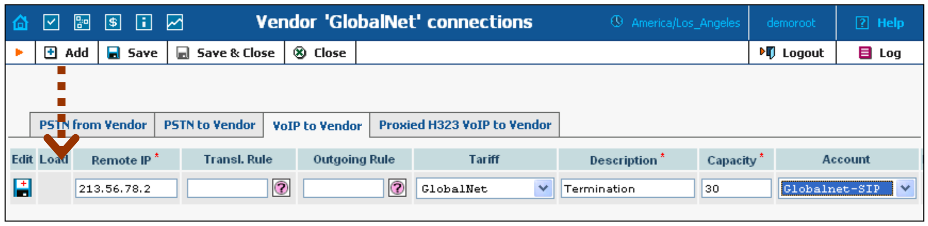


6. Click **Close** in order to return to the **Vendors** admin page.
7. Repeat steps 2-6 to add all of your vendors.

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** 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**.
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.
- **Type** – is it a reseller or retail (direct) customer? (Normally, most of your customers would be retail customers. Only if a customer is reselling your services, and you are providing services and billing to his subscribers, would he be created as a reseller).

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:

- **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 for this customer.
- **Balance Warning Threshold** – the customer can be notified by email when his balance is dangerously close to the credit limit and service will soon be blocked. Here you can enter the value for such a warning threshold. This can be entered:
  - As a percentage (e.g. 90%). The warning will be sent when the customer's balance exceeds that percentage of his credit limit. So, if the credit limit is USD 1000.00 and the threshold is 90%, a warning will be sent as soon as the balance exceeds USD 900.00. This is only applicable when the customer has a positive credit limit.
  - As an absolute value. The warning will be sent as soon as the balance goes above the specified value.

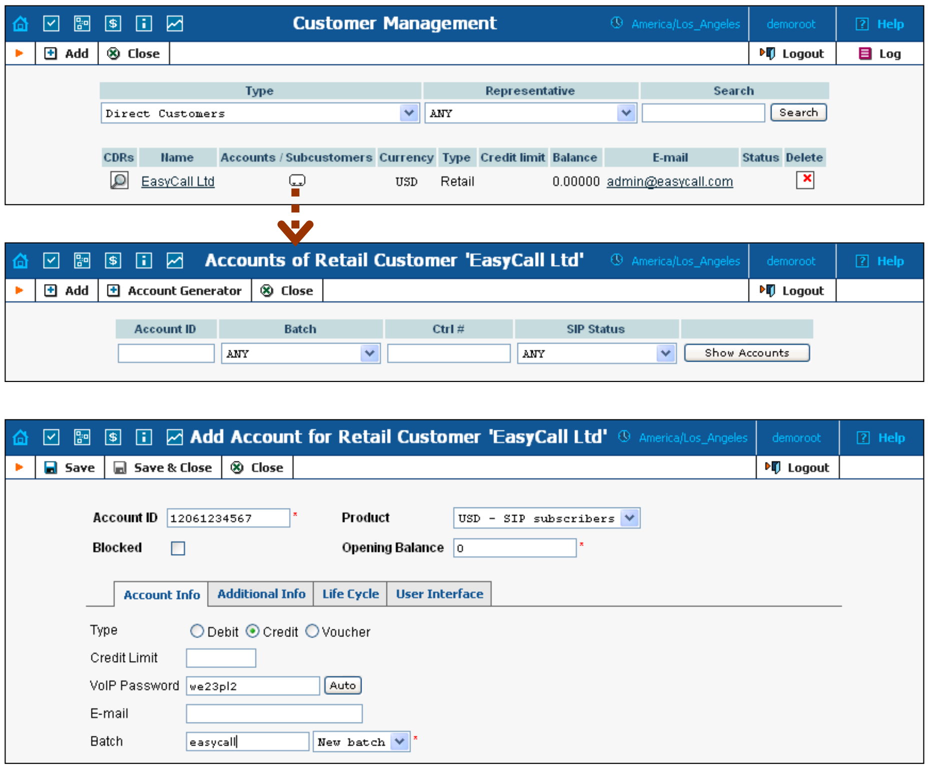
User Interface tab:

- **Time zone** – time zone, in which customer will see his CDRs and also the time zone, which will define his billing period. For example, if you choose **America/New_York** here and the billing period is **Monthly**, it means the billing period will start on the first day of the month, 00:00 New York time.
- **Web Interface Language** – language to be used on the customer self-care web interface.

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 **Add**.
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).
- **VoIP 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.
- **Redirect Number** – redirect number (discussed in the **Advanced features** section); leave this empty since it is not used by PortaSIP.
- **UM Enabled** – check the box if this account has unified messaging (e.g. voicemail) services enabled.
- **Follow Me Enabled** – check the box if this account has "follow me" feature enabled. If yes, account owner can define a list of the numbers where the incoming call to his UA will be redirected (for example his home phone, mobile, …).

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.

User Interface tab:

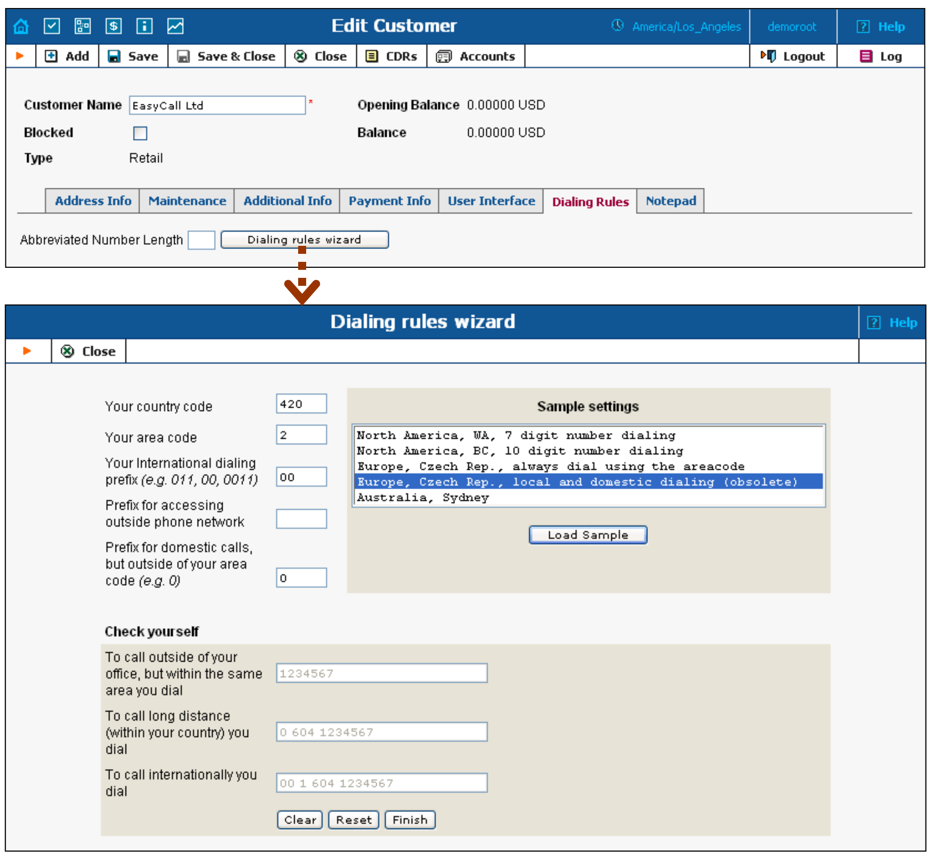
- **Login** – Account login to web self-care pages. Can be the same as account ID.
- **Password** – password for the web self-care pages.
- **Time zone** - When an account owner (pre-paid card user) accesses web self-care pages to see a list of his calls, we can show the time in the time zone most appropriate for him.

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

Set Up Dialing Rules for the Customer (optional)

It could be that your customer wishes to use his custom numbering format. For example, in order to make transition from PSTN/PBX to VoIP as easy as possible, he requires that his users should be able to dial the phone number in exactly the same way as they used to do it on their PBX: 9 for the outside line, then 00 for the international dialing or 0 for domestic, …

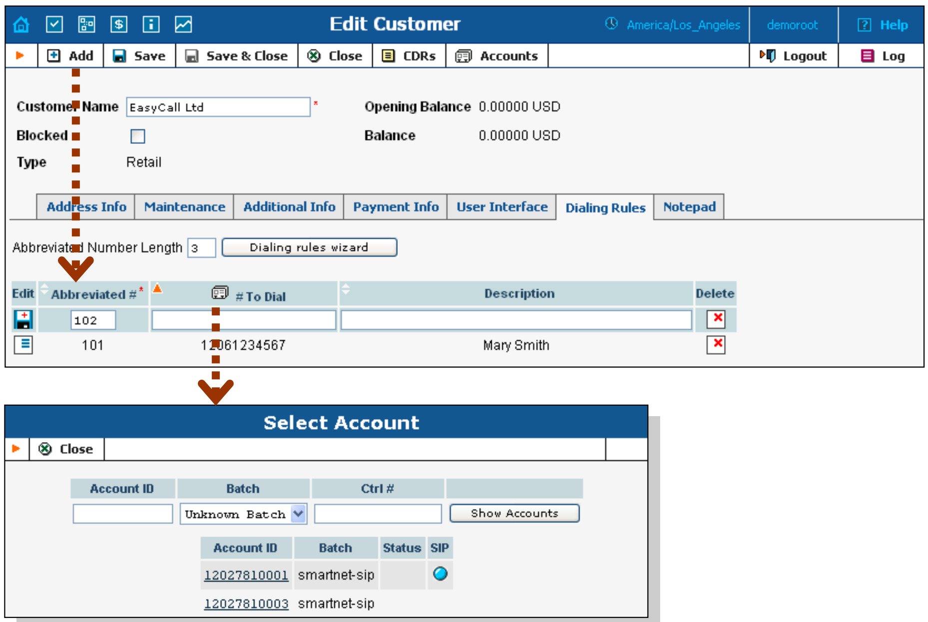
Clearly there is a need for the translation rule, and there is one – customer based translation rule. Moreover, to give the customer ability to manage his translation rule himself without the necessity to learn regular expressions – there is a wizard, which allows to construct the rule by just entering the main parameters, such as international dialing prefix.



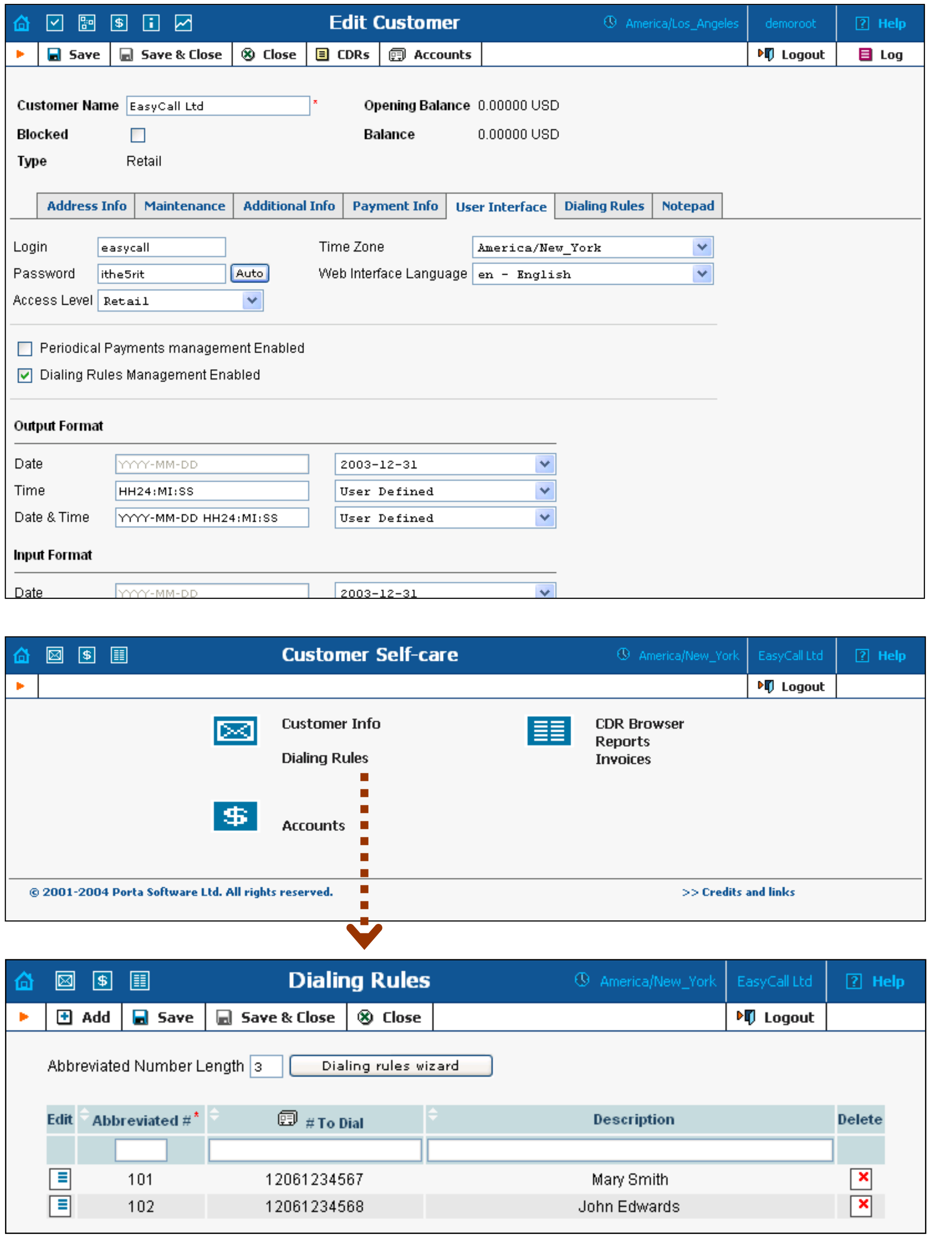
So when one of the accounts of this customer tries to make a call to 90042021234567, the SIP server will send a LOOKUP request to the billing. Billing can apply this customer's translation rule (if defined), or node translation rule (if it

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.



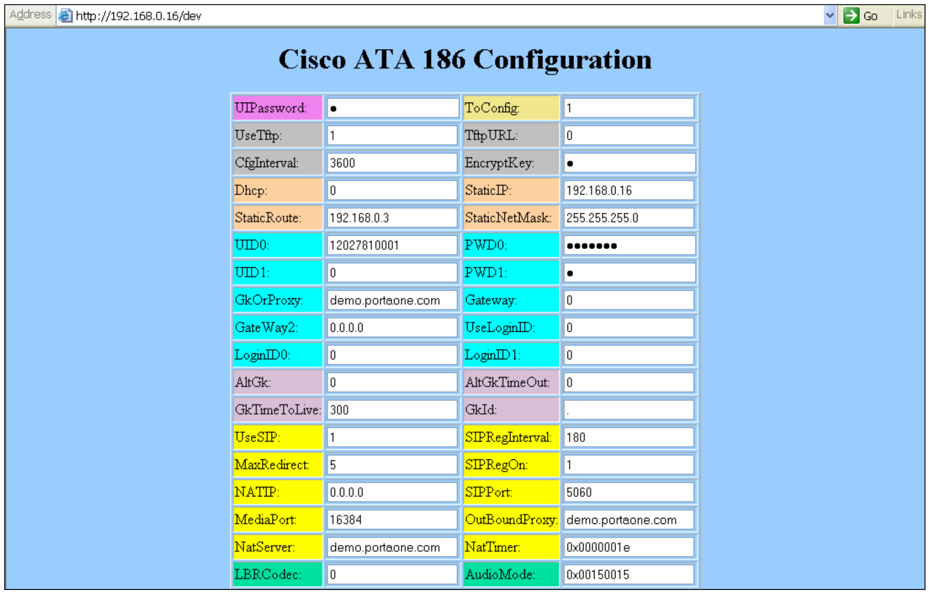
This is much better than programming every phone used in the organization. In addition, the customer himself can manage these dialing rules on his self-care pages, if you allow him.



Configure Cisco ATA Using ATA Expert (optional)

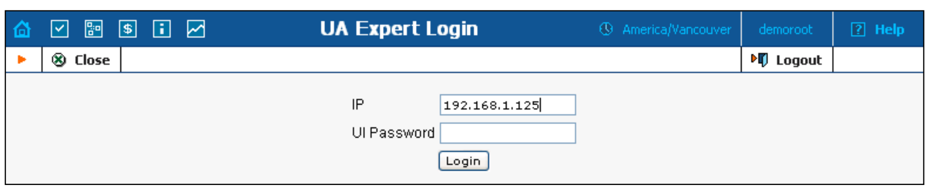
Cisco ATA could be configured from the web interface accessible at **http://<ata-IP-address>/dev**. However, this web interface is designed to be used by experts, and parameter values must be entered in the protocol-specific format (e.g. `0x00150015`). You may find more information at:
http://www.cisco.com/en/US/products/hw/gatecont/ps514/products_configuration_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

However, this complicated way of entering the parameters makes it virtually impossible for end-users to employ.



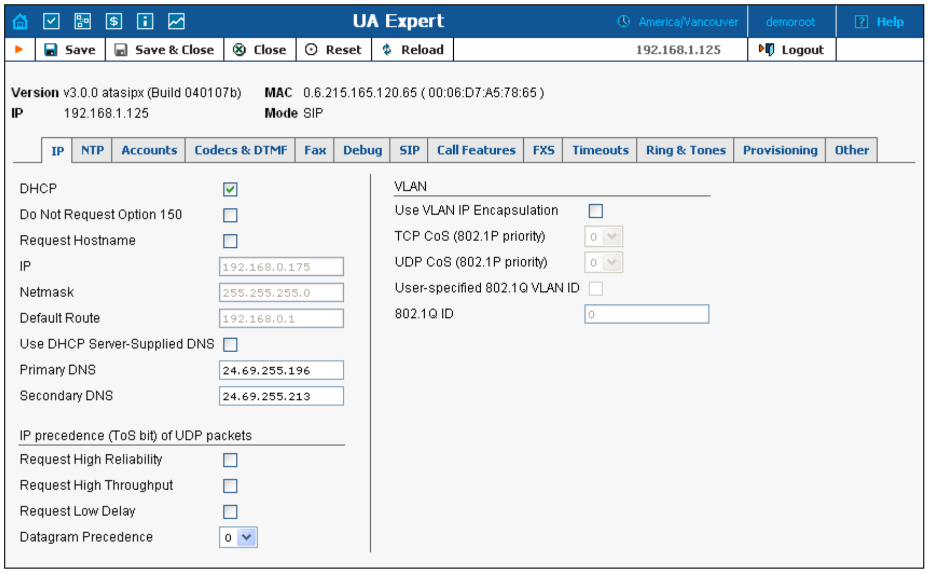
Fortunately, PortaBilling provides a safe and user-friendly way to configure your Cisco ATA from the web interface via **ATA Expert**:

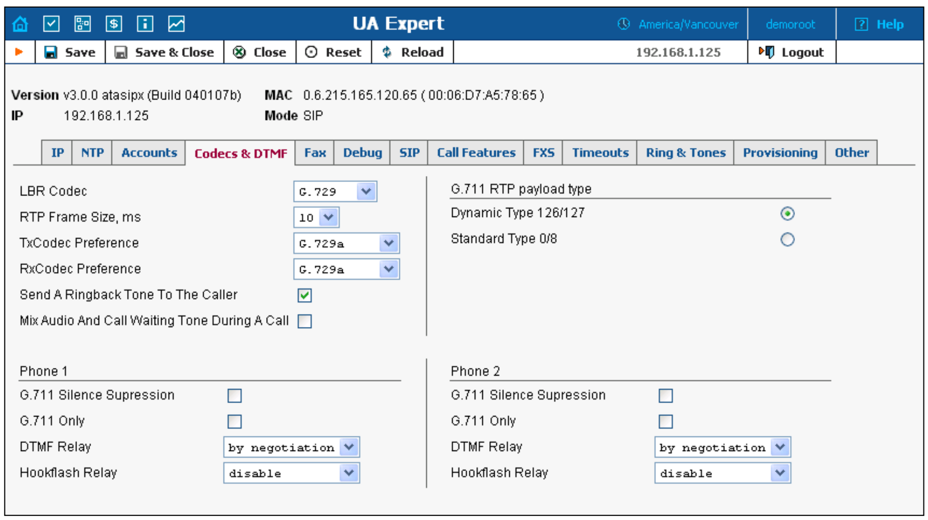
1. In the Management section of the Admin-Index page, choose **UA Expert.**
2. Type in your Cisco ATA IP address, as well as the administrator's password if you have set up one.



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.

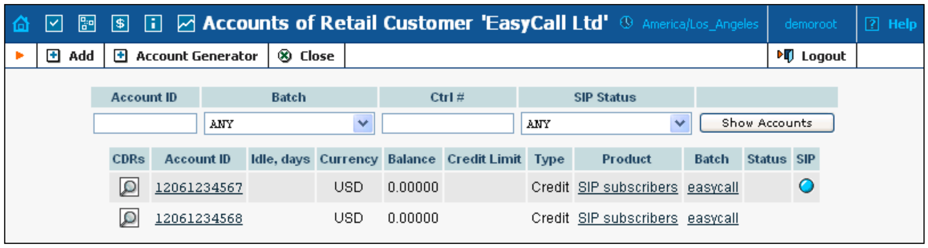




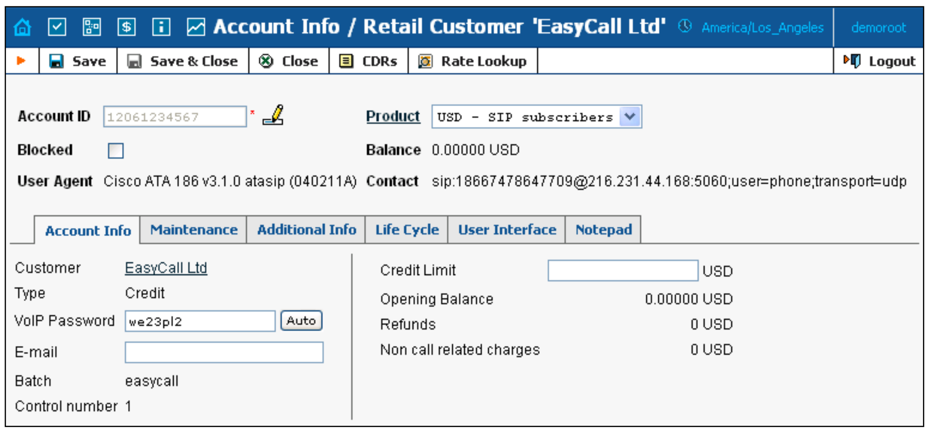
4. Press **Save** to save the new configuration to the ATA.

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). Some of the SIP request parameters have been removed for greater clarity.

SIP user agent attempt to make a call via the SIP server

```
11 Mar 07:50:09/GLOBAL/ser:RECEIVED message from 195.234.212.178:50535:
INVITE sip:380675028490@216.232.84.32 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.100:5060;branch=z9hG4bK-2ace6057
From: sipura <sip:160452152773@216.232.84.32>;tag=6ab7b6a32482b910
To: <sip:380675028490@216.232.84.32>
Call-ID: d454db7-1b62048f@192.168.0.100
CSeq: 101 INVITE
Max-Forwards: 70
Contact: sipura <sip:160452152773@192.168.0.100:5060>
Expires: 240
User-Agent: Sipura/SPA2000-1.0.31
Content-Length: 396
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Content-Type: application/sdp

v=0
o=- 7259 7259 IN IP4 192.168.0.100
s=-
c=IN IP4 192.168.0.100
t=0 0
m=audio 16416 RTP/AVP 18 0 8 96 2 97 98 101 100
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
```

```
a=rtpmap:2 G726-32/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:100 NSE/8000
a=ptime:20
a=sendrecv

11 Mar 07:50:09/d454db7-1b62048f@192.168.0.100/ser: processing INVITE
received from 195.234.212.178
11 Mar 07:50:09/d454db7-1b62048f@192.168.0.100/ser: no auth info or
auth failure, sending challenge
```

SIP proxy does not allow calls without authentication, UA is given a challenge

```
11 Mar 07:50:09/GLOBAL/ser: SENT message to 195.234.212.178:50535:
SIP/2.0 401 Unauthorized
…
WWW-Authenticate: Digest realm="216.232.84.32",
nonce="40508bdd65e380a6eeb900b13e2f2a52a4ef7a19"
…
```

UA re-sends the INVITE request with the challenge response

```
11 Mar 07:50:10/GLOBAL/ser:RECEIVED message from 195.234.212.178:50535:
INVITE sip:380675028490@216.232.84.32 SIP/2.0
…
Authorization: Digest
username="160452152773",realm="216.232.84.32",nonce="40508bdd65e380a6ee
b900b13e2f2a52a4ef7a19",uri="sip:380675028490@216.232.84.32",algorithm=
MD5,response="a9f89be52a1d8bb033f2e19174ee1962"
…
```

SIP server sends LOOKUP request to RADIUS (billing)

```
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/ser: processing INVITE
received from 195.234.212.178
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/ser: sending AAA request
for 160452152773, method LOOKUP:
Called-Station-Id                = '380675028490'
Calling-Station-Id               = '160452152773'
User-Name                        = '160452152773'
Digest-User-Name                 = '160452152773'
Digest-Realm                     = '216.232.84.32'
Digest-Nonce                     = '40508bdd65e380a6eeb900b13e2f2a52a'
Digest-URI                       = 'sip:380675028490@216.232.84.32'
Digest-Method                    = 'LOOKUP'
Digest-Algorithm                 = 'MD5'
Digest-Response                  = 'a9f89be52a1d8bb033f2e19174ee1962'
Service-Type                     = 'Sip-Session'
Cisco-AVPair                     = 'call-id=d454db7-1b62048f@192.168.'
```

RADIUS responds with the authorization confirmation

```
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/ser: AAA request
accepted, processing response:
h323-billing-model               = 'h323-billing-model=0'
Cisco-AVPair                     = 'h323-ivr-in=Tariff:Apollo SIP'
Cisco-AVPair                     = 'h323-ivr-in=PortaBilling_Complete'
h323-return-code                 = 'h323-return-code=13'
h323-currency                    = 'h323-currency=USD'
h323-preferred-lang              = 'h323-preferred-lang=en'
```

SIP server performs abbreviated number expansion

```
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/ser: got complete number
380675028490, rewriting URI
```

SIP proxy forwards INVITE to the B2BUA

```
11 Mar 07:50:10/GLOBAL/ser: SENT message to 216.232.84.32:5061:
INVITE sip:380675028490@216.232.84.32:5061 SIP/2.0
…
```

B2BUA receives the request

```
11 Mar 07:50:10/GLOBAL/b2bua: RECEIVED message from 216.232.84.32:5060:
INVITE sip:380675028490@216.232.84.32:5061 SIP/2.0
…
```

B2BUA creates a new answer call leg according to parameters, specified in the request

```
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: session received
from 195.234.212.178:50535, creating a new answering call leg
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: setting UID
(160452152773), realm (216.232.84.32), nonce
(40508bdd65e380a6eeb900b13e2f2a52a4ef7a19), method (INVITE), uri
(sip:380675028490@216.232.84.32), algorithm (MD5), response
(a9f89be52a1d8bb033f2e19174ee1962) from Authorization header
```

B2BUA prepares INVITE request and sends a provisional reply to SIP proxy

```
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: requesting
billing-assisted routing
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: sending AAA
request
11 Mar 07:50:10/GLOBAL/b2bua: SENT message to 216.232.84.32:5060:
SIP/2.0 100 Trying
…
```

SIP proxy receives the provisional reply

```
11 Mar 07:50:10/GLOBAL/ser: RECEIVED message from 216.232.84.32:5061:
SIP/2.0 100 Trying
…
```

B2BUA sends INVITE request to the RADIUS server (billing)

```
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: sending AAA
request:
User-Name                 = '160452152773'
Digest-Attributes         = 'Realm (1) = "216.232.84.32"'
Digest-Attributes         = 'Nonce (2) =
"40508bdd65e380a6eeb900b13e2f2a52a4ef7a19"'
Digest-Attributes         = 'Method (3) = "INVITE"'
Digest-Attributes         = 'URI (4) = "sip:380675028490@216.232.84.32"'
Digest-Attributes         = 'Algorithm (6) = "MD5"'
Digest-Attributes         = 'User-Name (10) = "160452152773"'
Digest-Response           = 'a9f89be52a1d8bb033f2e19174ee1962'
call-id                   = 'd454db7-1b62048f@192.168.0.100'
h323-remote-address       = '195.234.212.178'
Cisco-AVPair              = 'h323-session-protocol=sipv2'
h323-conf-id              = '50CAFBC4 0104E630 F0EA8D14 BEB08934'
h323-ivr-out              = 'PortaBilling_Routing:SIP'
Called-Station-Id         = '380675028490'
NAS-IP-Address            = '216.232.84.32'
Calling-Station-Id        = '160452152773'
```

RADIUS server replies with the authorization confirmation

```
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: AAA request
accepted, processing response:
h323-ivr-in                =
'PortaBilling_Routing:380675028490@76.104.130.201;auth=DDBB088F1316608E
1DEE58CEB70591BF7C352221D6945914;expires=300;credit-time=-1'
h323-billing-model         = '0'
h323-ivr-in=               = 'Tariff:Apollo SIP'
h323-ivr-in                = 'PortaBilling_CompleteNumber:380675028490'
h323-return-code           = '13'
h323-currency              = 'USD'
h323-preferred-lang        = 'en'
```

Now B2BUA has call routing and maximum call duration parameters, so it can place an outgoing call

```
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: authorization
accepted
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: call duration is
unlimited
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: got route:
380675028490@76.104.130.201;auth=DDBB088F1316608E1DEE58CEB70591BF7C3522
21D6945914;expires=300;credit-time=-1
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: got username from
route: X-Telecom, password: abc306k
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: got no answer
timeout from route 300 seconds
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: got maximum call
duration from route -1 seconds
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: placing outgoing
session to sip:380675028490@76.104.130.201;user=phone, h323-conf-
id=50CAFBC4 0104E630 F0EA8D14 BEB08934
11 Mar 07:50:10/d454db7-1b62048f@192.168.0.100/b2bua: no answer timeout
is 300 seconds
11 Mar 07:50:10/GLOBAL/b2bua: SENT message to 76.104.130.201:5060:
INVITE sip:380675028490@76.104.130.201;user=phone SIP/2.0
…
```

Vendor's SIP server sends a challenge, so the call must be authorized

```
11 Mar 07:50:11/GLOBAL/b2bua: RECEIVED message from
76.104.130.201:5060:
SIP/2.0 401 Unauthorized
….
```

B2BUA acknowledges this

```
11 Mar 07:50:11/GLOBAL/b2bua: SENT message to 76.104.130.201:5060:
ACK sip:380675028490@76.104.130.201;user=phone SIP/2.0
…
```

And sends another INVITE request, this time with the challenge response included

```
11 Mar 07:50:11/GLOBAL/b2bua: SENT message to 76.104.130.201:5060:
INVITE sip:380675028490@76.104.130.201;user=phone SIP/2.0
…
Authorization: Digest
nonce="40508d5eab94d36a5b370aa5297f1aa1f9f0481a",realm="216.232.84.32",
response="9cc4a553fa9f59ef8ab9a9393f3f27ca",uri="sip:380675028490@76.10
4.130.201;user=phone",username="PortaSoftware"
…
```

Vendor's gateway sends another provisional response, informing that the in-band alerting (usually a ring-back tone) is available

```
11 Mar 07:50:11/GLOBAL/b2bua: RECEIVED message from
76.104.130.201:5060:
SIP/2.0 100 trying -- your call is important to us
…

11 Mar 07:50:16/GLOBAL/b2bua: RECEIVED message from
76.104.130.201:5060:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 216.232.84.32:5061
To: <sip:380675028490@216.232.84.32>;tag=58c10677
From: sipura
<sip:160452152773@216.232.84.32:5061>;tag=c1f453229c5032dbb27baaf94a773
2b9
Call-ID: d454db7-1b62048f@192.168.0.100
CSeq: 2 INVITE
Record-Route: <sip:380675028490@76.104.130.201;lr>
Content-Type: application/sdp
Content-Length: 264

v=0
o=CiscoSystemsSIP-GW-UserAgent 4063 2930 IN IP4 212.119.160.51
s=SIP Call
c=IN IP4 212.119.160.51
t=0 0
m=audio 16948 RTP/AVP 18 101
c=IN IP4 212.119.160.51
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-15
```

B2BUA sends provisional response down to the SER

```
11 Mar 07:50:16/GLOBAL/b2bua: SENT message to 216.232.84.32:5060:
SIP/2.0 183 Session Progress
…
```

SER receives provisional response from the B2BUA

```
11 Mar 07:50:16/GLOBAL/ser: RECEIVED message from 216.232.84.32:5061:
SIP/2.0 183 Session Progress
…
```

SER sends provisional response to the UA

```
11 Mar 07:50:16/GLOBAL/ser: SENT message to 195.234.212.178:50535:
SIP/2.0 183 Session Progress
…
```

Vendor's gateway informs that audio session has been established by sending final positive response

```
11 Mar 07:50:25/GLOBAL/b2bua: RECEIVED message from
76.104.130.201:5060:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 216.232.84.32:5061
To: <sip:380675028490@216.232.84.32>;tag=58c10677
From: sipura
<sip:160452152773@216.232.84.32:5061>;tag=c1f453229c5032dbb27baaf94a773
2b9
Call-ID: d454db7-1b62048f@192.168.0.100
CSeq: 2 INVITE
```

```
Record-Route: <sip:380675028490@76.104.130.201;lr>
Contact: <sip:380675028490@76.104.130.201:5061>
Content-Type: application/sdp
Content-Length: 264

v=0
o=CiscoSystemsSIP-GW-UserAgent 4063 2930 IN IP4 212.119.160.51
s=SIP Call
c=IN IP4 212.119.160.51
t=0 0
m=audio 16948 RTP/AVP 18 101
c=IN IP4 212.119.160.51
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-15
```

B2BUA registers outgoing call leg as connected and sends the final positive response to SIP proxy

```
11 Mar 07:50:25/d454db7-1b62048f@192.168.0.100/b2bua: outgoing session
started
11 Mar 07:50:25/GLOBAL/b2bua: SENT message to 216.232.84.32:5060:
SIP/2.0 200 OK
…
```

SIP proxy receives this message and transmits it to the UA

```
11 Mar 07:50:25/GLOBAL/ser: RECEIVED message from 216.232.84.32:5061:
SIP/2.0 200 OK
…

11 Mar 07:50:25/GLOBAL/ser: SENT message to 195.234.212.178:50535:
SIP/2.0 200 OK
…
```

The call is now in progress

```
11 Mar 07:50:26/GLOBAL/ser:RECEIVED message from 195.234.212.178:50535:
ACK sip:380675028490@216.232.84.32:5061 SIP/2.0
…

11 Mar 07:50:26/GLOBAL/b2bua: RECEIVED message from 216.232.84.32:5060:
ACK sip:380675028490@216.232.84.32:5061 SIP/2.0
…
11 Mar 07:50:26/GLOBAL/ser: SENT message to 216.232.84.32:5061:
ACK sip:380675028490@216.232.84.32:5061 SIP/2.0
…
11 Mar 07:50:26/GLOBAL/b2bua: SENT message to 76.104.130.201:5060:
ACK sip:380675028490@76.104.130.201;lr;lr SIP/2.0
…
```

SIP proxy receives BYE message sent by the UA which indicates that the caller has hung up

```
11 Mar 07:50:36/GLOBAL/ser: RECEIVED message from
195.234.212.178:50535:
BYE sip:380675028490@216.232.84.32:5061 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.100:5060;branch=z9hG4bK-99c761cb
From: sipura <sip:160452152773@216.232.84.32>;tag=6ab7b6a32482b910
To: <sip:380675028490@216.232.84.32>;tag=0a11f5a9
Call-ID: d454db7-1b62048f@192.168.0.100
CSeq: 103 BYE
Max-Forwards: 70
Route: <sip:380675028490@216.232.84.32;lr>
```

```
Authorization: Digest
username="160452152773",realm="216.232.84.32",nonce="40508bdd65e380a6ee
b900b13e2f2a52a4ef7a19",uri="sip:380675028490@216.232.84.32",algorithm=
MD5,response="333e7c8556e53af7450ad2c8fd850c65"
User-Agent: Sipura/SPA2000-1.0.31
Content-Length: 0
```

SIP proxy sends BYE to the B2BUA

```
11 Mar 07:50:36/GLOBAL/ser: SENT message to 216.232.84.32:5061:
BYE sip:380675028490@216.232.84.32:5061 SIP/2.0
…
```

B2BUA receives BYE

```
11 Mar 07:50:36/GLOBAL/b2bua: RECEIVED message from 216.232.84.32:5060:
BYE sip:380675028490@216.232.84.32:5061 SIP/2.0
…
```

B2BUA terminates both call legs and sends accounting to the billing

```
11 Mar 07:50:36/d454db7-1b62048f@192.168.0.100/b2bua: outgoing session
ended successfuly
11 Mar 07:50:36/d454db7-1b62048f@192.168.0.100/b2bua: sending Acct Stop
(Answering)
11 Mar 07:50:36/d454db7-1b62048f@192.168.0.100/b2bua: sending Acct Stop
(Originate)
11 Mar 07:50:36/d454db7-1b62048f@192.168.0.100/b2bua: session duration
is 11
```

B2BUA confirms to SIP proxy that it received BYE

```
11 Mar 07:50:36/GLOBAL/b2bua: SENT message to 216.232.84.32:5060:
SIP/2.0 200 OK
…

11 Mar 07:50:36/GLOBAL/ser: RECEIVED message from 216.232.84.32:5061:
SIP/2.0 200 OK
…

11 Mar 07:50:36/GLOBAL/b2bua: SENT message to 76.104.130.201:5060:
BYE sip:380675028490@76.104.130.201;lr;lr SIP/2.0
…
```

B2BUA sends accounting about the answer call leg to the billing

```
11 Mar 07:50:36/d454db7-1b62048f@192.168.0.100/b2bua: sending AAA
request:
User-Name                = '160452152773'
NAS-IP-Address           = '216.232.84.32'
Calling-Station-Id       = '160452152773'
Acct-Status-Type         = 'Stop (2)'
Called-Station-Id        = '380675028490'
Acct-Session-Id          = 'd454db7-1b62048f@192.168.0.100'
Acct-Session-Time        = '11'
Acct-Delay-Time          = '15'
call-id                  = 'd454db7-1b62048f@192.168.0.100'
h323-setup-time          = '15:50:10 GMT Thu Mar 11 2004'
h323-connect-time        = '15:50:25 GMT Thu Mar 11 2004'
h323-disconnect-time     = '15:50:36 GMT Thu Mar 11 2004'
h323-voice-quality       = '0'
h323-remote-address      = '195.234.212.178'
h323-call-type           = 'VoIP'
```