PORTA ONE

Porta✓Switch™

PortaSwitch Handbook: SIP Services

Maintenance Release 19

Part I

www.portaone.com

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PortaSwitch Handbook: SIP Services, May 2009
V.1.19.2

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Table of Contents

Preface

This document provides PortaSIP (PortaSwitch) 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 PortaSwitch 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 in-between minor releases. The online copy of this guide is always up-to-date, integrating the latest changes to the product. You can access the latest copy of this guide at: **http://portaone.com/support/documentation/**

Conventions

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

Caution indicates that the described action might result in program malfunction or data loss.

NOTE: Notes contain helpful suggestions about or references to materials not contained in this manual.

Timesaver means that you can save time by performing the action described in the paragraph.

Tips provide information that might help you solve a problem.

1. Setting up Standard SIP Services

This chapter will demonstrate how to set up a basic SIP service so that your customers will be able to make outgoing calls via your SIP server.

Please refer to the **PortaBilling Web Reference Guide** PDF file for detailed instructions on how to navigate and operate the web interface, as well as detailed explanations of particular fields.

Basic SIP Service

Checklist

Print the following page and use it to check off the operations you have completed while performing system setup according to the instructions in this chapter. Please be sure to perform all of the operations (all of the boxes must be checked), otherwise the service will not work.

Operation	Done
General configuration	
Enter company data under Company Info and specify a base currency.	[ ]
For any other currency you plan to use, specify the exchange rate source and define exchange rates	[ ]
Create all the required destinations for off-net calls	[ ]
Create a destination for your SIP numbers	[ ]
Network configuration	
Create a node for your PortaSIP	[ ]
Rating configuration	
Create a tariff A, which will be applied to SIP subscribers	[ ]
Enter rates in tariff A for the destinations you plan to call (both off-net and SIP-to-SIP calls)	[ ]
Create a tariff B, which describes your termination costs and routing for off-net calls (make sure it has a **Routing** type!)	[ ]
Enter rates in tariff B for the destinations you plan to call	[ ]
Create a tariff C, which describes your termination costs for SIP-to-SIP calls	[ ]
Enter rates in tariff C for SIP destinations (zero cost)	[ ]
Create your SIP product	[ ]
Create one accessibility entry for this product, using the PortaSIP node and tariff A	[ ]
Create an off-net calls vendor	[ ]
Create a connection for this vendor using tariff B	[ ]
Create a fake SIP-to-SIP vendor	[ ]
Create a "VoIP to vendor" connection with the remote IP field containing the **SIP-UA** string for this vendor using tariff C	[ ]
Account provisioning	
Create a customer class for your SIP customers	[ ]
Create a retail customer who will use the SIP service	[ ]
Create several accounts for this customer, with account ID identical to the SIP phone number	[ ]
Testing	
Program the parameters (phone #, password, SIP server address) into the SIP phone and make a test call	[ ]

Initial Configuration of PortaSwitch

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 on the **Report Info** tab.
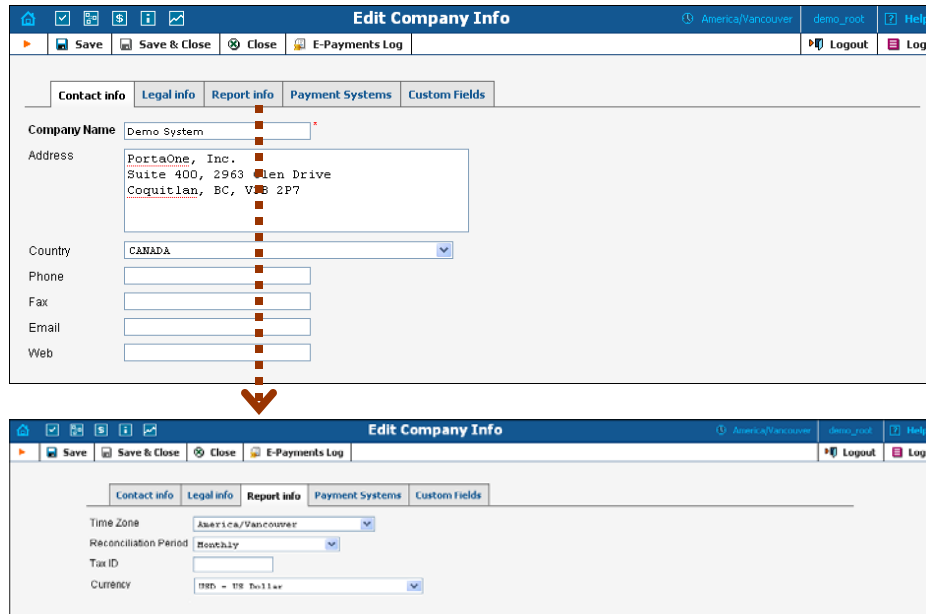

Naturally, this does not limit your operations to this currency only. However, on cost/revenue reports and the like different currencies will be converted to the one you specify here.

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 setup. Make sure you are able to login as the newly-created user, and change the password for the pb-root user.

If you plan to do billing in multiple currencies, define these in the **Currencies** section and specify exchange rates in **Exchange Rates.**
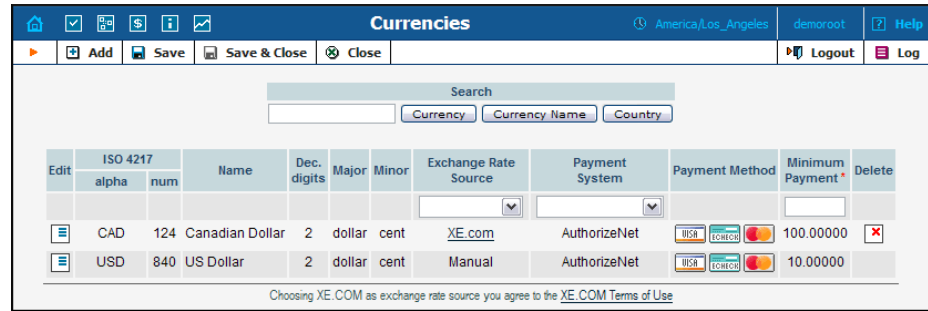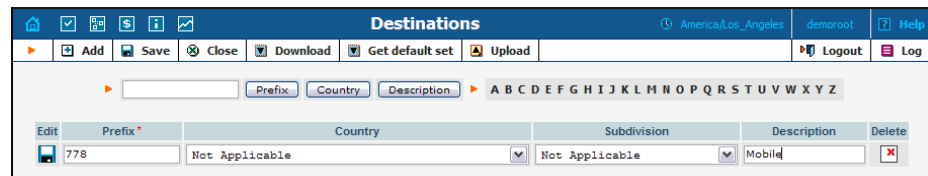

Create Destinations

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.
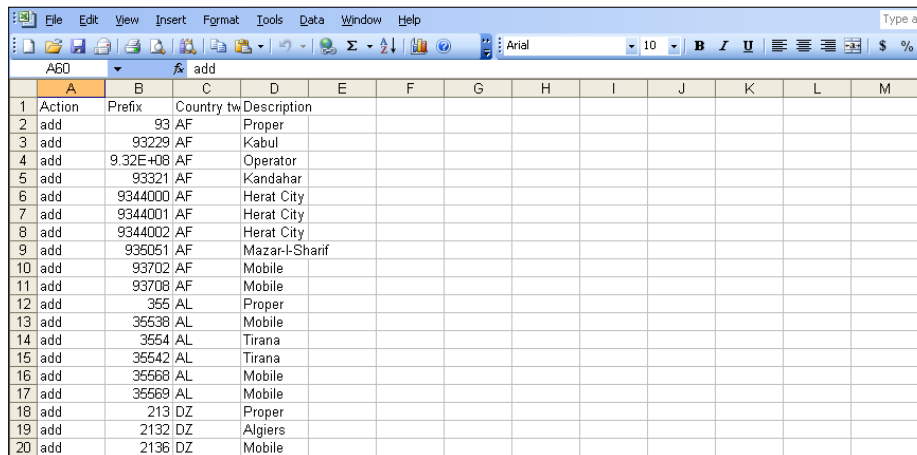
Creating destinations "one-by-one":

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.
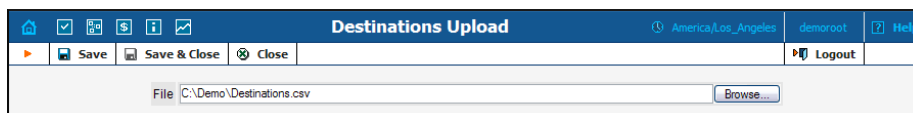
Uploading a set of destinations from a file:

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	E	F	G	H	I	J	K	L	M
1	Action	Prefix	Country tw	Description									
2	add	93	AF	Proper									
3	add	93229	AF	Kabul									
4	add	9.32E+08	AF	Operator									
5	add	93321	AF	Kandahar									
6	add	9344000	AF	Herat City									
7	add	9344001	AF	Herat City									
8	add	9344002	AF	Herat City									
9	add	935051	AF	Mazar-I-Sharif									
10	add	93702	AF	Mobile									
11	add	93708	AF	Mobile									
12	add	355	AL	Proper									
13	add	35538	AL	Mobile									
14	add	3554	AL	Tirana									
15	add	35542	AL	Tirana									
16	add	35568	AL	Mobile									
17	add	35569	AL	Mobile									
18	add	213	DZ	Proper									
19	add	2132	DZ	Algiers									
20	add	2136	DZ	Mobile									

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.

Destinations Upload	America/Los_Angeles	demoroot	? Help
▶ 💾 Save 💾 Save & Close ⊗ Close			▶🔲 Logout
File C:\Demo\Destinations.csv		Browse…	

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

If you plan to use PortaSwitch's Voice-On-Net rating feature, which enables rating of all calls between SIP phones in your system according to one predefined rate, you need to create a special **VOICEONNET** destination. (Please make sure this feature is enabled in your system.) A similar situation is with the Voice-VPN feature, when a special rating is applied for all calls going between accounts of the same customer (extensions within the same IP Centrex context) – you will need to create a **VOICEVPN** destination.

Another option would be to create one or more phone prefixes covering the actual phone numbers assigned to your SIP phones, as described below.
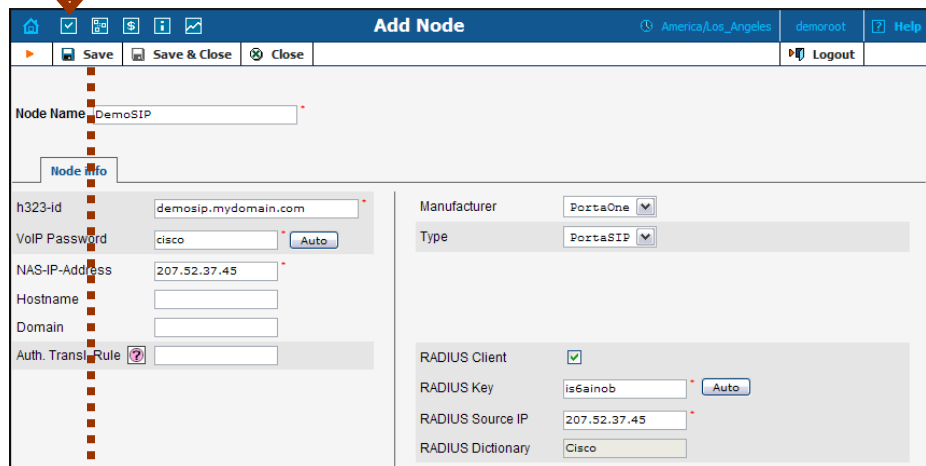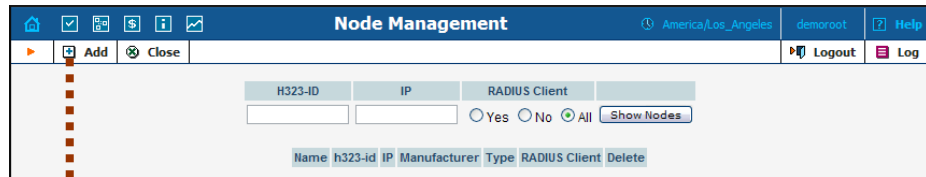
In order to receive an incoming call, an SIP user agent must be configured with a phone number. Normally, you will obtain a range of phone numbers from your local telecom, and you will be able to assign these to your customers. For example, you will be assigned range 12027810000 – 12027819999. It is, 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.
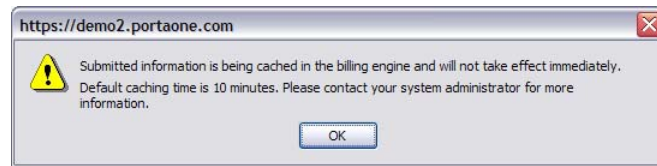
Even if you have not obtained an official phone prefix, it is highly recommended not to assign IDs to your SIP user agents at random. Choose a non-existing prefix, e.g. **777**, and create it as the destination with **N/A** country and the description **SIP phones**. Then use SIP IDs such as **77700001, 7770002, … 7770999, …**
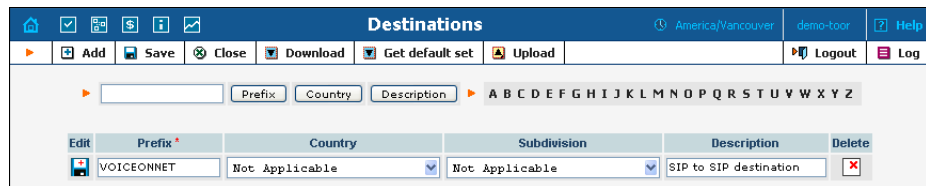
Create Nodes

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.

https://demo2.portaone.com

⚠ Submitted information is being cached in the billing engine and will not take effect immediately. Default caching time is 10 minutes. Please contact your system administrator for more information.

[OK]

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:
    o **Node name** – a short descriptive name for your SIP server (this will be used in the select menus).
    o **H323-ID** – PortaSIP server's hostname (recommended: `hostname.domainname`).
    o **VoIP 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.
    o **NAS-IP-Address** – the IP address of the SIP server.
    o **Auth. Translation rule** – leave this blank (you can use customer-based translation rules later to allow your customers to dial a number in their own numbering format).
    o **Manufacturer** – select **PortaOne**.
    o **Type** – VoIP node type; select **PortaSIP**.
    o **Radius Client** – check this, since PortaSIP will need to communicate with the billing.
    o **Radius Key** – enter the radius shared secret here; this must be the same **key** which you entered during the PortaSIP installation.
    o **Radius Source IP** – see the **Node ID, NAS IP address and Radius source IP** section in the **PortaBilling Administrator 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**.

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:
   o **Name** – A short name for the tariff object; this is the name you will then see in the select menus.
   o **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.

- o **Applied To** – Designates whether this tariff will be used to charge your customers (in this case, choose a **Customer**) or to calculate costs associated with your vendors (in this case, choose a **Vendor** and also turn on the **Routing** check-box, which will appear below).
- o **Managed By** – Choose "**Administrator Only**" here (this option is only visible after you select **Applied to: Customer** above).
- o **Service Type** – Choose **Voice Calls** here.
- o **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.
- o **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.
- o **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.
- o **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).
- o **Post-Call Surcharge** – Percentage of the amount charged for the call.
- o **Login Fee** – Amount to be charged immediately after the first user authentication (i.e. after the user enters his PIN).
- o **Connect Fee** – Amount to be charged for each connected call (call with a non-zero duration).
- o **Round Charged Amount** – Instead of calculating xDRs with a 5-decimal-place precision, round up xDR 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.
- o **Default 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.
- o **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.
- o **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
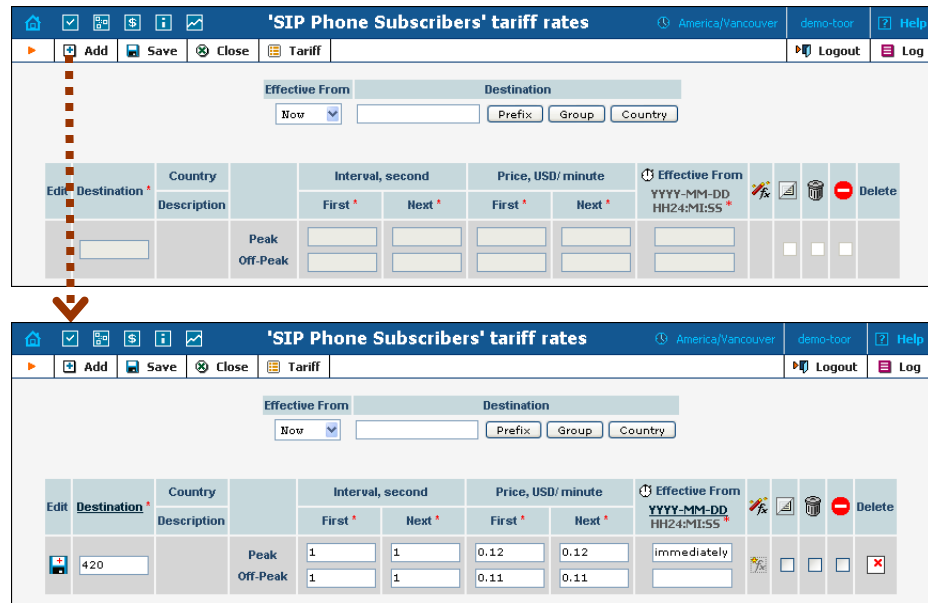
choose **Vendor** in the **Applied To** select menu and check the check-box **Routing** when creating tariffs for your vendors.

Enter Rates

Rates are per-destination prices. Please refer to the *Call Billing Parameters* section in the **PortaBilling Administrator Guide** for more information 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:
   o **Destination** – A destination prefix may be entered directly, e.g. **420** for Czech Republic, 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.

   o **Interval First** – first billing unit in seconds

- o **Interval Next** – next billing unit in seconds
- o **Price First** – per minute price for first interval
- o **Price Next** – per minute price for next interval
- o **Off-peak Interval First**– first billing unit in seconds for off - peak time
- o **Off-peak Interval Next** – next billing unit in seconds for off- peak time
- o **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.
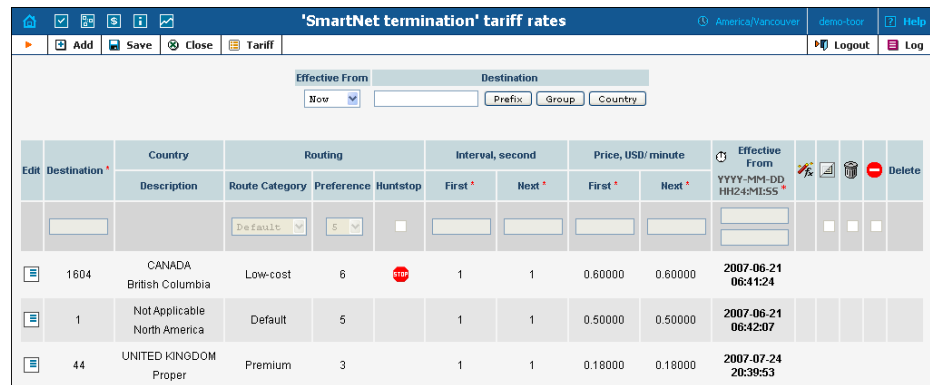
- o **Rate Formula Wizard** 🪄 – launches the wizard for creating a custom rating formula
- o **Effective from** – If you want this rate to take effect sometime in the future, you can either type in a date manually, or use the calendar (click on the DD-MM-YYYY link).

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.

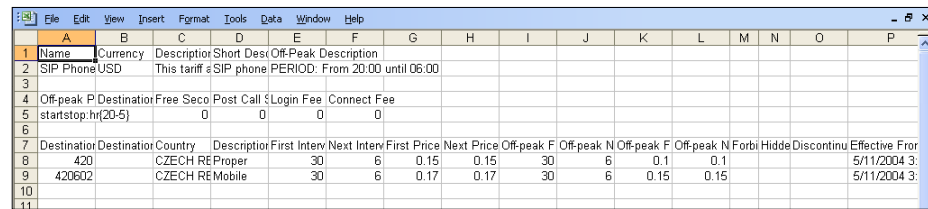| Edit	Destination*	Country	Routing			Interval, second		Price, USD/ minute		Effective From YYYY-MM-DD HH24:MI:SS *					Delete
		Description	Route Category	Preference	Huntstop	First*	Next*	First*	Next*						
			Default	5	☐						☐	☐	☐		
📄	1604	CANADA British Columbia	Low-cost	6	🛑 STOP	1	1	0.60000	0.60000	2007-06-21 06:41:24					
📄	1	Not Applicable North America	Default	5		1	1	0.50000	0.50000	2007-06-21 06:42:07					
📄	44	UNITED KINGDOM Proper	Premium	3		1	1	0.18000	0.18000	2007-07-24 20:39:53					

- o **Route category** – you can split your available routes into several categories, such as "Low-cost", "Premium", 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%.
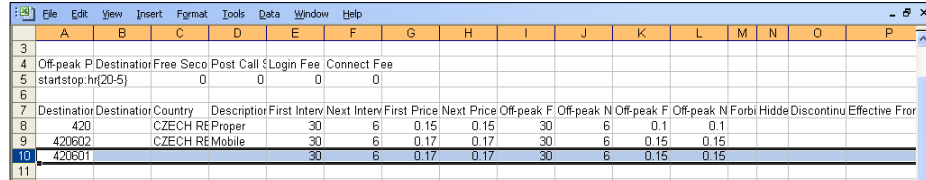
1. If you are not in Tariff Management for your tariff, go to the main menu, click on **Tariffs**, and then click on the tariff name.
2. In the Edit Tariff window, move the mouse over the **Download** button and hold it there until a popup menu appears. Choose the **Now** menu item and click on it. This 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.

	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P
3																
4	Off-peak P	Destination	Free Seco	Post Call S	Login Fee	Connect Fee										
5	startstop:hr{20-5}		0	0	0	0										
6																
7	Destination	Destination	Country	Description	First Interv	Next Interv	First Price	Next Price	Off-peak F	Off-peak N	Off-peak F	Off-peak N	Forbi	Hidde	Discontinu	Effective From
8	420		CZECH RE	Proper	30	6	0.15	0.15	30	6	0.1	0.1				
9	420602		CZECH RE	Mobile	30	6	0.17	0.17	30	6	0.15	0.15				
10	420601				30	6	0.17	0.17	30	6	0.15	0.15				
11																

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 billing scheme for your clients; these tariffs are created as "Applied to: Customer". For example, if you plan to have two different SIP call products with different rates, you will need two separate tariffs.
- A tariff with the termination costs for each termination partner you have; these tariffs are created as "Applied to: Vendor".
- If you have resellers, you should also create tariffs for charging each of them; these tariffs are created as "Applied to: Reseller". Do not create tariffs which will be applied to your resellers' subscribers yet. First create the resellers, and then return to this

step. When creating these subscriber tariffs, make sure you choose *Managed by NNN* in the menu, where *NNN* is the name of the corresponding reseller.

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.
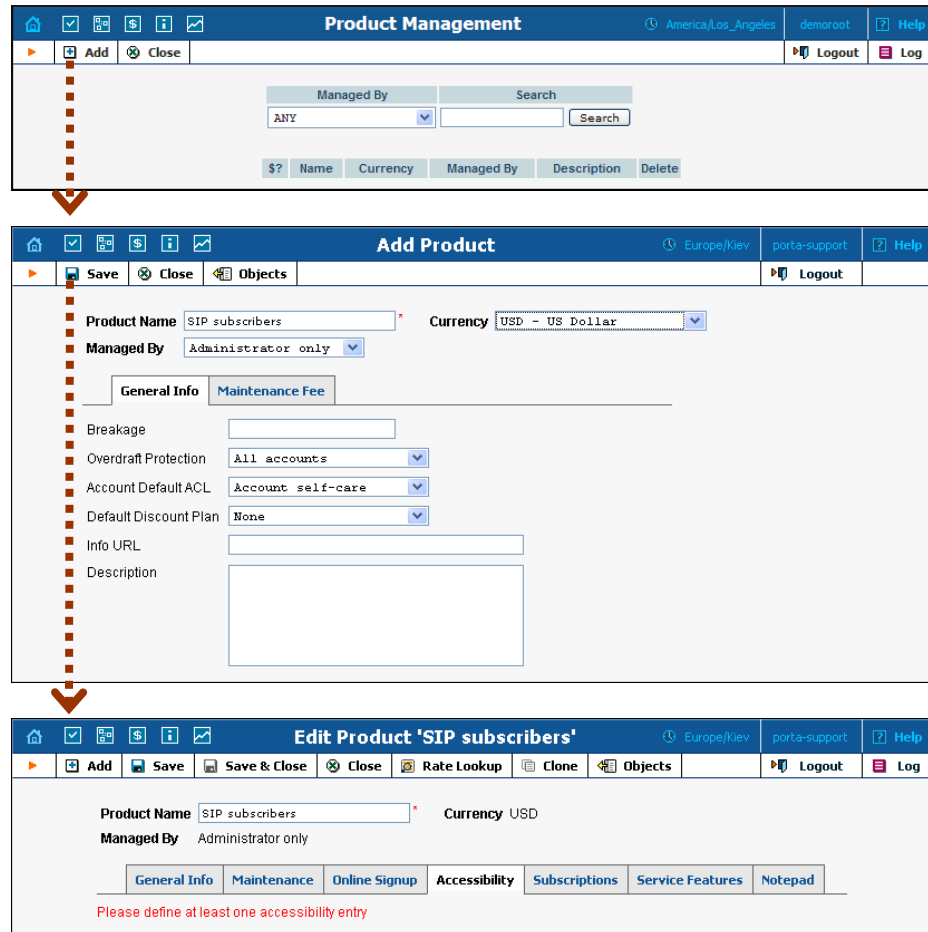

In the Management section of the Admin-Index page, choose **Products**.
1. On the Product Management page, click the ⊞ **Add** icon.
2. Fill in the "Add product" form:
    o **Product name** – product object name.
    o **Currency** – product currency; only tariffs which have the same currency will be permitted in the accessibility list.
    o **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.

- o **Breakage** – This parameter is typically used only for prepaid accounts, so leave it empty.
- o **Account Default ACL** – The access level assigned by default to new accounts created with this product. The ACL determines which operations may be performed by accounts on the self-care pages. The default value is "Account self-care" (pre-defined ACL), which allows all possible operations.
- o **Default Discount Plan** – Leave **None** as the selected entry if you do not plan to offer any discounted rates based on call volume. (In later sections of this document you will find examples for setting up volume discounts).
- o **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.
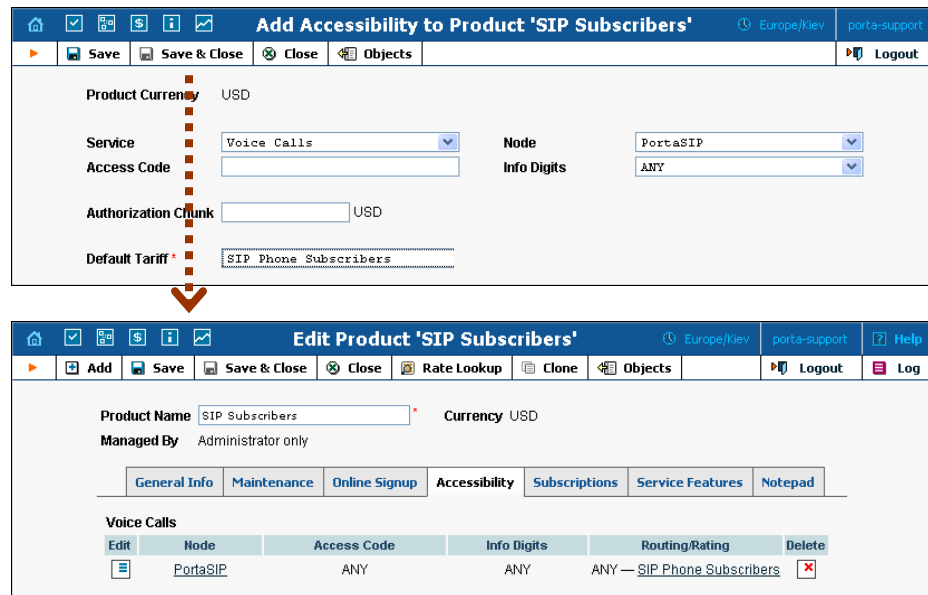
1. When the Accessibility tab is selected, click on the ⊞ **Add** icon.
2. Choose **Voice Calls** in the **Service** select menu.
3. In the **Node** select menu select the PortaSIP node.
4. The **Access Code** should be left empty for the basic SIP service.
5. The **Info Digits** field only makes sense when a call originates from your customer in a public telephony network. Therefore, just leave it empty for the SIP service.

6. In the **Default Tariff** menu, select the tariff that will be applied to your customers for outgoing calls.
7. Click ■ **Save&Close** to save this accessibility entry.

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)

o **Vendor name** – short name for the Vendor object; this will be used on the web interface.

o **Currency** – the currency in which this vendor charges you.
o **Opening Balance** – starting balance for the vendor; the default is zero.

Additional info:

o **Billing Period** – split period for vendor statistics.

User Interface:

o **Time Zone** – the time zone that the vendor uses for his billing period (when sending you an invoice). Statistics will be split into periods in this time zone, so your statistics will match the vendor's.
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**.
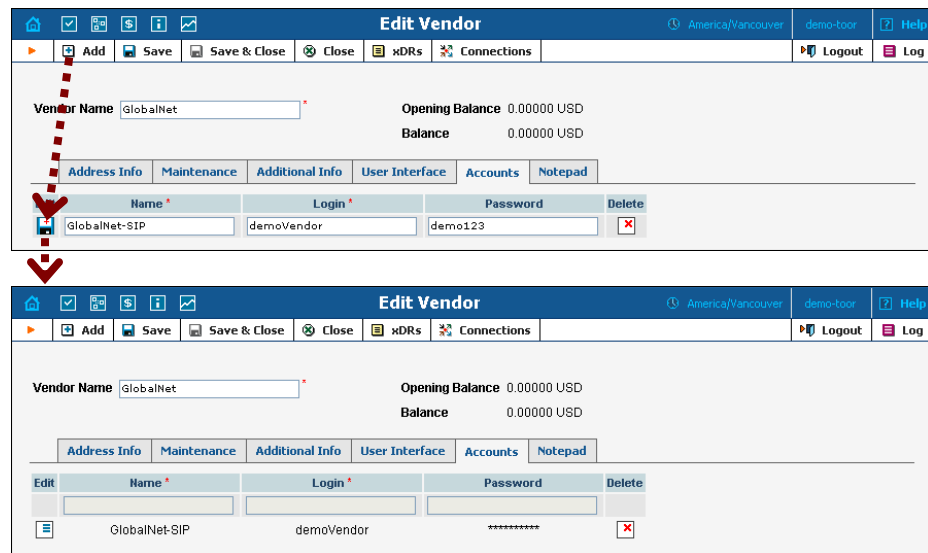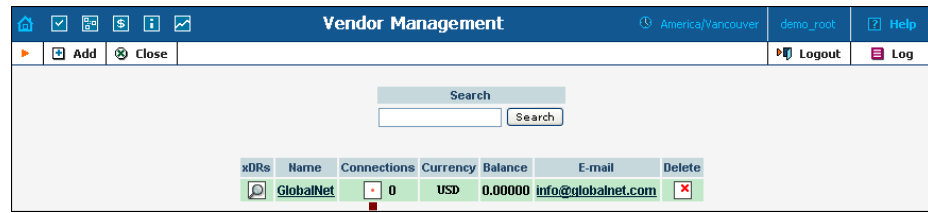

6. Click **Close** in order to return to the **Vendors** admin page.
7. Repeat steps 2-6 to add all of your vendors.

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.
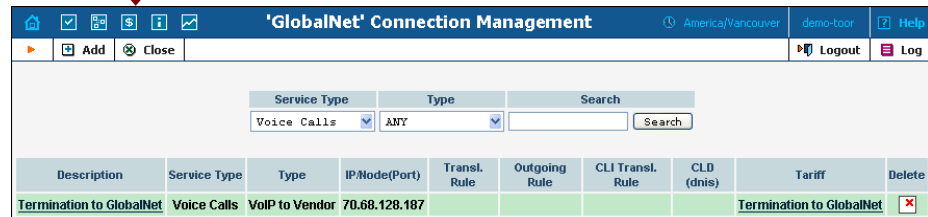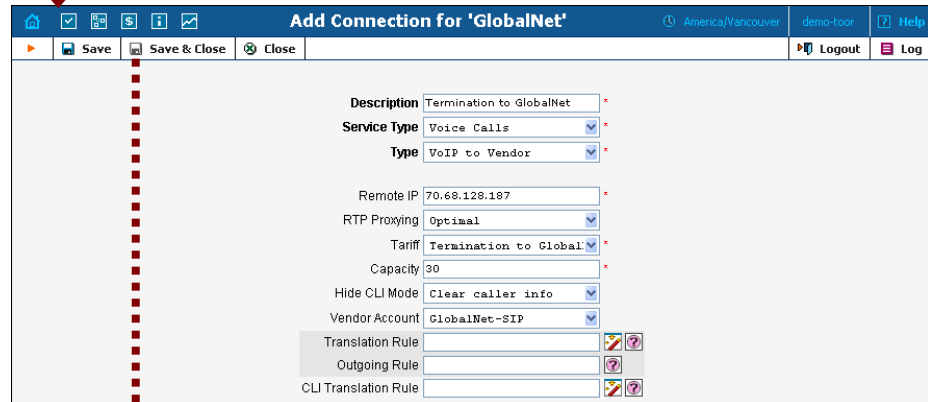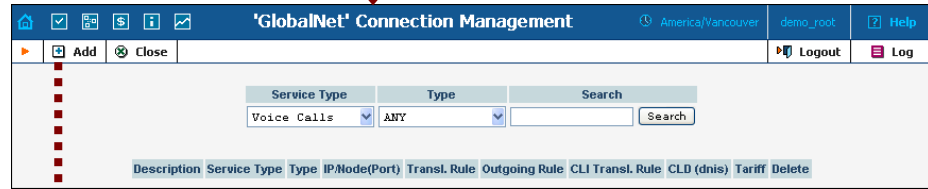
3. Press ⊞ **Add** to add a new connection.
4. Fill in the connection information. If you send traffic to a vendor via telephony, choose the node and enter the optional port pattern. If you send traffic via VoIP, enter the remote IP address (address of the vendor's gateway or SIP server). Choose the 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.
5. Click ◻ **Save&Close**.
6. Repeat steps 3-5 to add more connections to the same vendor, then click **Close** to exit to the **Vendor Management** screen.
7. Repeat steps 2-6 to add connections for other vendors.

Create Customer Class

Customer class provides the ability to define a group of parameters in a centralized way, then apply those parameters to many customers at once. (If you have already created a customer class that you can use for prepaid card distributors, skip this step and go to the next one.)

1. In the Management section of the Admin interface, choose **Customer Classes**.
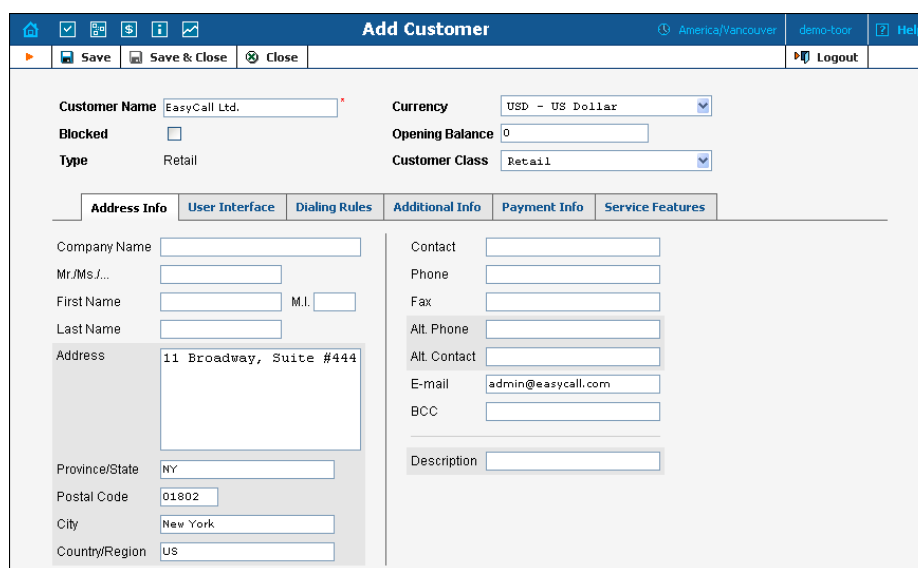2. On the Customer Class Management page, choose ➕ **Add**.

3. Fill in the customer class parameters:
   - **Name** – Short name for this customer class.
   - **Send Statistics** – Choose whether you want to send statistics to a customer.
   - **Invoice Template** – Select the appropriate template which will be used to create a PDF file with the invoice for your SIP customers. If you have not created a template yet, simply leave **Do not create invoice** for now. Later you can create an invoice template and assign it here.
   - **Terms** – Choose the appropriate description for the invoice payment terms.
   - **Send Invoices** – Check the box to automatically send an invoice to a customer at the end of the billing period.
   - Please consult the **PortaBilling Web Reference Guide** for a description of other parameters available here. For now you may leave these empty. If you change them later, these changes will automatically affect all customers who are already assigned to this customer class.
   - **Description** – Your comments about the intended use of this customer class.

4. Click ▪ **Save&Close**.

Create Customer

A customer is an owner of accounts. The customer's contact information is used to distribute account usage information, call statistics, invoices, and so on.

1. In the Management section of the Admin-Index page, choose **Customers**.
2. On the Customer Management page, choose ➕ **Add Customer**.
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.
- **Customer Class** – Choose the customer class you created in the previous step

Address info tab

- **Email** – An email address for the distribution of accounting information. After the billing period is over, a list of xDRs and other statistics will be sent to this address

23

o **Bcc** - Delivery to the specified email address of your account representative a copy of every outgoing email sent to the customer; this may be used for debug and archiving purposes.

Additional info tab

o **Billing period** – Frequency of invoice generation. For more details about the different available billing periods, see the **PortaBilling Administrator Guide**.

o **Send Statistics** - **Summary only** – Deliver a call summary only and do not attach a details file; this could be useful in the case of a large number of calls. Other options are **Full Statistics** (attach a complete list of xDRs) or **Do not send** (do not deliver call statistics to this customer via email at all).

Payment info tab

o **Credit Limit** – if left empty, then there is no credit limit for this customer.

o **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

o **Time Zone** – the time zone in which customer will see his xDRs and also that which defines 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.

o **Web Interface Language** – language to be used on the customer self-care web interface.

Service Features tab

Using this tab, you can activate/deactivate various features of the services provided to customers. Note that features are defined per *service type* (the physical service provided to the user such as Conferencing, Data Service, Voice Calls, Messaging etc.)

This tab provides you with a convenient tool for managing parameters which are to be the same for a whole set of accounts.. Instead of trying to configure them for each individual account, you can define them once at the customer level, then specify in the account configuration that the value in the customer's configuration be used. Should you wish to change this value later, you need only modify it once for the change to be automatically propagated to all accounts.

Please consult the **PortaBilling Web Reference Guide** for a description of other parameters available here. For now you may leave these with their default values. If you change them later, these changes will automatically affect all accountscreated under this Customer

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.
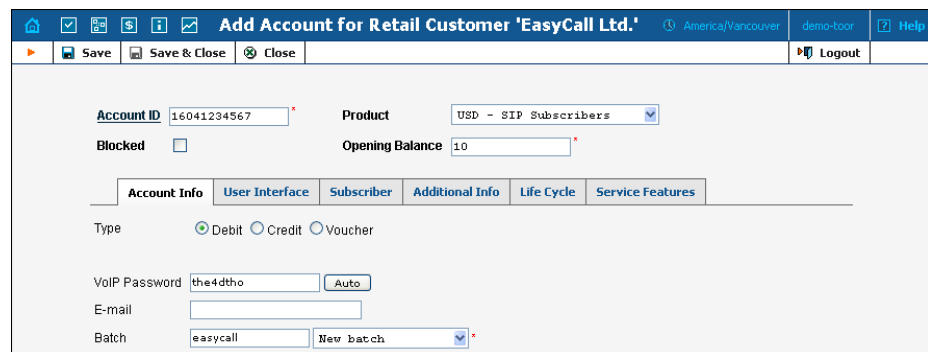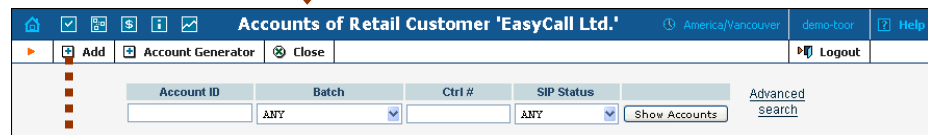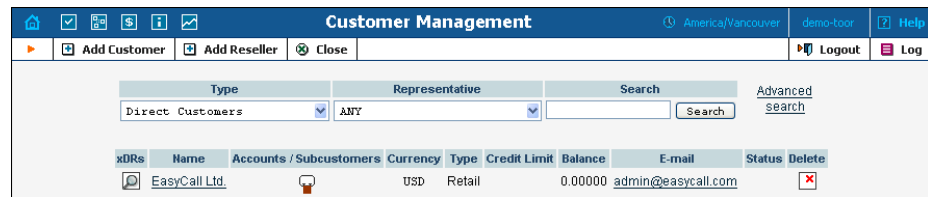
2. Next to the customer name, click on the ▦ icon (the one in the **Accounts** column) to go to the account management for that customer.
3. Click on ➕ **Add**.

4. Fill in the "Add account" form:
    o   **Account ID** – SIP ID, i.e. the phone number which will be used to login to the SIP server and receive incoming calls.
    o   **Product** – choose the product, which you would like your account to have.
    o   **Blocked** – you may create your account as blocked, although this is rarely done with SIP service accounts.
    o   **Opening Balance** – the initial balance on the account.

Account Info tab

o   **Type** – Account type; select credit for postpaid and debit for prepaid service.
o   **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).
o   **VoIP Password** – The account ID and this password will be used to authenticate SIP server login.
o   **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.
o   **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

o   **IP Phone** – When using the PortaSwitch auto-provisioning feature, here you may enter the name of the particular phone which this account will be provisioned to. Leave this field empty for basic SIP service, and use manual provisioning instead.
o   **IP Phone Port** – The specific port (phone line) on the IP phone which this account should be provisioned to (e.g. Sipura-2002 supports two phone lines, so two separate accounts can be provisioned on the same device on different lines). Leave this field empty for basic SIP service, and use manual provisioning instead.

Life Cycle tab

o   **Activation date** – Account activation date.
o   **Expiration date** – Account expiration date; since we are setting up a postpaid service, which should function for a long time, leave this field blank.
o   **Lifetime** – Relative expiration date; since we are setting up a postpaid service, which should function for a long time, leave this field blank.

User Interface tab

- o **Login** – Account login to web self-care pages. Can be the same as account ID.
- o **Password** – Password for the web self-care pages.
- o **Time Zone** - When an account owner (SIP services subscriber) 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.
- o **Web Interface Language** – The language to be used on the customer self-care web interface.

Service Features tab

Using this tab, you can activate/deactivate various features of the services provided to this account. Note that features are defined per *service type* (the physical service provided to the user such as Conferencing, Data Service, Voice Calls, Messaging etc.)

Please consult the **PortaBilling Web Reference Guide** for a description of parameters available here. For now you may leave these with their default values.

5. After clicking ◼ **Save&Close**, you will see a confirmation screen announcing that a new account has been created.
6. Repeat steps 3-5 if the customer has more than one phone line (extension).

Set up Dialing Rules for Customer (Optional)

Your customer may wish to use his own custom numbering format. For example, in order to make the transition from PSTN/PBX to VoIP as easy as possible, he will require that his users be able to dial phone numbers in exactly the same way as they used to on their PBX: 9 for an outside line, then 00 for international dialing or 0 for domestic, and so on.

Clearly a translation rule is needed here, and there is one – a customer-based translation rule. Likewise, in order to give customers the ability to manage their translation rules themselves without needing to use special terminology, there is a wizard allowing them to construct this rule by just entering the main parameters, such as the international dialing prefix.

Porta Switch

Thus, when one of this customer's accounts tries to make a call to 900420212345678, the SIP server will send an authentication request to the billing. Billing will then apply this customer's translation rule or node translation rule (if defined).
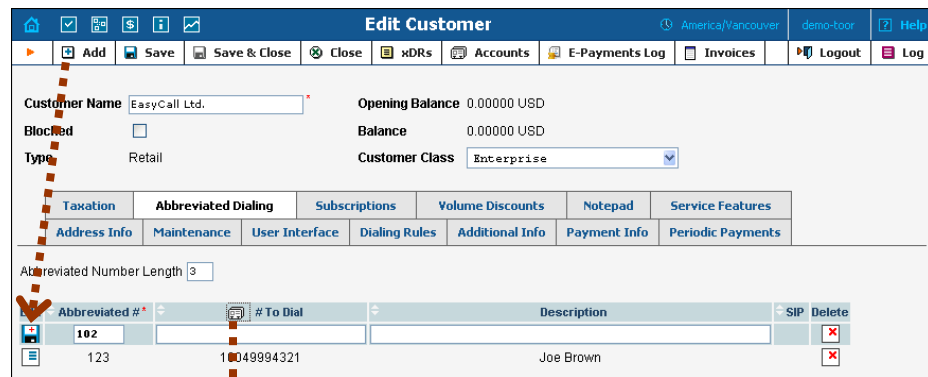
Set up Abbreviated Dialing for Customer (Optional)

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

1. In **Abbreviated Number Length** enter the maximum number of digits in the abbreviated number (e.g. if you plan to have extension numbers 401, 402 and so forth, the length will be 3). Click **Save**.

2. Now a table of abbreviated numbers will appear. Click on ⊞ **Add** to add a new extension.
3. Enter the abbreviated number and the actual phone number the call will be forwarded to. You may use a popup window to search for a specific account. Also note that **# To Dial** may contain any phone number (e.g. your partner's mobile phone number), and not just one of the SIP account IDs.

NOTE: If you enter an off-net PSTN number in **# To Dial**; it must be in the E.164 format, i.e. you cannot enter the number in the customer's dialing format.

4. Click the 🖫 **Save** button in the toolbar, or the 🖫 icon on the left side of the row.
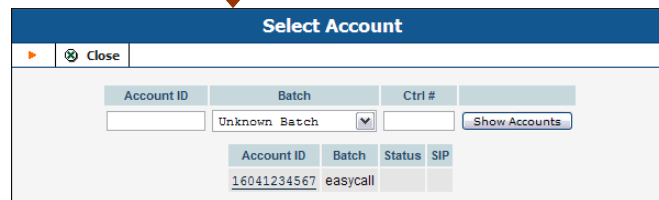5. Repeat steps 2-4 to add all the required abbreviated numbers.

🏠 ☑ 🖼 $ 🛈 📈	**Edit Customer**	🕓 America/Vancouver	demo-toor	? Help
▶ ⊞ Add 🖫 Save 🖫 Save & Close ⊗ Close 🗐 xDRs 🖳 Accounts 🖳 E-Payments Log 🗋 Invoices ▶🗖 Logout 🗐 Log				

Customer Name `EasyCall Ltd.` *   Opening Balance 0.00000 USD
Blocked ☐   Balance 0.00000 USD
Type   Retail   Customer Class `Enterprise ▾`

Taxation	**Abbreviated Dialing**	Subscriptions	Volume Discounts	Notepad	Service Features
Address Info	Maintenance	User Interface	Dialing Rules	Additional Info	Payment Info	Periodic Payments

Abbreviated Number Length `3`

	Abbreviated #*		🖳 # To Dial		Description	SIP	Delete
🖫	`102`		` `		` `		✖
🗐	123		16049994321		Joe Brown		✖

Select Account

▶ ⊗ Close	

Account ID	Batch	Ctrl #	
` `	`Unknown Batch ▾`	` `	Show Accounts

Account ID	Batch	Status	SIP
16041234567	easycall		

Testing 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 appendices 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:
    o   Go to the account list screen and see if the SIP indicator button (a blue circle) is on for this account.

o 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

Check Log Files

Browse the SIP server log file (`/var/log/sipenv-<sipserverIP>/log/sip.log` on the SIP server host) with a file display utility such as `less` or `more`. Some SIP auxiliary requests or parameters have been removed for greater clarity.

Using the SIP log viewer

You can also browse the SIP log file from the PortaBilling web interface using a SIP log viewer, which offers added convenience: for instance, automatically drawing call flow diagrams for you.
1. Choose the **SIP Log Viewer** item from the main menu.
2. Choose your PortaSIP node in the **PortaSIP node** select menu and select the time interval on the right side of the screen. Now you will see all call attempts in that interval. Press the **Generate** button to produce a call attempts list.

30

3. To obtain a detailed log view, click on the call-id for the call you are interested in. In the lower frame you will see a detailed copy of all SIP messages in this call. The upper frame provides a call-flow diagram, representing the main participating entities (SIP phone, SIP proxy, B2BUA, remote SIP UA, etc.). If you click on a particular dialogue element here, the lower frame scrolls automatically to display the corresponding SIP message.

SIP user agent attempts to make a call via the SIP server.

```
11 Jun 22:09:43/GLOBAL/ser[98218]: RECEIVED message from 216.231.44.168:9062:
INVITE sip:6831234@193.28.87.106 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-1035d24f
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
Max-Forwards: 70
Contact: John Doe <sip:16041234568@192.168.0.250:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 428
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura
Content-Type: application/sdp

v=0
o=- 16430652 16430652 IN IP4 192.168.0.250
s=-
c=IN IP4 192.168.0.250
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv

10 Jun 01:04:59/9154867a-2393e376@192.168.0.250/ser[53367]: processing INVITE
received from 216.231.44.168
```

SIP user agent is informed that his request is being processed.

```
11 Jun 22:09:43/GLOBAL/ser[98218]: SENDING message to 216.231.44.168:9062:
SIP/2.0 100 trying -- your call is important to us
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-
1035d24f;rport=9062;received=216.231.44.168
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
Server: Sip EXpress router (0.9.4 (i386/freebsd))
Content-Length: 0
```

Request is sent to B2BUA.

```
11 Jun 22:09:43/GLOBAL/ser[98218]: SENDING message to 193.28.87.106:5061:
INVITE sip:6831234@193.28.87.106:5061 SIP/2.0
Record-Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
Via: SIP/2.0/UDP
193.28.87.106;branch=z9hG4bK52d.65c85fbb7b48dc83837bf35f80ab19f5.0
Via: SIP/2.0/UDP
192.168.0.250:9062;rport=9062;received=216.231.44.168;branch=z9hG4bK-1035d24f
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
```

```
Max-Forwards: 16
Contact: John Doe <sip:16041234568@216.231.44.168:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 477
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura
Content-Type: application/sdp
PortaBilling-Notify: NAT

v=0
o=- 16430652 16430652 IN IP4 192.168.0.250
s=-
c=IN IP4 216.231.44.168
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

B2BUA receives this INVITE request.

```
11 Jun 22:09:43/GLOBAL/b2bua: RECEIVED message from 193.28.87.106:5060:
INVITE sip:6831234@193.28.87.106:5061 SIP/2.0
Record-Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
Via: SIP/2.0/UDP
193.28.87.106;branch=z9hG4bK52d.65c85fbb7b48dc83837bf35f80ab19f5.0
Via: SIP/2.0/UDP
192.168.0.250:9062;rport=9062;received=216.231.44.168;branch=z9hG4bK-1035d24f
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
Max-Forwards: 16
Contact: John Doe <sip:16041234568@216.231.44.168:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 477
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura
Content-Type: application/sdp
PortaBilling-Notify: NAT

v=0
o=- 16430652 16430652 IN IP4 192.168.0.250
s=-
c=IN IP4 216.231.44.168
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
```

```
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

In the rest of the log example, we will skip the request transmission between SER and B2BUA (request received by SER, request sent to B2BUA, request received by B2BUA) since this would only duplicate the same information.

```
11 Jun 22:09:43/GLOBAL/ser[]: SENDING message to 216.231.44.168:9062:
```

PortaSIP requests digest authentication from the SIP UA, providing a challenge.

```
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP
192.168.0.250:9062;received=216.231.44.168;rport=9062;branch=z9hG4bK-1035d24f
Record-Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
From: "John Doe" <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
Server: Sippy
WWW-Authenticate: Digest
realm="193.28.87.106",nonce="3051864d7d0c6578460cc4b0e28ad43b448c6a77"

11 Jun 22:09:43/GLOBAL/ser[98219]: RECEIVED message from 216.231.44.168:9062:
```

SIP UA acknowledges that it has received an authorization request (ACKs will be skipped in the rest of the document).

```
ACK sip:6831234@193.28.87.106 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-1035d24f
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 ACK
Max-Forwards: 70
Contact: John Doe <sip:16041234568@192.168.0.250:9062>
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 0
```

SER receives a reply to the authorization request, with a response to the challenge.

```
11 Jun 22:09:43/GLOBAL/ser[98219]: RECEIVED message from 216.231.44.168:9062:
INVITE sip:6831234@193.28.87.106 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-1628d42b
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 102 INVITE
Max-Forwards: 70
Authorization: Digest
username="16041234568",realm="193.28.87.106",nonce="3051864d7d0c6578460cc4b0e28ad4
3b448c6a77",uri="sip:
6831234@193.28.87.106",algorithm=MD5,response="54e0b42337ace33edf36d004f1037ebd"
Contact: John Doe <sip:16041234568@192.168.0.250:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 428
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura
Content-Type: application/sdp

v=0
```

```
o=- 16430652 16430652 IN IP4 192.168.0.250
s=-
c=IN IP4 192.168.0.250
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv

11 Jun 22:09:43/98dda488-69c74dd3@192.168.0.250/ser[98219]: processing INVITE
received from 216.231.44.168
11 Jun 22:09:43/GLOBAL/ser[98219]: SENDING message to 216.231.44.168:9062:
```

SIP UA is informed that the request has been received and is being processed (`100 Trying` responses will be omitted in the rest of the document).

```
SIP/2.0 100 trying -- your call is important to us
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-
1628d42b;rport=9062;received=216.231.44.168
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 102 INVITE
Server: Sip EXpress router (0.9.4 (i386/freebsd))
Content-Length: 0
```

This request is resent to B2BUA with several modifications (in particular, a `PortaBilling-Notify:NAT` flag is added to inform B2BUA of the NAT status of the device).

```
11 Jun 22:09:43/GLOBAL/ser[98219]: SENDING message to 193.28.87.106:5061:
INVITE sip:6831234@193.28.87.106:5061 SIP/2.0
Record-Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
Via: SIP/2.0/UDP
193.28.87.106;branch=z9hG4bK22d.30fef1504a66ee2bd3c0b9cfa4e4e09b.0
Via: SIP/2.0/UDP
192.168.0.250:9062;rport=9062;received=216.231.44.168;branch=z9hG4bK-1628d42b
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 102 INVITE
Max-Forwards: 16
Authorization: Digest
username="16041234568",realm="193.28.87.106",nonce="3051864d7d0c6578460cc4b0e28ad4
3b448c6a77",uri="sip:
6831234@193.28.87.106",algorithm=MD5,response="54e0b42337ace33edf36d004f1037ebd"
Contact: John Doe <sip:16041234568@216.231.44.168:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 477
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura
Content-Type: application/sdp
PortaBilling-Notify: NAT

v=0
o=- 16430652 16430652 IN IP4 192.168.0.250
s=-
c=IN IP4 216.231.44.168
```

```
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

B2BUA sends an authorization request to the billing.

```
11 Jun 22:09:43/98dda488-69c74dd3@192.168.0.250/b2bua: sending AAA request:
User-Name                        = '16041234568'
Digest-Realm                     = '193.28.87.106'
Digest-Nonce                     = '3051864d7d0c6578460cc4b0e28ad43b448c6a77'
Digest-Method                    = 'INVITE'
Digest-URI                       = 'sip:6831234@193.28.87.106'
Digest-Algorithm                 = 'MD5'
Digest-User-Name                 = '16041234568'
Digest-Response                  = '54e0b42337ace33edf36d004f1037ebd'
Calling-Station-Id               = '16041234568'
Called-Station-Id                = '6831234'
h323-conf-id                     = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id                          = '98dda488-69c74dd3@192.168.0.250'
h323-remote-address              = '216.231.44.168'
h323-session-protocol            = 'sipv2'
h323-ivr-out                     = 'PortaBilling_Routing:SIP'
h323-ivr-out                     = 'PortaBilling_AuthMethod:INVITE'
h323-ivr-out                     = 'PortaBilling_Notify:NAT'
h323-ivr-out                     = 'PortaBilling_Seed:144514807'

11 Jun 22:09:44/98dda488-69c74dd3@192.168.0.250/b2bua: AAA request accepted,
processing response:
```

Billing authorizes the call and provides information about call routing (5 possible routes are returned).

```
Cisco-AVPair        = 'h323-ivr-in=PortaBilling_Routing:@;g-
hunt=seq;expires=300;credit-time=29460;patience=20'
Cisco-AVPair        = 'h323-ivr-in=PortaBilling_Routing: 16046831234@193.28.187.3'
Cisco-AVPair        = 'h323-ivr-in=PortaBilling_Routing:
16046831234@70.68.128.186;auth=ipcall-test:test123'
Cisco-AVPair        = 'h323-ivr-in=PortaBilling_Routing:
16046831234@69.104.30.123;auth=PortaSoftware:PortaSoftware;rtpp=1'
Cisco-AVPair        = 'h323-ivr-in=PortaBilling_Routing: 16046831234@192.168.0.66'
Cisco-AVPair        = 'h323-ivr-in=PortaBilling_Routing: 16046831234@192.168.0.30'
h323-billing-model  = 'h323-billing-model=0'
Cisco-AVPair        = 'h323-ivr-in=Tariff:SIP Phone Subscribers'
Cisco-AVPair        = 'h323-ivr-in=PortaBilling_CLI:16041234568'
Cisco-AVPair        = 'h323-ivr-in=MOH:1'
Cisco-AVPair        = 'h323-ivr-in=PortaBilling_CompleteNumber:16046831234'
Cisco-AVPair        = 'h323-ivr-in=DURATION:29460'
h323-return-code    = 'h323-return-code=0'
h323-currency       = 'h323-currency=USD'
h323-credit-time    = 'h323-credit-time=29460'
h323-preferred-lang = 'h323-preferred-lang=en'
```

INVITE is sent to the first gateway/proxy in the route list.

```
11 Jun 22:09:44/GLOBAL/b2bua: SENDING message to 193.28.187.3:5060:
INVITE sip:16046831234@193.28.187.3:5060 SIP/2.0
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK1343caf4b64b53d7a6d0b68a51b554aa;rport
Max-Forwards: 70
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=52d266cda37f42d0c24541d5190acf71
To: <sip:16046831234@193.28.187.3>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 200 INVITE
Contact: Anonymous <sip:193.28.87.106:5061>
Expires: 300
User-Agent: Sippy
cisco-GUID: 2847286574-946067207-396958912-1950566603
h323-conf-id: 2847286574-946067207-396958912-1950566603
Content-Length: 475
Content-Type: application/sdp

v=0
o=Sippy 137112044 0 IN IP4 193.28.87.106
s=-
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
c=IN IP4 216.231.44.168
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

No reply is received from this gateway/proxy, so PortaSIP re-sends the invite message several times.

```
11 Jun 22:10:00/GLOBAL/b2bua: SENDING message to 193.28.187.3:5060:
INVITE sip:16046831234@193.28.187.3:5060 SIP/2.0
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK1343caf4b64b53d7a6d0b68a51b554aa;rport
Max-Forwards: 70
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=52d266cda37f42d0c24541d5190acf71
To: <sip:16046831234@193.28.187.3>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 200 INVITE
Contact: Anonymous <sip:193.28.87.106:5061>
Expires: 300
User-Agent: Sippy
cisco-GUID: 2847286574-946067207-396958912-1950566603
h323-conf-id: 2847286574-946067207-396958912-1950566603
Content-Length: 475
Content-Type: application/sdp

v=0
o=Sippy 137112044 0 IN IP4 193.28.87.106
s=-
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
c=IN IP4 216.231.44.168
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
```

```
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

Finally, PortaSIP decides that this route is non-functioning, and a failed accounting record is sent to the billing.

```
11 Jun 22:10:04/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Stop
(Originate):
h323-call-origin     = 'originate'
h323-call-type       = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time      = '19:09:44.000 GMT Sun Jun 11 2006'
User-Name            = '16041234568'
Calling-Station-Id   = '16041234568'
Called-Station-Id    = '16046831234'
h323-conf-id         = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id              = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id      = '98dda488-69c74dd3@192.168.0.250'
h323-remote-address  = '193.28.187.3'
h323-ivr-out         = 'DURATION:7200'
h323-ivr-out         = 'PortaBilling_Seed:144514807'
h323-disconnect-time = '19:10:04.000 GMT Sun Jun 11 2006'
h323-connect-time    = '19:10:04.000 GMT Sun Jun 11 2006'
Acct-Session-Time    = '0'
h323-disconnect-cause = '10'
Acct-Status-Type     = 'Stop'
```

After that, PortaSIP tries the next route in the list.

```
11 Jun 22:10:04/GLOBAL/b2bua: SENDING message to 70.68.128.186:5060:
INVITE sip:16046831234@70.68.128.186:5060 SIP/2.0
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bKade4b51964bd86e3026170c62ce471c6;rport
Max-Forwards: 70
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 200 INVITE
Contact: Anonymous <sip:193.28.87.106:5061>
Expires: 300
User-Agent: Sippy
cisco-GUID: 2847286574-946067207-396958912-1950566603
h323-conf-id: 2847286574-946067207-396958912-1950566603
Content-Length: 475
Content-Type: application/sdp

v=0
o=Sippy 137183084 0 IN IP4 193.28.87.106
s=-
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
c=IN IP4 216.231.44.168
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
```

```
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

This gateway is available, so we get a reply stating that it has started to process the call.

```
11 Jun 22:10:04/GLOBAL/b2bua: RECEIVED message from 70.68.128.186:5060:
SIP/2.0 100 trying -- your call is important to us
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bKade4b51964bd86e3026170c62ce471c6;rport=5061
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 200 INVITE
Server: Sip EXpress router (0.9.4 (i386/freebsd))
Content-Length: 0
```

Ringback is transferred to the SIP UA, so the user on the SIP phone will hear ringing.

```
11 Jun 22:10:04/GLOBAL/ser[98217]: SENDING message to 216.231.44.168:9062:
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP
192.168.0.250:9062;received=216.231.44.168;rport=9062;branch=z9hG4bK-1628d42b
Record-Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
From: "John Doe" <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>;tag=b293b17e775f2bdf6d192e545261bb1d
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 102 INVITE
Server: Sippy

11 Jun 22:10:04/GLOBAL/b2bua: RECEIVED message from 70.68.128.186:5060:
```

The remote SIP proxy requests authorization.

```
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 193.28.87.106:5061;
branch=z9hG4bKade4b51964bd86e3026170c62ce471c6;rport=5061
Record-Route: <sip:70.68.128.186;ftag=f74a7a2b122becfb3bb9ca65f75193f6;lr>
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 200 INVITE
Server: Sippy
WWW-Authenticate: Digest
realm="70.68.128.186",nonce="ec2f8a5c71f14e5a8e08fc77816b3341448c6a8c"

11 Jun 22:10:04/GLOBAL/b2bua: SENDING message to 70.68.128.186:5060:
```

PortaSIP computes the digest authentication response and sends back a reply.

```
INVITE sip:16046831234@70.68.128.186:5060 SIP/2.0
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK03ab045dae8f997ee66e34049ee07ee8;rport
Max-Forwards: 70
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>
Call-ID: 98dda488-69c74dd3@192.168.0.250
```

```
CSeq: 201 INVITE
Contact: Anonymous <sip:193.28.87.106:5061>
Expires: 300
User-Agent: Sippy
cisco-GUID: 2847286574-946067207-396958912-1950566603
h323-conf-id: 2847286574-946067207-396958912-1950566603
Authorization: Digest username="ipcall-
test",realm="70.68.128.186",nonce="ec2f8a5c71f14e5a8e08fc77816b3341448c6a8c",uri="
sip:
16046831234@70.68.128.186:5060",response="90b1b14cd90e62d2be3b8b6c10ae72f1"
Content-Length: 475
Content-Type: application/sdp

v=0
o=Sippy 137183084 0 IN IP4 193.28.87.106
s=-
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
c=IN IP4 216.231.44.168
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

The called party's phone starts ringing.

```
11 Jun 22:10:08/GLOBAL/b2bua: RECEIVED message from 70.68.128.186:5060:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK03ab045dae8f997ee66e34049ee07ee8;rport=5061
Record-Route: <sip:70.68.128.186;ftag=f74a7a2b122becfb3bb9ca65f75193f6;lr>
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>;tag=f55ecc2530650faffe5da956658086c9
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 201 INVITE
Server: Sippy
Content-Length: 176
Content-Type: application/sdp

v=0
o=NexTone-MSW 48600030 0 IN IP4 64.7.121.229
s=sip call
t=0 0
m=audio 35086 RTP/AVP 18 101
c=IN IP4 70.68.128.186
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

The called party answers the call.

```
11 Jun 22:10:09/GLOBAL/b2bua: RECEIVED message from 70.68.128.186:5060:
SIP/2.0 200 OK
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK03ab045dae8f997ee66e34049ee07ee8;rport=5061
Record-Route: <sip:70.68.128.186;ftag=f74a7a2b122becfb3bb9ca65f75193f6;lr>
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
```

```
To: <sip:16046831234@70.68.128.186>;tag=f55ecc2530650faffe5da956658086c9
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 201 INVITE
Server: Sippy
Contact: Anonymous <sip:70.68.128.186:5061>
Content-Length: 176
Content-Type: application/sdp

v=0
o=NexTone-MSW 48600030 0 IN IP4 64.7.121.229
s=sip call
t=0 0
m=audio 35086 RTP/AVP 18 101
c=IN IP4 70.68.128.186
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Start accounting for the outgoing (originate/VoIP) call leg is sent to the billing.

```
11 Jun 22:10:09/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Start
(Originate):
h323-call-origin      = 'originate'
h323-call-type        = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time       = '19:10:04.000 GMT Sun Jun 11 2006'
User-Name             = '16041234568'
Calling-Station-Id    = '16041234568'
Called-Station-Id     = '16046831234'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id               = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id       = '98dda488-69c74dd3@192.168.0.250'
h323-remote-address   = '70.68.128.186'
h323-ivr-out          = 'DURATION:7200'
h323-ivr-out          = 'PortaBilling_Seed:144514807'
h323-connect-time     = '19:10:09.000 GMT Sun Jun 11 2006'
alert-timepoint       = '19:10:08.000 GMT Sun Jun 11 2006'
Acct-Status-Type      = 'Start'
```

Start accounting for the incoming (answer/VoIP) call leg is sent to the billing.

```
11 Jun 22:10:09/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Start
(Answer):
h323-call-origin      = 'answer'
h323-call-type        = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time       = '19:09:44.000 GMT Sun Jun 11 2006'
User-Name             = '16041234568'
Calling-Station-Id    = '16041234568'
Called-Station-Id     = '6831234'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id               = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id       = '98dda488-69c74dd3@192.168.0.250'
h323-remote-address   = '216.231.44.168'
h323-ivr-out          = 'PortaBilling_Seed:144514807'
h323-connect-time     = '19:10:09.000 GMT Sun Jun 11 2006'
alert-timepoint       = '19:10:09.000 GMT Sun Jun 11 2006'
Acct-Status-Type      = 'Start'
```

One of the parties hangs up; the call termination process is started.

```
11 Jun 22:11:24/GLOBAL/ser[98217]: RECEIVED message from 216.231.44.168:9062:
BYE sip:193.28.87.106:5061 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-87f40044
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>;tag=b293b17e775f2bdf6d192e545261bb1d
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 103 BYE
```

```
Max-Forwards: 70
Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
Authorization: Digest
username="16041234568",realm="193.28.87.106",nonce="3051864d7d0c6578460cc4b0e28ad4
3b448c6a77",uri="sip:
193.28.87.106:5061",algorithm=MD5,response="4e3308a385aeff5159ad03f738f9bd31"
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 0
```

B2BUA sends stop accounting to the billing for the incoming call leg.

```
11 Jun 22:11:24/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Stop (Answer):
h323-call-origin      = 'answer'
h323-call-type        = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time       = '19:09:44.000 GMT Sun Jun 11 2006'
User-Name             = '16041234568'
Calling-Station-Id    = '16041234568'
Called-Station-Id     = '6831234'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id               = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id       = '98dda488-69c74dd3@192.168.0.250'
h323-remote-address   = '216.231.44.168'
h323-ivr-out          = 'PortaBilling_Seed:144514807'
h323-voice-quality    = '0'
Acct-Terminate-Cause  = 'User-Request'
h323-ivr-out          = 'PortaBilling_Session:unlock'
h323-disconnect-time  = '19:11:23.000 GMT Sun Jun 11 2006'
h323-connect-time     = '19:10:09.000 GMT Sun Jun 11 2006'
Acct-Session-Time     = '74'
h323-disconnect-cause = '0'
alert-timepoint       = '19:10:09.000 GMT Sun Jun 11 2006'
Acct-Status-Type      = 'Stop'
```

The outgoing call leg is closed.

```
11 Jun 22:11:24/GLOBAL/b2bua: SENDING message to 70.68.128.186:5060:
BYE sip:70.68.128.186:5061 SIP/2.0
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK13db39161f94a7d3e3eabcf209661a93;rport
Route: <sip:70.68.128.186;ftag=f74a7a2b122becfb3bb9ca65f75193f6;lr>
Max-Forwards: 70
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>;tag=f55ecc2530650faffe5da956658086c9
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 203 BYE
Contact: Anonymous <sip:193.28.87.106:5061>
Expires: 300
User-Agent: Sippy
cisco-GUID: 2847286574-946067207-396958912-1950566603
h323-conf-id: 2847286574-946067207-396958912-1950566603
```

B2BUA sends stop accounting to the billing for the incoming call leg.

```
11 Jun 22:11:24/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Stop
(Originate):
h323-call-origin      = 'originate'
h323-call-type        = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time       = '19:10:04.000 GMT Sun Jun 11 2006'
User-Name             = '16041234568'
Calling-Station-Id    = '16041234568'
Called-Station-Id     = '16046831234'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id               = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id       = '98dda488-69c74dd3@192.168.0.250'
```

```
h323-remote-address   = '70.68.128.186'
h323-ivr-out          = 'DURATION:7200'
h323-ivr-out          = 'PortaBilling_Seed:144514807'
h323-voice-quality    = '0'
Acct-Terminate-Cause  = 'User-Request'
h323-disconnect-time  = '19:11:23.000 GMT Sun Jun 11 2006'
h323-connect-time     = '19:10:09.000 GMT Sun Jun 11 2006'
Acct-Session-Time     = '74'
h323-disconnect-cause = '0'
alert-timepoint       = '19:10:08.000 GMT Sun Jun 11 2006'
Acct-Status-Type      = 'Stop'
```

The other party confirms call disconnection.

```
11 Jun 22:11:24/GLOBAL/b2bua: RECEIVED message from 70.68.128.186:5060:
SIP/2.0 200 OK
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK13db39161f94a7d3e3eabcf209661a93;rport=5061
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>;tag=f55ecc2530650faffe5da956658086c9
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 203 BYE
Server: Sippy
```

The call is finished.

4. Browse information in the PortaBilling log file. To do so, you can either:

   - Login to the PortaBilling master server and type "`less /var/log/porta-billing.log`"
   - Find this call using **Trace call** on the PortaBilling web interface, view the call details (press 🔍 icon in the leftmost column), and then click the 📄**View log** button in the tool bar.

PortaBilling receives the authorization request.

```
Jun 11 12:09:44: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address        = '193.28.87.106'
User-Name             = '16041234568'
Called-Station-Id     = '6831234'
Calling-Station-Id    = '16041234568'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id               = '98dda488-69c74dd3@192.168.0.250'
Digest-Attributes     = 'Realm = "193.28.87.106"'
Digest-Attributes     = 'Nonce = "3051864d7d0c6578460cc4b0e28ad43b448c6a77"'
Digest-Attributes     = 'Method = "INVITE"'
Digest-Attributes     = 'URI = "sip:6831234@193.28.87.106"'
Digest-Attributes     = 'Algorithm = "MD5"'
Digest-Attributes     = 'User-Name = "16041234568"'
Digest-Response       = '54e0b42337ace33edf36d004f1037ebd'
h323-remote-address   = '216.231.44.168'
h323-session-protocol = 'sipv2'
h323-ivr-out          = 'PortaBilling_Routing:SIP'
h323-ivr-out          = 'PortaBilling_AuthMethod:INVITE'
h323-ivr-out          = 'PortaBilling_Notify:NAT'
h323-ivr-out          = 'PortaBilling_Seed:144514807'
NAS-Port              = '5060'
Jun 11 12:09:44: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:09:44: H323/SIP call, use h323-conf-id, but remember call-id
Jun 11 12:09:44: Checking if this call comes through a VoIP from vendor connection
```

```
Jun 11 12:09:44: No VoIP from vendor connections were found
Jun 11 12:09:44: PrepareNexecute 'AccountAuth'
Jun 11 12:09:44: Found Account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000]
```

Account information is located in the database.

```
Jun 11 12:09:44: Account 16041234568 is not logged in yet
Jun 11 12:09:44: Verification of password using method 'digest_response': success
```

Password verification is successful.

```
Jun 11 12:09:44: Applying override translation rule on CLD ...
Jun 11 12:09:44: Translation 's/^\*3164\*//' applied: '6831234' unchanged
Jun 11 12:09:44: Applying customer dialing translation rule on CLD ...
Jun 11 12:09:44: Translation
'local_to_e164($_,{cc=>'1',ac=>'604',dp=>'1',ip=>'011',em=>'911',ex=>'411'});#%
cc=1 ac=604 dp=1 ip=011 em=911 ex=411' applied: '6831234' -> '16046831234'
```

The customer's number translation rule is applied, and the phone number is changed from the local format into E.164.

```
Jun 11 12:09:44: PrepareNexecute 'AccountAuth'
Jun 11 12:09:44: CLD '16046831234' is an off-net number
```

The dialed number is an off-net destination.

```
Jun 11 12:09:44: Setting up a new charge with tariff 'SIP Phone Subscribers' ...
Jun 11 12:09:44: Checked 2006-06-11 21:09:44 Europe/Prague against 'hr{20-5}': 1
Jun 11 12:09:44: Start of call is peak level 1
Jun 11 12:09:44: PrepareNexecute 'GetPricePerDestination*'
Jun 11 12:09:44: Maximum call duration: 29460 announced as 29460
(!+1x30x0.02+4905x6x0.02) by rate 101 using 144547388 as seed
```

Maximum call duration is calculated according to the available funds and rate.

```
Jun 11 12:09:44: Remote termination 'PortaSIP': Calculating routing for
16046831234
Jun 11 12:09:44: RTP Proxy with origination preference 'Undetermined', Calling
party behind NAT.
Jun 11 12:09:44: Looking up routes to '16046831234' using '<Default System
Routing>' routing plan
Jun 11 12:09:44: PrepareNexecute 'GetRoutingPerDestination*'
Jun 11 12:09:44: Using peak rate, since no off-peak is defined
Jun 11 12:09:44: Result routes to destination '16046831234':
        16046831234@193.28.187.3, prio = 9, cost = 0.07000, 'test' - to remote GW
        16046831234@70.68.128.186, prio = 7, cost = 0.01000, 'Termination to
Globalnet' - to remote GW
        16046831234@69.104.30.123, prio = 6, cost = 0.55000, 'VoIPio via SIP' - to
remote GW
        16046831234@192.168.0.66, prio = 6, cost = 0.60000, 'Premium VoIP->Vendor'
- to remote GW
        16046831234@192.168.0.30, prio = 6, cost = 0.60000, 'X-Telecom' - to
remote GW
```

There are five possible routes (sorted according to preference and cost).

```
Jun 11 12:09:44: Logging in account '16041234568'(103) to 'A9B6252E 3863D707
17A91CC0 744348CB'
Jun 11 12:09:44: Authentication acknowledge response
```

An authorization response is sent to PortaSIP.

```
Cisco-AVPair        = h323-ivr-in=PortaBilling_Routing:@;g-
hunt=seq;expires=300;credit-time=29460;patience=20
Cisco-AVPair        = h323-ivr-in=PortaBilling_Routing: 16046831234@193.28.187.3
Cisco-AVPair        = h323-ivr-in=PortaBilling_Routing:
16046831234@70.68.128.186;auth=C43B527B8BCCF31A5CB84F49D8D576DE613800D328115690
Cisco-AVPair        = h323-ivr-in=PortaBilling_Routing:
16046831234@69.104.30.123;auth=3741D17EED2A31B7990D8DFEC859D77EA2147BAEE44CD5DBBD9
C1C4D35DE9A99;rtpp=1
Cisco-AVPair        = h323-ivr-in=PortaBilling_Routing: 16046831234@192.168.0.66
Cisco-AVPair        = h323-ivr-in=PortaBilling_Routing: 16046831234@192.168.0.30
h323-billing-model  = 0
h323-ivr-in         = Tariff:SIP Phone Subscribers
h323-ivr-in         = PortaBilling_CLI:16041234568
h323-ivr-in         = MOH:1
h323-ivr-in         = PortaBilling_CompleteNumber:16046831234
h323-ivr-in         = DURATION:29460
h323-return-code    = 0
h323-currency       = USD
h323-credit-time    = 29460
h323-preferred-lang = en
Jun 11 12:09:44: ...Done.
```

Accounting for the failed outgoing call leg is received.

```
Jun 11 12:10:04: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address      = '193.28.87.106'
User-Name           = '16041234568'
Called-Station-Id   = '16046831234'
Calling-Station-Id  = '16041234568'
Acct-Status-Type    = 'Stop'
h323-call-origin    = 'originate'
h323-call-type      = 'VoIP'
h323-setup-time     = '19:09:44.000 GMT Sun Jun 11 2006'
h323-connect-time   = '19:10:04.000 GMT Sun Jun 11 2006'
h323-disconnect-time = '19:10:04.000 GMT Sun Jun 11 2006'
h323-disconnect-cause = '10'
h323-conf-id        = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id             = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id     = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Time   = '0'
Acct-Delay-Time     = '0'
h323-session-protocol = 'sipv2'
h323-remote-address = '193.28.187.3'
h323-ivr-out        = 'DURATION:7200'
h323-ivr-out        = 'PortaBilling_Seed:144514807'
NAS-Port            = '5060'
Exec-Program-Log    = 'porta-billing.pl'
Jun 11 12:10:04: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:10:04: Found a call in cache with such id
Jun 11 12:10:04: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request
```

The billing re-uses information in the call cache to speed up account info lookup.

```
Jun 11 12:10:04: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:10:04: End of the outgoing failed call for logged in account. Waiting
another outgoing call or hang up
Jun 11 12:10:04: Looking up vendor/connection
Jun 11 12:10:04: Trying to match connection for call
Jun 11 12:10:04: Looking for a connection VoIP/originate
Jun 11 12:10:04: Outgoing VoIP, matching by the remote IP address '193.28.187.3'
(env 4)
Jun 11 12:10:04: Found connection 4 'test' to vendor 'MCI Vendor'
```

Connection matched.

```
Jun 11 12:10:04: Found vendor/connection
Jun 11 12:10:04: Charging call ...
Jun 11 12:10:04: Zero duration call
Jun 11 12:10:04: Checked 2006-06-11 21:10:04 Europe/Prague against 'hr{20-5}': 1
Jun 11 12:10:04: Start of call is peak level 1
Jun 11 12:10:04: End of call is peak level 1
Jun 11 12:10:04: Can reuse the already initialized charge.
Jun 11 12:10:04: Calculating account's charge by tariff 'SIP Phone Subscribers'
Jun 11 12:10:04: Call to '16046831234' with duration 0 seconds will be charged for
0 seconds and cost is 0 (0s<1s) by rate 101 using 144547388 as seed
Jun 11 12:10:04: Setting up a new charge with tariff 'CT Tariff' ...
Jun 11 12:10:04: Using peak rate, since no off-peak is defined
Jun 11 12:10:04: PrepareNexecute 'GetPricePerDestination*'
Jun 11 12:10:04: SQL query 'GetPricePerDestination*' executed in 0.001555 seconds
Jun 11 12:10:04: Calculating vendor's charge by tariff 'CT Tariff'
Jun 11 12:10:04: Call to '16046831234' with duration 0 seconds will be charged for
0 seconds and cost is 0 (0s<1s) by rate 106 using 144547388 as seed
Jun 11 12:10:04: Updating account usage of '16041234568' with 'Sun Jun 11 12:10:04
2006'
Jun 11 12:10:04: PrepareNexecute 'UpdateAccountUsage'
Jun 11 12:10:04: Charging vendor for the call
Jun 11 12:10:04: Inserting fail CDR
Jun 11 12:10:04: PrepareNexecute 'InsertVendorCDRFail'
```

A failed CDR is inserted for the vendor.

```
Jun 11 12:10:04: Accounting response
Jun 11 12:10:04: ...Done.
```

The call is now established, and the start accounting record is sent.

```
Jun 11 12:10:09: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address            = '193.28.87.106'
User-Name                 = '16041234568'
Called-Station-Id         = '6831234'
Calling-Station-Id        = '16041234568'
Acct-Status-Type          = 'Start'
h323-call-origin          = 'answer'
h323-call-type            = 'VoIP'
h323-setup-time           = '19:09:44.000 GMT Sun Jun 11 2006'
h323-connect-time         = '19:10:09.000 GMT Sun Jun 11 2006'
h323-conf-id              = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id                   = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id           = '98dda488-69c74dd3@192.168.0.250'
Acct-Delay-Time           = '0'
h323-session-protocol     = 'sipv2'
h323-remote-address       = '216.231.44.168'
h323-ivr-out              = 'PortaBilling_Seed:144514807'
alert-timepoint           = '19:10:09.000 GMT Sun Jun 11 2006'
NAS-Port                  = '5060'
Exec-Program-Log          = 'porta-billing.pl'
Jun 11 12:10:09: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:10:09: Found a call in cache with such id
Jun 11 12:10:09: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request
```

The billing re-uses information in the call cache to speed up account info lookup.

```
Jun 11 12:10:09: Fixing time with duration 0 seconds from connect time
Jun 11 12:10:09: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:10:09: Looking up vendor/connection
Jun 11 12:10:09: Trying to match connection for call
Jun 11 12:10:09: Looking for a connection VoIP/answer
Jun 11 12:10:09: VoIP, matching by the node IP '193.28.87.106' and User-Name
'16041234568'
```

```
Jun 11 12:10:09: No VoIP from vendor connections were found
Jun 11 12:10:09: Connection to vendor not found
Jun 11 12:10:09: PrepareNexecute 'InsertActiveLeg'
```

The call has not been billed yet, but an entry is made in the table of active calls.

```
Jun 11 12:10:09: Accounting response
Jun 11 12:10:09: ...Done.
```

A start accounting record about the egress call leg is received.

```
Jun 11 12:10:09: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address       = '193.28.87.106'
User-Name            = '16041234568'
Called-Station-Id    = '16046831234'
Calling-Station-Id   = '16041234568'
Acct-Status-Type     = 'Start'
h323-call-origin     = 'originate'
h323-call-type       = 'VoIP'
h323-setup-time      = '19:10:04.000 GMT Sun Jun 11 2006'
h323-connect-time    = '19:10:09.000 GMT Sun Jun 11 2006'
h323-conf-id         = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id              = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id      = '98dda488-69c74dd3@192.168.0.250'
Acct-Delay-Time      = '0'
h323-session-protocol = 'sipv2'
h323-remote-address  = '70.68.128.186'
h323-ivr-out         = 'DURATION:7200'
h323-ivr-out         = 'PortaBilling_Seed:144514807'
alert-timepoint      = '19:10:08.000 GMT Sun Jun 11 2006'
NAS-Port             = '5060'
Exec-Program-Log     = 'porta-billing.pl'
Jun 11 12:10:09: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:10:09: Found a call in cache with such id
Jun 11 12:10:09: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request
```

The billing re-uses information in the call cache to speed up account info lookup.

```
Jun 11 12:10:09: Fixing time with duration 0 seconds from connect time
Jun 11 12:10:09: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:10:09: SQL query 'GetActiveLegIdByAcct' executed in 0.00115 seconds
Jun 11 12:10:09: Looking up vendor/connection
Jun 11 12:10:09: Trying to match connection for call
Jun 11 12:10:09: Looking for a connection VoIP/originate
Jun 11 12:10:09: Outgoing VoIP, matching by the remote IP address '70.68.128.186'
(env 4)
Jun 11 12:10:09: Found connection 11 'Termination to Globalnet' to vendor
'GlobalNet'
Jun 11 12:10:09: Found vendor/connection
Jun 11 12:10:09: PrepareNexecute 'InsertActiveLeg'
```

The call has not been billed yet, but an entry is made in the table of active calls.

```
Jun 11 12:10:09: Accounting response
Jun 11 12:10:09: ...Done.
```

The call is terminated, and stop accounting for one of the call legs is received.

```
Jun 11 12:11:24: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address       = '193.28.87.106'
User-Name            = '16041234568'
```

```
Called-Station-Id     = '6831234'
Calling-Station-Id    = '16041234568'
Acct-Status-Type      = 'Stop'
h323-call-origin      = 'answer'
h323-call-type        = 'VoIP'
h323-setup-time       = '19:09:44.000 GMT Sun Jun 11 2006'
h323-connect-time     = '19:10:09.000 GMT Sun Jun 11 2006'
h323-disconnect-time  = '19:11:23.000 GMT Sun Jun 11 2006'
h323-disconnect-cause = '0'
h323-voice-quality    = '0'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id               = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Time     = '74'
Acct-Delay-Time       = '0'
h323-session-protocol = 'sipv2'
h323-remote-address   = '216.231.44.168'
h323-ivr-out          = 'PortaBilling_Seed:144514807'
Acct-Terminate-Cause  = 'User-Request'
h323-ivr-out          = 'PortaBilling_Session:unlock'
alert-timepoint       = '19:10:09.000 GMT Sun Jun 11 2006'
NAS-Port              = '5060'
Exec-Program-Log      = 'porta-billing.pl'
Jun 11 12:11:24: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:11:24: Found a call in cache with such id
Jun 11 12:11:24: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request
Jun 11 12:11:24: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:11:24: PrepareNexecute 'DeleteActiveLeg'
Jun 11 12:11:24: Force unlock requested by NAS
Jun 11 12:11:24: Scheduling 16041234568 for logout, call lifetime reduced to 15
Jun 11 12:11:24: Logging out account '16041234568'(103) from 'A9B6252E 3863D707
17A91CC0 744348CB'
Jun 11 12:11:24: Set lifetime with 15s to Sun Jun 11 12:11:39 2006
Jun 11 12:11:24: Looking up vendor/connection
Jun 11 12:11:24: Trying to match connection for call
Jun 11 12:11:24: Looking for a connection VoIP/answer
Jun 11 12:11:24: VoIP, matching by the node IP '193.28.87.106' and User-Name
'16041234568'
Jun 11 12:11:24: Connection to vendor not found
Jun 11 12:11:24: No connection from vendor
```

This is an on-net call leg (while the call is still traveling on the network), so it is ignored.

```
Jun 11 12:11:24: Accounting response
Jun 11 12:11:24: ...Done.
```

Accounting for the second (outgoing) call leg is received.

```
Jun 11 12:11:24: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address        = '193.28.87.106'
User-Name             = '16041234568'
Called-Station-Id     = '16046831234'
Calling-Station-Id    = '16041234568'
Acct-Status-Type      = 'Stop'
h323-call-origin      = 'originate'
h323-call-type        = 'VoIP'
h323-setup-time       = '19:10:04.000 GMT Sun Jun 11 2006'
h323-connect-time     = '19:10:09.000 GMT Sun Jun 11 2006'
h323-disconnect-time  = '19:11:23.000 GMT Sun Jun 11 2006'
h323-disconnect-cause = '0'
h323-voice-quality    = '0'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id               = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id       = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Time     = '74'
Acct-Delay-Time       = '0'
```

```
h323-session-protocol = 'sipv2'
h323-remote-address   = '70.68.128.186'
h323-ivr-out          = 'DURATION:7200'
h323-ivr-out          = 'PortaBilling_Seed:144514807'
Acct-Terminate-Cause  = 'User-Request'
alert-timepoint       = '19:10:08.000 GMT Sun Jun 11 2006'
NAS-Port              = '5060'
Exec-Program-Log      = 'porta-billing.pl'
Jun 11 12:11:24: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:11:24: Found a call in cache with such id
Jun 11 12:11:24: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request
Jun 11 12:11:24: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:11:24: PrepareNexecute 'DeleteActiveLeg'
Jun 11 12:11:24: End of the outgoing call for logged in account. Waiting another
outgoing call or hang up
Jun 11 12:11:24: Set lifetime with 15s to Sun Jun 11 12:11:39 2006
Jun 11 12:11:24: Looking up vendor/connection
Jun 11 12:11:24: Trying to match connection for call
Jun 11 12:11:24: Looking for a connection VoIP/originate
Jun 11 12:11:24: Outgoing VoIP, matching by the remote IP address '70.68.128.186'
(env 4)
Jun 11 12:11:24: Found connection 11 'Termination to Globalnet' to vendor
'GlobalNet'
```

This call leg crosses a connection to the vendor.

```
Jun 11 12:11:24: Found vendor/connection
Jun 11 12:11:24: Charging call ...
Jun 11 12:11:24: Checked 2006-06-11 21:10:09 Europe/Prague against 'hr{20-5}': 1
Jun 11 12:11:24: Checked 2006-06-11 21:11:23 Europe/Prague against 'hr{20-5}': 1
Jun 11 12:11:24: Start of call is peak level 1
Jun 11 12:11:24: End of call is peak level 1
Jun 11 12:11:24: Can reuse the already initialized charge.
Jun 11 12:11:24: Calculating account's charge by tariff 'SIP Phone Subscribers'
Jun 11 12:11:24: Call to '16046831234' with duration 74 seconds will be charged
for 78 seconds and cost is 0.03 (1x30x0.02+8x6x0.02^2) by rate 101 using 144547388
as seed
```

PortaBilling calculates how much the account should be charged for this call.

```
Jun 11 12:11:24: Setting up a new charge with tariff 'GlobalNet Termination' ...
Jun 11 12:11:24: Using peak rate, since no off-peak is defined
Jun 11 12:11:24: PrepareNexecute 'GetPricePerDestination*'
Jun 11 12:11:24: Calculating vendor's charge by tariff 'GlobalNet Termination'
Jun 11 12:11:24: Call to '16046831234' with duration 74 seconds will be charged
for 74 seconds and cost is 0.01234 (1x1x0.01+73x1x0.01) by rate 104 using
144547388 as seed
```

PortaBilling also calculates the termination costs for this call.
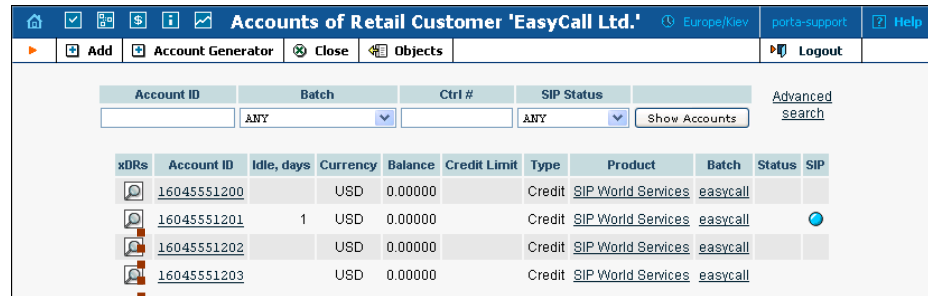
```
Jun 11 12:11:24: Charging account for the call
Jun 11 12:11:24: Inserting CDR
Jun 11 12:11:24: PrepareNexecute 'InsertAccountCDR'
Jun 11 12:11:24: Charging credit account 16041234568 0.03
Jun 11 12:11:24: PrepareNexecute 'UpdateAccountBalance'
Jun 11 12:11:24: Charging account's owner for the call
Jun 11 12:11:24: Charging customer 3 'EasyCall Ltd.' 0.03
Jun 11 12:11:24: PrepareNexecute 'UpdateCustomerBalance'
Jun 11 12:11:24: Charging vendor for the call
Jun 11 12:11:24: Charging vendor 9 'GlobalNet' 0.01234
Jun 11 12:11:24: Inserting CDR
Jun 11 12:11:24: PrepareNexecute 'InsertVendorCDR'
Jun 11 12:11:24: PrepareNexecute 'UpdateVendorBalance'
```

CDRs are inserted and balances are modified.

```
Jun 11 12:11:24: Accounting response
Jun 11 12:11:24: ...Done.
```
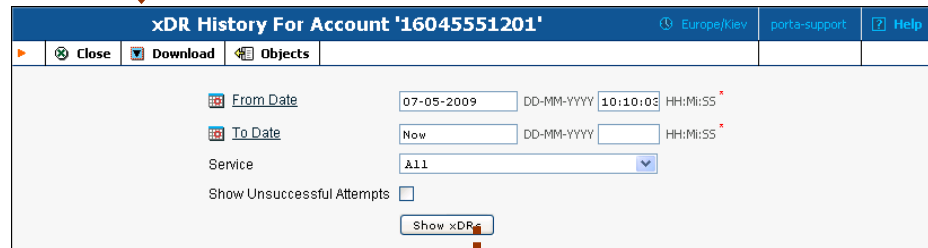
Verify Call History for Account

To view the xDR of an account, go to Customers, select the customer owning the account, and click on the Accounts icon; or, alternatively, select the Account Info link from the Main Menu.

You can also view the xDR history when going to the account self-care page (accessible via the **Accounts** menu item in the **Home** popup menu). The only difference is that account's owner does not have the option of seeing incomplete (failed) calls when using the self-care pages. Login with the account's web access login and password. After that you will be able to see the account's dashboard interface with functional drop-down menus upper left. On the **Statisitcs** menu, click **xDR Browser**.

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

Check Call History

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

- ▪ In the Helpdesk section of Admin-Index, click **Trace a Call**.
- ▪ Fill in the check phone number form:
  - o **h323-conf-id** – if you need to trace a specific call, enter h323-conf-id here; otherwise this leave empty.
  - o **Destination** – the phone number you are looking for or a destination pattern (first digits and a percent sign, for example **380%**).
  - o **From, To Date** – the date range. Click on the "10 min." icon to limit the time interval to the last 10 minutes only.
- ▪ Click **Trace Call.**

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

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

2. Setting up PSTN-to-SIP Services

This chapter will demonstrate how to set up a PSTN-to-SIP service so that your customers will be able to receive calls on their SIP phones when someone dials their number from a fixed-line or mobile phone.

Please refer to the **PortaBilling100 Web Reference Guide** PDF file for detailed instructions on how to navigate and operate the web interface, as well as detailed explanations of particular fields.

Incoming DID Calls (from PSTN)

If you receive calls from your vendor via a PRI interface (E1/T1), you will need to send them to a VoIP gateway (such as Cisco or Quintum) and then forward them to PortaSIP.

The simplest and the most efficient way to set this up is to **not** define this gateway as your node. Instead, simply describe it as a "VoIP from Vendor" connection, as explained in the next chapter.

However, if you do need to connect this gateway to PortaBilling as a node (e.g. you are running some other services there, such as prepaid cards), please follow the instructions in the "Incoming DID calls (from PSTN)" section in older versions of this guide (MR18 and older).

Incoming DID Calls (from VoIP Vendor)

Checklist

Print the following page and use it to check off operations you have completed while performing system setup according to the instructions in this chapter. Please make sure that you perform all of the operations (all of the boxes must be checked), otherwise the service will not work.

The checklist below assumes that you have already configured the basic SIP service according to the instructions provided in the earlier chapter.

Operation	Done
Initial configuration	
General and network configuration has already been performed according to the instructions in the *Basic SIP Service* section.	[ ]
Basic SIP service configuration has already been performed according to the instructions in the *Basic SIP Service* section, and you are able to make SIP-to-SIP calls.	[ ]
Rating configuration	
Create a tariff A, which will apply to SIP subscribers for their incoming calls.	[ ]
Enter rates in tariff A for the destinations covering your incoming DID range, i.e. the rates you plan to charge your customers.	[ ]
Create a tariff B, which describes the costs of incoming DIDs from this DID provider (it should not be of the **Routing** type).	[ ]
Enter rates in tariff B for the incoming DID range.	[ ]

Create a new accessibility entry in your SIP product, using the PortaSIP node, tariff A and the access code INCOMING.	[ ]
Create a vendor for the DID supplier.	[ ]
Create a "VoIP from Vendor" connection for this vendor using tariff B.	[ ]

Account provisioning

Check that the retail SIP service customer is already created according to the instructions in the *Basic SIP Service* section.	[ ]
Create a new account for this customer, with the account ID identical to the SIP phone number (DID).	[ ]

Testing

Program the parameters (phone #, password, SIP server address) into the SIP phone and make a test call.	[ ]

Set up PSTN-to-SIP Service

Configure the basic SIP service according to the instructions in the previous section, and ensure that you are able to make calls between SIP phones (i.e. your SIP phones can receive incoming calls).

Create Tariff to Charge Your Subscribers

The tariff is a single price list for calling services; in this case, for the incoming call service for your customers.
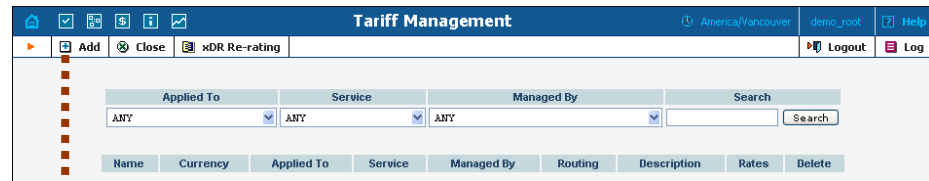
1. In the Management section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, choose ⊞ **Add**.
3. Fill in the **Add Tariff** form:
    - **Name** – A short name for the tariff object; this is the name you will then see in the select menus.
    - **Currency** – Indicates 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.

    - **Applied To** – Choose **Customer** here, since this tariff will be applied to your subscribers.
    - **Managed By** – Choose "**Administrator Only**" here (this option is only visible after you select **Applied to: Customer** above).
    - **Service Type** – Choose **Voice Calls** here.
    - **Off-peak Period** – Defines the off-peak period. Click on the Off-peak period wizard icon (![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.

- o **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).
- o **Post-Call Surcharge** – Percentage of the amount charged for the call.
- o **Login Fee** – Amount to be charged immediately after the first user authentication (i.e. after the user enters his PIN).
- o **Connect Fee** – Amount to be charged for each connected call (call with a non-zero duration).
- o **Round Charged Amount** – Instead of calculating xDRs with a 5-decimal-place precision, round up xDR 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.
- o **Default 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.
- o **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.
- o **Description** – An extended tariff description.
4. Click 💾 **Save**.

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:
    o   **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.

    o   **Interval First** – first billing unit in seconds
    o   **Interval Next** – next billing unit in seconds
    o   **Price First** – per minute price for first interval
    o   **Price Next** – per minute price for next interval
    o   **Off-peak Interval First**– first billing unit in seconds for off - peak time
    o   **Off-peak Interval Next** – next billing unit in seconds for off- peak time
    o   **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   **Rate Formula Wizard** 🎨 – 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.

Uploading a rate list from a file

Please consult the instructions provided in the earlier chapters and **PortaBilling Templates Guide** for more information regarding this.

Create Tariff for Incoming DID Costs

A tariff is a single price list for calling services; in this case, for calculating how much your DID provider will charge you for delivering incoming calls to your network.
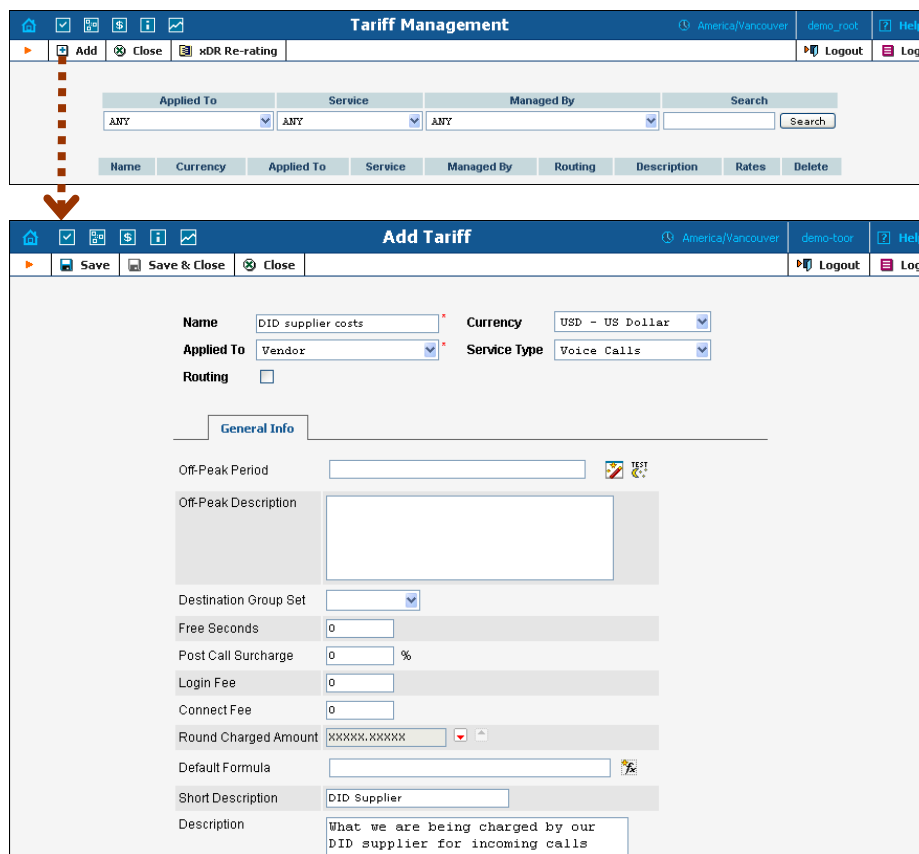
1. In the Management section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, choose ⊞ **Add**.
3. Fill in the **Add Tariff** form (please refer to the previous section for a detailed description of its elements). In the **Applied To** select menu choose **Vendor** then clear the check-box **Routing**, since no routing is actually done for this vendor, i.e. the vendor will be sending calls to your network.

4. Click ![save] **Save**.

Enter Rates

Enter the rates applied to you by the DID provider.

1. On the Tariff Management page you will see a list of the available tariffs. Click the ![icon] **Rates** icon next to the name of the tariff. When you have entered Tariff Management for a particular tariff, click on **Rates** in the toolbar.
2. On the **Edit Rates** screen, click ![add] **Add**.
3. Fill in the required information; please refer to the previous section for a detailed description of the available columns.
4. Click the ![save] **Save** button in the toolbar, or the ![icon] icon on the left end of the row.
5. Repeat these steps if you need to enter more rates.

Modify Accessibility for Product

Now you must specify when the tariff you have created to charge your customers for incoming calls will actually be applied to them.

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

1. Click on the product name to enter the Edit Product screen.
2. After selecting the Accessibility tab, click the ⊞ **Add** icon.
3. Choose **Voice Calls** in the **Service** select menu.
4. In the **Node** select menu select the PortaSIP.
5. Type INCOMING in the **Access Code** column.
6. **Info Digits** do not apply to this type of service (i.e. you do not differentiate between calls your customers receive from fixed or mobile networks). Therefore, just leave this field empty.
7. In the **Default Tariff** menu, select the tariff which applies to your customers for receiving incoming calls.
8. Click ⊟ **Save&Close** to save this accessibility entry.

Create DID Supplier Vendor

This entity is required in order to keep track of your incoming DID expenses, and also to provide an adequate level of security on your network. Since every incoming call to your network must be authorized, you can create an account under the vendor which will be used for such authorization.

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 refer to the instructions provided in the *Basic SIP Service* chapter for a detailed description.
4. Click 🖫 **Save**.
5. Click the Accounts tab.
6. Click ⊞ **Add** to enter a record for the account which will be used by this vendor to send calls to your network.

7. Enter the following values:
   - **Name** – A short name for this account information (visible when associating an account with a connection).
   - **Login** – Choose a username to be used by the vendor's proxy or gateway to access your network. If the vendor does not support digest authentication, and you are using IP-based
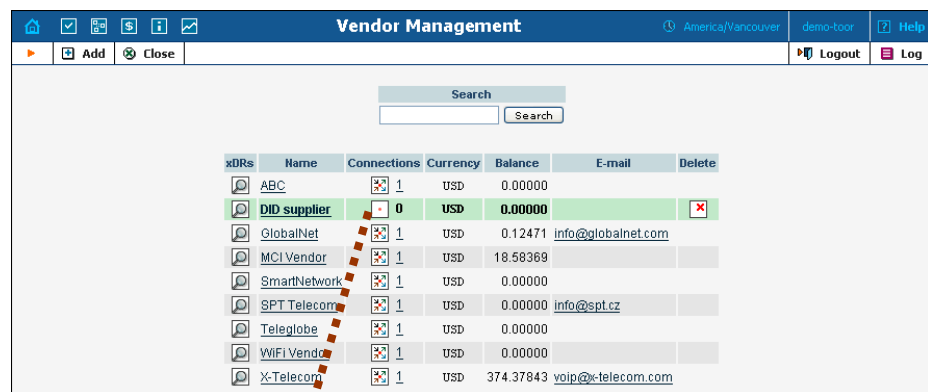
authentication, enter the IP address of the vendor's gateway here and use **cisco** as the password.

- **Password** – Choose a secure password (difficult to guess or crack) for the vendor's account.

8. Click the 🖫 **Save** button in the toolbar, or the 🖫 icon on the left end of the row.
9. Click **Close** in order to return to the **Vendors** admin page.

Define Connections

Connections are points at which calls leave or enter a network. They are directed to or from vendors, whereby costing occurs. In this case, a connection is the point where calls enter your network via the PSTN gateway.

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

3. Press 🞣 **Add** to add a new connection.

4. Fill in the connection information. Choose VoIP from Vendor as the **Type** of connection and your PortaSIP node as the **Node**, then select the tariff which defines your incoming DID costs for this vendor. In the **Vendor Account** select menu, choose the account you previously created in the vendor configuration; this account will be used to authorize incoming calls from the vendor. In **RTP Proxying,** specify this vendor's capabilities regarding NAT traversal (if you are not sure whether this vendor can do NAT traversal, select **OnNAT**). **Description** and **Capacity** are mandatory for all connection types.
5. Click 💾 **Save**.

Create Accounts

1. Go to the **Customers** screen (the one containing a list of customers). It should resemble the screenshot below:
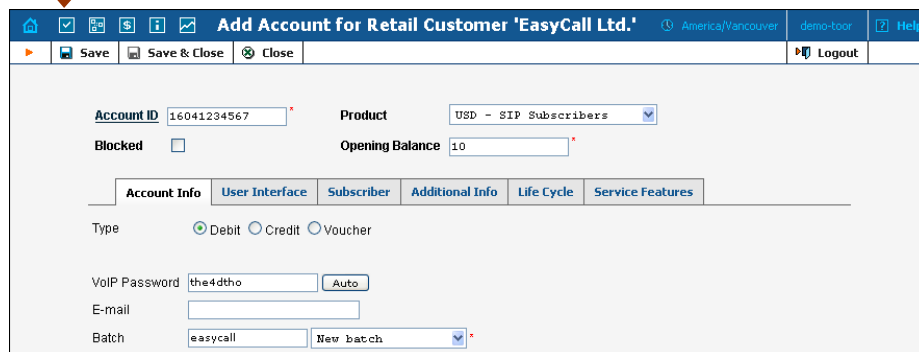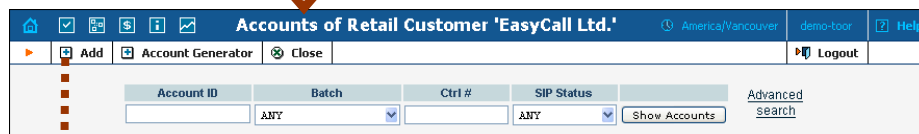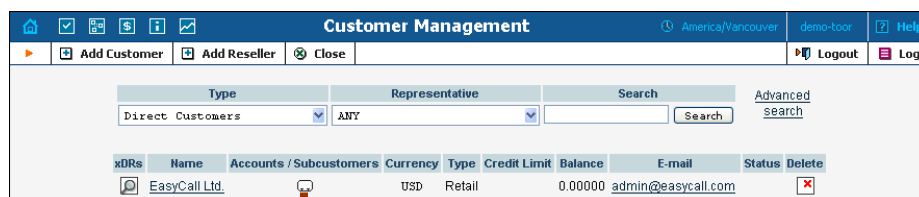
2. Next to the customer name, click the 🖥 icon (the one in the **Accounts** column) to go to account management for that customer.
3. Click on ➕ **Add**.
4. Fill in the Add Account form:
   o **Account ID** – SIP ID, i.e. the phone number which will be used to login to the SIP server and receive incoming calls.
   o **Product** – Choose the product you would like your account to have.
   o **Blocked** – You may create the account as blocked, although this is rarely done with SIP service accounts.

o   **Opening Balance** – The initial balance on the account.

Account Info tab

o   **Account Type** – Select credit for postpaid, and debit for prepaid service.
o   **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).
o   **VoIP Password** – This password is used for SIP services as well. Together with the account ID, this password will be used to authenticate SIP server login.
o   **Email** – Enter the account owner's email address here. If he ever forgets his password for the web self-care pages, the password may be reset, and a new one will be sent to this email address. You can also just leave this field empty.
o   **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

o   **IP Phone, IP Phone Port** – Leave these fields empty for now and use manual provisioning instead. SIP phone auto-provisioning is described in the *Setting up Auto-provisioning of IP* chapter.

Life Cycle tab

o   **Activation Date** – Account activation date.
o   **Expiration Date** – Account expiration date; since we are setting up a postpaid service, which should function for a long time, leave this field blank.
o   **Lifetime** – Relative expiration date; since we are setting up a postpaid service, which should function for a long time, leave this field blank.

User Interface tab

o   **Login** – Account login to web self-care pages. This may be identical to account ID.
o   **Password** – Password for the web self-care pages.
o   **Time Zone** - When an account owner (prepaid card user) accesses the web self-care pages to see a list of his calls, the time may be shown in the time zone most appropriate for him.
o   **Web Interface Language** – The language to be used on the customer self-care web interface.

Service Features tab

Using this tab, you can activate/deactivate various features of the services provided to this account. Note that features are defined per *service type* (the physical service provided to the user such as Conferencing, Data Service, Voice Calls, Messaging etc.)

Please consult the **PortaBilling Web Reference Guide** for a description of parameters available here. For now you may leave these with their default values.

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

3. Calls Between SIP Phones

Rating for Calls Between SIP Phones

Checklist

Print the following page and use it to check off operations you have completed while performing system setup according to the instructions in this chapter. Please make sure that you perform all of the operations (all of the boxes must be checked), otherwise the service will not work.

The checklist below assumes that you have already configured the basic SIP service according to the instructions provided in the earlier chapters.

Operation	Done
Initial configuration	
General and network configuration has already been performed according to the instructions in the *Basic SIP Service* section.	[ ]
Basic SIP service configuration has already been performed according to the instructions in the *Basic SIP Service* section, and you are able to make SIP-to-SIP calls.	[ ]
Rating configuration	
Enter rates in tariff A (which applies to your customers' outgoing calls) for the special destinations describing phone-to-phone calls, i.e. the rates you plan to charge your customers for on-net calls.	[ ]
Enter rates in tariff B (which defines the costs of on-net calls, and is associated with the SIP-UA connection) for the special destination describing phone-to-phone calls.	[ ]
Make sure that tariff A is assigned to the PortaSIP node with the access code OUTGOING in your SIP product's accessibility list.	[ ]
Account provisioning	
Check that the retail SIP service customer has already been created according to the instructions in the *Basic SIP Service* section.	[ ]
If you wish, enable the VoiceVPN feature for this customer.	[ ]
Ensure that SIP accounts have already been created for this customer (if not, follow the instructions in the *Basic SIP Service* section).	[ ]
Testing	
Program the parameters (phone #, password, SIP server address) into the SIP phone and make a test call.	[ ]

Set up Basic SIP Service

Configure the basic SIP service according to the instructions in the previous sections.

Modify Tariff to Charge Your Subscribers

In this tariff you will specify rates for outgoing calls made by your customers to both on-net (SIP phones on your network) and off-net (calls to landlines, mobile networks and foreign countries) destinations.

1. In the Management section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, click on the ▦ icon in the table row containing your tariff for outgoing SIP calls.

3. On the **Edit Rates** screen, click ⊞ **Add**.

4. Fill in the required information
   - **Destination** – In this case there are two destination prefixes you will use. VOICEVPN refers to all calls made to IP phones of the **same** customer (within a single IP Centrex environment). VOICEONNET specifies calls made to any IP phone within this environment.
   - The rest of the parameters are identical to those you used when setting up the basic SIP service.
5. Click the 🖫 **Save** button in the toolbar, or the 🖫 icon on the left end of the row.

Uploading a rate list from a file

Please consult the instructions provided in the earlier chapters and **PortaBilling Templates Guide** for more information regarding this.

Tariff for SIP-to-SIP Call Costs

You have probably already created this tariff when setting up the basic SIP service. This tariff contains your on-net call cost information, which normally will be zero, as you are not actually being charged for these calls by any telco. However, you may wish to calculate the approximate cost of such calls based on the amount of bandwidth being used on your network, and enter it here for better internal cost tracking.

1. In the Management section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page you will see a list of the available tariffs. Click the ☰ **Rates** icon next to the name of a tariff. When you have entered Tariff Management for a particular tariff, click on **Rates** in the toolbar.
3. On the **Edit Rates** screen, click ➕ **Add**.

Porta Switch

4. Fill in the required information; use VOICEONNET as the **Destination**. Please refer to the previous section for a detailed description of the available columns.
5. Click the 💾 **Save** button in the toolbar, or the icon on the left end of the row.
6. Repeat these steps if you need to enter more rates.

Modify the Accessibility for Product

Now you must specify when the tariff you have created to charge your customers for outgoing calls will actually be applied to them. (Again, there is a single tariff for all outgoing calls, both on-net and off-net.)

1. In the Management section of the Admin-Index page, choose **Products**.
2. Click on the product name to enter the Edit Product screen.
3. After selecting the Accessibility tab, click the **Add** icon.
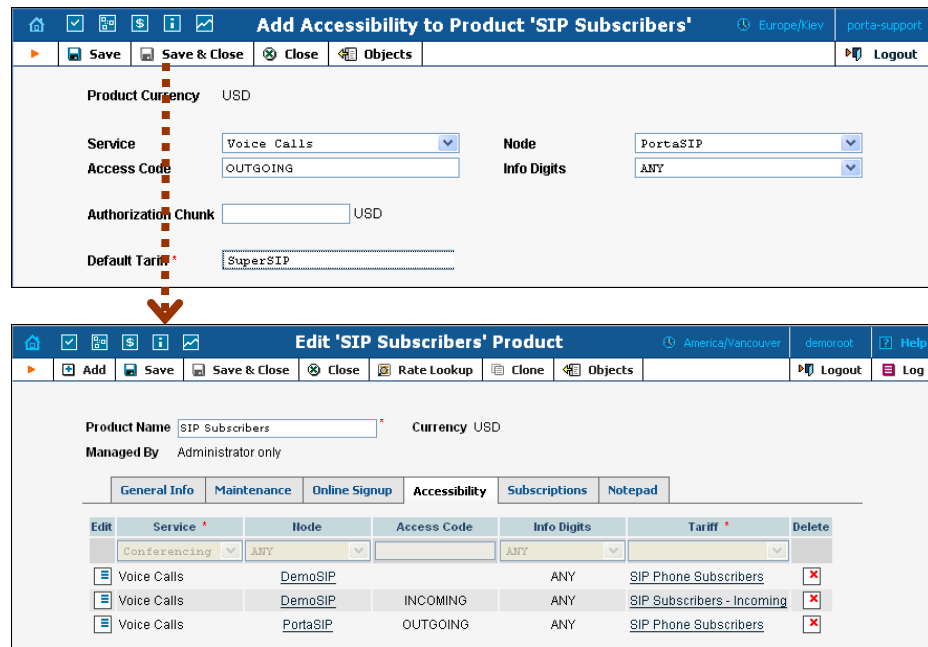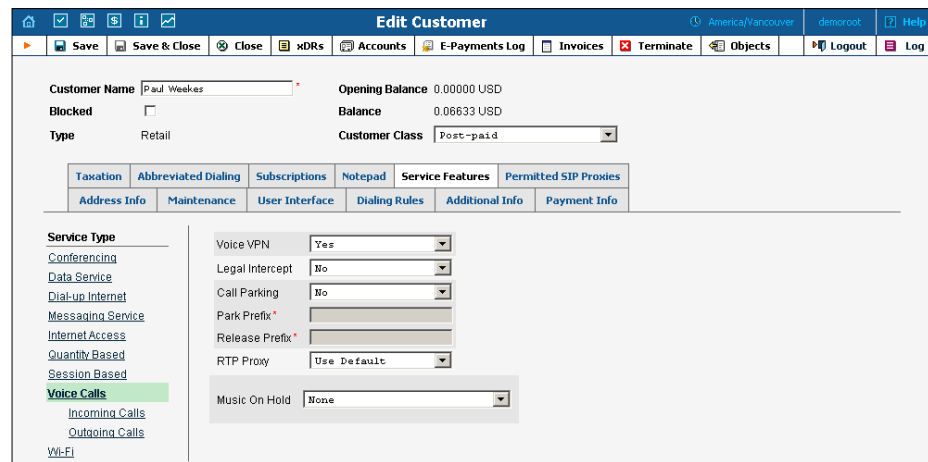
4. Choose **Voice Calls** in the **Service** select menu.
5. In the **Node** select menu, select the PortaSIP node.
6. Type OUTGOING in the **Access Code** column.
7. **Info Digits** do not apply to this type of service (i.e. you do not differentiate between calls your customers receive from fixed or mobile networks). Therefore, just leave this field empty.
8. In the **Default Tariff** menu, select the tariff which applies to your customers' outgoing calls.
9. Click 🖫 **Save&Close** to save this accessibility entry.

Enable Voice VPN Feature for Customer (Optional)

If you do not enable this feature for a particular customer, all of his on-net calls will be charged according to the VOICEONNET rate. If this feature is enabled, you may use a special rating for calls made between the customer's IP phones.

1. In the Management section of the Admin-Index page, choose **Customers**.
2. On the Customer Management page, click on the customer name to edit the customer data.
3. Select "Voice Calls" in the **Service Type** list on the left side.
4. Select **Yes** for the **VoiceVPN** feature.
5. Click 🖫 **Save&Close**.

4. Setting up Auto-provisioning of IP Phones

This chapter demonstrates how to set up SIP phone auto-provisioning so that you can easily control the parameters and configuration of thousands of IP phones from the PortaBilling100 web interface, without having to provision each unit individually.

Please refer to the **PortaBilling Web Reference Guide** PDF file for detailed instructions on how to navigate and operate the web interface, as well as detailed explanations of particular fields.

Setting up Auto-provisioning of IP Phones

The following section will assist you in setting up automatic provisioning of IP phones by PortaSwitch, so that every phone will automatically download all the required configuration parameters, such as phone number or password, from the provisioning server.

Checklist

Print the following page and use it to check off the operations you have completed while performing system setup according to the instructions in this section. Please make sure that you perform all of the operations (all of the boxes must be checked), otherwise the service will not work.

Operation	Done
General configuration	
Create a new IP phone profile for the required type of IP phone	[ ]
In the IP phone profile, fill in all the required parameters (e.g. SIP proxy address)	[ ]
IP phone inventory	
Create a record for your IP phone in the IP phone inventory, making sure to enter the correct MAC address	[ ]
Assigning a phone number to the IP phone	
Make sure your SIP service is provisioned according to the instructions given above in this guide	[ ]
Create a new account with the product you allocated for the SIP service, with an account ID identical to the phone number	[ ]
Assign this account for provisioning on a certain IP phone	[ ]
IP phone settings	
Connect the IP phone to the Internet	[ ]
If the phone has not been pre-configured for your provisioning server by the vendor, enter provisioning information into the phone manually	[ ]
Wait until the configuration files are updated on the provisioning server	[ ]
Testing	
After your phone downloads the configuration from the server and successfully registers on the SIP server for the first time, you will receive a "first login greeting" call	[ ]
Make a test phone call	[ ]

Create IP Phone Profile

1. In the Management section of the Admin-Index page, choose **IP Phone Profiles.**
2. In the IP Phone Profile management window, click the ✚ **Add** icon.
3. Fill in the Add IP Phone profile:
    *   **Name** – A short descriptive name for this profile.
    *   **Managed By** – If you plan to use this profile for a certain reseller's customers, choose the reseller from the select menu; otherwise leave this as **Administrator Only**.
    *   **Type** – The hardware type of the IP phone.
    *   **As Copy Of** – This will allow you to create new profiles based on already existing ones; for now, leave this as **None**.
    *   **Effective From** – Leave the value in this field as **immediately**.
4. Click 💾 **Save.**

IP Phone Profile Settings

After clicking 💾 **Save** on the previous page, you will go to the Profile Edit page, where you can edit the generic device settings. These configuration parameters are dependent on the specific model of your IP device. The example below uses a Sipura-2000 device, but most of the settings should be the same for other Sipura/Linksys VoIP products.

Field	Description
Enable Web Server	Enable/disable the built-in web server (for device monitoring)
Enable Web Admin Access	Enable/disable the built-in web server (for device administration and changing configuration parameters)
Admin Password	The password for administrator access
User Password	The password for user access
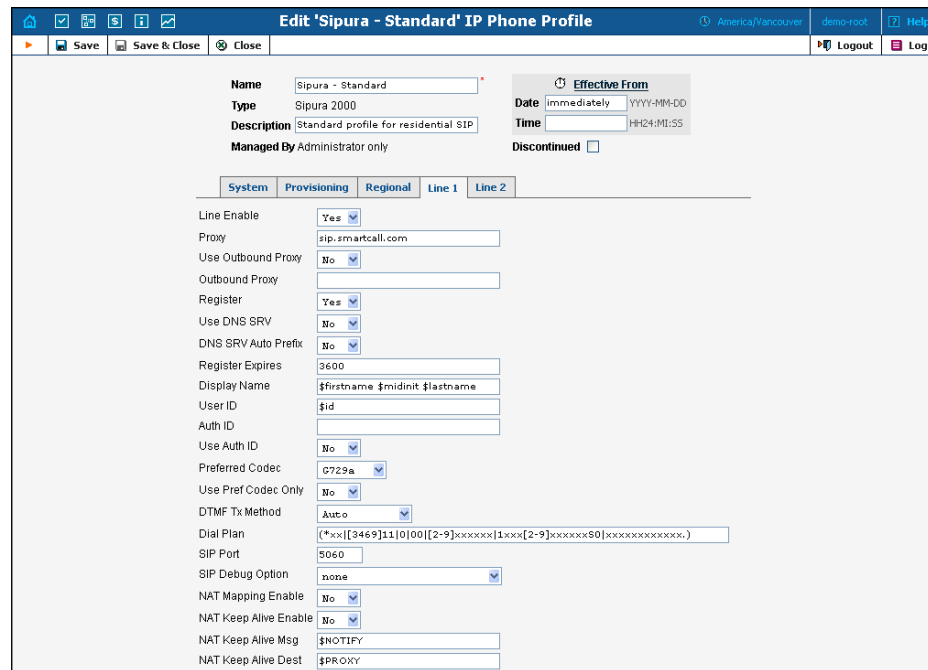
Field	Description
Profile Rule	Path to the profile configuration file. Replace PB_SLAVE_SERVER with the actual hostname or IP address of your PortaBilling100 slave server; for instance, the actual value in the field might be: [--

	key $B]http://demo.portaone.com/$A/$MA.cfg. The example on the screen assumes provisioning via the HTTP protocol; change `http:` to `tftp:` for TFTP provisioning. When an individual IP phone will make a request for a configuration file, $MA will be automatically replaced by it's MAC-address, so leave it as is (otherwise if you put the actual MAC address in, you cannot reuse the same profile for multiple phones).
GPP A, GPP B, GPP C	Dynamic variables. During config file generation, $i_env will be replaced by a unique environment ID, and $ascii_key by the crypt key from the IP phone. GPP A corresponds to $A variable in the **Profile Rule** field, GPP B to $B, etc.

There is no need to change any of the values for dynamic variables.

In the **Line 1** and **Line 2** tabs, you can specify parameters for both phone ports of your IP phone.

Field	Description
Proxy	SIP server for registration and placing outgoing calls.
Outbound Proxy	SIP Server for placing outgoing calls (if different from the SIP registration server).

Use Outbound Proxy	If set to "no", the SIP server defined by the Proxy parameter will be used for all registrations and outgoing calls.
Register	Whether or not the IP phone should register with the SIP server; this is required to receive incoming calls.
Register Expires	Registration lifetime in seconds (the Expires value in a REGISTER request). The IP phone will periodically renew registration shortly before the current registration expires.
Display Name	The subscriber's display name, which appears in the caller-id for his outgoing calls; by default First Name + M Name + Last Name.
User ID	The phone number assigned to this port. The dynamic variable $id will be automatically replaced by the account's ID (phone number).
Auth ID	Authentication ID (if different from the phone number).
Use Auth ID	If set to "yes", the combination Auth ID and Password will be used for SIP authentication. Otherwise, User ID and Password are used.
Preferred Codec	Select a preferred codec for all calls. However, the actual codec used in a call will still depend on the outcome of the codec negotiation protocol. Possible values: G711u, G711a, G726-16, G726-24, G726-32, G726-40, G729a, G723.
Use Pref Codec Only	Only use the preferred codec for all calls. The call will fail if the other endpoint does not support this codec.
DTMF Tx Method	Method for transmitting DTMF signals to the far end: Inband = Send DTMF using audio path; INFO = Use the SIP INFO method, AVT = Send DTMF as AVT events; Auto = Use Inband or AVT based on outcome of codec negotiation.
Dial Plan	Per-line dial plan script; see the product manual for a detailed description of the syntax.

Create IP Phone Entry

1. In the Management section of the Admin-Index page, choose **IP Phone Inventory.**
2. In the IP Phone Inventory management window, click the ⬆ **Add** icon.
3. Fill in the Add IP Phone form:
   - **Name** – A unique ID for the IP phone.
   - **Type** – The hardware model of the phone.
   - **Profile** – The IP phone profile you previously created.
   - **MAC Address** – The hardware ID of the IP phone (typically printed on the back of the device), 6 hexadecimal numbers, separated by colons. Make sure you enter the full value of the MAC address, including the colons!
   - **Ports** – How many phone lines are available on this device.
4. Click ⬅ **Save & Close**.

Provisioning an Account on an IP Phone

1. Open the Account Info form for an existing account, and go to the **Additional Info** tab:

2. Select the **IP Phone** from the list.
3. Choose a specific port from the **IP Phone Port** select menu if the device has multiple phone ports.
4. Click ![save icon] **Save & Close**.

Note: The **IP Phone** select field shows a list of phones that have not been used before in other accounts, or phones with available (unused) ports.

After the automated script creates a new configuration file, it will place it on the provisioning server. The file generation process runs every hour (or every time you update a specific IP phone profile).

All device configuration files are stored in */home/porta-admin/apache/htdocs/* on the PortaBilling100 slave server, in subdirectories with a name identical to the environment ID (i_env): 1, 2, 3, and so on.

IP Phone Device Configuration

If your IP phone/ATA was pre-configured by the vendor to retrieve a configuration from your provisioning server, you need only connect the phone to the Internet. Otherwise, you must enter information regarding the provisioning server into the IP phone manually:

1. Connect the Sipura device to the LAN, and find the IP address assigned to it by DHCP: connect an analog phone to phone port 1 and then dial **** on the phone, so that you enter the configuration menu. Dial 110# and listen to the IP address announced.
2. On the PC connected to the same LAN as the Sipura device, open a new web browser window and enter the URL http://<IP>, where <IP> is the IP address which was announced to you by the IVR.
3. Click on **Admin login**, then go to **Advanced** view and the **Provisioning** tab.

4. Enter the provisioning information.
    *   In the **Profile Rule** field, enter:

        ```
        [--key $B]http://PB_SLAVE_SERVER/$A/$MA.cfg
        ```

        (replace PB_SLAVE_SERVER with the actual hostname or IP
        address of your PortaBilling100 slave server; for instance, the
        actual value in the field might be: [--key
        $B]http://demo.portaone.com/$A/$MA.cfg).
    *   In the **GPP A** field, enter the numeric ID of the environment
        (i_env).
    *   If you plan to use encryption for the configuration files, enter the
        secret key in the **GPP B** field.
5. Apply the changes by clicking **Submit All Changes** at the bottom of
    the page. After reboot, the device will request the configuration file
    from the server via the http protocol, and all profile parameters will
    be applied.

Advanced Provisioning Tips

In order to understand provisioning in depth, you have to know
something about the internal processes of configuration file generation.

Auto-provisioning is a built-in function of an IP phone, allowing the
device to download its configuration from an external server via the

HTTP or TFTP protocols. Different IP phones use a different set of configuration parameters and a different format for the configuration file. In addition, many IP phone manufacturers require the configuration file to be processed by some proprietary utility (profile compiler) before it can be supplied to the IP phone.

PortaBilling initially processes each account with an associated IP phone and creates a parameter-value plain-text file located in `/usr/home/porta-admin/profile`. This file then becomes the source for the manufacturer's configuration compiler. The result of the compilation (i.e. the file to be downloaded by the IP phone) is placed in `/usr/home/porta-admin/apache`. For convenience in the initial setup, `/usr/home/porta-admin/apache/htdocs` is the root directory for the default web host, so any IP phone can access its configuration file just by sending a request to the http-port of the PortaBilling100 slave server.

For phones that do not support http provisioning, you may use the TFTP protocol. The main disadvantage of TFTP is that it has a higher chance of being blocked by a firewall. There is no support for subdirs in the TFTP provisioning server path, so all provision configs are stored without a $env-subfolder in */home/porta-admin/apache/htdocs*, unlike other UA profile configs.

You may change the default location for storing configuration files:
1. To change the http-root dir, edit the `DocumentRoot` parameter for port 80 in the Apache host's configuration file:
    ```
    /usr/local/etc/apache/porta.httpd.conf:
            <VirtualHost _default_:80>
              DocumentRoot "/home/porta-admin/apache/htdocs/"
              Options ExecCGI
              DirectoryIndex index.pl
            </VirtualHost>
    ```

Note: If you change this value you must make corresponding changes in an additional list of configuration files. Do not forget to restart the Apache server afterwards.

2. To change the PortaBilling output directory for compiled profiles, edit the section `[UA_Profiles] ResultDir` parameter in the `/usr/home/porta-admin/etc/porta-admin.conf` file.
3. To change the PortaBilling result dir for intermediary (non yet compiled) text files, edit the section `[UA_Profiles] Dir` parameter in the `/usr/home/porta-admin/etc/porta-admin.conf` file.
4. To change the tftp-server root path for devices which only have TFTP-protocol support, edit the `/etc/inetd.conf` file and the -s inline parameter for tftpd-server startup. Do not forget to restart inetd afterwards.

5. Appendices

APPENDIX A. Client's Sipura Configuration for PortaSIP

1. First, you need to know the SPA IP address. Via a touchtone telephone attached to the phone port on the SPA, press the star key four times (****). Then type 110# and the IP address will be announced.
2. Run a Web browser application on the same network as the SPA. Open a session in the SPA by typing http://<spa ip address>/admin/advanced.
3. Choose the specific phone port (click on **Line 1**, **Line 2** or another tab).
4. Provide values for the required parameters, which include:
    a. in **Proxy and Registration**:
        i. **Proxy** – PortaSIP address (or hostname)
        ii. **Register** – yes
    b. in the **Subscriber** information part:
        i. **Display Name** – your identification (e.g. John Doe; this will be seen by the called party)
        ii. **User ID** – SIP account ID
        iii. **Password** – VoIP password for your SIP account
        iv. **Use Auth ID** – no
5. Submit all the changes and update the SPA configuration.

Porta✔**Switch**

APPENDIX B. SJLabs Softphone Configuration for PortaSIP

1. First, you need to install the SJPhone on your machine. Following installation, launch the SJPhone software. The following login screen will be displayed.

2. Key in the account ID and password for PortaSIP and press OK. The SJPhone display should be similar to the one in the illustration below, showing the account balance in "Ready to call" state. The phone is now ready to be used.
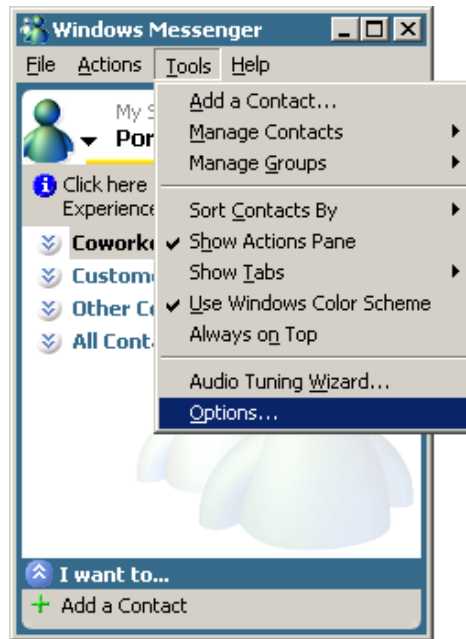

3. Right click on the softphone and press "Login…" to change or make corrections to the account/password.

APPENDIX C. Configuring Windows Messenger for Use as a SIP User Agent

The following instructions apply to Windows Messenger version 5.0.
1. Start Windows Messenger, and select "Options…" from the "Tools" menu
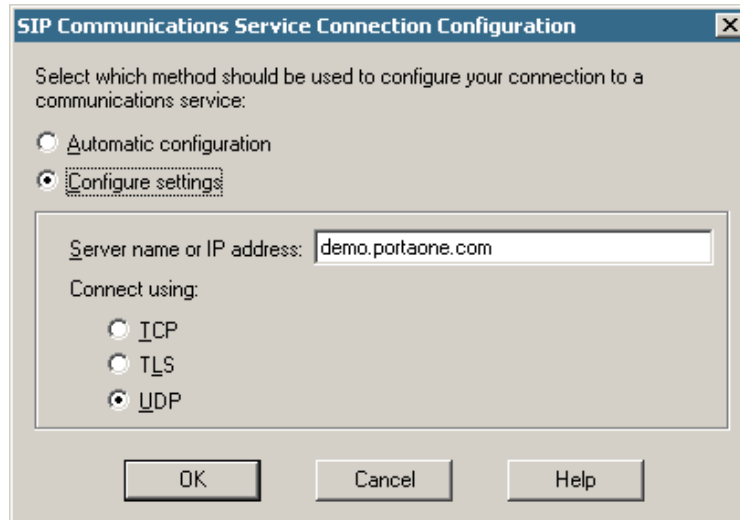
2. Check the "My contacts include users of a SIP Communication Service" check box. Enter your "Sign-in name" as shown, in the form *username@address*, where *username* is the name of the appropriate account in PB and *address* is either the IP address of the PortaSIP server or its name in DNS. Then click the "Advanced…" button.
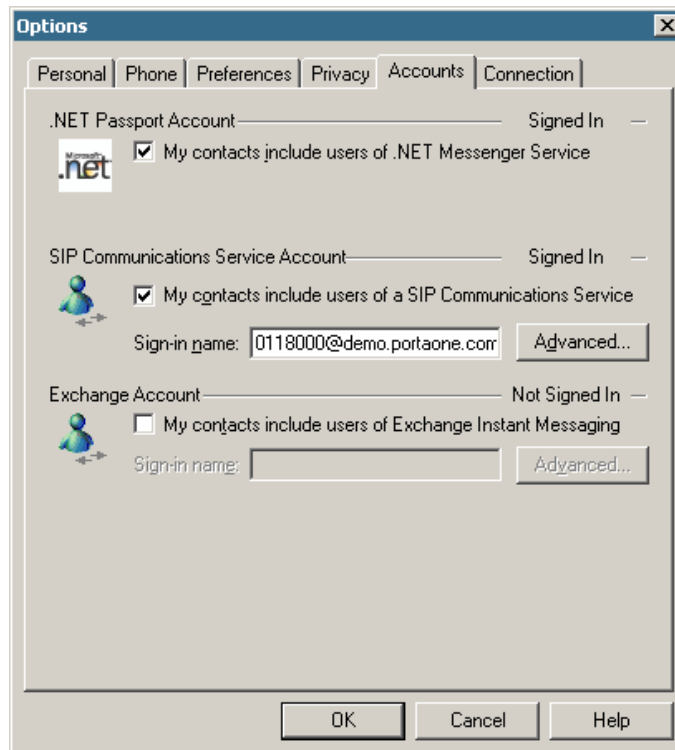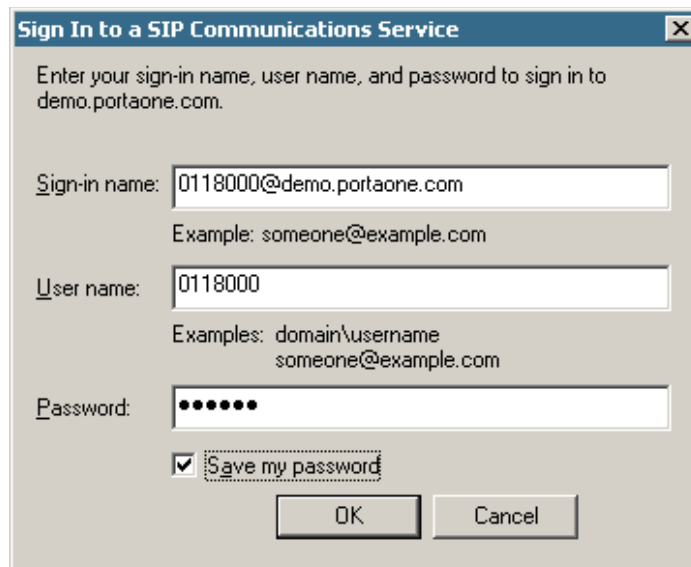
3. Click the "Configure settings" radio button and enter the "Server name of IP address" using either the IP address of the PortaSIP server or its name in DNS. Make sure that the "UDP" radio button is selected, then click OK.


4. Sign out and then sign in again. You should see the pop-up dialog below. Fill it in as follows: "Sign-in name" in the form *username@address*, where *username* is the name of the appropriate account in PB and *address* is either the IP address of the PortaSIP server or its name in DNS. Enter the name of the appropriate PB account as the "User Name" and the appropriate account password as the "Password", then click OK. You should now see your status change to online.

5. To make a call, click the "Action" item in the main menu, then select "Start Voice Conversation". Click the "Other" tab, making sure that "Communications Service" is selected in the drop-down Service box, and enter the phone number in the "Enter e-mail address:" field, as shown below. Finally, click "OK" to place a call.
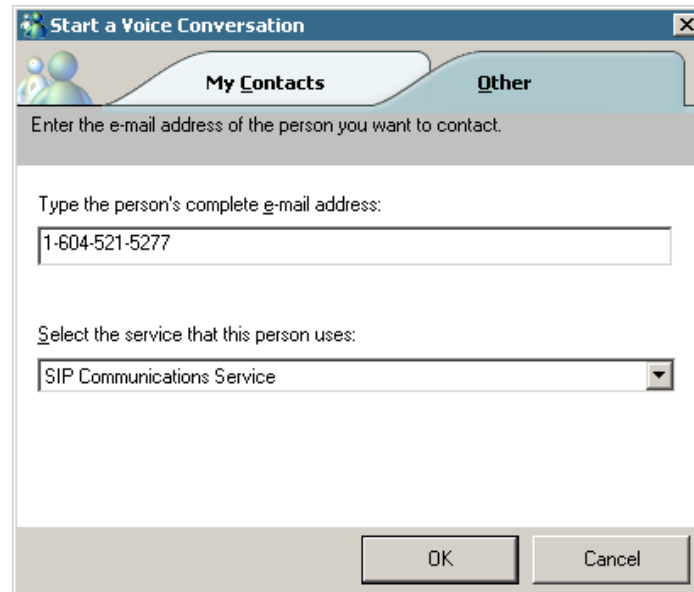
APPENDIX D. Auto-provisioned IP Phones and Adapters

Currently the following IP devices can be auto-provisioned via PortaSwitch:

- Cisco ATA 186 (firmware versions 2 and 3)
- Sipura 1001
- Sipura 2000
- Sipura 2100
- Sipura 3000
- Linksys PAP2
- Linksys RTP-300
- Linksys/Sipura SPA-2102
- Linksys SPA-941
- Linksys SPA-962
- Linksys WRT54GP2
- GrandStream GXW400x
- GrandStream HT286
- GrandStream HT486
- GrandStream HT488

- GrandStream HT496

We are constantly working to extend the list of supported IP devices. If the IP phone you plan to use is not listed here, please contact us – it may already be scheduled for a future release, or we may include it at your request.