PORTA ONE

Porta Switch®

Handbook:

Residential VoIP Services
Maintenance Release 23

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Preface

This document provides PortaSwitch® users with the most common examples and guidelines for setting up a VoIP network.

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`

The **exclamation mark** draws your attention to important information or actions.

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

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

Tips provide information that might help you solve a problem.

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1. Basic Residential VoIP Service

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 wildcard '|' destination	[ ]
Create a destination for your SIP numbers	[ ]
Network configuration	
Create a node for your PortaSIP	[ ]
Rating configuration (Vendor)	
Create a tariff (referred to as **tariff A** later on), which describes your termination costs and routing for off-net calls (make sure it has a **Routing** type!)	[ ]
Enter rates in tariff A for the destinations your customers will call	[ ]
Create a tariff (referred to as **tariff B** later on), which describes your termination costs for SIP-to-SIP calls (make sure it is NOT a **Routing** type!)	[ ]
Enter a zero cost rate for the wildcard destination (|) in tariff B	[ ]
Create an off-net calls vendor	[ ]
Create a connection for this vendor using tariff A	[ ]
Create an internal (fake) vendor for keeping track of SIP-to-SIP calls	[ ]
Create a "VoIP to vendor" connection with the Remote Gateway ID field containing the **SIP-UA** string for this vendor using tariff B	[ ]
Rating configuration (Customer)	
Create a tariff (referred to as **tariff C** later on), which will be applied to SIP subscribers	[ ]
Enter rates in tariff C for the destinations your customers will call (both off-net and SIP-to-SIP calls)	[ ]
Create your SIP product	[ ]
Create one rating entry for this product, using the PortaSIP node and 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 **My Company** 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 them and specify exchange rates in **Currencies** section**.**

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 **Rating** 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 **Rating** section of Admin-Index, choose **Destinations**.
2. Click on **Get default set** to download a set of destinations as a CSV (Comma-Separated Values) file.

3. Open this file in Microsoft Excel or any other suitable program. Edit the data if necessary.
4. Save the file and close it in Excel.
5. Switch back to the PortaBilling® web interface, and click 🔺 **Upload** on the Destinations screen.

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

Destinations for SIP Phones

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.) 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 **VOICEONNETRX** 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, …**
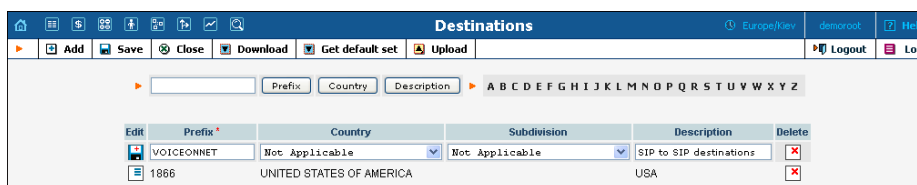
The Wildcard Destination

Sometimes you do not need to differentiate between individual destinations, all calls should be rated the same. A good example of a situation where this is required would be processing CDRs for on-net calls for your internal vendor. There is no direct cost, so you basically want every CDR to be rated at $0.0/min. One way of doing this is to create a rate for every country in the world, but a more elegant way is to use a wild-card rate. Destination | ('pipe' symbol) is special - when a rate is created for it in a tariff, it would match any dialed number (unless of course there is more specific rate available).
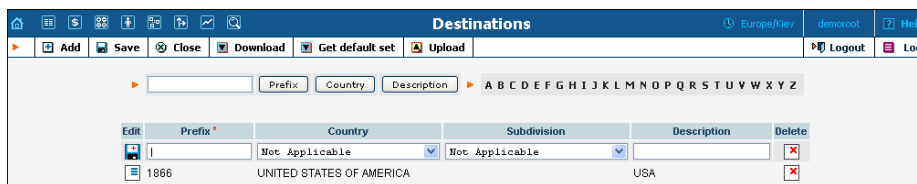
So please create this destination now – it will come handy when working with SIP-to-SIP tariffs.

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, Node ID, Radius shared secret, and so on.

1. In the **Networking** 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 **Manufacturer** – select **PortaOne**.
   o **Type** – VoIP node type; select **PortaSIP**.
   o **Node ID** – PortaSIP® server's hostname (recommended: `hostname.domainname`).
   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 **Radius Client** – Make sure this check-box is enabled, since the gateway (node) must be able to communicate with PortaBilling® via RADIUS.
   o **Radius Key** – Enter the RADIUS secret key you have specified in the configuration of the gateway.
   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.
   o **POD Server** – This checkbox relates to ISP billing. Leave it unchecked for now.
4. Click 🖫 **Save&Close**.

Create a Vendor 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 **Rating** 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 the currency, in which the vendor charges you.

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

    - **Applied To** – Choose **Vendor** in the **Applied To** select menu.
    - **Routing** – Leave the **Routing** box checked.
    - **Service Type** – Choose **Voice Calls** here.
    - **Default Off-peak Period** – If you do not differentiate between peak and off-peak rates, just choose **Not defined**; otherwise, select one of the previously defined off-peak periods.
    - **Short Description** – A short tariff description.
    - **Description** – An extended tariff description.
4. Click 🖫 **Save**.

Enter Rates for Your Vendor Tariff

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 in front of the name of the tariff. When you are in Tariff Management for a particular tariff, click on **Rates** in the toolbar.
2. On the **Edit Rates** screen, click ⊞ **Add**.
3. Fill in the required information:
   - **Destination** – A destination prefix may be entered directly, e.g. **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.

   - **Route category** – You can split your available routes into several categories, such as "High quality", "Premium", etc., then create routing plans for your customers. Use the **Default** route category for now.
   - **Preference** – The routing priority for the specific destination. 10 is the highest priority, 0 is the lowest (i.e. do not use this 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).
   - **Huntstop** – Do not try any routes with a lower preference.
   - **Interval First** – first billing unit in seconds.
   - **Interval Next** – next billing unit in seconds.
   - **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 **Payback Rate**, **Hidden**, **Forbidden** or **Discontinued** flags are optional.
4. Click the **Save** button in the toolbar, or the icon on the left side of the row.
5. Repeat these steps if you need to enter more rates.

Managing Rates Offline

You also can upload rates from a .CSV or .XLS file – please consult the *Rate Import* chapter of the **PortaSwitch Handbook: Wholesale Services** for more details.

Create Internal Vendor Tariff

Follow the steps described in the *Create a Vendor Tariff* section above. When creating this tariff, make sure it does not have the Routing option turned on.

Enter Rates for Internal Vendor Tariff

1. Click the **Rates** icon next to the name of the internal vendor tariff.
2. On the **Edit Rates** screen, click ⊞ **Add**.
3. Insert '**|**' (pipe) in the **Destination** field. Leave the other fields in the form as they are, because there is no need to define actual prices for

calls going between individual IP phones. This will create a special rate that matches any dialed phone number.

4. Click 🔲 **Save&Close**.

⧗ Perform the *Create Tariff* and *Enter Rates* steps described above until you have created a tariff with the termination costs for each termination partner you have; these tariffs are created as "Applied to: Vendor".

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 **Participants** 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 three 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. Three more tabs now appear on the screen. If you plan to terminate your calls to the vendor's SIP server, typically he will provide you with a username / password authorizing you to send calls to his server. In this case, to create the account for digest authorization:
   1. Click the **Accounts** tab, then click ⊞ **Add** and enter this information.
   2. Click the 🔲 **Save** button in the toolbar, or the 🔲 icon on the left end of the row.
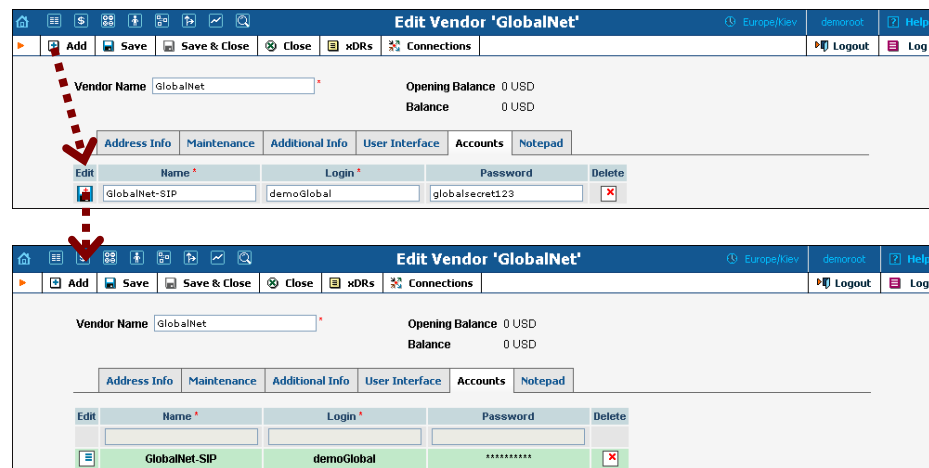
6. Click **Close** in order to return to the **Vendors** admin page.
7. Repeat steps 2-6 to add all of your vendors.

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 **Participants** 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 a Customer Tariff

1. In the **Rating** 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 of the tariff object; this is the name you will then see in the select menus.
   o **Currency** – Indicates the currency in which you charge your customers.

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

   o **Applied To** – Choose "**Customer**", as this tariff will be used to charge your customers.
   o **Managed By** – Choose "**Administrator Only**" here (this option is only visible after you select **Applied to: Customer** above).
   o **Service** – Choose **Voice Calls**.
   o **Default Off-peak Period** – If you do not differentiate between peak and off-peak rates, just choose **Not defined**; otherwise, select one of the previously defined off-peak periods.
   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 **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

1. On the Tariff Management page you will see a list of the available tariffs. Click the 🔲 **Rates** icon in front of 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 (please refer to the *Enter Rates for Your Vendor Tariff* section above for a detailed description of the available rate attributes).
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.

Perform the *Create Tariff* and *Enter Rates* steps described above until you have created a tariff for each account's billing scheme; these tariffs are

created as "Applied to: Customer". For example, if you plan to charge your customers more when they access toll-free lines instead of local ones, you need two tariffs, i.e. "Normal" and "Using Toll-free line".

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 rating list.

In the **Rating** 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 rating 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 **Administrator only** 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 the **PortaSwitch Handbook: Converged Services** you can find examples for setting up volume discounts).
- o **Description** – your comments about the intended use of this product.

3. Click ![save icon] **Save**.
4. Click on the **Services and Rating** tab to edit this product's rating list.

Enter Node and Tariff into the Product's Rating List

The rating 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 **Services and Rating** tab is selected, click on the ![add icon] **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 **Tariff** menu, select the tariff that will be applied to your customers for outgoing calls.
7. Configure overdraft protection for this product on the **Overdraft Protection** tab. (Please consult the *Rating* section in the **PortaBilling Web Reference Guide** for a description of the parameters available here.)
8. Click 🖫 **Save&Close** to save this rating entry.

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 **Billing** 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**.
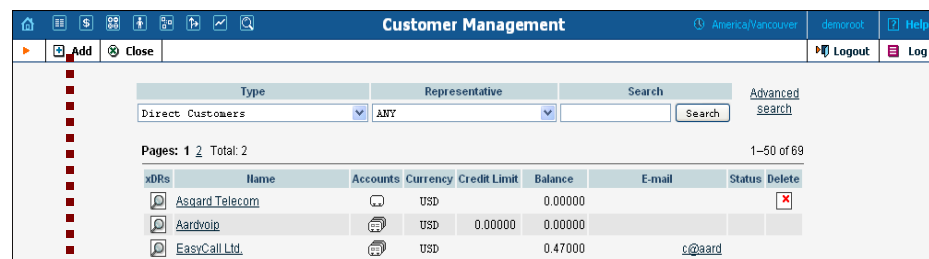
Create Customer

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.

In this example we will describe how to create a new customer manually, without using a Quick Form.

1. In the **Participants** section of the Admin-Index page, choose **Customers**.
2. On the Customer Management page, click ⊞ **Add Customer** and choose Manually.

NOTE: If no Quick Forms have been created, you will go directly to the **Add Customer** page.

xDRs	Name	Accounts	Currency	Credit Limit	Balance	E-mail	Status	Delete
🔍	Asgard Telecom		USD		0.00000			❌
🔍	Aardvoip		USD	0.00000	0.00000			
🔍	EasyCall Ltd.		USD		0.47000	c@aard		

Customer Management — Type: Direct Customers, Representative: ANY, Search. Pages: 1 2 Total: 2, 1–50 of 69
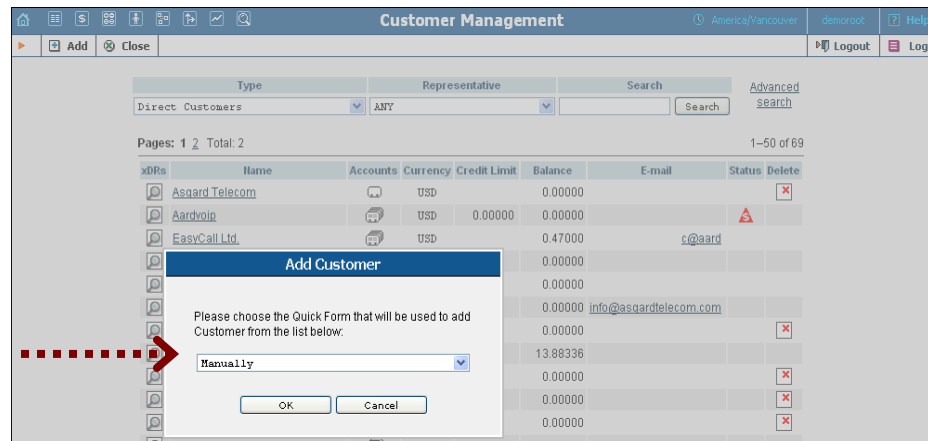
3. Fill in the **New Customer** form. Please note that there are several tabs with extra information available on the screen.
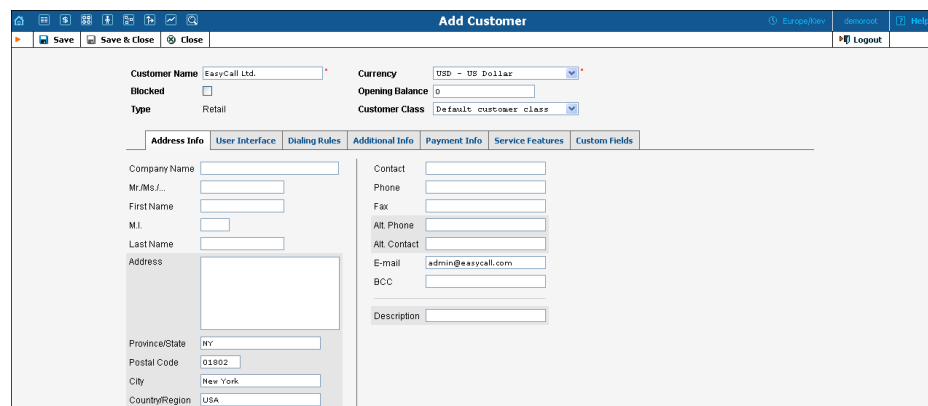
The most important fields are:

Main form (top)

- o **Customer Name** – short name for the customer object; this will be used on the web interface.
- o **Currency** – the currency in which this customer will be billed.
- o **Opening Balance** – a starting balance for the customer; the default is zero.
- o **Customer Class** – Choose the customer class you created in the previous step.

Address Info tab

- o **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.
- 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 accounts created under this Customer.

4. Click 💾 **Save&Close**.

Creating a new Customer using a Quick Form:

1. In the **Participants** section of the Admin-Index page, choose **Customers**.
2. On the Customer Management page, click ➕ **Add Customer** and choose one of the available Quick Forms.
3. Fill in the **Add Customer** form.

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. In this example, we assume that you are creating the account manually, so choose **Manually** from the drop-down menu (do not use any of the Quick Form options).

5. 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 **Service Password** – The account ID and this password will be used to authenticate SIP server login.
o **E-mail** – 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. If a new batch name is provided in the text field, the created account will be placed into a new batch with the given name. Alternatively, an existing batch could be selected from the drop-down list. If you leave this field empty, the created account will not be tied to any specific 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 **Life Time** – 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.
- o **IP Centrex Self-Care Enabled (demo)** – If enabled, account owners will use a new self-care interface to view their profile data and manage settings. To switch back to the default account self-care interface, simply disable this option.

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.

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

Creating a new account using a Quick Form:

1. Go to the **Customers** screen (the one containing the list of customers).

2. Next to the customer name, click on the icon (the one in the **Accounts** column) to go to account management for that customer.
3. Click on ⊞ **Add** and choose one of the available Quick Forms.
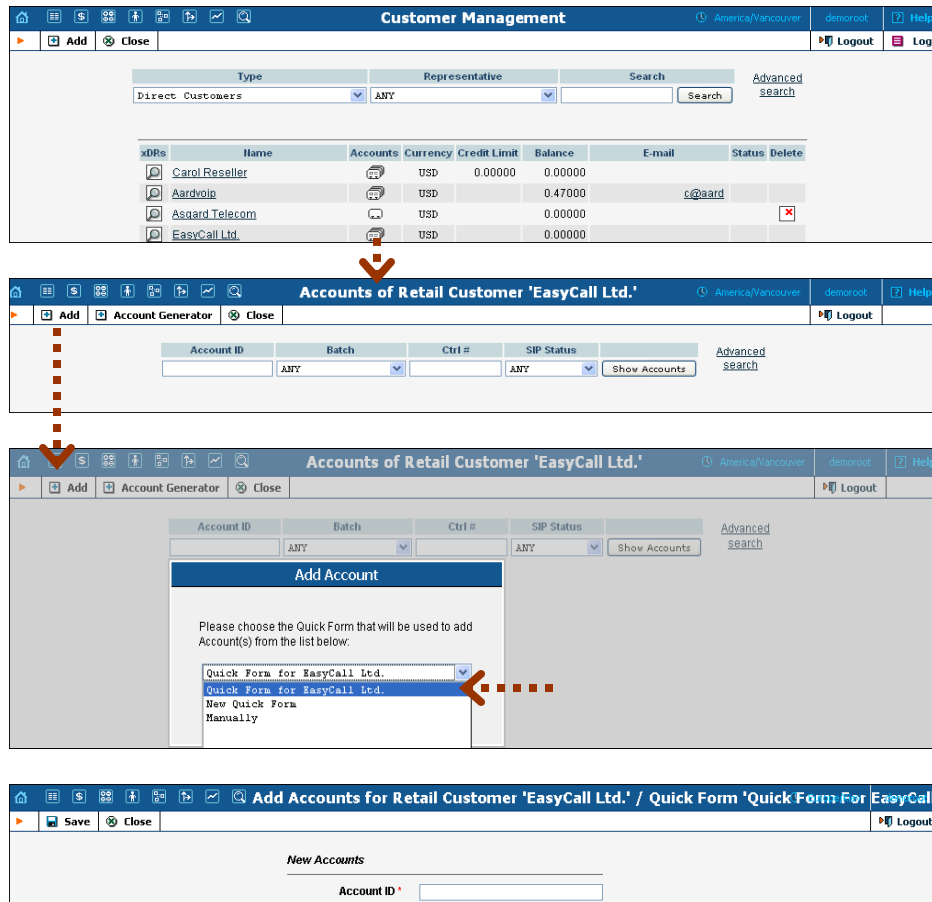
Set up Dialing Rules for a Customer (Optional)

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.

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 a 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.

Test the Whole System

1. Make sure the PortaBilling® radius and PortaSIP® servers are running.
2. Configure your SIP user agent with the account ID and password. (See 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 with SIP Log Viewer

You can 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.

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.

```
27 May 06:44:46.198/GLOBAL/proxy[25662]: RECEIVED message from 192.168.224.94:5060
at 192.168.193.113:5060:
        INVITE sip:16046831234@192.168.193.113 SIP/2.0
        Via: SIP/2.0/UDP 192.168.224.94:5060;branch=z9hG4bK-84c0a0d1
        From: "John Doe" <sip:16041234568@192.168.193.113>;tag=2425c0fc84fdf3f0o1
        To: <sip:16046831234@192.168.193.113>
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        CSeq: 101 INVITE
        Max-Forwards: 70
        Contact: "John Doe" <sip:16041234568@192.168.224.94:5060>
        Expires: 240
```

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```
User-Agent: Linksys/SPA941-5.1.8
Content-Length: 401
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: replaces
Content-Type: application/sdp

v=0
o=- 7990845 7990845 IN IP4 192.168.224.94
s=-
c=IN IP4 192.168.224.94
t=0 0
m=audio 16434 RTP/AVP 18 0 2 4 8 96 97 98 101
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:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
```

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

```
27 May 06:44:46.224/GLOBAL/proxy[25662]: SENDING message to 192.168.224.94:5060
from 192.168.193.113:5060:
      SIP/2.0 100 Trying
      Via: SIP/2.0/UDP 192.168.224.94:5060;branch=z9hG4bK-84c0a0d1;rport=5060
      To: <sip:16046831234@192.168.193.113>
      From: "John Doe"<sip:16041234568@192.168.193.113>;tag=2425c0fc84fdf3f0o1
      Call-ID: df629eec-35fd2ec0@192.168.224.94
      CSeq: 101 INVITE
      Content-Length: 0
```

B2BUA receives this INVITE request transmitted by proxy.

```
27 May 06:44:46.225/GLOBAL/b2bua[26273]: RECEIVED message from
192.168.193.113:5060 at 192.168.193.113:5061:
      INVITE sip:16046831234@192.168.193.113 SIP/2.0
      Via: SIP/2.0/UDP 192.168.193.113:5060;branch=z9hG4bK-d8754z-
1b561f52b7e12427-1---d8754z-;rport
      Via: SIP/2.0/UDP 192.168.224.94:5060;branch=z9hG4bK-84c0a0d1;rport=5060
      Max-Forwards: 69
      Record-Route: <sip:192.168.193.113;lr>
      Contact: "John Doe"<sip:16041234568@192.168.224.94:5060>
      To: <sip:16046831234@192.168.193.113>
      From: "John Doe"<sip:16041234568@192.168.193.113>;tag=2425c0fc84fdf3f0o1
      Call-ID: df629eec-35fd2ec0@192.168.224.94
      CSeq: 101 INVITE
      Expires: 240
      Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
      Content-Type: application/sdp
      Supported: replaces
      User-Agent: Linksys/SPA941-5.1.8
      PortaBilling-Notify: NAT
      Content-Length: 418

v=0
o=- 7990845 7990845 IN IP4 192.168.224.94
s=-
c=IN IP4 192.168.224.94
t=0 0
a=oldmediaip:192.168.224.94
m=audio 16434 RTP/AVP 18 0 2 4 8 96 97 98 101
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:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

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

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

```
27 May 06:44:46.227/GLOBAL/proxy[25662]: RECEIVED message from
192.168.193.113:5061 at 192.168.193.113:5060:
        SIP/2.0 401 Unauthorized
        Via: SIP/2.0/UDP 192.168.193.113:5060;branch=z9hG4bK-d8754z-
1b561f52b7e12427-1---d8754z-;rport=5060
        Via: SIP/2.0/UDP 192.168.224.94:5060;branch=z9hG4bK-84c0a0d1;rport=5060
        From: "John Doe" <sip:16041234568@192.168.193.113>;tag=2425c0fc84fdf3f0o1
        Server: Sippy
        To: <sip:16046831234@192.168.193.113>
        CSeq: 101 INVITE
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        Record-Route: <sip:192.168.193.113;lr>
        WWW-Authenticate: Digest
realm="192.168.193.113",nonce="95e358ee265a0fc2be454f43a7d0cbe25567"
```

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

```
27 May 06:44:46.268/GLOBAL/proxy[25662]: RECEIVED message from 192.168.224.94:5060
at 192.168.193.113:5060:
        ACK sip:16046831234@192.168.193.113 SIP/2.0
        Via: SIP/2.0/UDP 192.168.224.94:5060;branch=z9hG4bK-84c0a0d1
        From: "John Doe" <sip:16041234568@192.168.193.113>;tag=2425c0fc84fdf3f0o1
        To: <sip:16046831234@192.168.193.113>
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        CSeq: 101 ACK
        Max-Forwards: 70
        Contact: "John Doe" <sip:16041234568@192.168.224.94:5060>
        User-Agent: Linksys/SPA941-5.1.8
        Content-Length: 0
```

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

```
27 May 06:44:46.278/GLOBAL/proxy[25662]: RECEIVED message from 192.168.224.94:5060
at 192.168.193.113:5060:
        INVITE sip:16046831234@192.168.193.113 SIP/2.0
        Via: SIP/2.0/UDP 192.168.224.94:5060;branch=z9hG4bK-cd7c90c9
        From: "John Doe" <sip:16041234568@192.168.193.113>;tag=2425c0fc84fdf3f0o1
        To: <sip:16046831234@192.168.193.113>
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        CSeq: 102 INVITE
        Max-Forwards: 70
        Authorization: Digest
username="16041234568",realm="192.168.193.113",nonce="95e358ee265a0fc2be454f43a7d0
cbe25567",uri="sip:16046831234@192.168.193.113",algorithm=MD5,response="76ed4c20e5
845ae33c541c508f88d489"
        Contact: "John Doe" <sip:16041234568@192.168.224.94:5060>
        Expires: 240
        User-Agent: Linksys/SPA941-5.1.8
        Content-Length: 401
```

```
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: replaces
Content-Type: application/sdp

v=0
o=- 7990845 7990845 IN IP4 192.168.224.94
s=-
c=IN IP4 192.168.224.94
t=0 0
m=audio 16434 RTP/AVP 18 0 2 4 8 96 97 98 101
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:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
```

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.)

```
27 May 06:44:46.304/GLOBAL/proxy[25662]: SENDING message to 192.168.224.94:5060
from 192.168.193.113:5060:
        SIP/2.0 100 Trying
        Via: SIP/2.0/UDP 192.168.224.94:5060;branch=z9hG4bK-cd7c90c9;rport=5060
        To: <sip:16046831234@192.168.193.113>
        From: "John Doe"<sip:16041234568@192.168.193.113>;tag=2425c0fc84fdf3f0o1
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        CSeq: 102 INVITE
        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.)

```
27 May 06:44:46.225/GLOBAL/b2bua[26273]: RECEIVED message from
192.168.193.113:5060 at 192.168.193.113:5061:
        INVITE sip:16046831234@192.168.193.113 SIP/2.0
        Via: SIP/2.0/UDP 192.168.193.113:5060;branch=z9hG4bK-d8754z-
1b561f52b7e12427-1---d8754z-;rport
        Via: SIP/2.0/UDP 192.168.224.94:5060;branch=z9hG4bK-84c0a0d1;rport=5060
        Max-Forwards: 69
        Record-Route: <sip:192.168.193.113;lr>
        Contact: "John Doe"<sip:16041234568@192.168.224.94:5060>
        To: <sip:16046831234@192.168.193.113>
        From: "John Doe"<sip:16041234568@192.168.193.113>;tag=2425c0fc84fdf3f0o1
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        CSeq: 101 INVITE
        Expires: 240
        Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
        Content-Type: application/sdp
        Supported: replaces
        User-Agent: Linksys/SPA941-5.1.8
        PortaBilling-Notify: NAT
        Content-Length: 418

v=0
o=- 7990845 7990845 IN IP4 192.168.224.94
s=-
c=IN IP4 192.168.224.94
t=0 0
a=oldmediaip:192.168.224.94
m=audio 16434 RTP/AVP 18 0 2 4 8 96 97 98 101
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:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

B2BUA sends an authorization request to the billing.

```
27 May 06:44:46.309/df629eec-35fd2ec0@192.168.224.94/b2bua[26273]: sending AAA
request:
        b2bua_request_serial        = 74F9C7E4
        User-Name                   = "16041234568"
        Digest-Realm                = "192.168.193.113"
        Digest-Nonce                = "95e358ee265a0fc2be454f43a7d0cbe25567"
        Digest-Method               = "INVITE"
        Digest-URI                  = "sip:16046831234@192.168.193.113"
        Digest-Algorithm            = "MD5"
        Digest-User-Name            = "16041234568"
        Digest-Response             = "76ed4c20e5845ae33c541c508f88d489"
        Calling-Station-Id          = "16041234568"
        Called-Station-Id           = "16046831234"
        h323-conf-id                = "366C2DAA 884D11E0 8935000C 2938E4DC"
        call-id                     = "df629eec-35fd2ec0@192.168.224.94"
        h323-remote-address         = "192.168.224.94"
        h323-session-protocol       = "sipv2"
        h323-ivr-out                = "PortaBilling_AuthMethod:INVITE"
        h323-ivr-out                =
"PortaBilling_Routing:format=PortaSIP;version=21.0"
        h323-ivr-out                =
"PortaBilling_Notify:rtpp_same_nat=1;NAT"
        h323-ivr-out                = "PortaBilling_Seed:1090853988"
        h323-ivr-out                = "PortaBilling_CLN:John%20Doe"
        h323-ivr-out                =
"PortaSIP_Aux_Info:auth_life=124549/1/0;auth_cnt=1;CC_now=124549/1;pid=26273;reque
st_type=routing"
```

Billing authorizes the call and provides information about call routing (3 possible routes are
returned.)

```
27 May 06:44:46.375/df629eec-35fd2ec0@192.168.224.94/b2bua[26273]: AAA request
accepted, processing response:
        b2bua_request_serial  = 74F9C7E4
        h323-return-code       = "h323-return-code=13"
        Cisco-AVPair   = "h323-ivr-in=Product:Product_Standard"
        Cisco-AVPair   = "h323-ivr-in=Tariff:SIP Phone Subscribers"
        Cisco-AVPair   = "h323-ivr-in=PortaBilling_NodeId:sipenv-10.portaone.com"
        Cisco-AVPair   = "h323-ivr-in=PortaBilling_AccessCode:OUTGOING"
        Cisco-AVPair   = "h323-ivr-in=PortaBilling_Seed:1090853988"
        Cisco-AVPair   = "h323-ivr-in=PortaBilling_RatePattern:16046831234"
        Cisco-AVPair   = "h323-ivr-in=PortaBilling_Authorize:base=session-
time;avail=-1;expires=1306493086"
        Cisco-AVPair   = "h323-ivr-in=PortaBilling_CompleteNumber:16046831234"
        Cisco-AVPair   = "h323-ivr-in=PortaBilling_CLI:16041234568"
        Cisco-AVPair   = "h323-ivr-in=PortaBilling_ACLI:16041234568"
        Cisco-AVPair   = "h323-ivr-in=PortaBilling_CLN:John%20Doe"
        Cisco-AVPair   = "h323-ivr-in=MOH:104"
        h323-billing-model = "h323-billing-model=0"
        h323-currency      = "h323-currency=USD"
        h323-preferred-lang = "h323-preferred-lang=en"
        Framed-Route       = "r=;g-hunt=seq"
        Framed-Route       = "P_=;g-hunt=skip;expires=300;patience=10"
        Framed-Route       = "p__=193.28.187.3;auth=Partner:123test;spec="
        Framed-Route       = "p__=193.28.87.182;auth=demoGlobal:123test;spec="
        Framed-Route       = "p__=192.168.224.243"
```

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

```
27 May 06:44:46.379/df629eec-35fd2ec0@192.168.224.94/b2bua[26273]: SENDING message
to 193.28.187.3:5060 from 192.168.193.113:5061:
        INVITE sip:16046831234@193.28.187.3:5060 SIP/2.0
        P-Asserted-Identity: "John Doe" <sip:16041234568@192.168.193.113>
        h323-conf-id: 913059242-2286752224-2301952012-691594460
        Via: SIP/2.0/UDP 192.168.193.113:5061;branch=z9hG4bK-
lu3uftz3fxkt2ubw;rport
        From: "John Doe" <sip:16041234568@192.168.193.113>;tag=yrmqlf52qpm3kqcm.o
        Content-disposition: session
        Expires: 300
        User-Agent: Sippy
        To: <sip:16046831234@193.28.187.3>
        Remote-Party-ID: "John Doe"
<sip:16041234568@192.168.193.113>;party=calling
        Contact: Anonymous <sip:192.168.193.113:5061>
        CSeq: 118 INVITE
        cisco-GUID: 913059242-2286752224-2301952012-691594460
        Max-Forwards: 70
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        Content-Length: 419
        Content-Type: application/sdp

        v=0
        o=Sippy 372416384 0 IN IP4 192.168.193.113
        s=-
        t=0 0
        m=audio 16434 RTP/AVP 18 0 2 4 8 96 97 98 101
        c=IN IP4 192.168.224.94
        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:101 telephone-event/8000
        a=fmtp:101 0-15
        a=sendrecv
        a=oldmediaip:192.168.224.94
```

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

```
27 May 06:44:46.879/GLOBAL/b2bua[26273]: SENDING message to 193.28.187.3:5060 from
192.168.193.113:5061:
        INVITE sip:16046831234@193.28.187.3:5060 SIP/2.0
        P-Asserted-Identity: "John Doe" <sip:16041234568@192.168.193.113>
        h323-conf-id: 913059242-2286752224-2301952012-691594460
        Via: SIP/2.0/UDP 192.168.193.113:5061;branch=z9hG4bK-
lu3uftz3fxkt2ubw;rport
        From: "John Doe" <sip:16041234568@192.168.193.113>;tag=yrmqlf52qpm3kqcm.o
        Content-disposition: session
        Expires: 300
        User-Agent: Sippy
        To: <sip:16046831234@193.28.187.3>
        Remote-Party-ID: "John Doe"
<sip:16041234568@192.168.193.113>;party=calling
        Contact: Anonymous <sip:192.168.193.113:5061>
        CSeq: 118 INVITE
        cisco-GUID: 913059242-2286752224-2301952012-691594460
        Max-Forwards: 70
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        Content-Length: 419
        Content-Type: application/sdp

        v=0
```

```
o=Sippy 372416384 0 IN IP4 192.168.193.113
s=-
t=0 0
m=audio 16434 RTP/AVP 18 0 2 4 8 96 97 98 101
c=IN IP4 192.168.224.94
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:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=oldmediaip:192.168.224.94
```

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

```
27 May 06:44:56.380/df629eec-35fd2ec0@192.168.224.94/b2bua[26273]: sending Acct
Stop (Originate):
        h323-call-origin            = "originate"
        h323-call-type              = "VoIP"
        h323-session-protocol       = "sipv2"
        h323-setup-time             = "10:44:46.376 GMT Fri May 27 2011"
        User-Name                   = "16041234568"
        Calling-Station-Id          = "16041234568"
        Called-Station-Id           = "16046831234"
        h323-conf-id                = "366C2DAA 884D11E0 8935000C 2938E4DC"
        call-id                     = "df629eec-35fd2ec0@192.168.224.94"
        Acct-Session-Id             = "df629eec-35fd2ec0@192.168.224.94"
        h323-ivr-out                =
"PortaSIP_Aux_Info:CC=124549/2;route_seqnum=3;route_id=372405960;cc_id=373102280;p
id=26273"
        h323-ivr-out                = "DURATION:7200"
        h323-ivr-out                = "PortaBilling_RatePattern:16046831234"
        h323-ivr-out                = "PortaBilling_AccessCode:OUTGOING"
        h323-ivr-out                = "PortaBilling_Seed:1090853988"
        h323-remote-address         = "193.28.187.3"
        h323-disconnect-time        = "10:44:56.376 GMT Fri May 27 2011"
        h323-connect-time           = "10:44:56.376 GMT Fri May 27 2011"
        Acct-Session-Time           = "0"
        h323-disconnect-cause       = "12"
        Acct-Status-Type            = "Stop"
```

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

```
27 May 06:44:56.382/df629eec-35fd2ec0@192.168.224.94/b2bua[26273]: SENDING message
to 193.28.87.182:5060 from 192.168.193.113:5061:
        INVITE sip:16046831234@193.28.87.182:5060 SIP/2.0
        P-Asserted-Identity: "John Doe" <sip:16041234568@192.168.193.113>
        h323-conf-id: 913059242-2286752224-2301952012-691594460
        Via: SIP/2.0/UDP 192.168.193.113:5061;branch=z9hG4bK-
tguhkhgmefy34qym;rport
        From: "John Doe" <sip:16041234568@192.168.193.113>;tag=rqmpcttcnc3hhmck.o
        Content-disposition: session
        Expires: 300
        User-Agent: Sippy
        To: <sip:16046831234@193.28.87.182>
        Remote-Party-ID: "John Doe"
<sip:16041234568@192.168.193.113>;party=calling
        Contact: Anonymous <sip:192.168.193.113:5061>
        CSeq: 188 INVITE
        cisco-GUID: 913059242-2286752224-2301952012-691594460
        Max-Forwards: 70
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        Content-Length: 419
```

```
Content-Type: application/sdp

v=0
o=Sippy 372517416 0 IN IP4 192.168.193.113
s=-
t=0 0
m=audio 16434 RTP/AVP 18 0 2 4 8 96 97 98 101
c=IN IP4 192.168.224.94
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:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=oldmediaip:192.168.224.94
```

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

```
27 May 06:44:56.410/GLOBAL/b2bua[26273]: RECEIVED message from 193.28.87.182:5060
at 192.168.193.113:5061:
        SIP/2.0 100 Trying
        Via: SIP/2.0/UDP 192.168.193.113:5061;branch=z9hG4bK-
tguhkhgmefy34qym;rport=5061
        To: <sip:16046831234@193.28.87.182>
        From: "John Doe"<sip:16041234568@192.168.193.113>;tag=rqmpcttcnc3hhmck.o
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        CSeq: 188 INVITE
        Content-Length: 0
```

The remote SIP proxy requests authorization.

```
27 May 06:44:56.439/GLOBAL/b2bua[26273]: RECEIVED message from 193.28.87.182:5060
at 192.168.193.113:5061:
        SIP/2.0 401 Unauthorized
        Via: SIP/2.0/UDP 192.168.193.113:5061;branch=z9hG4bK-
tguhkhgmefy34qym;rport=5061
        Record-Route: <sip:193.28.87.182;lr>
        To: <sip:16046831234@193.28.87.182>
        From: "John Doe"<sip:16041234568@192.168.193.113>;tag=rqmpcttcnc3hhmck.o
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        CSeq: 188 INVITE
        Server: Sippy
        WWW-Authenticate: Digest
realm="193.28.87.182",nonce="9b9e876958fe07668f8e45d3e8deb2810f33"
        Content-Length: 0
```

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

```
27 May 06:44:56.463/df629eec-35fd2ec0@192.168.224.94/b2bua[26273]: SENDING message
to 193.28.87.182:5060 from 192.168.193.113:5061:
        INVITE sip:16046831234@193.28.87.182:5060 SIP/2.0
        P-Asserted-Identity: "John Doe" <sip:16041234568@192.168.193.113>
        h323-conf-id: 913059242-2286752224-2301952012-691594460
        Via: SIP/2.0/UDP 192.168.193.113:5061;branch=z9hG4bK-
n3s7oyw5jsnktidj;rport
        From: "John Doe" <sip:16041234568@192.168.193.113>;tag=rqmpcttcnc3hhmck.o
        Expires: 300
        User-Agent: Sippy
        To: <sip:16046831234@193.28.87.182>
        Remote-Party-ID: "John Doe"
<sip:16041234568@192.168.193.113>;party=calling
        Contact: Anonymous <sip:192.168.193.113:5061>
        CSeq: 189 INVITE
```

```
             cisco-GUID: 913059242-2286752224-2301952012-691594460
             Max-Forwards: 70
             Call-ID: df629eec-35fd2ec0@192.168.224.94
             Authorization: Digest
username="demoGlobal",realm="193.28.87.182",nonce="9b9e876958fe07668f8e45d3e8deb28
10f33",uri="sip:16046831234@193.28.87.182:5060",response="77307dd8e6237d49ba04c3e2
ae2ba909"
             Content-Length: 419
             Content-Type: application/sdp

             v=0
             o=Sippy 372517416 0 IN IP4 192.168.193.113
             s=-
             t=0 0
             m=audio 16434 RTP/AVP 18 0 2 4 8 96 97 98 101
             c=IN IP4 192.168.224.94
             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:101 telephone-event/8000
             a=fmtp:101 0-15
             a=sendrecv
             a=oldmediaip:192.168.224.94
```

The remote SIP proxy sends a ringback, so the user on the SIP phone will hear ringing.

```
27 May 06:44:56.683/GLOBAL/b2bua[26273]: RECEIVED message from 193.28.87.182:5060
at 192.168.193.113:5061:
             SIP/2.0 180 Ringing
             Via: SIP/2.0/UDP 192.168.193.113:5061;branch=z9hG4bK-
n3s7oyw5jsnktidj;rport=5061
             Record-Route: <sip:193.28.87.182;lr>
             To: <sip:16046831234@193.28.87.182>;tag=vbflpoyv6l4ukomw.i
             From: "John Doe"<sip:16041234568@192.168.193.113>;tag=rqmpcttcnc3hhmck.o
             Call-ID: df629eec-35fd2ec0@192.168.224.94
             CSeq: 189 INVITE
             Server: Sippy
             Content-Length: 0
```

The called party answers the call.

```
27 May 06:44:59.278/GLOBAL/b2bua[26273]: RECEIVED message from 193.28.87.182:5060
at 192.168.193.113:5061:
             SIP/2.0 200 OK
             Via: SIP/2.0/UDP 192.168.193.113:5061;branch=z9hG4bK-
n3s7oyw5jsnktidj;rport=5061
             Record-Route: <sip:193.28.87.182;lr>
             Contact: "Anonymous"<sip:193.28.87.182:5061>
             To: <sip:16046831234@193.28.87.182>;tag=vbflpoyv6l4ukomw.i
             From: "John Doe"<sip:16041234568@192.168.193.113>;tag=rqmpcttcnc3hhmck.o
             Call-ID: df629eec-35fd2ec0@192.168.224.94
             CSeq: 189 INVITE
             Content-Type: application/sdp
             Server: Sippy
             H323-credit-time: 4970
             Content-Length: 413

             v=0
             o=Sippy 462269184 1 IN IP4 193.28.87.182
             s=CounterPath X-Lite 4.0
             t=0 0
             a=ice-ufrag:a8de07
             a=ice-pwd:5900f5ccc6e5fb1098eb44f3b13028c8
             m=audio 61502 RTP/AVP 0 8 101
```

```
c=IN IP4 192.168.192.223
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=candidate:1 1 UDP 659136 192.168.192.223 61502 typ host
a=candidate:1 2 UDP 659134 192.168.192.223 61503 typ host
a=oldmediaip:192.168.192.223
```

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

```
27 May 06:44:59.294/df629eec-35fd2ec0@192.168.224.94/b2bua[26273]: sending Acct
Start (Originate):
        h323-call-origin           = "originate"
        h323-call-type             = "VoIP"
        h323-session-protocol      = "sipv2"
        h323-setup-time            = "10:44:56.380 GMT Fri May 27 2011"
        User-Name                  = "16041234568"
        Calling-Station-Id         = "16041234568"
        Called-Station-Id          = "16046831234"
        h323-conf-id               = "366C2DAA 884D11E0 8935000C 2938E4DC"
        call-id                    = "df629eec-35fd2ec0@192.168.224.94"
        Acct-Session-Id            = "df629eec-35fd2ec0@192.168.224.94"
        h323-ivr-out               =
"PortaSIP_Aux_Info:CC=124549/2;route_seqnum=4;route_id=372405672;cc_id=373102280;p
id=26273"
        h323-ivr-out               = "DURATION:7200"
        h323-ivr-out               = "PortaBilling_RatePattern:16046831234"
        h323-ivr-out               = "PortaBilling_AccessCode:OUTGOING"
        h323-ivr-out               = "PortaBilling_Seed:1090853988"
        h323-remote-address        = "193.28.87.182"
        h323-connect-time          = "10:44:59.294 GMT Fri May 27 2011"
        alert-timepoint            = "10:44:56.684 GMT Fri May 27 2011"
        Acct-Status-Type           = "Start"
```

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

```
27 May 06:44:59.365/df629eec-35fd2ec0@192.168.224.94/b2bua[26273]: sending Acct
Start (Answer):
        h323-call-origin           = "answer"
        h323-call-type             = "VoIP"
        h323-session-protocol      = "sipv2"
        h323-setup-time            = "10:44:46.375 GMT Fri May 27 2011"
        User-Name                  = "16041234568"
        Calling-Station-Id         = "16041234568"
        Called-Station-Id          = "16046831234"
        h323-conf-id               = "366C2DAA 884D11E0 8935000C 2938E4DC"
        call-id                    = "df629eec-35fd2ec0@192.168.224.94"
        Acct-Session-Id            = "df629eec-35fd2ec0@192.168.224.94"
        h323-ivr-out               = "PortaBilling_Seed:1090853988"
        h323-ivr-out               =
"PortaSIP_Aux_Info:CC=124549/1;cc_id=373102280;pid=26273"
        h323-ivr-out               = "PortaBilling_UserName:16041234568"
        h323-ivr-out               = "PortaBilling_AccessCode:OUTGOING"
        h323-remote-address        = "192.168.224.94"
        h323-connect-time          = "10:44:59.365 GMT Fri May 27 2011"
        alert-timepoint            = "10:44:59.365 GMT Fri May 27 2011"
        Acct-Status-Type           = "Start"
```

The called party hangs up; the call termination process is started.

```
27 May 06:45:09.914/GLOBAL/b2bua[26273]: RECEIVED message from 193.28.87.182:5060
at 192.168.193.113:5061:
        BYE sip:192.168.193.113:5061 SIP/2.0
        Via: SIP/2.0/UDP 193.28.87.182:5060;branch=z9hG4bK-d8754z-
0f92a220a8ba967a-1---d8754z-;rport
        Via: SIP/2.0/UDP 193.28.87.182:5061;branch=z9hG4bK-
32ovb5x567ju3dpy;rport=5061
        Max-Forwards: 69
```

```
Contact: "Anonymous"<sip:193.28.87.182:5061>
To: "John Doe"<sip:16041234568@192.168.193.113>;tag=rqmpcttcnc3hhmck.o
From: <sip:16046831234@193.28.87.182>;tag=vbflpoyv6l4ukomw.i
Call-ID: df629eec-35fd2ec0@192.168.224.94
CSeq: 475 BYE
User-Agent: Sippy
h323-conf-id: 913059242-2286752224-2301952012-691594460
cisco-GUID: 913059242-2286752224-2301952012-691594460
Content-Length: 0
```

PortaSIP confirms call disconnection.

```
27 May 06:45:09.915/GLOBAL/b2bua[26273]: SENDING message to 193.28.87.182:5060
from 192.168.193.113:5061:
    SIP/2.0 200 OK
    Via: SIP/2.0/UDP 193.28.87.182:5060;branch=z9hG4bK-d8754z-
0f92a220a8ba967a-1---d8754z-;rport=5060
    Via: SIP/2.0/UDP 193.28.87.182:5061;branch=z9hG4bK-
32ovb5x567ju3dpy;rport=5061
    From: <sip:16046831234@193.28.87.182>;tag=vbflpoyv6l4ukomw.i
    Server: Sippy
    To: "John Doe" <sip:16041234568@192.168.193.113>;tag=rqmpcttcnc3hhmck.o
    CSeq: 475 BYE
    Call-ID: df629eec-35fd2ec0@192.168.224.94
```

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

```
27 May 06:45:10.117/df629eec-35fd2ec0@192.168.224.94/b2bua[26273]: sending Acct
Stop (Answer):
    h323-call-origin            = "answer"
    h323-call-type              = "VoIP"
    h323-session-protocol       = "sipv2"
    h323-setup-time             = "10:44:46.375 GMT Fri May 27 2011"
    User-Name                   = "16041234568"
    Calling-Station-Id          = "16041234568"
    Called-Station-Id           = "16046831234"
    h323-conf-id                = "366C2DAA 884D11E0 8935000C 2938E4DC"
    call-id                     = "df629eec-35fd2ec0@192.168.224.94"
    Acct-Session-Id             = "df629eec-35fd2ec0@192.168.224.94"
    h323-ivr-out                = "PortaBilling_Seed:1090853988"
    h323-ivr-out                =
"PortaSIP_Aux_Info:CC=124549/1;cc_id=373102280;pid=26273"
    h323-ivr-out                = "PortaBilling_UserName:16041234568"
    h323-ivr-out                = "PortaBilling_AccessCode:OUTGOING"
    h323-remote-address         = "192.168.224.94"
    h323-voice-quality          = "0"
    Acct-Terminate-Cause        = "User-Request"
    h323-disconnect-time        = "10:45:10.116 GMT Fri May 27 2011"
    h323-connect-time           = "10:44:59.365 GMT Fri May 27 2011"
    Acct-Session-Time           = "11"
    h323-disconnect-cause       = "0"
    alert-timepoint             = "10:44:59.365 GMT Fri May 27 2011"
    Acct-Status-Type            = "Stop"
```

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

```
27 May 06:45:09.944/df629eec-35fd2ec0@192.168.224.94/b2bua[26273]: sending Acct
Stop (Originate):
    h323-call-origin            = "originate"
    h323-call-type              = "VoIP"
    h323-session-protocol       = "sipv2"
    h323-setup-time             = "10:44:56.380 GMT Fri May 27 2011"
    User-Name                   = "16041234568"
    Calling-Station-Id          = "16041234568"
    Called-Station-Id           = "16046831234"
    h323-conf-id                = "366C2DAA 884D11E0 8935000C 2938E4DC"
    call-id                     = "df629eec-35fd2ec0@192.168.224.94"
```

```
        Acct-Session-Id                 = "df629eec-35fd2ec0@192.168.224.94"
        h323-ivr-out                    =
"PortaSIP_Aux_Info:CC=124549/2;route_seqnum=4;route_id=372405672;cc_id=373102280;p
id=26273"
        h323-ivr-out                    = "DURATION:7200"
        h323-ivr-out                    = "PortaBilling_RatePattern:16046831234"
        h323-ivr-out                    = "PortaBilling_AccessCode:OUTGOING"
        h323-ivr-out                    = "PortaBilling_Seed:1090853988"
        h323-remote-address             = "193.28.87.182"
        h323-voice-quality              = "0"
        Acct-Terminate-Cause            = "User-Request"
        h323-disconnect-time            = "10:45:09.916 GMT Fri May 27 2011"
        h323-connect-time               = "10:44:59.294 GMT Fri May 27 2011"
        Acct-Session-Time               = "11"
        h323-disconnect-cause           = "0"
        alert-timepoint                 = "10:44:56.684 GMT Fri May 27 2011"
        Acct-Status-Type                = "Stop"
```

The incoming call leg is closed.

```
27 May 06:45:10.142/GLOBAL/proxy[25662]: SENDING message to 192.168.224.94:5060
from 192.168.193.113:5060:
        BYE sip:16041234568@192.168.224.94:5060 SIP/2.0
        Via: SIP/2.0/UDP 192.168.193.113:5060;branch=z9hG4bK-d8754z-
8bf1b65243387770-1---d8754z-;rport
        Via: SIP/2.0/UDP 192.168.193.113:5061;branch=z9hG4bK-
bvbsn3maosxgcgcv;rport=5061
        Max-Forwards: 69
        Contact: "Anonymous"<sip:192.168.193.113:5061>
        To: "John Doe"<sip:16041234568@192.168.193.113>;tag=2425c0fc84fdf3f0o1
        From: <sip:16046831234@192.168.193.113>;tag=qh7hepd544zznp5i.i
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        CSeq: 502 BYE
        User-Agent: Sippy
        h323-conf-id: 913058492-2286752224-2301952012-691594460
        cisco-GUID: 913058492-2286752224-2301952012-691594460
        Content-Length: 0
```

The other party confirms call disconnection.

```
27 May 06:45:10.163/GLOBAL/proxy[25662]: RECEIVED message from 192.168.224.94:5060
at 192.168.193.113:5060:
        SIP/2.0 200 OK
        To: "John Doe"<sip:16041234568@192.168.193.113>;tag=2425c0fc84fdf3f0o1
        From: <sip:16046831234@192.168.193.113>;tag=qh7hepd544zznp5i.i
        Call-ID: df629eec-35fd2ec0@192.168.224.94
        CSeq: 502 BYE
        Via: SIP/2.0/UDP 192.168.193.113:5060;branch=z9hG4bK-d8754z-
8bf1b65243387770-1---d8754z-
        Via: SIP/2.0/UDP 192.168.193.113:5061;branch=z9hG4bK-
bvbsn3maosxgcgcv;rport=5061
        Server: Linksys/SPA941-5.1.8
        Content-Length: 0
```

The call is finished.

👓

1. Browse information in the PortaBilling® log file. On the page where the SIP log is displayed, click the H323-Conf-ID for this call, as shown on the screenshot below. The BE log will open in a new browser tab.

PortaBilling receives the authorization request.

```
May 27 06:44:46 [0]: Processing request (BE ver:mr23-1-1-be,smp:8389)...
May 27 06:44:46 [3]: Attributes:
NAS-IP-Address                = '192.168.193.113'
User-Name                     = '16041234568'
Called-Station-Id             = '16046831234'
Calling-Station-Id            = '16041234568'
h323-conf-id                  = '366C2DAA 884D11E0 8935000C 2938E4DC'
call-id                       = 'df629eec-35fd2ec0@192.168.224.94'
Digest-Attributes             = 'Realm = "192.168.193.113"'
Digest-Attributes             = 'Nonce =
"95e358ee265a0fc2be454f43a7d0cbe25567"'
Digest-Attributes             = 'Method = "INVITE"'
Digest-Attributes             = 'URI = "sip:16046831234@192.168.193.113"'
Digest-Attributes             = 'Algorithm = "MD5"'
Digest-Attributes             = 'User-Name = "16041234568"'
Digest-Response-Draft         = '76ed4c20e5845ae33c541c508f88d489'
h323-remote-address           = '192.168.224.94'
h323-session-protocol         = 'sipv2'
h323-ivr-out                  = 'PortaBilling_AuthMethod:INVITE'
h323-ivr-out                  =
'PortaBilling_Routing:format=PortaSIP;version=21.0'
h323-ivr-out                  = 'PortaBilling_Notify:rtpp_same_nat=1;NAT'
h323-ivr-out                  = 'PortaBilling_Seed:1090853988'
h323-ivr-out                  = 'PortaBilling_CLN:John%20Doe'
h323-ivr-out                  =
'PortaSIP_Aux_Info:auth_life=124549/1/0;auth_cnt=1;CC_now=124549/1;pid=26273;reque
st_type=routing'
NAS-Port                      = '5060'
May 27 06:44:46 [4]: Mod(psip): fixup: Detected draft-styled digest authorization
May 27 06:44:46 [5]: Setting up routing features: local-lookup=1;cli=1;bill-
to=1;max-depth=15;version=21.0;sip=1;pattern=1;format=PortaSIP;incoming-
charge=1;cld=1;criteria=1
May 27 06:44:46 [4]: Registering new session
May 27 06:44:46 [0]: <Session id="366C2DAA 884D11E0 8935000C 2938E4DC" i_env="10"
created="May 27 06:44:46" expires="May 27 06:45:16" valid="1"/>
May 27 06:44:46 [4]: Looking for a connection with matching class 3/VoIP/answer in
environment 10
May 27 06:44:46 [3]: Processing account authorization: <Authenticate::INVITE
nas="sipenv-10.portaone.com" at="May 27 06:44:46" from="16041234568" serial="0"/>
```

Account information is located in the database.

```
May 27 06:44:46 [5]: SQL query 'FindAccountById' on connection 'porta-billing-
master' executed in 0.000860 seconds
May 27 06:44:46 [5]: SQL query 'GetCustomer' on connection 'porta-billing-master'
executed in 0.000200 seconds
May 27 06:44:46 [4]: Found account: <Account i="559" id="16041234568"
model="credit" balance="1.06169"/> of <Customer i="94" name="EasyCall Ltd."
balance="3.10174"/>
May 27 06:44:46 [5]: SQL query 'GetAllRoutingPlans' on connection 'porta-billing-
master' executed in 0.000130 seconds
May 27 06:44:46 [5]: SQL query 'FindAccessibility' on connection 'porta-billing-
master' executed in 0.000115 seconds
May 27 06:44:46 [5]: SQL query 'GetTariff' on connection 'porta-billing-master'
executed in 0.000105 seconds
May 27 06:44:46 [5]: SQL query 'GetServiceFeatures' on connection 'porta-billing-
master' executed in 0.000131 seconds
May 27 06:44:46 [5]: SQL query 'GetServiceFeaturesConfiguration' on connection
'porta-billing-master' executed in 0.000128 seconds
```

Password verification is successful.

```
May 27 06:44:46 [4]: Verification of password using method 'digest_response':
success
```

The customer's number translation rule is applied; the phone number is not changed as it is already in the E.164 format.

```
May 27 06:44:46 [5]: SQL query 'GetSessionDeposit' on connection 'porta-billing-
master' executed in 0.000443 seconds
May 27 06:44:46 [4]: Gained the new deposit for this session: <Session::Deposit
i_account="559" i_customer="94" session_id="366C2DAA 884D11E0 8935000C 2938E4DC"
customer_lock="1" created="May 27 06:44:46"/>
May 27 06:44:46 [4]: Mod(voice): Applying override translation rule on CLD ...
May 27 06:44:46 [4]: Translation 's/^\*3164\*//;' applied: '16046831234' unchanged
May 27 06:44:46 [5]: SQL query 'FindCentrexExtensionbyId' on connection 'porta-
billing-master' executed in 0.000124 seconds
May 27 06:44:46 [4]: Applying customer dialing translation rule on CLD ...
May 27 06:44:46 [4]: Translation
'local_to_e164($_,{cc=>'1',ac=>'604,778',ia=>'1',dp=>'1',ip=>'011',em=>'911',ex=>'
411'});#% cc=1 ac=604,778 ia=1 dp=1 ip=011 em=911 ex=411' applied: '16046831234'
unchanged
May 27 06:44:46 [3]: Mod(voice): Applying account matching translation rule on CLD
...
May 27 06:44:46 [4]: Translation '' applied: '16046831234' unchanged
```

The dialed number is an off-net destination.

```
May 27 06:44:46 [5]: SQL query 'FindAccountById' on connection 'porta-billing-
master' executed in 0.000548 seconds
May 27 06:44:46 [4]: No account with id '16046831234'
May 27 06:44:46 [5]: SQL query 'LoadAllVoiceAppAccessNumbers' on connection
'porta-billing-master' executed in 0.000122 seconds
May 27 06:44:46 [5]: SQL query 'FindNumberPortability' on connection 'porta-
billing-master' executed in 0.000156 seconds
```

Maximum session time is calculated according to the available funds and rate.

```
May 27 06:44:46 [4]: Setting up rating in Charge <Tariff i="126" name="SIP Phone
Subscribers"/> of Account ...
May 27 06:44:46 [3]: Looking up rate for '16046831234' in <Tariff i="126"
name="SIP Phone Subscribers"/>
May 27 06:44:46 [5]: SQL query 'GetPricePerDestination' on connection 'porta-
billing-master' executed in 0.001409 seconds
May 27 06:44:46 [4]: Allocating unlimited deposit amount for this session
```

```
May 27 06:44:46 [4]: Computing account allowed session-time by <Rate::Standard
i="28492" pattern="16046831234" seed="1090853988" destination="1604"/> ...
May 27 06:44:46 [4]: No credit limit, unlimited seconds of session-time
May 27 06:44:46 [5]: SQL query 'SetSessionDeposit' on connection 'porta-billing-
master' executed in 0.000676 seconds
May 27 06:44:46 [4]: Successfull creation of the session deposit:
<Session::Deposit i_account="559" i_customer="94" session_id="366C2DAA 884D11E0
8935000C 2938E4DC" customer_lock="1" opened="May 27 06:44:46" expires="May 27
06:44:48"/>
```

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

```
May 27 06:44:46 [4]: Looking up routes in <Route::PointList route-lookup-
cld="16046831234" cld="16046831234" sip="1" i_routing_plan="-1"
routing_plan_name="All Available Routes"/>
May 27 06:44:46 [5]: SQL query 'GetRoutingPerDestination' on connection 'porta-
billing-master' executed in 0.001689 seconds
May 27 06:44:46 [5]: SQL query 'GetExchangeRate' on connection 'porta-billing-
master' executed in 0.000229 seconds
May 27 06:44:46 [5]: Mod(psip): Setting up route <Route::Point p="193.28.187.3"
cld="16046831234" prio="5" cost="0.00900" description="Termination to
MegaNet"/>...
May 27 06:44:46 [4]: Mod(psip): Don't use RTP proxy according to policy 'Optimal'
=> 'Direct'. NAT conditions ignored
May 27 06:44:46 [5]: Mod(psip): Setting up route <Route::Point p="193.28.87.182"
cld="16046831234" prio="5" cost="0.01234" description="Termination to
GlobalNet"/>...
May 27 06:44:46 [4]: Mod(psip): Don't use RTP proxy according to policy 'Optimal'
=> 'Direct'. NAT conditions ignored
May 27 06:44:46 [5]: Mod(psip): Setting up route <Route::Point p="192.168.224.243"
cld="16046831234" prio="5" cost="0.02500" description="Termination to
MultiVendor"/>...
May 27 06:44:46 [4]: Mod(psip): Don't use RTP proxy according to policy 'Optimal'
=> 'Direct'. NAT conditions ignored
May 27 06:44:46 [4]: Routing result:
<Route::PointList route-lookup-cld="16046831234" cld="16046831234" sip="1"
i_routing_plan="-1" routing_plan_name="All Available Routes">
        <Route::Point p="193.28.187.3" cld="16046831234" prio="5" cost="0.00900"
description="Termination to MegaNet"/>
        <Route::Point p="193.28.87.182" cld="16046831234" prio="5" cost="0.01234"
description="Termination to GlobalNet"/>
        <Route::Point p="192.168.224.243" cld="16046831234" prio="5"
cost="0.02500" description="Termination to MultiVendor"/>
</Route::PointList>
```

An authorization response is sent to PortaSIP.

```
May 27 06:44:46 [3]: Computing routing
May 27 06:44:46 [3]: AAA::Authentication acknowledge response, attributes:
h323-return-code              = 13
h323-ivr-in                   = Product:Product_Standard
h323-ivr-in                   = Tariff:SIP Phone Subscribers
h323-ivr-in                   = PortaBilling_NodeId:sipenv-10.portaone.com
h323-ivr-in                   = PortaBilling_AccessCode:OUTGOING
h323-ivr-in                   = PortaBilling_Seed:1090853988
h323-ivr-in                   = PortaBilling_RatePattern:16046831234
h323-ivr-in                   = PortaBilling_Authorize:base=session-
time;avail=-1;expires=1306493086
h323-ivr-in                   = PortaBilling_CompleteNumber:16046831234
h323-ivr-in                   = PortaBilling_CLI:16041234568
h323-ivr-in                   = PortaBilling_ACLI:16041234568
h323-ivr-in                   = PortaBilling_CLN:John%20Doe
h323-ivr-in                   = MOH:104
h323-billing-model            = 0
h323-currency                 = USD
h323-preferred-lang           = en
Framed-Route                  = r=;g-hunt=seq
Framed-Route                  = P_=;g-hunt=skip;expires=300;patience=10
Framed-Route                  =
```

```
p__=193.28.187.3;auth=090E1D08112549896C9E824BA024C3A2;spec=
Framed-Route                    =
p__=193.28.87.182;auth=B0E9C94B60D6A562460FC83FE68943B680BE9CDF9ECC0BCC;spec=
Framed-Route                    = p__=192.168.224.243
May 27 06:44:46 [4]: This session expires in cache at May 27 06:45:16 (in 30s)
May 27 06:44:46 [0]: ...Done Authorize::INVITE in 0.043429 seconds, 0.007065 spent
in SQL queries. Dispatch+activation: 0.000079+0.000469 seconds.
```

Accounting for the failed outgoing call leg is received.

```
May 27 06:44:56 [0]: Processing request (BE ver:mr23-1-1-be,smp:8389)...
May 27 06:44:56 [3]: Attributes:
NAS-IP-Address             = '192.168.193.113'
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            = '10:44:46.376 GMT Fri May 27 2011'
h323-connect-time          = '10:44:56.376 GMT Fri May 27 2011'
h323-disconnect-time       = '10:44:56.376 GMT Fri May 27 2011'
h323-disconnect-cause      = '12'
h323-conf-id               = '366C2DAA 884D11E0 8935000C 2938E4DC'
call-id                    = 'df629eec-35fd2ec0@192.168.224.94'
Acct-Session-Id            = 'df629eec-35fd2ec0@192.168.224.94'
Acct-Session-Time          = '0'
Acct-Delay-Time            = '0'
h323-session-protocol      = 'sipv2'
h323-ivr-out               =
'PortaSIP_Aux_Info:CC=124549/2;route_seqnum=3;route_id=372405960;cc_id=373102280;p
id=26273'
h323-ivr-out               = 'DURATION:7200'
h323-ivr-out               = 'PortaBilling_RatePattern:16046831234'
h323-ivr-out               = 'PortaBilling_AccessCode:OUTGOING'
h323-ivr-out               = 'PortaBilling_Seed:1090853988'
h323-remote-address        = '193.28.187.3'
NAS-Port                   = '5060'
```

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

```
May 27 06:44:56 [4]: The session already exists in cache
May 27 06:44:56 [0]: <Session id="366C2DAA 884D11E0 8935000C 2938E4DC" i_env="10"
created="May 27 06:44:46" modified="May 27 06:44:56" expires="May 27 06:45:16"
valid="1"/>
May 27 06:44:56 [3]: Processing accounting: <Accounting::VOICE-originate-
VoIP::Stop nas="sipenv-10.portaone.com" at="May 27 06:44:56" from="16041234568"
cld="16046831234" serial="1"/>
May 27 06:44:56 [3]: Account and charge info found in <Authorize::INVITE
nas="sipenv-10.portaone.com" at="May 27 06:44:46" from="16041234568"
cld="16046831234" serial="0"/>
May 27 06:44:56 [3]: Copied account: <Account i="559" id="16041234568"
model="credit" balance="1.06169"/> of <Customer i="94" name="EasyCall Ltd."
balance="3.10174"/> with charge info into current request
May 27 06:44:56 [4]: End of originate leg on access node
May 27 06:44:56 [5]: SQL query 'GetSessionDeposit' on connection 'porta-billing-
master' executed in 0.000574 seconds
May 27 06:44:56 [4]: Ignore zombie session deposit: <Session::Deposit
i_account="559" i_customer="94" session_id="366C2DAA 884D11E0 8935000C 2938E4DC"
customer_lock="1" opened="May 27 06:44:46" expires="May 27 06:44:48"/>
May 27 06:44:56 [4]: Looking for a connection with matching class 3/VoIP/originate
in environment 10
```

Connection matched.

```
May 27 06:44:56 [3]: Found connection matched by remoteip('193.28.187.3'),
prefix(''): <Connection i="66" name="Termination to MegaNet"/>
May 27 06:44:56 [3]: Found vendor: <Vendor i="46" name="MegaNet"/>
```

```
May 27 06:44:56 [3]: Mod(voice): No alert-timepoint present. Using h323-connect-
time '10:44:56.376 GMT Fri May 27 2011' as alert-timepoint for PDD statistics
May 27 06:44:56 [3]: This request isn't marked as billable, hold on till cleanup
time
May 27 06:44:56 [5]: SQL query 'GetActiveLegByAcct' on connection 'porta-billing-
master' executed in 0.000708 seconds
May 27 06:44:56 [5]: SQL query 'InsertActiveLeg' on connection 'porta-billing-
master' executed in 0.000362 seconds
May 27 06:44:56 [5]: prepare 'GetLastInsertId' on connection 'porta-billing-
master'
```

A failed CDR is inserted for the vendor.

```
May 27 06:44:56 [3]: AAA::Accounting response, no attributes.
May 27 06:44:56 [4]: This session expires in cache at May 27 06:45:26 (in 30s)
May 27 06:44:56 [0]: ...Done Accounting::VOICE-originate-VoIP::Stop in 0.012006
seconds, 0.001644 spent in SQL queries. Dispatch+activation: 0.000081+0.000434
seconds.
```

The call is now established, and a start accounting record about the egress call leg is received.

```
May 27 06:44:59 [0]: Processing request (BE ver:mr23-1-1-be,smp:8389)...
May 27 06:44:59 [3]: Attributes:
NAS-IP-Address                 = '192.168.193.113'
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                = '10:44:56.380 GMT Fri May 27 2011'
alert-timepoint                = '10:44:56.684 GMT Fri May 27 2011'
h323-connect-time              = '10:44:59.294 GMT Fri May 27 2011'
h323-conf-id                   = '366C2DAA 884D11E0 8935000C 2938E4DC'
call-id                        = 'df629eec-35fd2ec0@192.168.224.94'
Acct-Session-Id                = 'df629eec-35fd2ec0@192.168.224.94'
Acct-Delay-Time                = '0'
h323-session-protocol          = 'sipv2'
h323-ivr-out                   =
'PortaSIP_Aux_Info:CC=124549/2;route_seqnum=4;route_id=372405672;cc_id=373102280;p
id=26273'
h323-ivr-out                   = 'DURATION:7200'
h323-ivr-out                   = 'PortaBilling_RatePattern:16046831234'
h323-ivr-out                   = 'PortaBilling_AccessCode:OUTGOING'
h323-ivr-out                   = 'PortaBilling_Seed:1090853988'
h323-remote-address            = '193.28.87.182'
NAS-Port                       = '5060'
```

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

```
May 27 06:44:59 [4]: The session already exists in cache
May 27 06:44:59 [0]: <Session id="366C2DAA 884D11E0 8935000C 2938E4DC" i_env="10"
created="May 27 06:44:46" modified="May 27 06:44:59" expires="May 27 06:45:26"
valid="1"/>
May 27 06:44:59 [3]: Processing accounting: <Accounting::VOICE-originate-
VoIP::Start nas="sipenv-10.portaone.com" at="May 27 06:44:59" from="16041234568"
cld="16046831234" serial="2"/>
May 27 06:44:59 [3]: Account and charge info found in <Accounting::VOICE-
originate-VoIP::Stop nas="sipenv-10.portaone.com" at="May 27 06:44:56"
from="16041234568" cld="16046831234" serial="1"/>
May 27 06:44:59 [3]: Copied account: <Account i="559" id="16041234568"
model="credit" balance="1.06169"/> of <Customer i="94" name="EasyCall Ltd."
balance="3.10174"/> with charge info into current request
May 27 06:44:59 [5]: SQL query 'GetSessionDeposit' on connection 'porta-billing-
master' executed in 0.000700 seconds
May 27 06:44:59 [4]: This is an update to session with existent deposit:
<Session::Deposit i_account="559" i_customer="94" session_id="366C2DAA 884D11E0
8935000C 2938E4DC" customer_lock="1" opened="May 27 06:44:46" expires="May 27
```

```
06:44:48"/>
May 27 06:44:59 [5]: SQL query 'UpdateSessionDeposit' on connection 'porta-
billing-master' executed in 0.001418 seconds
May 27 06:44:59 [4]: Successfull update of the session deposit: <Session::Deposit
i_account="559" i_customer="94" session_id="366C2DAA 884D11E0 8935000C 2938E4DC"
customer_lock="1" opened="May 27 06:44:46" expires="May 27 06:47:01"/>
May 27 06:44:59 [5]: SQL query 'GetActiveLegByAcct' on connection 'porta-billing-
master' executed in 0.000804 seconds
May 27 06:44:59 [4]: Looking for a connection with matching class 3/VoIP/originate
in environment 10
May 27 06:44:59 [3]: Found connection matched by remoteip('193.28.87.182'),
prefix(''): <Connection i="52" name="Termination to GlobalNet"/>
May 27 06:44:59 [5]: SQL query 'InsertActiveLeg' on connection 'porta-billing-
master' executed in 0.000495 seconds
May 27 06:44:59 [5]: prepare 'GetLastInsertId' on connection 'porta-billing-
master'
```

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

```
May 27 06:44:59 [3]: AAA::Accounting response, no attributes.
May 27 06:44:59 [4]: This session expires in cache at May 27 06:45:29 (in 30s)
May 27 06:44:59 [0]: ...Done Accounting::VOICE-originate-VoIP::Start in 0.092460
seconds, 0.003417 spent in SQL queries. Dispatch+activation: 0.000095+0.000717
seconds.
```

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

```
May 27 06:44:59 [0]: Processing request (BE ver:mr23-1-1-be,smp:8389)...
May 27 06:44:59 [3]: Attributes:
NAS-IP-Address              = '192.168.193.113'
User-Name                   = '16041234568'
Called-Station-Id           = '16046831234'
Calling-Station-Id          = '16041234568'
Acct-Status-Type            = 'Start'
h323-call-origin            = 'answer'
h323-call-type              = 'VoIP'
h323-setup-time             = '10:44:46.375 GMT Fri May 27 2011'
alert-timepoint             = '10:44:59.365 GMT Fri May 27 2011'
h323-connect-time           = '10:44:59.365 GMT Fri May 27 2011'
h323-conf-id                = '366C2DAA 884D11E0 8935000C 2938E4DC'
call-id                     = 'df629eec-35fd2ec0@192.168.224.94'
Acct-Session-Id             = 'df629eec-35fd2ec0@192.168.224.94'
Acct-Delay-Time             = '0'
h323-session-protocol       = 'sipv2'
h323-ivr-out                = 'PortaBilling_Seed:1090853988'
h323-ivr-out                =
'PortaSIP_Aux_Info:CC=124549/1;cc_id=373102280;pid=26273'
h323-ivr-out                = 'PortaBilling_UserName:16041234568'
h323-ivr-out                = 'PortaBilling_AccessCode:OUTGOING'
h323-remote-address         = '192.168.224.94'
NAS-Port                    = '5060'
```

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

```
May 27 06:44:59 [4]: The session already exists in cache
May 27 06:44:59 [0]: <Session id="366C2DAA 884D11E0 8935000C 2938E4DC" i_env="10"
created="May 27 06:44:46" modified="May 27 06:44:59" expires="May 27 06:45:29"
valid="1"/>
May 27 06:44:59 [3]: Processing accounting: <Accounting::VOICE-answer-VoIP::Start
nas="sipenv-10.portaone.com" at="May 27 06:44:59" from="16041234568"
cld="16046831234" serial="3"/>
May 27 06:44:59 [3]: Account and charge info found in <Accounting::VOICE-
originate-VoIP::Stop nas="sipenv-10.portaone.com" at="May 27 06:44:56"
from="16041234568" cld="16046831234" serial="1"/>
May 27 06:44:59 [3]: Copied account: <Account i="559" id="16041234568"
model="credit" balance="1.06169"/> of <Customer i="94" name="EasyCall Ltd."
balance="3.10174"/> with charge info into current request
May 27 06:44:59 [5]: SQL query 'GetSessionDeposit' on connection 'porta-billing-
```

```
master' executed in 0.001913 seconds
May 27 06:44:59 [4]: This is an update to session with existent deposit:
<Session::Deposit i_account="559" i_customer="94" session_id="366C2DAA 884D11E0
8935000C 2938E4DC" customer_lock="1" opened="May 27 06:44:46" expires="May 27
06:47:01"/>
May 27 06:44:59 [5]: SQL query 'UpdateSessionDeposit' on connection 'porta-
billing-master' executed in 0.000450 seconds
May 27 06:44:59 [4]: Successful update of the session deposit: <Session::Deposit
i_account="559" i_customer="94" session_id="366C2DAA 884D11E0 8935000C 2938E4DC"
customer_lock="1" opened="May 27 06:44:46" expires="May 27 06:47:01"/>
May 27 06:44:59 [5]: SQL query 'GetActiveLegByAcct' on connection 'porta-billing-
master' executed in 0.000567 seconds
May 27 06:44:59 [4]: Looking for a connection with matching class 3/VoIP/answer in
environment 10
May 27 06:44:59 [5]: SQL query 'InsertActiveLeg' on connection 'porta-billing-
master' executed in 0.000338 seconds
May 27 06:44:59 [5]: prepare 'GetLastInsertId' on connection 'porta-billing-
master'
```

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

```
May 27 06:44:59 [3]: AAA::Accounting response, no attributes.
May 27 06:44:59 [4]: This session expires in cache at May 27 06:45:29 (in 30s)
May 27 06:44:59 [0]: ...Done Accounting::VOICE-answer-VoIP::Start in 0.010833
seconds, 0.003268 spent in SQL queries. Dispatch+activation: 0.000081+0.000431
seconds.
```

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

```
May 27 06:45:10 [0]: Processing request (BE ver:mr23-1-1-be,smp:8389)...
May 27 06:45:10 [3]: Attributes:
NAS-IP-Address              = '192.168.193.113'
User-Name                   = '16041234568'
Called-Station-Id           = '16046831234'
Calling-Station-Id          = '16041234568'
Acct-Status-Type            = 'Stop'
h323-call-origin            = 'answer'
h323-call-type              = 'VoIP'
h323-setup-time             = '10:44:46.375 GMT Fri May 27 2011'
alert-timepoint             = '10:44:59.365 GMT Fri May 27 2011'
h323-connect-time           = '10:44:59.365 GMT Fri May 27 2011'
h323-disconnect-time        = '10:45:10.116 GMT Fri May 27 2011'
h323-disconnect-cause       = '0'
h323-voice-quality          = '0'
h323-conf-id                = '366C2DAA 884D11E0 8935000C 2938E4DC'
call-id                     = 'df629eec-35fd2ec0@192.168.224.94'
Acct-Session-Id             = 'df629eec-35fd2ec0@192.168.224.94'
Acct-Session-Time           = '11'
Acct-Delay-Time             = '0'
h323-session-protocol       = 'sipv2'
h323-ivr-out                = 'PortaBilling_Seed:1090853988'
h323-ivr-out                =
'PortaSIP_Aux_Info:CC=124549/1;cc_id=373102280;pid=26273'
h323-ivr-out                = 'PortaBilling_UserName:16041234568'
h323-ivr-out                = 'PortaBilling_AccessCode:OUTGOING'
h323-remote-address         = '192.168.224.94'
Acct-Terminate-Cause        = 'User-Request'
NAS-Port                    = '5060'
May 27 06:45:10 [4]: The session already exists in cache
May 27 06:45:10 [0]: <Session id="366C2DAA 884D11E0 8935000C 2938E4DC" i_env="10"
created="May 27 06:44:46" modified="May 27 06:45:10" expires="May 27 06:45:40"
valid="1"/>
May 27 06:45:10 [3]: Processing accounting: <Accounting::VOICE-answer-VoIP::Stop
nas="sipenv-10.portaone.com" at="May 27 06:45:10" from="16041234568"
cld="16046831234" serial="5"/>
May 27 06:45:10 [3]: Account and charge info found in <Accounting::VOICE-
originate-VoIP::Stop nas="sipenv-10.portaone.com" at="May 27 06:45:10"
from="16041234568" cld="16046831234" serial="4"/>
May 27 06:45:10 [3]: Copied account: <Account i="559" id="16041234568"
```

```
model="credit" balance="1.08431"/> of <Customer i="94" name="EasyCall Ltd."
balance="3.12436"/> with charge info into current request
May 27 06:45:10 [4]: End of session via answer leg on access node
May 27 06:45:10 [5]: SQL query 'GetSessionDeposit' on connection 'porta-billing-
master' executed in 0.002112 seconds
May 27 06:45:10 [4]: Unlock existent session deposit: <Session::Deposit
i_account="559" i_customer="94" session_id="366C2DAA 884D11E0 8935000C 2938E4DC"
customer_lock="1" opened="May 27 06:44:46" expires="May 27 06:47:01"/>
May 27 06:45:10 [5]: SQL query 'UpdateSessionDeposit' on connection 'porta-
billing-master' executed in 0.000409 seconds
May 27 06:45:10 [4]: Successfull update of the session deposit: <Session::Deposit
i_account="559" i_customer="94" session_id="366C2DAA 884D11E0 8935000C 2938E4DC"
customer_lock="1" opened="May 27 06:44:46" closing="May 27 06:45:12"/>
May 27 06:45:10 [4]: Looking for a connection with matching class 3/VoIP/answer in
environment 10
May 27 06:45:10 [4]: No connection to charge this access node answer leg.
May 27 06:45:10 [5]: SQL query 'GetActiveLegByAcct' on connection 'porta-billing-
master' executed in 0.000609 seconds
May 27 06:45:10 [5]: SQL query 'UpdateActiveLeg' on connection 'porta-billing-
master' executed in 0.000515 seconds
```

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

```
May 27 06:45:10 [3]: AAA::Accounting response, no attributes.
May 27 06:45:10 [4]: This session closes in cache at May 27 06:45:40 (in 30s)
May 27 06:45:10 [0]: ...Done Accounting::VOICE-answer-VoIP::Stop in 0.013503
seconds, 0.003645 spent in SQL queries. Dispatch+activation: 0.000079+0.000334
seconds.
```

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

```
May 27 06:45:10 [0]: Processing request (BE ver:mr23-1-1-be,smp:8389)...
May 27 06:45:10 [3]: Attributes:
NAS-IP-Address               = '192.168.193.113'
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              = '10:44:56.380 GMT Fri May 27 2011'
alert-timepoint              = '10:44:56.684 GMT Fri May 27 2011'
h323-connect-time            = '10:44:59.294 GMT Fri May 27 2011'
h323-disconnect-time         = '10:45:09.916 GMT Fri May 27 2011'
h323-disconnect-cause        = '0'
h323-voice-quality           = '0'
h323-conf-id                 = '366C2DAA 884D11E0 8935000C 2938E4DC'
call-id                      = 'df629eec-35fd2ec0@192.168.224.94'
Acct-Session-Id              = 'df629eec-35fd2ec0@192.168.224.94'
Acct-Session-Time            = '11'
Acct-Delay-Time              = '0'
h323-session-protocol        = 'sipv2'
h323-ivr-out                 =
'PortaSIP_Aux_Info:CC=124549/2;route_seqnum=4;route_id=372405672;cc_id=373102280;p
id=26273'
h323-ivr-out                 = 'DURATION:7200'
h323-ivr-out                 = 'PortaBilling_RatePattern:16046831234'
h323-ivr-out                 = 'PortaBilling_AccessCode:OUTGOING'
h323-ivr-out                 = 'PortaBilling_Seed:1090853988'
h323-remote-address          = '193.28.87.182'
Acct-Terminate-Cause         = 'User-Request'
NAS-Port                     = '5060'
May 27 06:45:10 [4]: The session already exists in cache
May 27 06:45:10 [0]: <Session id="366C2DAA 884D11E0 8935000C 2938E4DC" i_env="10"
created="May 27 06:44:46" modified="May 27 06:45:10" expires="May 27 06:45:29"
valid="1"/>
May 27 06:45:10 [3]: Processing accounting: <Accounting::VOICE-originate-
VoIP::Stop nas="sipenv-10.portaone.com" at="May 27 06:45:10" from="16041234568"
cld="16046831234" serial="4"/>
```

```
May 27 06:45:10 [3]: Account and charge info found in <Accounting::VOICE-
originate-VoIP::Stop nas="sipenv-10.portaone.com" at="May 27 06:44:56"
from="16041234568" cld="16046831234" serial="1"/>
May 27 06:45:10 [3]: Copied account: <Account i="559" id="16041234568"
model="credit" balance="1.06169"/> of <Customer i="94" name="EasyCall Ltd."
balance="3.10174"/> with charge info into current request
May 27 06:45:10 [4]: End of originate leg on access node
May 27 06:45:10 [5]: SQL query 'GetSessionDeposit' on connection 'porta-billing-
master' executed in 0.027293 seconds
May 27 06:45:10 [4]: Keep existent session deposit: <Session::Deposit
i_account="559" i_customer="94" session_id="366C2DAA 884D11E0 8935000C 2938E4DC"
customer_lock="1" opened="May 27 06:44:46" expires="May 27 06:47:01"/>
May 27 06:45:10 [4]: Looking for a connection with matching class 3/VoIP/originate
in environment 10
```

This call leg crosses a connection to the vendor.

```
May 27 06:45:10 [3]: Found connection matched by remoteip('193.28.87.182'),
prefix(''): <Connection i="52" name="Termination to GlobalNet"/>
May 27 06:45:10 [3]: Found vendor: <Vendor i="39" name="GlobalNet"/>
May 27 06:45:10 [5]: SQL query 'FindNumberPortability' on connection 'porta-
billing-master' executed in 0.000695 seconds
May 27 06:45:10 [3]: Charge <Tariff i="126" name="SIP Phone Subscribers"/> of
Account. Building XDR ...
```

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

```
May 27 06:45:10 [5]: Reusing existing <Rate::Standard i="28492"
pattern="16046831234" seed="1090853988" destination="1604"/>
May 27 06:45:10 [4]: Creating new charge for 11 seconds of session-time. 11 base
units.
May 27 06:45:10 [3]: Charge built:
<XDR::Account pay="0.02262 USD" use="11 second" history="+11x1x0.1234">
     <Charge pay="0.02262 USD" use="11 second" history="+11x1x0.1234"/>
</XDR::Account>
```

PortaBilling also calculates the termination costs for this call.

```
May 27 06:45:10 [3]: Charge <Tariff i="127" name="Vendor GlobalNet"/> of Vendor.
Building XDR ...
May 27 06:45:10 [3]: Looking up rate for '16046831234' in <Tariff i="127"
name="Vendor GlobalNet"/>
May 27 06:45:10 [5]: SQL query 'GetPricePerDestination' on connection 'porta-
billing-master' executed in 0.001393 seconds
May 27 06:45:10 [4]: Creating new charge for 11 seconds of session-time. 11 base
units.
May 27 06:45:10 [3]: Charge built:
<XDR::Vendor pay="0.00226 USD" use="11 second" history="+11x1x0.01234">
     <Charge pay="0.00226 USD" use="11 second" history="+11x1x0.01234"/>
</XDR::Vendor>
```

CDRs are inserted and balances are modified.

```
May 27 06:45:10 [5]: SQL query 'InsertAccountCDR' on connection 'porta-billing-
master' executed in 0.001019 seconds
May 27 06:45:10 [5]: prepare 'GetLastInsertId' on connection 'porta-billing-
master'
May 27 06:45:10 [5]: SQL query 'UpdateAccountBalanceByXdr' on connection 'porta-
billing-master' executed in 0.028954 seconds
May 27 06:45:10 [5]: SQL query 'UpdateCustomerBalanceByXdr' on connection 'porta-
billing-master' executed in 0.002504 seconds
May 27 06:45:10 [5]: SQL query 'UpdateAccountUsageByXdr' on connection 'porta-
billing-master' executed in 0.000340 seconds
May 27 06:45:10 [5]: Transaction 'StoreXDR::Account' on connection 'porta-billing-
master' committed in 0.035141 seconds
May 27 06:45:10 [5]: SQL query 'InsertVendorCDR' on connection 'porta-billing-
```



```
master' executed in 0.000688 seconds
May 27 06:45:10 [5]: prepare 'GetLastInsertId' on connection 'porta-billing-
master'
May 27 06:45:10 [5]: SQL query 'UpdateVendorBalanceByXdr' on connection 'porta-
billing-master' executed in 0.001336 seconds
May 27 06:45:10 [5]: Transaction 'StoreXDR::Vendor' on connection 'porta-billing-
master' committed in 0.050652 seconds
May 27 06:45:10 [5]: SQL query 'UpdateSessionDeposit' on connection 'porta-
billing-master' executed in 0.000516 seconds
May 27 06:45:10 [5]: SQL query 'GetActiveLegByAcct' on connection 'porta-billing-
master' executed in 0.022602 seconds
May 27 06:45:10 [5]: prepare 'UpdateActiveLeg' on connection 'porta-billing-
master'
May 27 06:45:10 [5]: SQL query 'UpdateActiveLeg' on connection 'porta-billing-
master' executed in 0.000797 seconds
May 27 06:45:10 [3]: AAA::Accounting response, no attributes.
May 27 06:45:10 [4]: This session expires in cache at May 27 06:45:40 (in 30s)
May 27 06:45:10 [0]: ...Done Accounting::VOICE-originate-VoIP::Stop in 0.239396
seconds, 0.173930 spent in SQL queries. Dispatch+activation: 0.000074+0.000917
seconds.
```

PortaBilling cleans up the call.

```
May 27 06:46:08 [0]: Removing session 366C2DAA 884D11E0 8935000C 2938E4DC/10 (BE
ver:mr23-1-1-be,smp:8389)
May 27 06:46:08 [0]: <Session id="366C2DAA 884D11E0 8935000C 2938E4DC" i_env="10"
created="May 27 06:44:46" modified="May 27 06:45:10" closing="May 27 06:45:40"/>
May 27 06:46:08 [3]: Cleaning up the call
May 27 06:46:08 [3]: Processing <Accounting::VOICE-originate-VoIP::Stop
nas="sipenv-10.portaone.com" at="May 27 06:45:10" from="16041234568"
cld="16046831234" serial="4"/>
May 27 06:46:08 [4]: Account charge already processed
May 27 06:46:08 [4]: Vendor charge already processed
May 27 06:46:08 [3]: Charged 1 of 1 found outgoing sessions
May 27 06:46:08 [3]: Processing <Accounting::VOICE-originate-VoIP::Stop
nas="sipenv-10.portaone.com" at="May 27 06:44:56" from="16041234568"
cld="16046831234" serial="1"/>
May 27 06:46:08 [5]: SQL query 'FindNumberPortability' on connection 'porta-
billing-master' executed in 0.000351 seconds
May 27 06:46:08 [3]: Charge <Tariff i="164" name="Vendor MegaNet"/> of Vendor.
Building XDR ...
May 27 06:46:08 [3]: Looking up rate for '16046831234' in <Tariff i="164"
name="Vendor MegaNet"/>
May 27 06:46:08 [5]: SQL query 'GetPricePerDestination' on connection 'porta-
billing-master' executed in 0.001279 seconds
May 27 06:46:08 [4]: Creating new charge for 0 seconds of session-time. 0 base
units.
May 27 06:46:08 [3]: Charge built:
<XDR::Vendor pay="0 USD" use="0 second" history="fail charge"/>
May 27 06:46:08 [5]: SQL query 'InsertVendorCDRFail' on connection 'porta-billing-
master' executed in 0.000421 seconds
May 27 06:46:08 [5]: prepare 'GetLastInsertId' on connection 'porta-billing-
master'
May 27 06:46:08 [5]: Transaction 'StoreXDR::Vendor' on connection 'porta-billing-
master' committed in 0.087715 seconds
May 27 06:46:08 [3]: Processed 1 of 1 found failed outgoing sessions
May 27 06:46:08 [5]: Session 366C2DAA 884D11E0 8935000C 2938E4DC/10 deleted from
the cache
May 27 06:46:08 [0]: Session 366C2DAA 884D11E0 8935000C 2938E4DC/10 removed
```

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 Session** function.

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

The advantage of this method is that you can view all call attempts, including unsuccessful calls, with the disconnect reasons displayed. You can also see the billing history for a call.

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

2. Incoming Calls from PSTN

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 **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.

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	
Make sure configuration of general and network settings has already been performed according to the instructions in the *Basic SIP Service* section.	[ ]
Make sure 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 (Vendor)	
Create a tariff A, which describes the costs of incoming DIDs from this DID provider (it should not be of the **Routing** type).	[ ]
Enter rates in tariff A for the incoming DID range.	[ ]
Create a vendor for the DID supplier.	[ ]
Create a "VoIP from Vendor" connection for this vendor using tariff A.	[ ]
Rating configuration (Customer)	
Create a tariff B, which will be applied to SIP subscribers for their incoming calls.	[ ]
Enter rates in tariff B for the destinations covering your incoming DID range, i.e. the rates you plan to charge your customers.	[ ]
Create a new rating entry in your SIP product, using the PortaSIP node, tariff B and the access code INCOMING.	[ ]
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 a Vendor 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 **Rating** 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 *Basic SIP Service* section for a detailed description of the available tariff attributes). 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**.

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 🔳 **Rates** icon previous 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**.
3. Fill in the required information; please refer to the *Basic SIP Service* section for a detailed description of the available rate attributes.
4. Click the 💾 **Save** button in the toolbar, or the 💾 icon on the left end of the row.
5. Repeat these steps if you need to enter more rates.

Create a 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 **Participants** 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* section for a detailed description.
4. Click 🖫 **Save**.
5. Three more tabs now appear on the screen. As a rule, the provider will assign you a username / password authorizing you to receive calls from their server. In this case, to create the account for digest authorization:
   1. Click the **Accounts** tab, then click ➕ **Add** and enter this information.
   2. Click the 🖫 **Save** button in the toolbar, or the icon on the left end of the row.

6. 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 **Participants** section of the admin interface, choose **Vendors**.
2. Click on the **Connections** icon next to the vendor name.
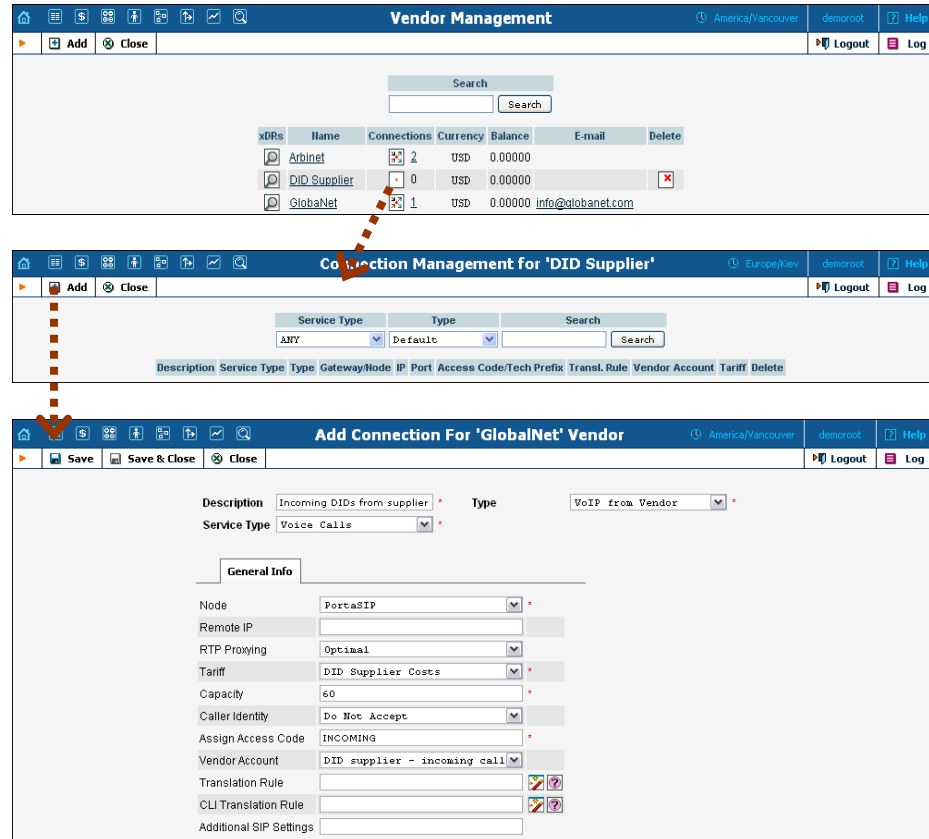
3. Press ![+] **Add** to add a new connection.
4. Fill in the connection information:
   o Choose VoIP from Vendor as the **Type** of connection and your PortaSIP® node as the **Node**, then select the tariff which defines your incoming DID costs for this vendor.
   o For VoIP connections where you have been assigned a login name and password, select the corresponding vendor account in the **Vendor Account** menu.
   o In **RTP Proxying,** specify this vendor's capabilities regarding NAT traversal (if you are not sure whether this vendor can do NAT traversal, select **On Nat**).
   o If you would like to use different rate plans for the same service based on where the call comes from, specify the corresponding access code in the **Assign Access Code** field. This allows the appropriate entry specified in the product's **Services and Rating** tab to be selected.
   For example, the default value for a "VoIP from Vendor" connection (applied to calls to the PortaSIP server arriving from
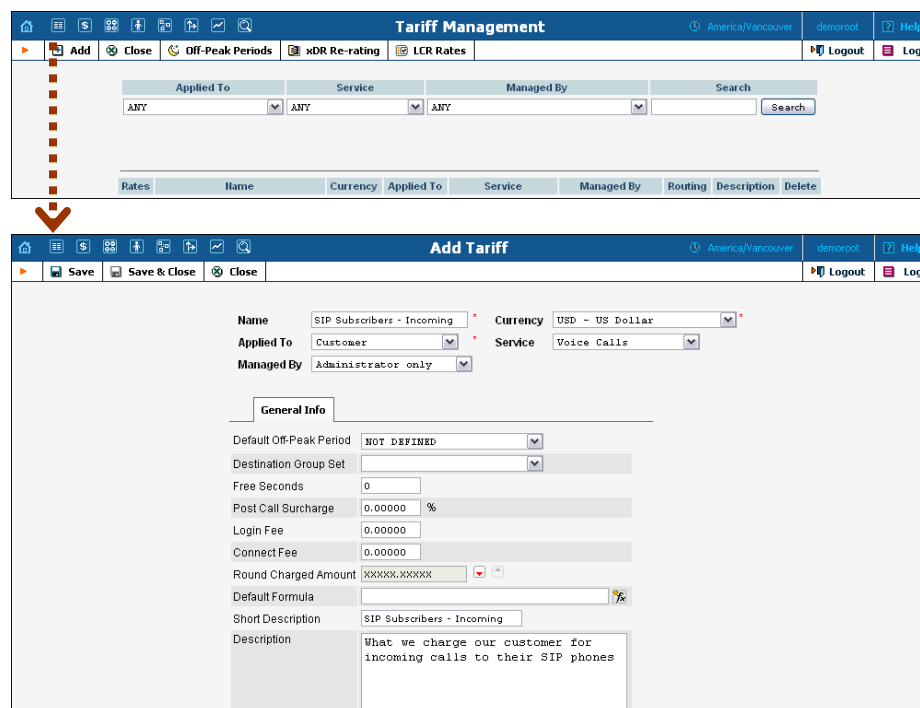
outside your network and terminated to one of your SIP phones) should be set to 'INCOMING'.

Description and **Capacity** are mandatory for all connection types.

5. Click ◼ **Save**.

Create a Customer 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 **Rating** 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 the currency, in which you charge your customers.

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

   o **Applied To** – Choose **Customer** here, since this tariff will be applied to your subscribers.

- 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 **Default Off-peak Period** – If you do not differentiate between peak and off-peak rates, just choose **Not defined**; otherwise, select one of the previously defined off-peak periods.
- 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 **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 *Call Billing Parameters* section in the **PortaBilling Administrator Guide** 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.
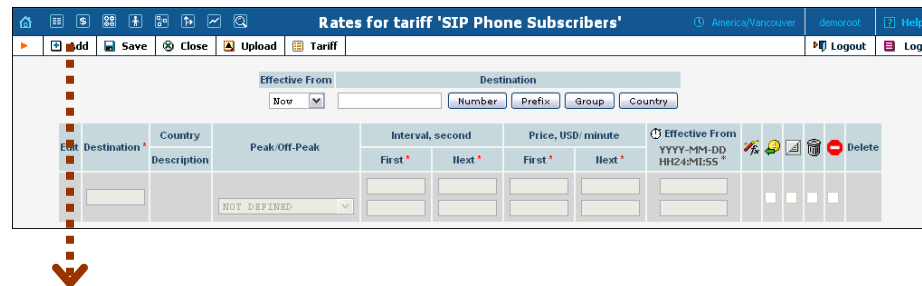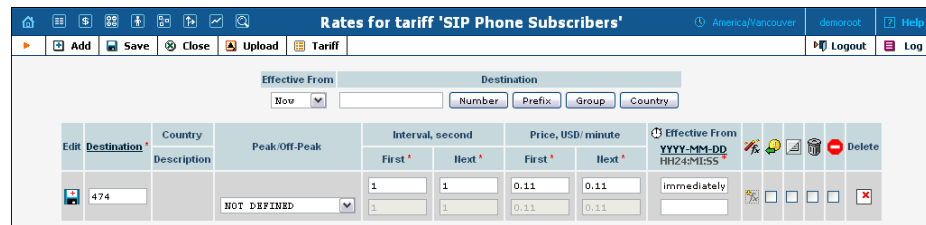
1. On the Tariff Management page you will see a list of the available tariffs. Click the ⊞ **Rates** icon previous to the name of the tariff. When you are in Tariff Management for a particular tariff, click on **Rates** in the toolbar.
2. On the **Edit Rates** screen, click ⊞ **Add**.
3. Fill in the 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 **Payback Rate**, **Hidden**, **Forbidden** or **Discontinued** flags are optional.

4. Click the ![save] **Save** button in the toolbar, or the ![icon] 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 *Rate Import* chapter of the **PortaSwitch Handbook: Wholesale Services** and **PortaBilling Templates Guide** for more information regarding this.

Modify Rating List for a Product

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

1. In the **Rating** section of the Admin-Index page, choose **Products**.
2. Click on the product name to enter the Edit Product screen.

3. After selecting the **Services and Rating** tab, click the ⊞ **Add** icon.
4. Choose **Voice Calls** in the **Service** select menu.
5. In the **Node** select menu select the PortaSIP®.
6. Type INCOMING 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 **Tariff** menu, select the tariff which applies to your customers for receiving incoming calls.
9. Click 🖫 **Save&Close** to save this rating entry.

Create Accounts

1. Go to the **Customers** screen (the one containing a list of customers). It should resemble the screenshot below:
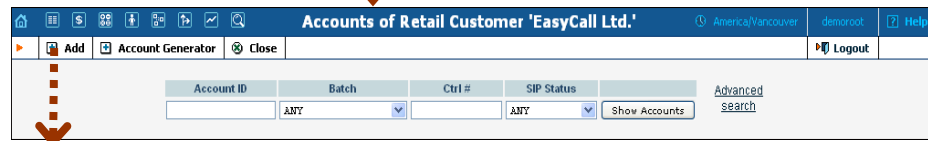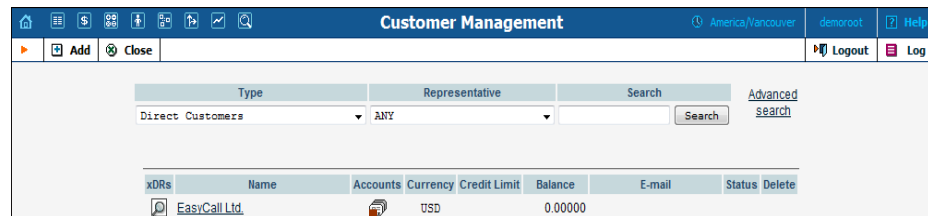
2. Next to the customer name, click the 🖼 icon (the one in the **Accounts** column) to go to account management for that customer.
3. Click on ⊞ **Add**.
4. In this example, we assume that you are creating accounts manually, so choose **Manually** from the drop-down menu (do not use any of the Quick Form options).

5. 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 **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 **Service 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 **E-mail** – 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. If a new batch name is provided in the text field, the created account will be placed into a new batch with the given name. Alternatively, an existing batch could be selected from the drop-down list. If you leave this field empty, the created account will not be tied to any specific 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 Phones* section.

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 **Life Time** – 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.

o **IP Centrex Self-Care Enabled (demo)** – If enabled, account owners will use a new self-care interface to view their profile data and manage settings. To switch back to the default account self-care interface, simply disable this option.

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.

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

3. Calls between SIP Phones (On-Net Calls)

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	
Make sure configuration of general and network settings has already been performed according to the instructions in the *Basic SIP Service* section.	[　]
Make sure 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 (Vendor)	
Enter rates in tariff A (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.	[　]
Rating configuration (Customer)	
Enter rates in tariff B (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.	[　]
Make sure that tariff B is assigned to the PortaSIP node with the access code OUTGOING in your SIP product's rating list.	[　]
Account provisioning	
Check that the retail SIP service customer has already been created according to the instructions in the *Basic SIP Service* section.	[　]
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.

Vendor 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 **Rating** 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 previous to the name of the 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**.

4. Fill in the required information; insert '**|**' (pipe) in the **Destination** field. Leave the other fields in the form as they are, because there is no need to define actual prices for calls going between individual IP phones. This will create a special rate that matches any dialed phone number.

5. Click the ![save] **Save** button in the toolbar, or the ![icon] icon on the left end of the row.

Modify a Customer 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.
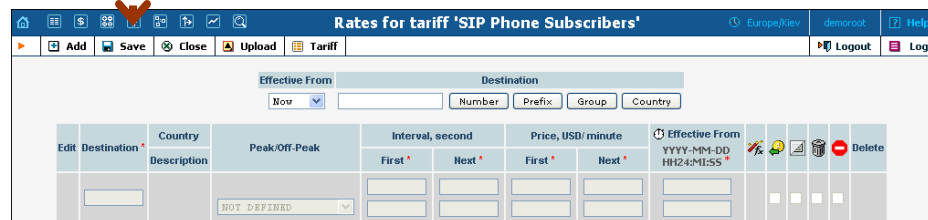
1. In the **Rating** section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, click on the ![icon] icon in the table row containing your tariff for outgoing SIP calls.
3. On the **Edit Rates** screen, click ![add] **Add**.
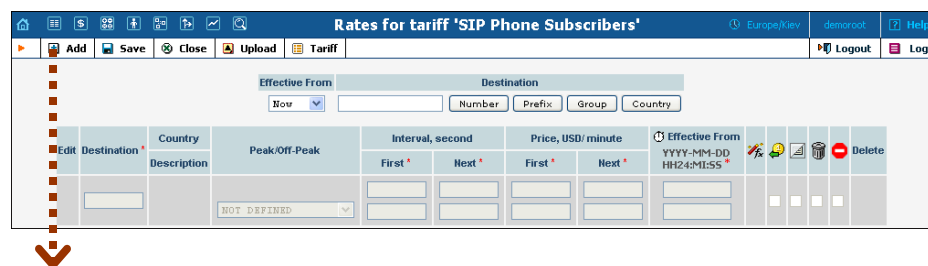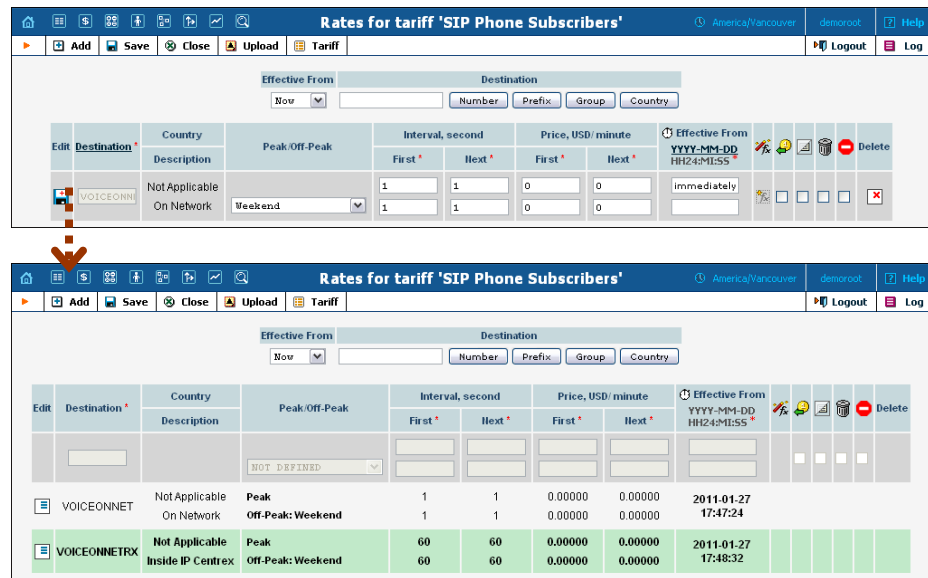
4. Fill in the required information
   - **Destination** – In this case there are two destination prefixes you will use. VOICEONNETRX refers to all calls made to IP phones of the **same** customer (within a single IP Centrex environment). VOICEONNET specifies calls made to any IP phone within this environment.
   - 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 *Rate Import* chapter of the **PortaSwitch Handbook: Wholesale Services** and **PortaBilling Templates Guide** for more information regarding this.

Modify Rating List for a 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 Rating section of the Admin-Index page, choose **Products**.
2. Click on the product name to enter the Edit Product screen.
3. After selecting the **Services and Rating** tab, click the ⊞ **Add** icon.
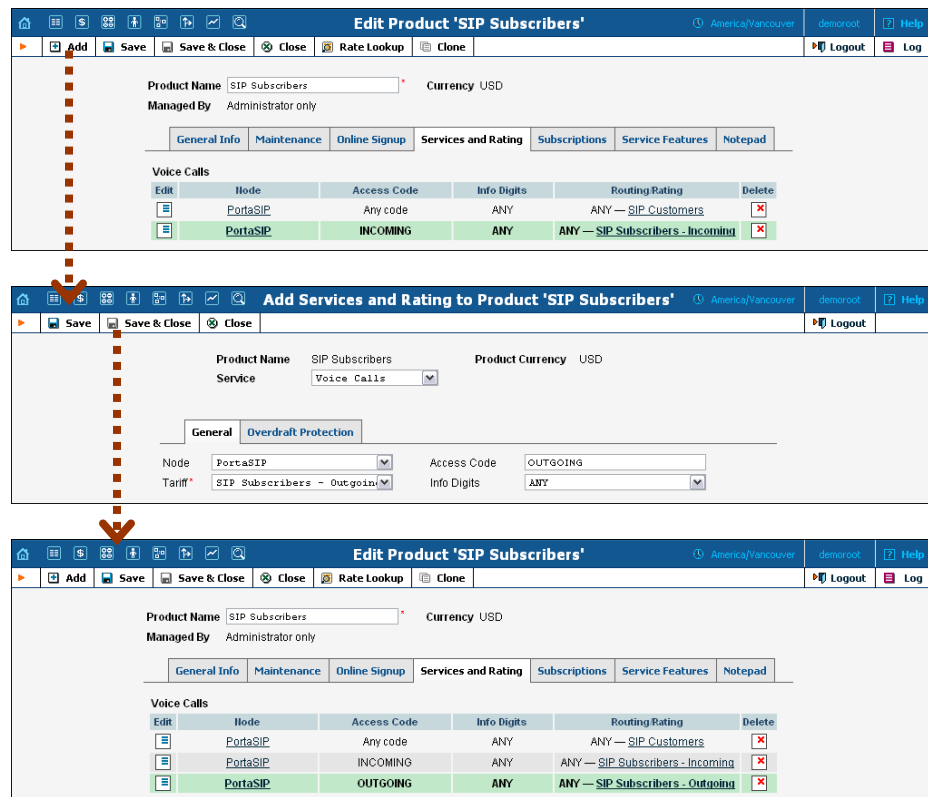
4. Choose **Voice Calls** in the **Service** select menu.
5. In the **Node** select menu, select the PortaSIP® node.
6. Type OUTGOING in the **Access Code** column.
7. **Info Digits** do not apply to this type of service (i.e. you do not differentiate between calls your customers receive from fixed or mobile networks). Therefore, just leave this field empty.
8. In the **Tariff** menu, select the tariff which applies to your customers' outgoing calls.
9. Configure overdraft protection for this product on the **Overdraft Protection** tab. (Please consult the *Rating* section in the **PortaBilling Web Reference Guide** for a description of the parameters available here.)
10. Click ▤ **Save&Close** to save this rating entry.

4. Managing Available Phone Numbers (DID Inventory)

This chapter demonstrates how to efficiently manage a large number of phone numbers (DIDs) used for your SIP service. It will assist you in importing a set of DID numbers into PortaBilling®, managing them, allocating them to your distributors or resellers and, finally, provisioning them on SIP phones.

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.

Using DID Inventory To Manage Phone Number Allocation

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.

⚠️ The checklist below assumes that you have already configured the basic SIP and PSTN-to-SIP services according to the instructions provided in the previous chapters.

Operation	Done
Initial configuration	
Make sure configuration of general and network settings has already been done according to the instructions in previous chapters.	[ ]
Make sure basic SIP service configuration has already been done according to the instructions in previous chapters.	[ ]
PSTN-to-SIP service configuration has already been done according to the instructions in previous chapters.	[ ]
Rating configuration	
Create a DID supplier vendor.	[ ]
Create a DID batch for this vendor.	[ ]
Upload the DID list to this batch.	[ ]
Create an owner batch.	[ ]
Allocate some DIDs to this batch.	[ ]
Account provisioning	
Check that the retail SIP service customer has already been created according to the instructions in previous chapters.	[ ]
Create a new account for this customer, with an 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.	[ ]

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Create a DID Supplier Vendor

This entity is required in order to keep track of your incoming DID expenses, as well as to provide an adequate level of security on your network. If you have already created a vendor while setting up a PSTN-to-SIP service, please proceed to the next step.

1. In the **Participants** 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 section for a detailed description.
4. Click 💾 **Save**.
5. If you need to enter a vendor-specific configuration (connections, vendor accounts), please refer to the instructions in the previous chapters.

Create a DID Batch for a Vendor

A DID batch groups together a set of phone numbers (typically from the same city or area) provided by a particular vendor.

1. In the **Networking** section of the admin interface, choose **DID Inventory**.
2. On the DID Inventory page, choose 📋 **Vendor Batches**.
3. Press ➕ **Add** to add a new batch.
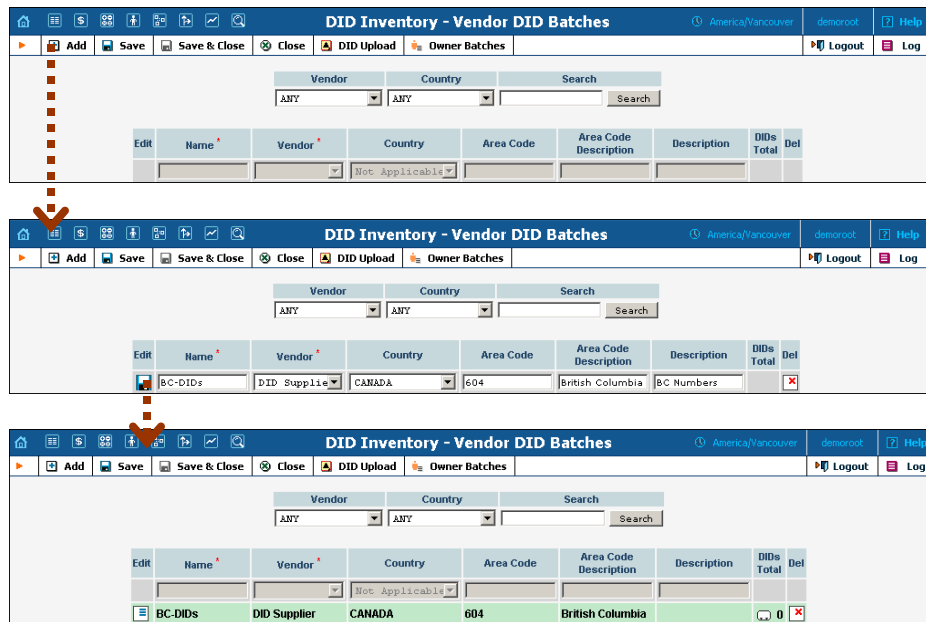
4. Click the 🔲 **Save** button in the toolbar, or the 🔲 icon on the left end of the row.
5. Repeat steps 1-4 if you need to create more DID batches.

Upload DID Numbers

Now we must fill in the DID batch with actual numbers. This is done by uploading a CSV file containing the numbers into PortaBilling®. (There is no operation for adding a single number into the batch from the web interface, as this is practically never needed; every DID supplier will be selling you numbers by the hundreds or even thousands.)

1. Prepare a CSV file. This file has a very simple structure, with only two columns:

    *   The first column contains the DID phone number (in E.164 format).
    *   The second column contains the batch name; make sure the name matches that of the batch you have created.

    See the sample file below:
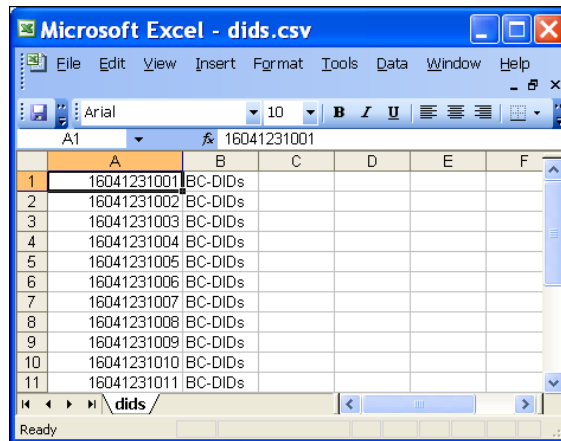
2. In the DID Inventory page, click ▲ **DID Upload** in the toolbar.
3. Either enter the name of your file manually, or click **Browse…** and choose the file.


4. Click 🖫 **Save&Close**.
5. You will see a DID management screen displaying the numbers you have just uploaded, similar to the one below.
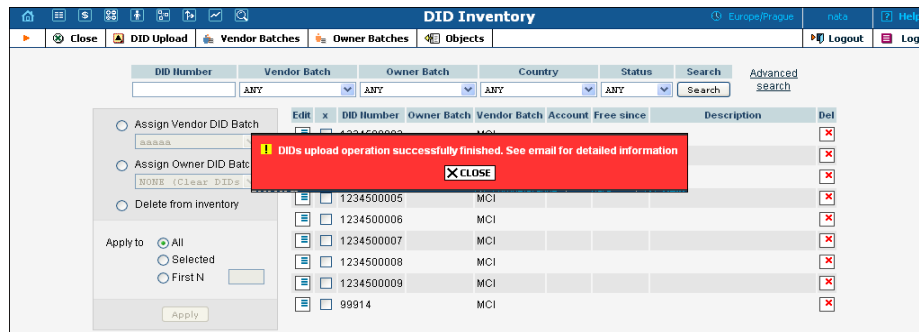
Create an Owner DID Batch

The owner DID batch is a set of DIDs to be used for a particular purpose (e.g. to be distributed by a particular reseller). Even if resellers are not being used, this allows you to clearly identify which DID numbers should be used by your administrators, and which should be kept in reserve.

1. In the **Networking** section of the admin interface, choose **DID Inventory**.
2. On the DID Inventory page, choose ᵻ **Owner Batches**.
3. Press ✚ **Add** to add a new batch.
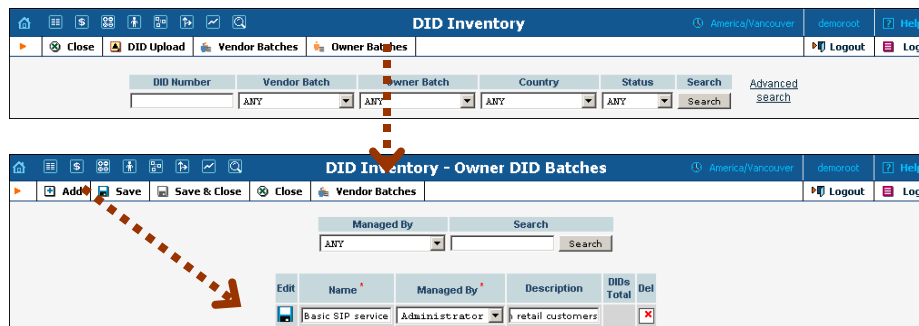
4. Click the **Save** button in the toolbar, or the icon on the left end of the row.
5. Repeat steps 1-4 if you need to create more owner DID batches.

Allocate DIDs to the Owner Batch

Now you just need to specify which DIDs will be included in this batch.

1. In the **Networking** section of the admin interface, choose **DID Inventory**.
2. On the DID Inventory page, choose a specific vendor batch in the **Vendor Batch** select menu and press **Search**.
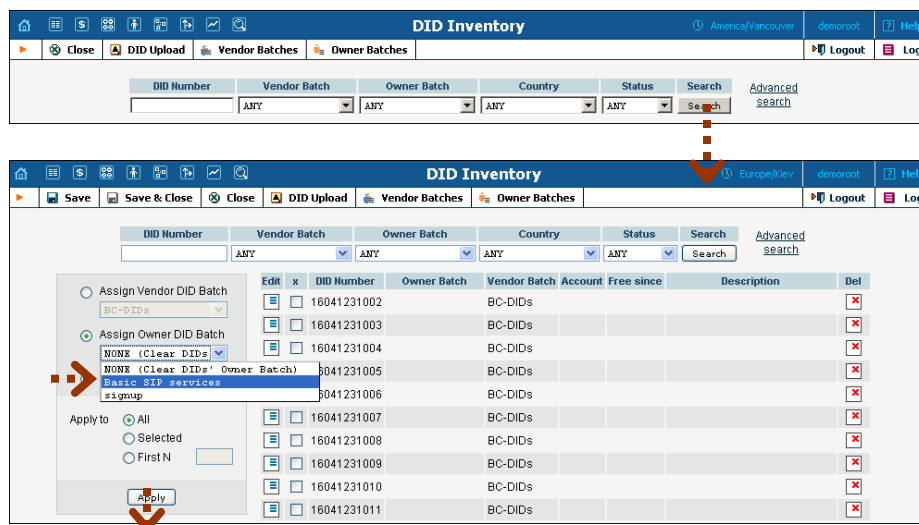

3. Select the appropriate DIDs or, alternatively, use the **First N** option to automatically process a large number of DIDs at once.
4. Choose the desired operation; in this case, choose Assign Owner DID Batch and select which owner batch these DIDs should be assigned to.
5. Click the **Apply** button. On the refreshed screen you will see that these DIDs have now been assigned to the owner batch.
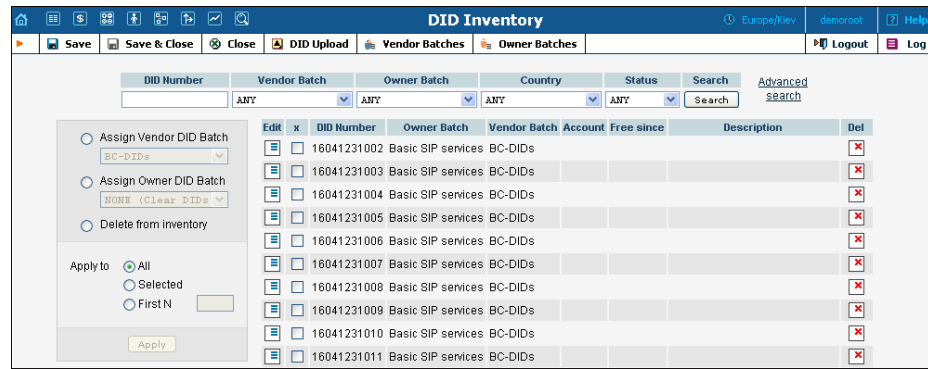
Create an Account

1. Go to the **Customers** screen (the one containing the list of customers).
2. Next to the customer name, click the ![icon] icon (the one in the **Accounts** column) to go to account management for that customer.
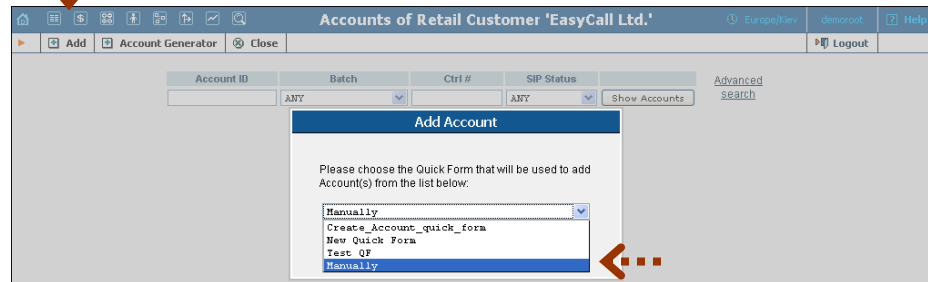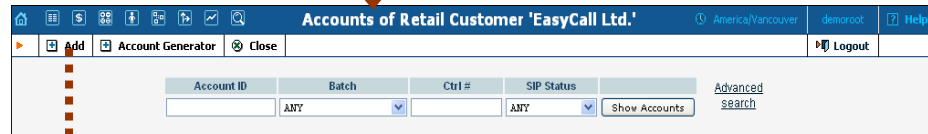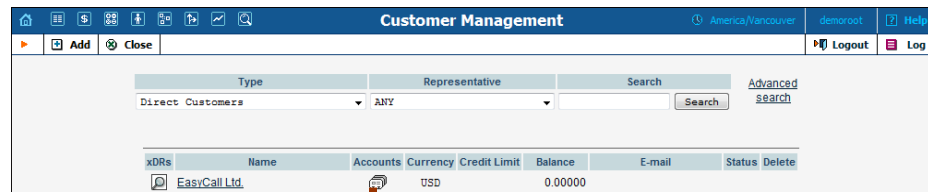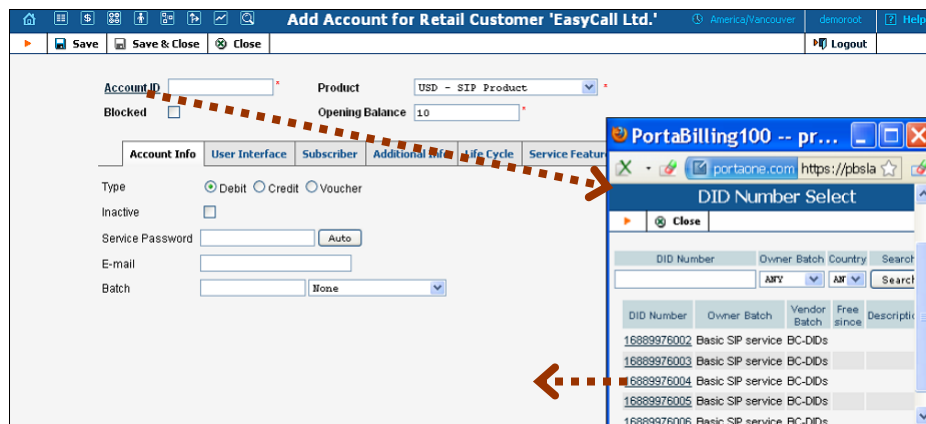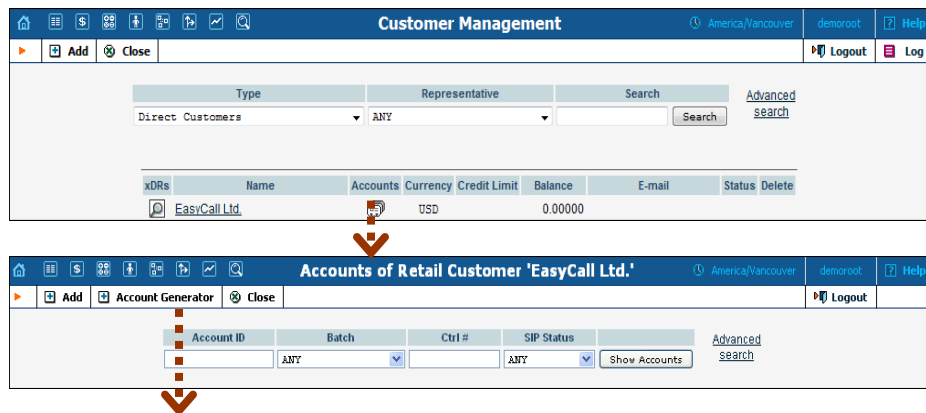
3. Click on ➕ **Add**.
4. In this example, we assume that you are creating the account manually, so choose **Manually** from the drop-down menu (do not use any of the Quick Form options).
5. When entering data for a new account, you will notice that Account ID is now a link. Click on it and a popup window appears, from which you can choose one of the available DID numbers.
6. Finish creating the account according to the instructions given in the previous chapters.

Create a Group of Accounts

If a customer (e.g. a small enterprise) buys multiple phone numbers, there is a convenient way of creating many accounts at once and allocating proper DID numbers to all of them.

1. Go to the **Customers** screen (the one containing the list of customers).
2. Next to the customer name, click the 🖅 icon (the one in the **Accounts** column) to go to account management for that customer.
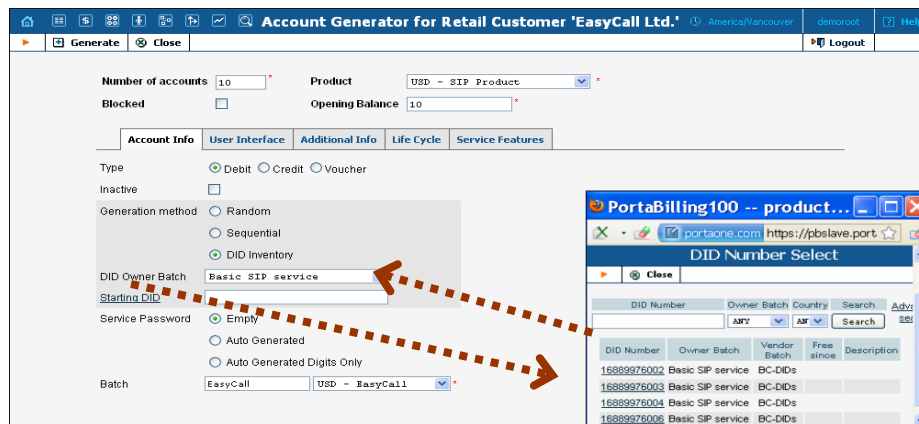
3. Click on ➕ **Account Generator**.
4. Choose DID Inventory as the **Generation Method**.
5. Select a corresponding DID batch from the **DID Owner Batch** menu.
6. If you wish, you can enter the starting DID manually or use the popup menu, e.g. if you want to make sure that the customer's phone range starts with a "round" phone number such as …00, with the following numbers …01, …02 and so on.
7. Specify the other parameters according to the instructions given in the **PortaBilling Web Reference Guide**.
8. Click ➕ **Generate** to start the account generation process.

5. 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 PortaBilling® 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 **Networking** 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 (e.g. [--key $B]http://PB_SLAVE_SERVER/$A/$MA.cfg for Sipura / Linksys VoIP products, [--key $B]http://PB_SLAVE_SERVER/cfg$MA for

	Grandstream.) Replace PB_SLAVE_SERVER with the actual hostname or IP address of your PortaBilling® 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 makes a request for a configuration file, $MA will be automatically replaced by its MAC-address, so leave this as is (if you enter the actual MAC address, you will not be able to 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.
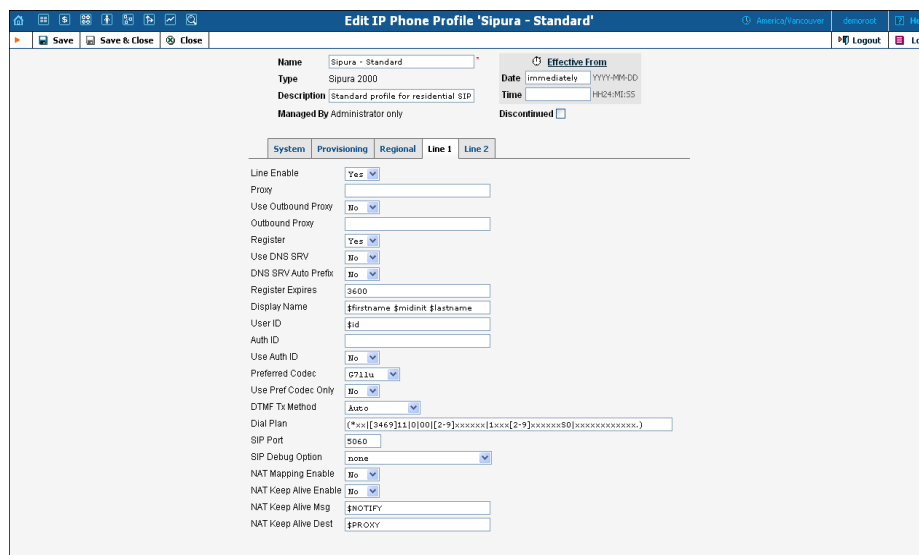
In the **Line 1** and **Line 2** tabs, you can specify parameters for both phone ports of your IP phone.

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.

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Create IP Phone Entry

1. In the **Networking** 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. Click on the **IP Phone** link, and select the IP phone on the page that opens in a new browser tab.
3. Choose a specific port from the **IP Phone Port** drop-down box, if the device has multiple phone ports.
4. Click 🖫 **Save & Close**.

Note: The **Select IP Phone** page 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 `/porta_var/<Server_IP>/tftp` on the PortaBilling® 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 (for Sipura
   / Linksys VoIP products),
   [--key $B]http://PB_SLAVE_SERVER/cfg$MA (for
   Grandstream.)
   ```
   Replace PB_SLAVE_SERVER with the actual hostname or IP address of your PortaBilling® 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 `/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 `/porta_var/<Server_IP>/tftp`. All provision configuration files are stored with a $env-subfolder. Any IP phone can access its configuration file just by sending a request to the http-port of the PortaBilling® 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.

6. 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** – Service password for your SIP account
      iv. **Use Auth ID** – no
5. Submit all the changes and update the SPA configuration.

SIPURA
technology, inc.

Sipura Phone Adapter Configuration

Info | System | SIP | Provisioning | Regional | Line 1 | Line 2 | User 1 | User 2

User Login   basic | advanced

System Information

DHCP:	Enabled	Current IP:	192.168.0.88
Host Name:	SipuraSPA	Domain:	portaone.com
Current Netmask:	255.255.255.0	Current Gateway:	192.168.0.192
Primary DNS:	192.168.0.192		
Secondary DNS:	207.102.99.66 207.102.99.82		

Product Information

Product Name:	SPA-2000	Serial Number:	88012BA66086
Software Version:	2.0.10(e)	Hardware Version:	2.0.1(0905)
MAC Address:	000E08AB4638	Client Certificate:	Installed

System Status

Current Time:	1/8/2003 14:17:56	Elapsed Time:	4 days and 02:23:13
Broadcast Pkts Sent:	0	Broadcast Bytes Sent:	0
Broadcast Pkts Recv:	560688	Broadcast Bytes Recv:	34980083
Broadcast Pkts Dropped:	0	Broadcast Bytes Dropped:	0
RTP Packets Sent:	3074	RTP Bytes Sent:	120568
RTP Packets Recv:	2341	RTP Bytes Recv:	54292
SIP Messages Sent:	1724	SIP Bytes Sent:	1167889
SIP Messages Recv:	362	SIP Bytes Recv:	166405
External IP:			

**Line 1 Status**

Hook State:	On	Registration State:	Registered
Last Registration At:	1/8/2003 14:07:33	Next Registration In:	2947 s
Message Waiting:	No	Call Back Active:	No
Last Called Number:	16044680035	Last Caller Number:	
Mapped SIP Port:			
Call 1 State:	Idle	Call 2 State:	Idle
Call 1 Tone:	None	Call 2 Tone:	None
Call 1 Encoder:		Call 2 Encoder:	
Call 1 Decoder:		Call 2 Decoder:	
Call 1 FAX:		Call 2 FAX:	
Call 1 Type:		Call 2 Type:	
Call 1 Remote Hold:		Call 2 Remote Hold:	
Call 1 Callback:		Call 2 Callback:	
Call 1 Peer Name:		Call 2 Peer Name:	
Call 1 Peer Phone:		Call 2 Peer Phone:	

Network Settings

SIP TOS/DiffServ Value:	0x68	Network Jitter Level:	high
RTP TOS/DiffServ Value:	0xb8		

SIP Settings

SIP Port:	5060	SIP 100REL Enable:	no
EXT SIP Port:		Auth Resync-Reboot:	yes
SIP Debug Option:	none		

Call Feature Settings

Blind Attn-Xfer Enable:	no	MOH Server:	
Xfer When Hangup Conf:	yes		

Proxy and Registration

Proxy:	216.231.44.168	Use Outbound Proxy:	no
Outbound Proxy:		Use OB Proxy In Dialog:	yes
Register:	yes	Make Call Without Reg:	no
Register Expires:	3600	Ans Call Without Reg:	no
Use DNS SRV:	no	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	3600		

Subscriber Information

Display Name:		User ID:	1206001236
Password:	*************	Use Auth ID:	no
Auth ID:			
Mini Certificate:			
SRTP Private Key:			

Supplementary Service Subscription

Call Waiting Serv:	yes	Block CID Serv:	yes
Block ANC Serv:	yes	Dist Ring Serv:	yes
Cfwd All Serv:	yes	Cfwd Busy Serv:	yes
Cfwd No Ans Serv:	yes	Cfwd Sel Serv:	yes
Cfwd Last Serv:	yes	Block Last Serv:	yes
Accept Last Serv:	yes	DND Serv:	yes
CID Serv:	yes	CWCID Serv:	yes
Call Return Serv:	yes	Call Back Serv:	yes
Three Way Call Serv:	yes	Three Way Conf Serv:	yes
Attn Transfer Serv:	yes	Unattn Transfer Serv:	yes

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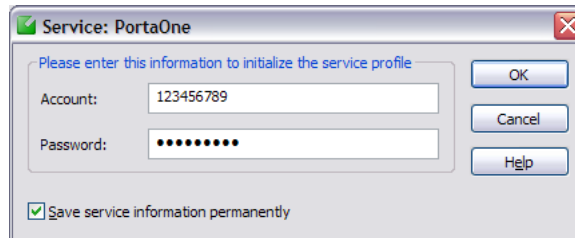
2. Key in the account ID and password for PortaSIP® and press OK. The SJPhone display should be similar to the one in the illustration below, showing the account balance in "Ready to call" state. The phone is now ready to be used.
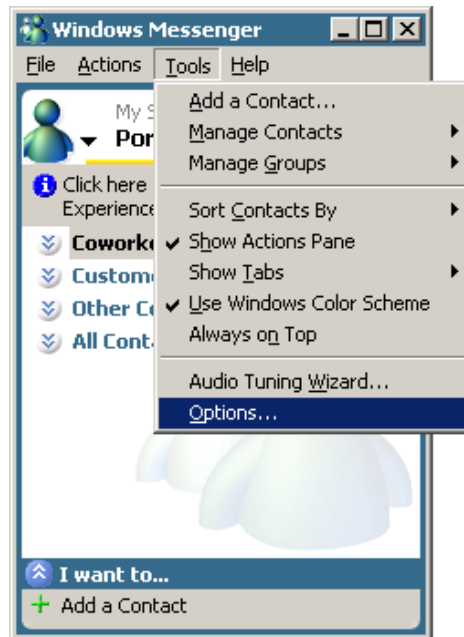

3. Right click on the softphone and press "Login…" to change or make corrections to the account / password.

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.
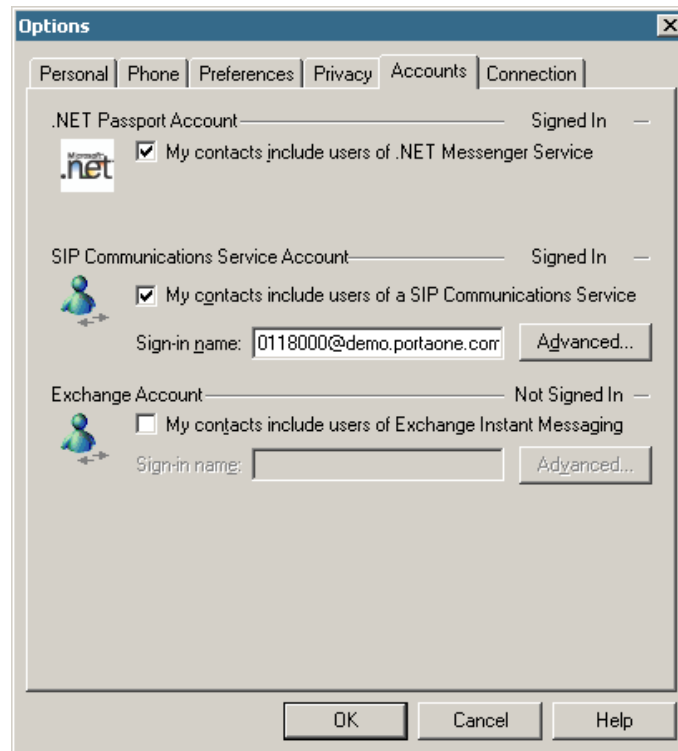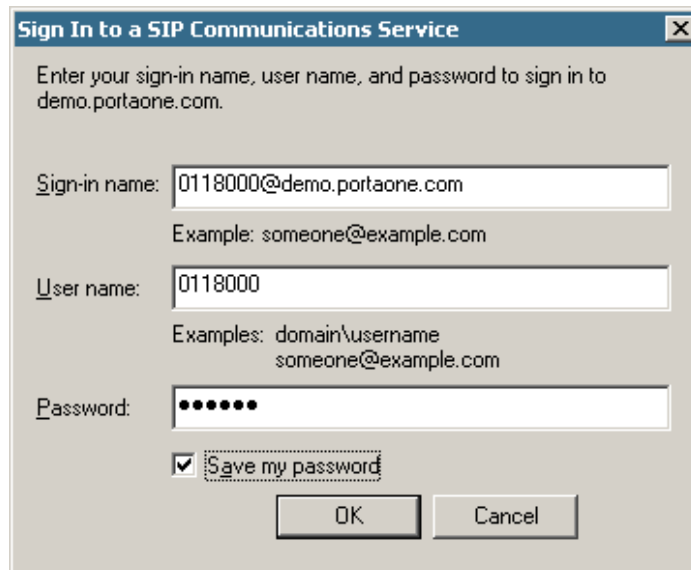
3. Click the "Configure settings" radio button and enter the "Server name of IP address" using either the IP address of the PortaSIP® server or its name in DNS. Make sure that the "UDP" radio button is selected, then click OK.
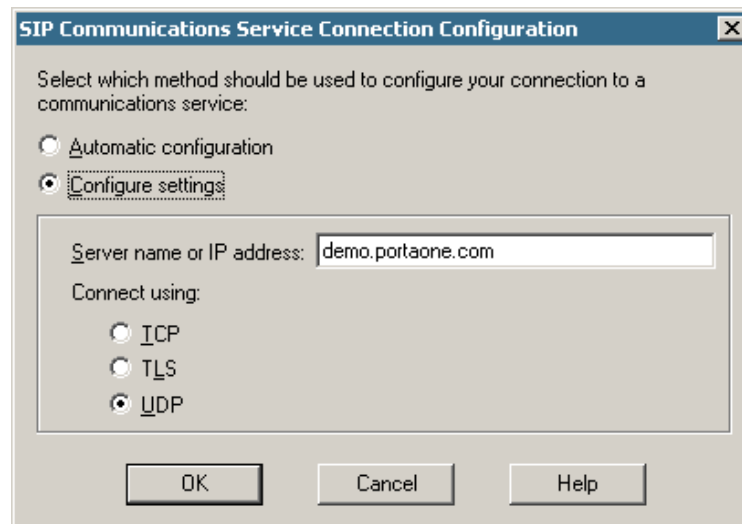
4. Sign out and then sign in again. You should see the pop-up dialog below. Fill it in as follows: "Sign-in name" in the form *username@address*, where *username* is the name of the appropriate account in PB and *address* is either the IP address of the PortaSIP® server or its name in DNS. Enter the 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.

Sign In to a SIP Communications Service   ☒

Enter your sign-in name, user name, and password to sign in to
demo.portaone.com.

Sign-in name: 0118000@demo.portaone.com

            Example: someone@example.com

User name: 0118000

            Examples: domain\username
                           someone@example.com

Password: ••••••

       ☑ Save my password

            [OK]   [Cancel]

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.

Start a Voice Conversation   ☒

      My Contacts   **Other**

Enter the e-mail address of the person you want to contact.

Type the person's complete e-mail address:

1-604-521-5277

Select the service that this person uses:

SIP Communications Service   ▼

             [OK]   [Cancel]

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 2002
- Sipura 2100
- Sipura 3000
- Linksys PAP2
- Linksys RTP-300
- Linksys/Sipura SPA-2102
- Linksys SPA-942
- Linksys SPA-921
- Linksys SPA-922
- Linksys SPA-3102
- Linksys SPA-941
- Linksys SPA-962
- Linksys WRT54GP2
- GrandStream GXW400x
- GrandStream HT286
- GrandStream HT486
- GrandStream HT488
- GrandStream HT496
- GrandStream HT502
- Thomson TWG850 (only eMTA part)

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.

APPENDIX E. Configuring Interoperability with DIDX

If your customers plan that calls to DID numbers provided by DIDX will be forwarded to their SIP phones, you need to configure interoperability with DIDX.

Configuration on the PortaSwitch Side

Create a tariff for incoming DID costs and define rates

1. In the **Rating** section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, choose ⊞ **Add**.
3. Fill in the **Add Tariff** form. In the **Applied To** select menu, choose **Vendor**. Then clear the **Routing** check-box, since no routing is actually being done for this vendor, i.e. the vendor will be sending calls to your network.

4. Click 🖫 **Save.**

5. Click on the Rates button, then click ⊞ **Add.** Enter the rates applied to you by the DID provider.
6. Click 🖫 **Save.**

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

Create a DIDX vendor

This entity is required in order to keep track of your incoming DIDX 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 that will be used for such authorization.

1. In the **Participants** 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* section for a detailed description.
4. Click ▣ **Save**.
5. Click on 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 description (visible when associating the account with a connection).
   - **Login** – Enter one of the IP addresses of the DIDX gateways, and use **cisco** as the password. For an IP address

> list, refer to the DIDX Frequently Asked Questions: For Buyers
> page.
>   - **Password** – Enter **cisco**.
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

A connection is the point where calls enter your network via the DIDX gateway.

1. On the Edit Vendor page, press the 🎲 **Connections** button.
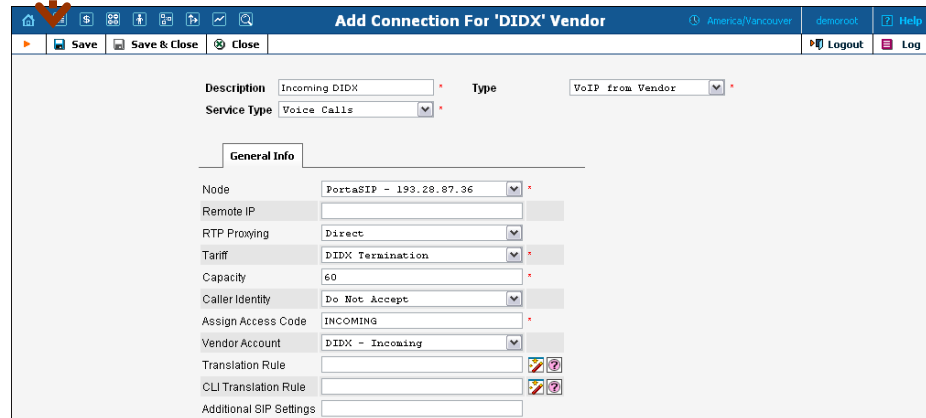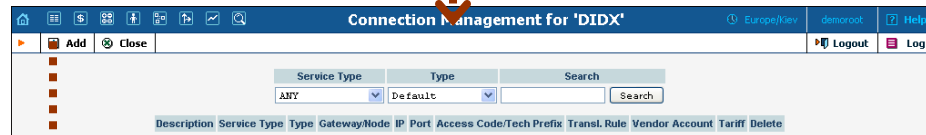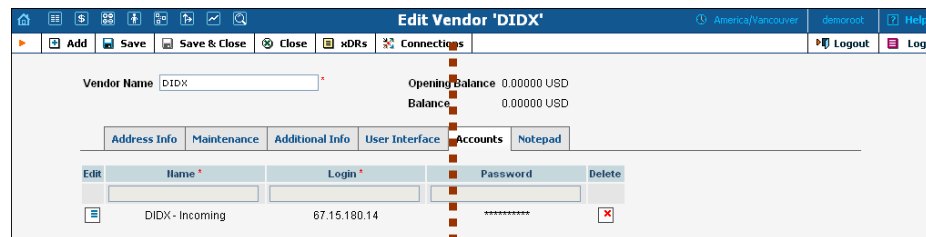

2. Click ➕ **Add** to add a new connection.
3. Fill in the connection information. Choose **VoIP from Vendor** as the **Type** of connection and your PortaSIP® node as the **Node**, then select the tariff which defines your incoming DIDX costs.
4. In the **Vendor Account** select menu, choose the account you previously created in the vendor configuration; this account will be used to authorize incoming calls from the vendor.
5. In **RTP Proxying,** specify this vendor's capabilities for NAT traversal (if you are not sure whether this vendor can perform NAT traversal, select **On Nat**).
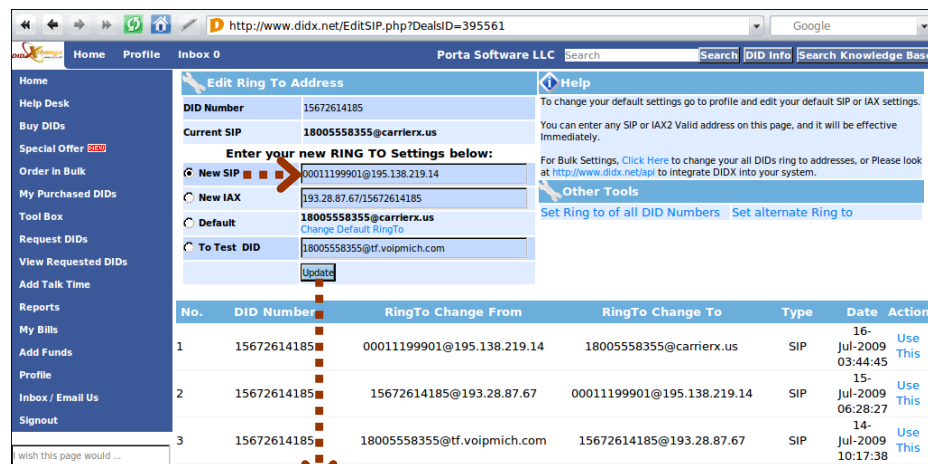
6. If you would like to use different rate plans for the same service based on where the call comes from, specify the corresponding access code in the **Assign Access Code** field. This allows the appropriate entry specified in the product's **Services and Rating** tab to be selected. For example, the default value for a "VoIP from Vendor" connection (applied to calls to the PortaSIP server arriving from outside your network and terminated to one of your SIP phones) should be set to 'INCOMING'.

7. **Description** and **Capacity** are mandatory for all connection types.

8. Click ![save icon] **Save**.

Configuration on the DIDX Side

1. Sign in to the DIDX members' area using the membership ID and password you received from DIDX.

2. Click on **My Purchased DIDs**.

3. View your list of DIDs:
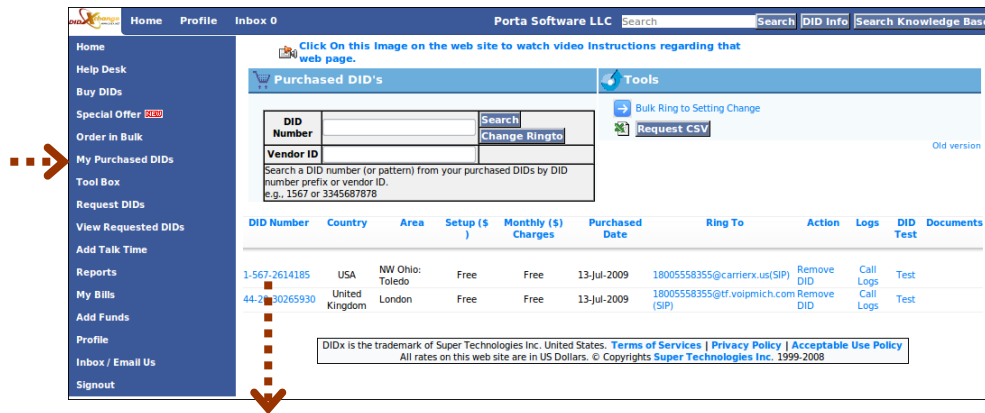
4. Click on the number that you want to point to PortaSwitch®.

5. Select the **New SIP** option, then enter the SIP address in the format: `<account>@<PortaSIP_proxy>` and click **Update**:

6. Make a test call.