

**Porta  Switch™**



**Handbook:**

Residential VoIP and IP Centrex Services  
Maintenance Release 21

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**PortaSwitch Handbook:  
Residential VoIP and IP Centrex Services, August 2010  
Maintenance Release 21  
V.1.21.4**

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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 performing the action described in the paragraph.



**Tips** provide information that might help you solve a problem.

# 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.

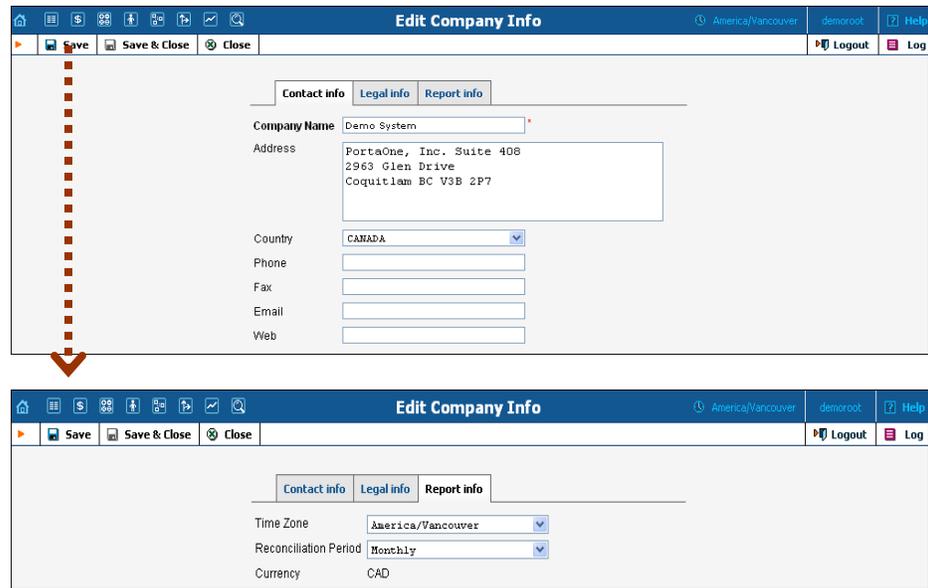
Operation	Done
<b>General configuration</b>	
Enter company data under Company Info and specify a base currency.	[ ]
For any other currency you plan to use, specify the exchange rate source and define exchange rates	[ ]
Create all the required destinations for off-net calls	[ ]
Create a wildcard ‘ ’ destination	[ ]
Create a destination for your SIP numbers	[ ]
<b>Network configuration</b>	
Create a node for your PortaSIP	[ ]
<b>Rating configuration</b>	
Create a tariff (referred to as <b>tariff A</b> later on), which will be applied to SIP subscribers	[ ]
Enter rates in tariff A for the destinations you plan to call (both off-net and SIP-to-SIP calls)	[ ]
Create a tariff (referred to as <b>tariff B</b> later on), which describes your termination costs and routing for off-net calls (make sure it has a <b>Routing</b> type!)	[ ]
Enter rates in tariff B for the destinations you plan to call	[ ]
Create a tariff (referred to as <b>tariff C</b> later on), which describes your termination costs for SIP-to-SIP calls (make sure it is NOT a <b>Routing</b> type!)	[ ]
Enter a zero cost rate for the wildcard destination ( ) in tariff C	[ ]
Create your SIP product	[ ]
Create one accessibility entry for this product, using the PortaSIP node and tariff A	[ ]
Create an off-net calls vendor	[ ]
Create a connection for this vendor using tariff B	[ ]
Create 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 <b>SIP-UA</b> string for this vendor using tariff C	[ ]
<b>Account provisioning</b>	
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	[ ]
<b>Testing</b>	

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 these in the **Currencies** section and specify exchange rates in **Exchange Rates**.



Edit	ISO 4217	Name	Dec. digits	Major	Minor	Exchange Rate Source	Payment System	Payment Method	Minimum Payment	Delete
	alpha num									
	CAD 124	Canadian Dollar	2			Manual	AuthorizeNet		100.00000	
	USD 532	US Dollar	2	dollar	cent	XE.com	AuthorizeNet		10.00000	

## Create Destinations

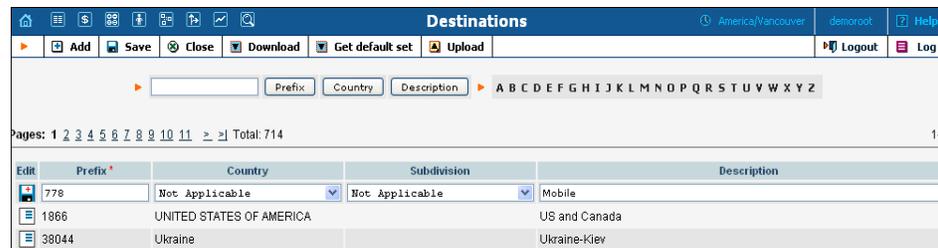
This step is only required if you have not previously defined the necessary destinations. There are two ways to insert a new destination into the system:

- 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).



Edit	Prefix	Country	Subdivision	Description
	778	Not Applicable	Not Applicable	Mobile
	1866	UNITED STATES OF AMERICA		US and Canada
	38044	Ukraine		Ukraine-Kiev

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.

Action	Prefix	Two-letter ISO Country Code	Description
add	670	TL	Proper
add	1684	AS	Proper new
add	1829	DO	Proper new
add	93	AF	Proper
add	93229	AF	Kabul
add	93290090	AF	Operator
add	93321	AF	Kandahar
add	9344000	AF	Herat City
add	9344001	AF	Herat City
add	9344002	AF	Herat City
add	936051	AF	Mazar-I-Sharif
add	93702	AF	Mobile
add	93708	AF	Mobile
add	355	AL	Proper
add	35538	AL	Mobile
add	3554	AL	Tirana
add	35542	AL	Tirana
add	35568	AL	Mobile
add	35569	AL	Mobile

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

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

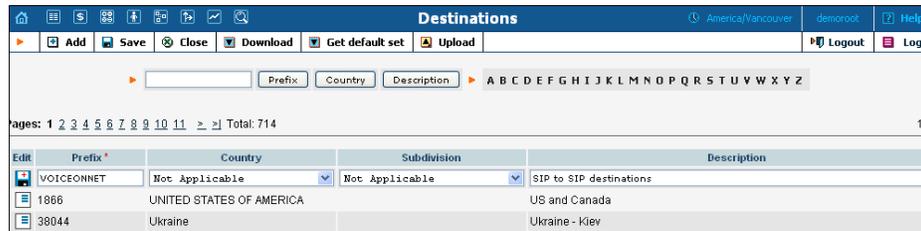
### Destinations for SIP phones

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

Another option would be to create one or more phone prefixes covering the actual phone numbers assigned to your SIP phones, as described below.

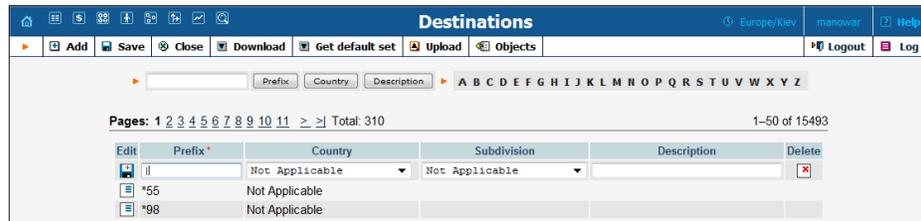
In order to receive an incoming call, an SIP user agent must be configured with a phone number. Normally, you will obtain a range of phone numbers from your local telecom, and you will be able to assign these to your customers. For example, you will be assigned range 12027810000 – 12027819999. It is, therefore, a good idea to create a special destination **1202781**. This prefix will cover all of your SIP phones, and thus its actual purpose is to set up your pricing or routing.

Even if you have not obtained an official phone prefix, it is highly recommended not to assign IDs to your SIP user agents at random. Choose a non-existing prefix, e.g. **777**, and create it as the destination with **N/A** country and the description **SIP phones**. Then use SIP IDs such as **77700001, 77700002, ... 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.

The top screenshot shows the 'Node Management' interface. It has a header with 'Node Management' and user information. Below the header is a toolbar with 'Add' and 'Close' buttons. The main area contains a table with columns: Name, Node ID, IP, Manufacturer, Type, RADIUS Client, and Delete. Above the table are input fields for Node ID, IP, and RADIUS Client, along with radio buttons for 'Yes', 'No', and 'All', and a 'Show Nodes' button.

The bottom screenshot shows the 'Add Node' form. It has a header with 'Add Node' and user information. Below the header is a toolbar with 'Save', 'Save & Close', and 'Close' buttons. The form contains several sections:
 

- Node Name:** A text input field with the value 'DemoSIP'.
- Manufacturer:** A dropdown menu with 'PortaOne' selected.
- Type:** A dropdown menu with 'PortaSIP' selected.
- Node info:** A section containing:
  - Node ID:** A text input field with the value 'demosip.mydomain.com'.
  - NAS-IP-Address:** A text input field with the value '207.52.37.45'.
  - Auth. Transl. Rule:** A text input field.
- RADIUS Client:** A checked checkbox.
- RADIUS Key:** A text input field with the value 'ditthead' and an 'Auto' button.
- RADIUS Source IP:** A text input field with the value '207.52.37.45'.
- RADIUS Dictionary:** A text input field with the value 'Cisco'.
- POD Server:** An unchecked checkbox.

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:
  - **Node Name** – a short descriptive name for your SIP server (this will be used in the select menus).
  - **Manufacturer** – select **PortaOne**.
  - **Type** – VoIP node type; select **PortaSIP**.
  - **Node ID** – PortaSIP server’s hostname (recommended: `hostname.domainname`).
  - **NAS-IP-Address** – the IP address of the SIP server.
  - **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).
  - **Radius Client** – check this, since PortaSIP will need to communicate with the billing.
  - **Radius Key** – enter the radius shared secret here; this must be the same **key** which you entered during the PortaSIP installation.
  - **Radius Source IP** – see the **Node ID, NAS IP address and Radius source IP** section in 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.
  - **POD Server** – this checkbox relates to ISP billing. Leave it unchecked for now.
4. Click **Save&Close**.

## Create Tariff

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

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

The top screenshot shows the 'Tariff Management' page. It has a search bar with 'Applied To', 'Service', and 'Managed By' dropdowns, all set to 'ANY'. Below is a table with columns: Rates, Name, Currency, Applied To, Service, Managed By, Routing, Description, Delete.

The bottom screenshot shows the 'Add Tariff' form. Fields include:
 

- Name: SIP Phone Subscribers
- Applied To: Customer
- Managed By: Administrator only
- Currency: USD - US Dollar
- Service: Voice Calls

 The 'General Info' section includes:
 

- Default Off-Peak Period: startstop:hr(20-5)
- Destination Group Set: (empty)
- Free Seconds: 0
- Post Call Surcharge: 0 %
- Login Fee: 0
- Connect Fee: 0
- Round Charged Amount: XXXXX.XXXXX
- Default Formula: (empty)
- Short Description: SIP Phone Subscriber's Tariff
- Description: This tariff applies to all SIP Phone subscribers

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 in which currency pricing information is defined. All pricing information for a single tariff must be defined in the same currency.

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

- **Applied To** – Designates whether this tariff will be used to charge your customers (in this case, choose a **Customer**) or to calculate costs associated with your vendors (in this case, choose a **Vendor** and also turn on the **Routing** check-box, which will appear below).

- **Managed By** – Choose “**Administrator Only**” here (this option is only visible after you select **Applied to: Customer** above).
  - **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.
  - **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.
  - **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.
  - **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.
  - **Short Description** – A short tariff description. This will be shown in the rate lookup on the admin interface and the self-care pages for your accounts and customers.
  - **Description** – An extended tariff description.
4. Click  **Save**.
  5. Repeat steps 1-4 until you have entered all of the tariffs. You will need at least two tariffs – one, which you will use to charge your customers, and another, which describes your termination costs. Make sure you choose **Vendor** in the **Applied To** select menu and check the checkbox **Routing** when creating tariffs for your vendors.

## Enter Rates

Rates are per-destination prices. Please refer to the *Call Billing Parameters* section in the **PortaBilling Administrator Guide** for more information on billing parameters.

## Managing rates online

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





1. On the Tariff Management page you will see a list of the available tariffs. Click the  **Rates** icon 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:
  - o **Destination** – A destination prefix may be entered directly, e.g. **420** for Czech Republic, or you can access the destinations directory by clicking the **Destination** link (in the column header). Here you can find the desired prefix by country name.

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

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

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

- o **Rate Formula Wizard**  – launches the wizard for creating a custom rating formula
- o **Effective from** – If you want this rate to take effect sometime in the future, you can either type in a date manually, or use the calendar (click on the DD-MM-YYYY link).

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

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

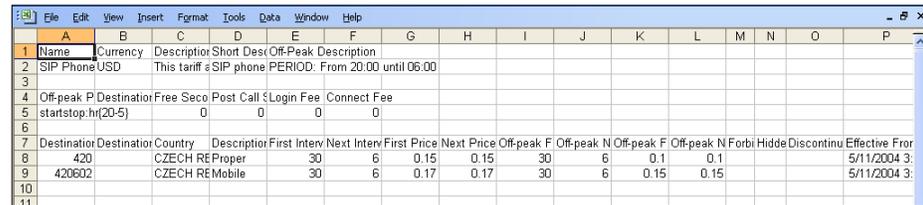
- Click the  **Save** button in the toolbar, or the  icon on the left side of the row.
- Repeat these steps if you need to enter more rates.

### Managing rates offline

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

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

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



Name	Currency	Descriptor	Short Desc	Off-Peak	Description											
SIP Phone USD		This tariff is SIP phone PERIOD: From 20:00 until 06:00														
Off-peak P	Destination	Free Seco	Post Call	Login Fee	Connect Fee											
startstop:hr(20-5)		0	0	0	0											
Destination	Destination	Country	Descriptor	First Interv	Next Interv	First Price	Next Price	Off-peak F	Off-peak N	Off-peak F	Off-peak N	Forbi	Hide	Discontin	Effective	Fr
420	CZECH RE	Proper		30	6	0.15	0.15	30	6	0.1	0.1				5/11/2004	3
420602	CZECH RE	Mobile		30	6	0.17	0.17	30	6	0.15	0.15				5/11/2004	3

- Edit the file by adding more rows with rate data, so that it resembles the screenshot below.
- If you want some of your rates to be effective in the future, enter a future date in the **Effective from** column. If you are retaining past dates, make sure to check the box **Rates with 'effective from' date in the past should be uploaded as effective immediately** when uploading the file (see step 10 below). Otherwise, these rates will fail to upload.



Destination	Destination	Country	Descriptor	First Interv	Next Interv	First Price	Next Price	Off-peak F	Off-peak N	Off-peak F	Off-peak N	Forbi	Hide	Discontin	Effective	Fr
420	CZECH RE	Proper		30	6	0.15	0.15	30	6	0.1	0.1				5/11/2004	3
420602	CZECH RE	Mobile		30	6	0.17	0.17	30	6	0.15	0.15				5/11/2004	3
420601				30	6	0.17	0.17	30	6	0.15	0.15					



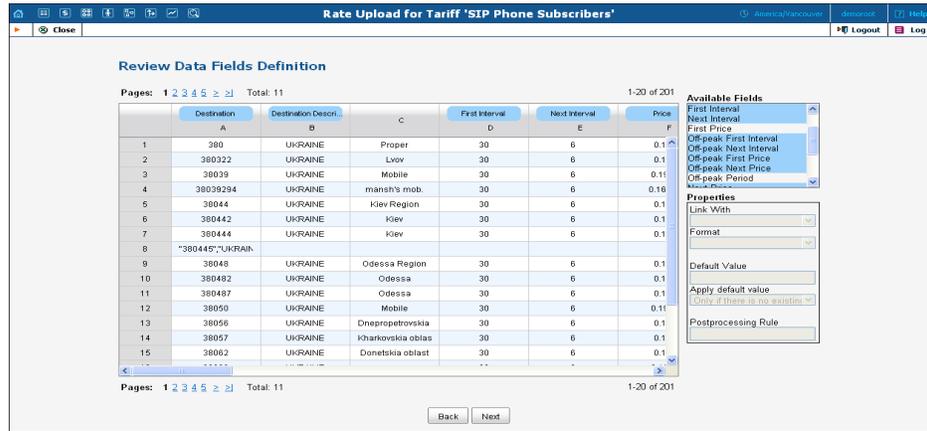
7. Save the file in Excel. You will probably get a warning from Excel that your file “*may contain features that are not compatible with CSV (Comma delimited)*”. Ignore this, and choose **Yes** to retain the CSV format.
8. Close the file in Excel. If you performed step 6, then disregard the message “*Do you want to save the changes you made?*”, since this arises only because your format is not the default Excel XLS format.
9. Go back to the PortaBilling web interface, and then go to the **Tariff** screen.
10. Click on the  **Upload** button.
11. Select one of the options on the screen to add rates from the file to the existing ones, or to replace the existing ones, as shown on the screenshot below:

12. Either enter the name of your file manually, or click **Browse...** and choose the file; then click Next.
13. On the Review File Parsing page, you can verify whether the file was parsed correctly and, if necessary, change file parsing parameters such as delimiters.

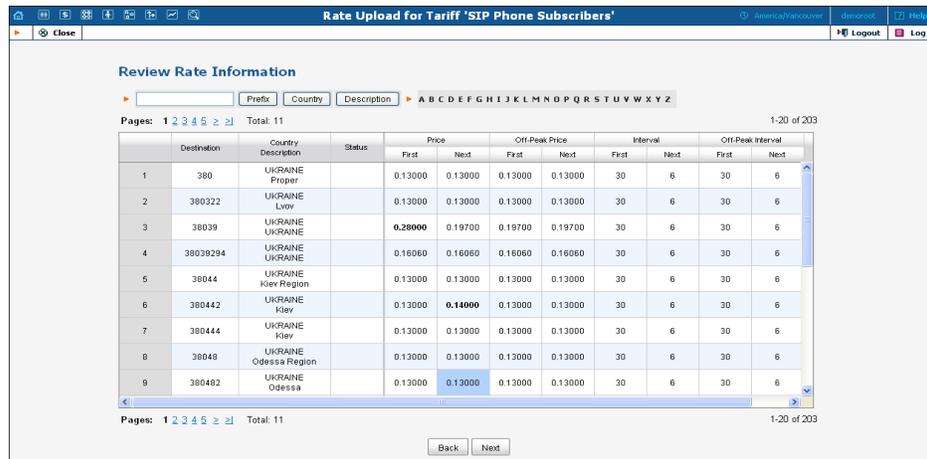
	A	B	C	D	E	F	G
	Destination	Country	Description	Interval 1	Interval N	Price 1	Price N
2	380	UKRAINE	Proper	30	6	0.13	0.13
3	380322	UKRAINE	Lviv	30	6	0.13	0.13
4	38039	UKRAINE	Mobile	30	6	0.197	0.197
5	38039294	UKRAINE	man'sh mob.	30	6	0.1606	0.1606
6	38044	UKRAINE	Kiev Region	30	6	0.13	0.13
7	380442	UKRAINE	Kiev	30	6	0.13	0.13
8	380444	UKRAINE	Kiev	30	6	0.13	0.13
9	"380444";UKRAIN						
10	38048	UKRAINE	Odessa Region	30	6	0.13	0.13
11	380482	UKRAINE	Odessa	30	6	0.13	0.13
12	380487	UKRAINE	Odessa	30	6	0.13	0.13
13	38050	UKRAINE	Mobile	30	6	0.197	0.197
14	38056	UKRAINE	Dnepropetrovska	30	6	0.13	0.13

14. On the Review Data Fields Definition page, you can specify where individual data fields (e.g. destination or price) are located in the file. In order to associate a certain column in the file with a specific data

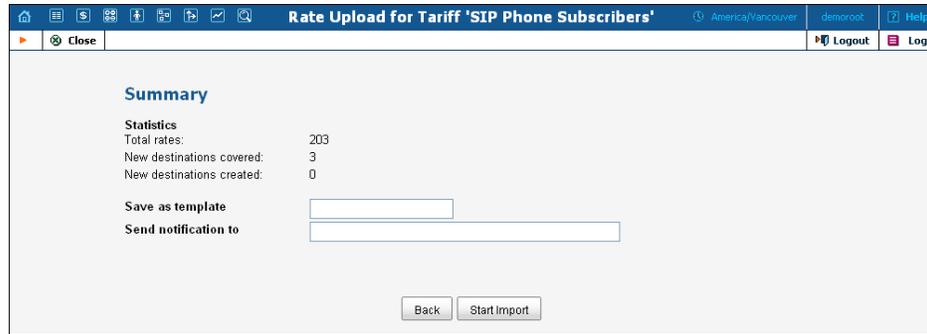
field, drag and drop that field from the list on the right onto the column header.



- On the Review Rate Information page, you can view information about rates and adjust the value of prices (just point your mouse at the window you want to change, click to modify the value, and press Save; the changed value will be in bold).



- On the Create New Destinations page, you can assign a country and destination group to destinations that were not in the system before, and which will be automatically created now.
- On the Summary page, you can find information about the number of rate records to be processed. You can also save the rate import options you have defined as a template, so that you can re-use them in the future.



18. Click Start Import. You will receive an email confirmation of the tariff upload. If any operations have failed, you will receive a notification about the error.

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

## Tariffs for Your Vendors

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

Edit	Destination *	Country	Routing				Interval, second		Price, USD/minute		Effective From YYYY-MM-DD HH24:MI:SS	Actions
			Description	Route Category	Preference	Huntstop	First *	Next *	First *	Next *		
			De default	5								
	420	CZECH REPUBLIC Proper	High quality	6		1	1	0.13000	0.13000	2010-01-14 13:02:34		
	1866	Not Applicable USA	Default	5		1	1	0.10000	0.10000	2010-01-14 13:03:44		
	38044	UKRAINE Kiev Region	Premium	5		1	1	0.05000	0.05000	2010-01-14 13:03:11		

- **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.

## Internal Vendor Tariff

When creating this tariff, make sure it does not have the Routing option turned on.

**Add Tariff**

Name: SIP on-net (Internal) | Currency: USD - US Dollar

Applied To: Vendor | Service: Voice Calls

Routing:

**General Info**

Default Off-Peak Period: NOT DEFINED

Destination Group Set: [Dropdown]

Free Seconds: 0

Post Call Surcharge: 0 %

Login Fee: 0

Connect Fee: 0

Round Charged Amount: xxxxxx.xxxxxx

Default Formula: [Text]

Short Description: [Text]

Description: [Text Area]

**Tariff Management**

Applied To	Service	Managed By	Search
ANY	ANY	ANY	[Search]

Rates	Name	Currency	Applied To	Service	Managed By	Routing	Description	Delete
[Icon]	Callshop	ADP	Customer	Voice Calls	Administrator only			[X]
[Icon]	DID Supplier Costs	USD	Vendor	Voice Calls		No		[X]
[Icon]	GlobalNet Termination	EUR	Vendor	Voice Calls				[X]
[Icon]	Aardvoip	USD	Customer	Voice Calls	Administrator only			
[Icon]	ABC Shuttle Prepaid Cards	USD	Customer	Voice Calls	ABC Shuttle Ltd.		ABC Shuttle Prepaid Cards Tariff	
[Icon]	SIP on-net (internal)	USD	Vendor	Voice Calls		No		[X]
[Icon]	SIP Phone Subscribers	USD	Customer	Voice Calls	Administrator only			

**Rates for tariff 'SIP on-net (internal)'**

Effective From: Now | Destination: [Dropdown]

Edit	Destination *	Country	Interval, second		Price, USD/minute		Effective From	Delete
			First *	Next *	First *	Next *		
[Icon]	[Text]	[Text]	1	1	0	0	Immediately	[X]

**Rates for tariff 'SIP on-net (internal)'**

Effective From: Now | Destination: [Dropdown]

Edit	Destination *	Country	Interval, second		Price, USD/minute		Effective From	Delete
			First *	Next *	First *	Next *		
[Icon]	[Text]	[Text]	[Text]	[Text]	[Text]	[Text]	[Text]	[Text]
[Icon]	[Text]	Not Applicable ALL Destinations	1	1	0.00000	0.00000	2010-04-26 01:04:23	[Text]

1. After the tariff is created, 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**.

## Create All Required Tariffs



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

- A tariff for each billing scheme for your clients; these tariffs are created as “Applied to: Customer”. For example, if you plan to have two different SIP call products with different rates, you will need two separate tariffs.
- A tariff with the termination costs for each termination partner you have; these tariffs are created as “Applied to: Vendor”.
- If you have resellers, you should also create tariffs for charging each of them; these tariffs are created as “Applied to: Reseller”. Do not create tariffs which will be applied to your resellers’ subscribers yet. First create the resellers, and then return to this step. When creating these subscriber tariffs, make sure you choose *Managed by NNN* in the menu, where *NNN* is the name of the corresponding reseller.

## Create Product

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



The image displays two screenshots of the PortaSwitch web interface. The top screenshot is the 'Add Product' form for 'SIP Subscribers'. It includes fields for Product Name, Currency (USD - US Dollar), Managed By (Administrator only), Breakage, Overdraft Protection (Debit accounts only), Account Default ACL (Access Level new), Default Discount Plan (None), Info URL, and Description. The bottom screenshot is the 'Edit Product' page for 'SIP Subscribers', showing tabs for General Info, Maintenance, Online Signup, Accessibility, Subscriptions, Service Features, and Notepad. A red arrow points from the 'Save' button in the 'Add Product' form to the 'Add' button in the 'Edit Product' page.

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:
  - **Product Name** – Product object name.
  - **Currency** – Product currency; only tariffs which have the same currency will be permitted in the accessibility list.
  - **Managed by** – If you want this product to be used for your reseller’s accounts, so the reseller himself can change the parameters of this tariff and create new accounts using this product, choose a customer name from the menu. Otherwise, choose **Administrator only** here.
  - **Breakage** – This parameter is typically used only for prepaid accounts, so leave it empty.
  - **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.
  - **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 [PortaSIP Handbook: Advanced SIP Services](#) you can find examples for setting up volume discounts).
  - **Description** – your comments about the intended use of this product.
3. Click **Save**.
4. Click on the **Accessibility** tab to edit this product’s accessibility.

## Enter Node and Tariff into the product's accessibility list

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

The top screenshot shows the 'Add Accessibility to Product 'SIP Subscribers'' form. The fields are: Product Currency (USD), Service (Voice Calls), Node (DemoSIP), Access Code (empty), Info Digits (ANY), Authorization Chunk (empty), and Tariff (SIP Subscribers).

The bottom screenshot shows the 'Edit Product 'SIP Subscribers'' interface. The 'Accessibility' tab is selected, showing a table of voice call access points:

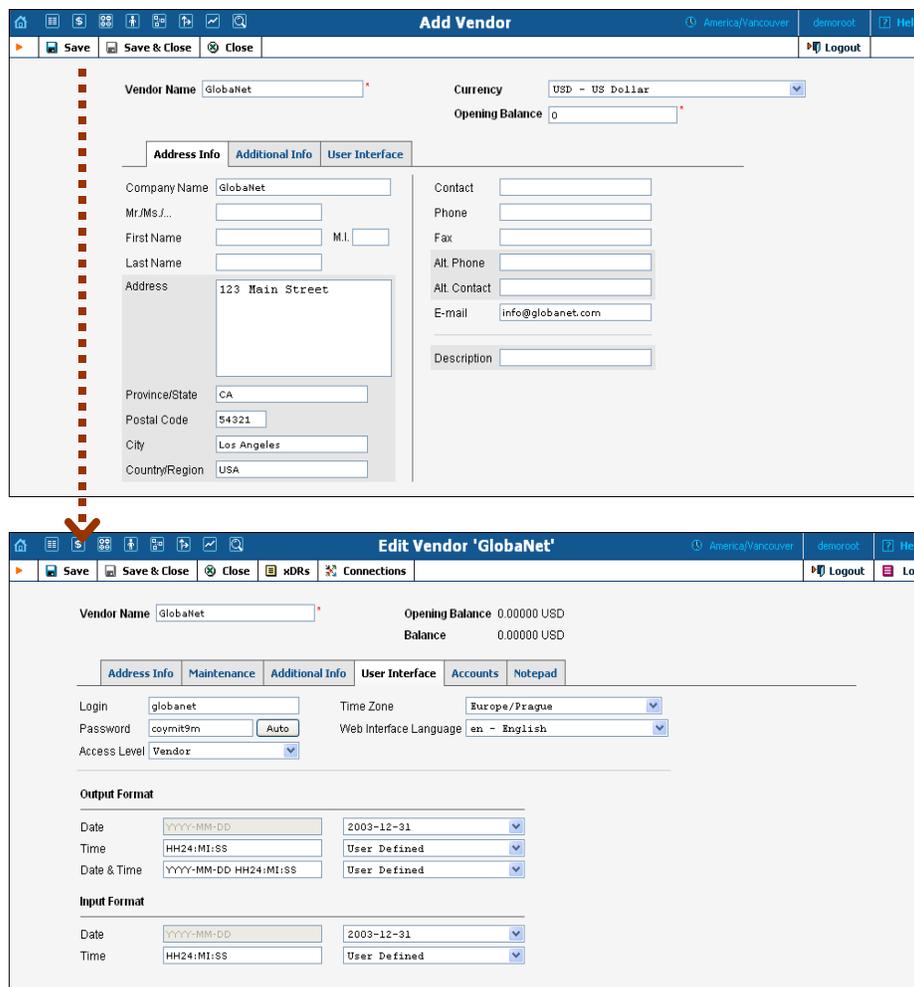
Edit	Node	Access Code	Info Digits	Routing Rating	Delete
	DemoSIP	ANY	ANY	ANY — SIP Subscribers	X

1. When the Accessibility tab is selected, click on the **Add** icon.
2. Choose **Voice Calls** in the **Service** select menu.
3. In the **Node** select menu select the PortaSIP node.
4. The **Access Code** should be left empty for the basic SIP service.
5. The **Info Digits** field only makes sense when a call originates from your customer in a public telephony network. Therefore, just leave it empty for the SIP service.
6. In the **Default Tariff** menu, select the tariff that will be applied to your customers for outgoing calls.
7. Click **Save&Close** to save this accessibility entry.

## Create Vendors

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

1. In the **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)

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

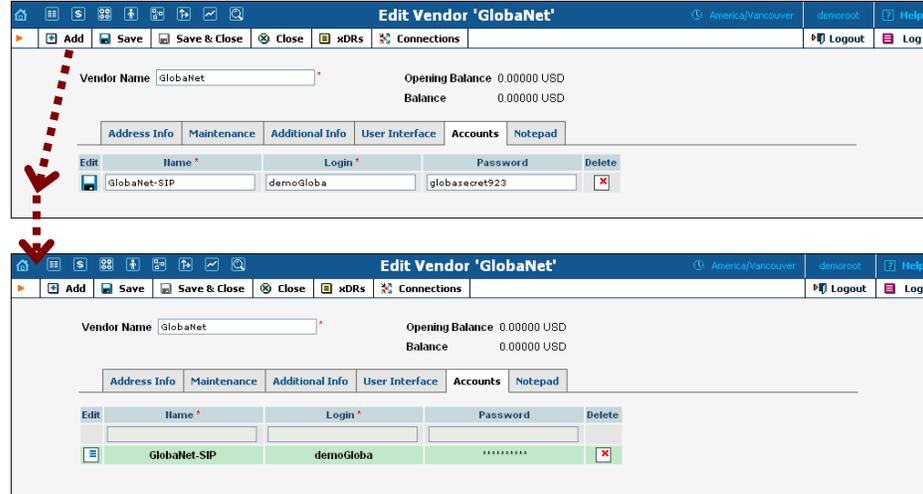
Additional info:

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

User Interface:

- **Time Zone** – 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. Click the **Accounts** tab, then click **Add** and enter this information.



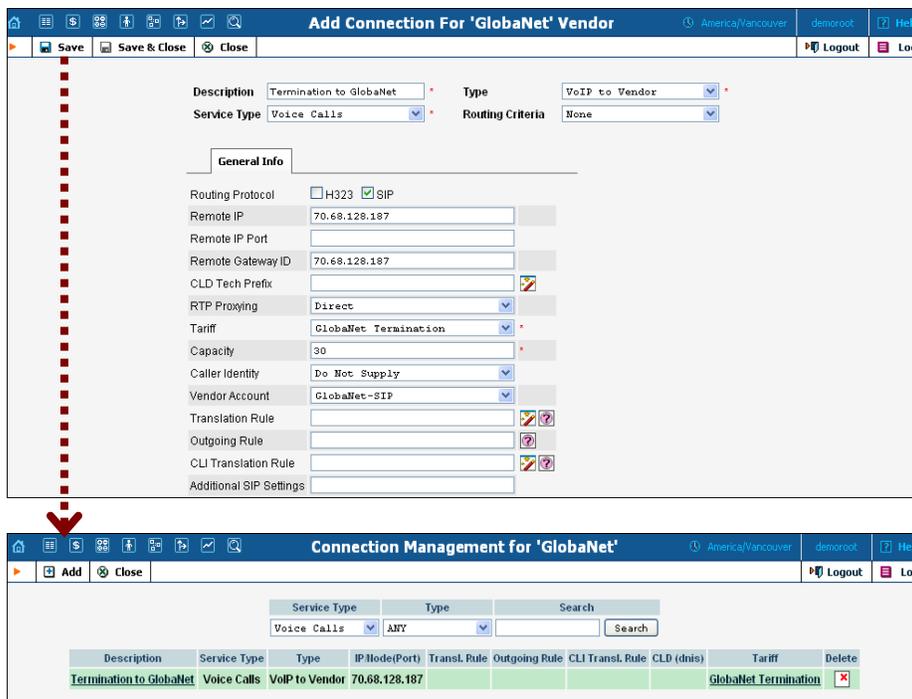
6. Click **Close** in order to return to the **Vendors** admin page.
7. Repeat steps 2-6 to add all of your vendors.

## 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 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**.

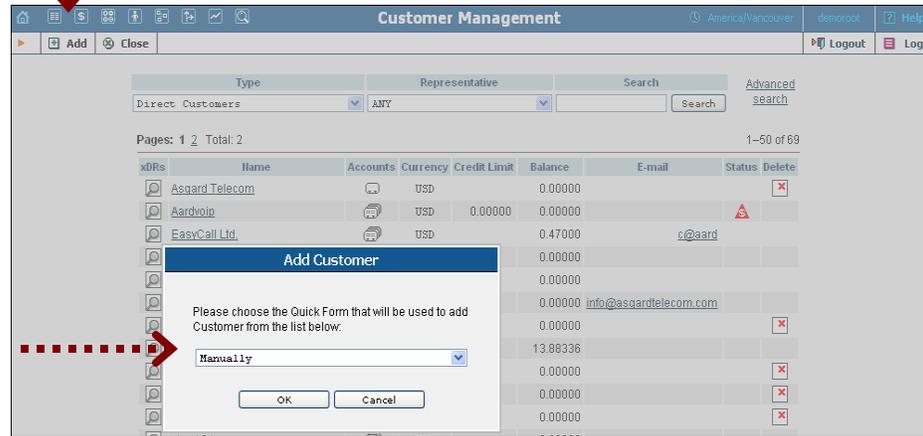
## Create Customer

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.

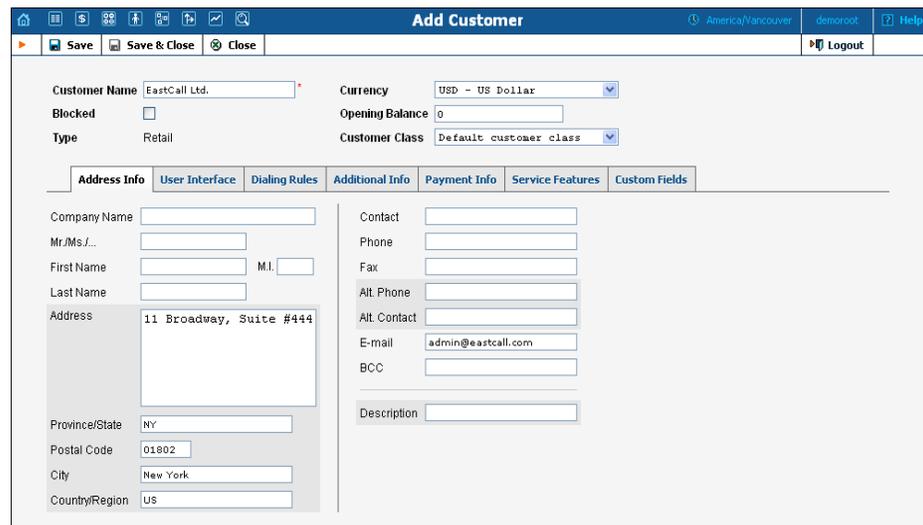
In this example we will describe how to create a new customer manually, without using a Quick Form.

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.



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)

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

#### Address info tab

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

- **Billing period** – Frequency of invoice generation. For more details about the different available billing periods, see the [PortaBilling Administrator Guide](#).
- **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

- **Credit Limit** – if left empty, then there is no credit limit for this customer.
- **Balance Warning Threshold** – the customer can be notified by email when his balance is dangerously close to the credit limit and service will soon be blocked. Here you can enter the value for such a warning threshold. This can be entered:
  - as a percentage (e.g. 90%). The warning will be sent when the customer's balance exceeds that percentage of his credit limit. So, if the credit limit is USD 1000.00 and the threshold is 90%, a warning will be sent as soon as the balance exceeds USD 900.00. This is only applicable when the customer has a positive credit limit.
  - as an absolute value. The warning will be sent as soon as the balance goes above the specified value.

### User Interface tab

- **Time Zone** – 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.
- **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.

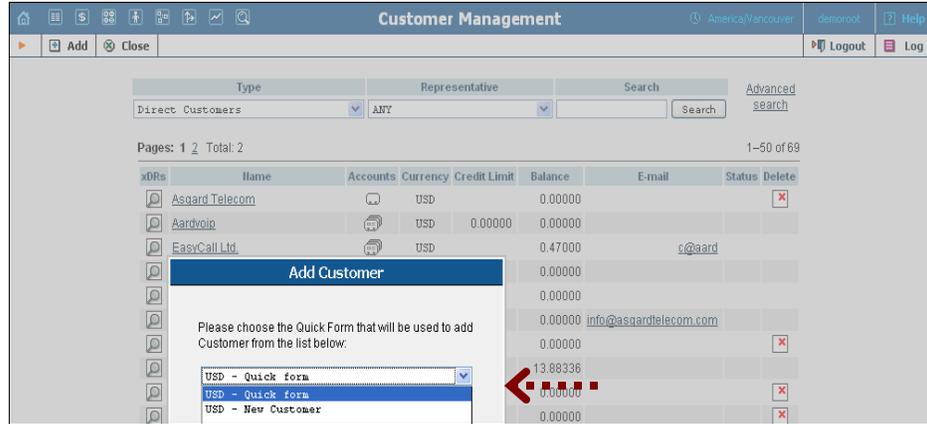
Please consult the [PortaBilling Web Reference Guide](#) for a description of other parameters available here. For now you may leave these with their default values. If you change them later, these changes will automatically affect all accounts created under this Customer.

4. Click  **Save&Close**.

### Creating a new Customer using a Quick Form:

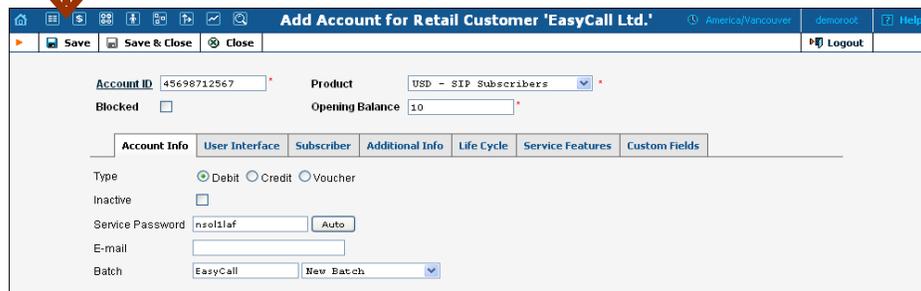
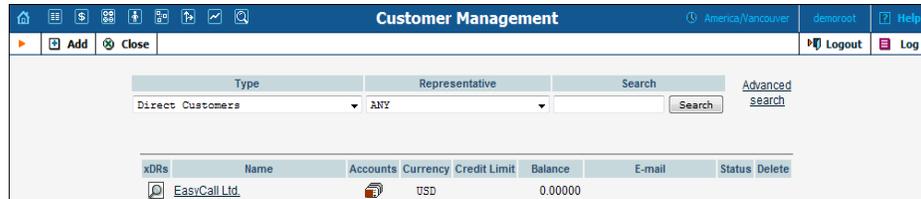
1. In the **Participants** section of the Admin-Index page, choose **Customers**.
2. On the Customer Management page, click  **Add Customer** and choose one of the available Quick Forms.
3. Fill in the **Add Customer** form.





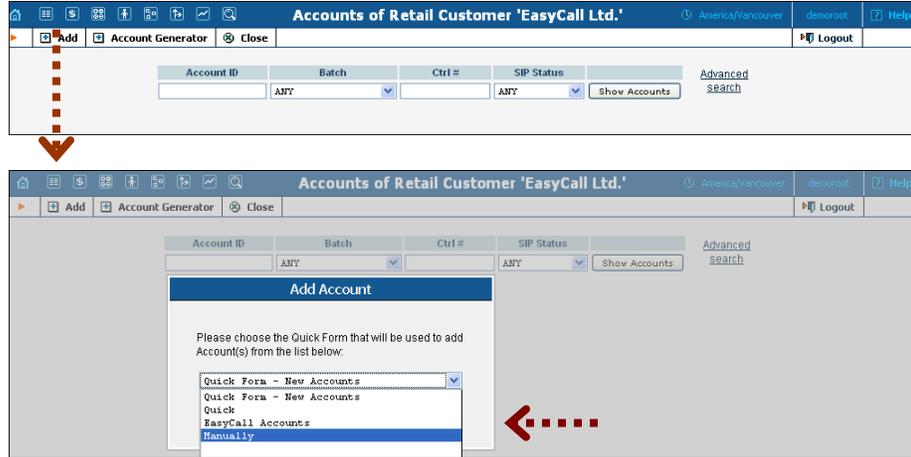
## Create Accounts

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

#### Account Info tab

- **Type** – Account type; select credit for postpaid and debit for prepaid service.
- **Credit limit** – For a credit account, specify the credit limit. If you leave this field empty, it means there is no credit limit for this account (but a customer credit limit may still apply).
- **Service Password** – The account ID and this password will be used to authenticate SIP server login.
- **Email** – Enter the account owner’s email address here. If he ever forgets his password for the web self-care pages, he will be able to reset it, and a new password will be sent to this email address. You can also just leave this field empty.
- **Batch** – A batch is a management unit for accounts. The batch name is alphanumeric. You can type a new name here, or use an existing name in order to generate more accounts for the same batch.

### Additional Info tab

- **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.
- **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

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

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

### Service Features tab

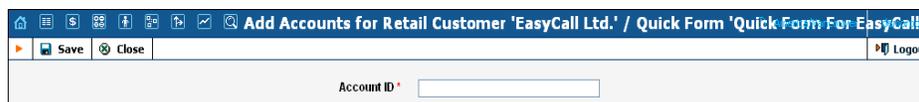
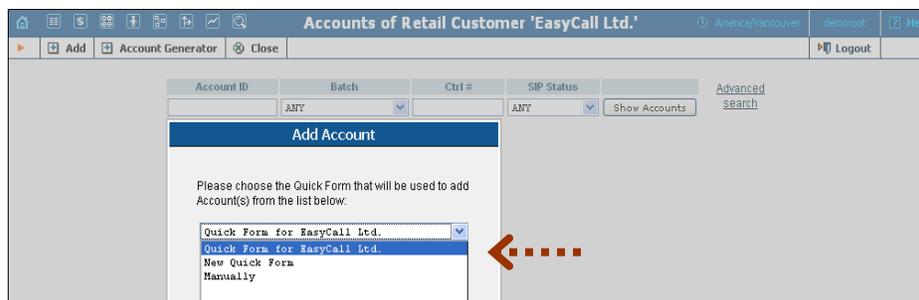
Using this tab, you can activate/deactivate various features of the services provided to this account. Note that features are defined per *service type* (the physical service provided to the user such as Conferencing, Data Service, Voice Calls, Messaging etc.)

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

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.

**Edit Customer 'EasyCall Ltd.'**

Customer Name: EasyCall Ltd. Opening Balance: 0.00000 USD  
 Blocked:  Balance: 193.80000 USD  
 Type: Retail Customer Class: Enterprise

Taxation | Abbreviated Dialing | Subscriptions | Trouble Tickets | Notepad | Service Features | Permitted SIP Proxies  
 Address Info | Maintenance | User Interface | **Dialing Rules** | Additional Info | Payment Info | Custom Fields

Enable Dialing Rules:  Dialing Rules Wizard

Load Sample:

Routing plan selection enabled:   
 Apply these dialing rules to the number that follows the selection code:   
 Your country code:   
 Your area code(s):   
 Always dial the area code as a part of the number:   
 Prefix for accessing the outside phone network:   
 Prefix for domestic calls, but outside of your area code (e.g. 1, 0):   
 International dialing prefix (e.g. 011, 00, 0011):   
 Emergency numbers (e.g. 911, 112):   
 Exceptions (e.g. \*98):   
 National Exceptions (e.g. \*98):   
 Local dialing number length:   
 Convert ANI (CLI) for incoming calls into this dialing format:



**Dialing Rules Wizard**

Routing plan selection enabled:   
 Apply these dialing rules to the number that follows the selection code:   
 Your country code:   
 Your area code(s):   
 Always dial the area code as a part of the number:   
 Prefix for accessing the outside phone network:   
 Prefix for domestic calls, but outside of your area code (e.g. 1, 0):   
 International dialing prefix (e.g. 011, 00, 0011):   
 Emergency numbers (e.g. 911, 112):   
 Exceptions (e.g. \*98):   
 National Exceptions (e.g. \*98):   
 Local dialing number length:   
 Convert ANI (CLI) for incoming calls into this dialing format:

**Sample Settings**

- Australia, Sydney
- E.164
- Europe, Czech Rep., always dial using the areacode
- Europe, Czech Rep., local and domestic dialing (obsolete)
- North America, 10 digit dialing
- North America, BC, 10 digit dialing
- North America, WA, 7 digit dialing

Buttons: Clear, Reset, Load Sample

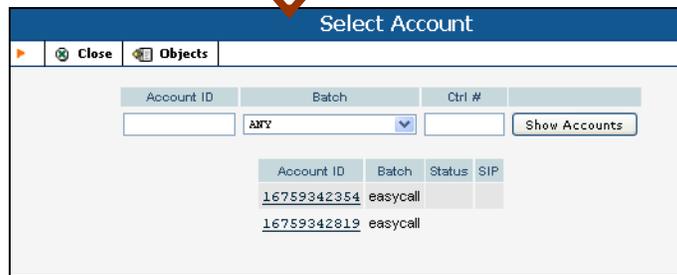
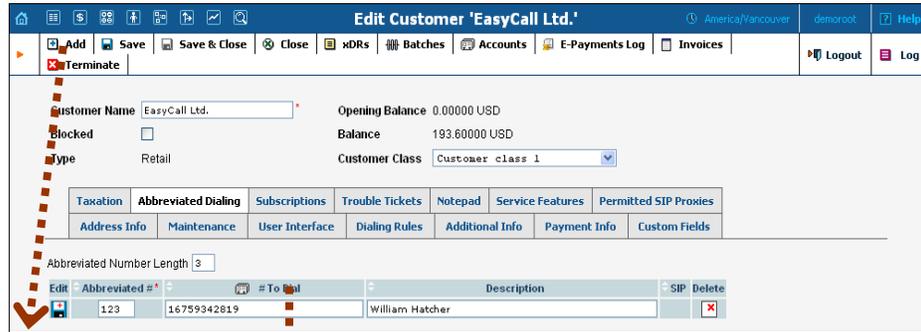
**Check Yourself**  
 To call 1234567 outside of your office, but within the same area you dial:

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 **Save** 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.

Account ID	Batch	Ctrl #	SIP Status	Product
160404373456	ANY		ANY	EasyCall
160416651839				SIP Subscribers
160420822883				SIP Customers
160428202360				EasyCall

- 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.

Account ID: 160404373456 | Product: USD - EasyCall | Balance: 9.68000 USD

User Agent: Linksys/SPA941-5.1.5 | Contact: sip:16759342354@192.168.1.1

Life Cycle	Subscriptions	Volume Discounts	Notepad	Service Features	Follow Me	Call Processing
Account Info	Maintenance	User Interface	Subscriber	Aliases	Additional Info	Payment Info

Customer: EasyCall Ltd. | Credit Limit: 10.00000 USD

Type: Credit | Opening Balance: 0.00000 USD

Service Password: ekfmpkpx | Refunds: 0.00000 USD

E-mail: | Non Call Related Charges: 9.68000 USD

Batch: EasyCall | Control Number: 5

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.

- Choose the **SIP Log Viewer** item from the main menu.
- 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.

PortaSIP Node: 193.28.87.36 - PortaSIP .36

Trace a call

Call-ID:  *but not H323-ConfID!*

Another Call-ID:  *optional, e.g. for callback calls*

Search log file for: --SELECT--

Output format:  log with call diagram,  only text log

Show call attempts

For last: 30 minutes

Filter:  do not show selftest calls

Generate



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

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

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

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

Request is sent to B2BUA.

```
11 Jun 22:09:43/GLOBAL/ser[98218]: SENDING message to 193.28.87.106:5061:
INVITE sip:6831234@193.28.87.106:5061 SIP/2.0
Record-Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
Via: SIP/2.0/UDP
193.28.87.106;branch=z9hG4bK52d.65c85fbb7b48dc83837bf35f80ab19f5.0
Via: SIP/2.0/UDP
192.168.0.250:9062;rport=9062;received=216.231.44.168;branch=z9hG4bK-1035d24f
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
Max-Forwards: 16
Contact: John Doe <sip:16041234568@216.231.44.168:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 477
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura
Content-Type: application/sdp
PortaBilling-Notify: NAT
```

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

```
a=direction:active
a=oldmediaip:192.168.0.250
```

**B2BUA receives this INVITE request.**

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

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

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

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

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

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

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

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

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

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

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

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

11 Jun 22:09:43/98dda488-69c74dd3@192.168.0.250/ser[98219]: processing INVITE received from 216.231.44.168

11 Jun 22:09:43/GLOBAL/ser[98219]: SENDING message to 216.231.44.168:9062:

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

SIP/2.0 100 trying -- your call is important to us

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

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

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

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

**B2BUA sends an authorization request to the billing.**

```
11 Jun 22:09:43/98dda488-69c74dd3@192.168.0.250/b2bua: sending AAA request:
User-Name = '16041234568'
Digest-Realm = '193.28.87.106'
Digest-Nonce = '3051864d7d0c6578460cc4b0e28ad43b448c6a77'
Digest-Method = 'INVITE'
Digest-URI = 'sip:6831234@193.28.87.106'
Digest-Algorithm = 'MD5'
Digest-User-Name = '16041234568'
```

```
Digest-Response = '54e0b42337ace33edf36d004f1037ebd'  
Calling-Station-Id = '16041234568'  
Called-Station-Id = '6831234'  
h323-conf-id = 'A9B6252E 3863D707 17A91CC0 744348CB'  
call-id = '98dda488-69c74dd3@192.168.0.250'  
h323-remote-address = '216.231.44.168'  
h323-session-protocol = 'sipv2'  
h323-ivr-out = 'PortaBilling_Routing:SIP'  
h323-ivr-out = 'PortaBilling_AuthMethod:INVITE'  
h323-ivr-out = 'PortaBilling_Notify:NAT'  
h323-ivr-out = 'PortaBilling_Seed:144514807'
```

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

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

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

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

```
11 Jun 22:09:44/GLOBAL/b2bua: SENDING message to 193.28.187.3:5060:  
INVITE sip:16046831234@193.28.187.3:5060 SIP/2.0  
Via: SIP/2.0/UDP  
193.28.87.106:5061;branch=z9hG4bK1343caf4b64b53d7a6d0b68a51b554aa;rport  
Max-Forwards: 70  
From: "John Doe"  
<sip:16041234568@193.28.87.106>;tag=52d266cda37f42d0c24541d5190acf71  
To: <sip:16046831234@193.28.187.3>  
Call-ID: 98dda488-69c74dd3@192.168.0.250  
CSeq: 200 INVITE  
Contact: Anonymous <sip:193.28.87.106:5061>  
Expires: 300  
User-Agent: Sippy  
cisco-GUID: 2847286574-946067207-396958912-1950566603  
h323-conf-id: 2847286574-946067207-396958912-1950566603  
Content-Length: 475  
Content-Type: application/sdp  
  
v=0  
o=Sippy 137112044 0 IN IP4 193.28.87.106  
s=-  
t=0 0  
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101  
c=IN IP4 216.231.44.168  
a=rtpmap:18 G729a/8000  
a=rtpmap:0 PCMU/8000  
a=rtpmap:2 G726-32/8000  
a=rtpmap:4 G723/8000  
a=rtpmap:8 PCMA/8000
```

```

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

```

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

```

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

```

```

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

```

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

```

11 Jun 22:10:04/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Stop
(Originate):
h323-call-origin      = 'originate'
h323-call-type       = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time      = '19:09:44.000 GMT Sun Jun 11 2006'
User-Name             = '16041234568'
Calling-Station-Id   = '16041234568'
Called-Station-Id    = '16046831234'
h323-conf-id         = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id              = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id      = '98dda488-69c74dd3@192.168.0.250'

```

```

h323-remote-address = '193.28.187.3'
h323-ivr-out        = 'DURATION:7200'
h323-ivr-out        = 'PortaBilling_Seed:144514807'
h323-disconnect-time = '19:10:04.000 GMT Sun Jun 11 2006'
h323-connect-time   = '19:10:04.000 GMT Sun Jun 11 2006'
Acct-Session-Time   = '0'
h323-disconnect-cause = '10'
Acct-Status-Type    = 'Stop'

```

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

```

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

```

```

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

```

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

```

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

```

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

```

11 Jun 22:10:04/GLOBAL/ser[98217]: SENDING message to 216.231.44.168:9062:

```

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP
192.168.0.250:9062;received=216.231.44.168;rport=9062;branch=z9hG4bK-1628d42b
Record-Route: <sip:193.28.87.106;ftag=873d0427882f8700;lr>
From: "John Doe" <sip:16041234568@193.28.87.106>;tag=873d0427882f8700
To: <sip:6831234@193.28.87.106>;tag=b293b17e775f2bdf6d192e545261bb1d
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 102 INVITE
Server: Sippy
```

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

The remote SIP proxy requests authorization.

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

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

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

```
INVITE sip:16046831234@70.68.128.186:5060 SIP/2.0
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK03ab045dae8f997ee66e34049ee07ee8;rport
Max-Forwards: 70
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 201 INVITE
Contact: Anonymous <sip:193.28.87.106:5061>
Expires: 300
User-Agent: Sippy
cisco-GUID: 2847286574-946067207-396958912-1950566603
h323-conf-id: 2847286574-946067207-396958912-1950566603
Authorization: Digest username="ipcall-
test",realm="70.68.128.186",nonce="ec2f8a5c71f14e5a8e08fc77816b3341448c6a8c",uri="
sip:
16046831234@70.68.128.186:5060",response="90b1b14cd90e62d2be3b8b6c10ae72f1"
Content-Length: 475
Content-Type: application/sdp
```

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

```

a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250

```

### The called party's phone starts ringing.

```

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

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

```

### The called party answers the call.

```

11 Jun 22:10:09/GLOBAL/b2bua: RECEIVED message from 70.68.128.186:5060:
SIP/2.0 200 OK
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK03ab045dae8f997ee66e34049ee07ee8;rport=5061
Record-Route: <sip:70.68.128.186;ftag=f74a7a2b122becfb3bb9ca65f75193f6;lr>
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>;tag=f55ecc2530650faffe5da956658086c9
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 201 INVITE
Server: Sippy
Contact: Anonymous <sip:70.68.128.186:5061>
Content-Length: 176
Content-Type: application/sdp

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

```

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

```

11 Jun 22:10:09/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Start
(Originate):
h323-call-origin      = 'originate'
h323-call-type        = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time       = '19:10:04.000 GMT Sun Jun 11 2006'
User-Name              = '16041234568'
Calling-Station-Id     = '16041234568'
Called-Station-Id      = '16046831234'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'

```

```

call-id = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id = '98dda488-69c74dd3@192.168.0.250'
h323-remote-address = '70.68.128.186'
h323-ivr-out = 'DURATION:7200'
h323-ivr-out = 'PortaBilling_Seed:144514807'
h323-connect-time = '19:10:09.000 GMT Sun Jun 11 2006'
alert-timepoint = '19:10:08.000 GMT Sun Jun 11 2006'
Acct-Status-Type = 'Start'

```

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

```

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

```

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

```

11 Jun 22:11:24/GLOBAL/ser[98217]: RECEIVED message from 216.231.44.168:9062:
BYE sip:193.28.87.106:5061 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-87f40044
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>;tag=b293b17e775f2bdf6d192e545261bb1d
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 103 BYE
Max-Forwards: 70
Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
Authorization: Digest
username="16041234568",realm="193.28.87.106",nonce="3051864d7d0c6578460cc4b0e28ad4
3b448c6a77",uri="sip:
193.28.87.106:5061",algorithm=MD5,response="4e3308a385aef5159ad03f738f9bd31"
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 0

```

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

```

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

```

```

Acct-Session-Time      = '74'
h323-disconnect-cause = '0'
alert-timepoint       = '19:10:09.000 GMT Sun Jun 11 2006'
Acct-Status-Type      = 'Stop'

```

The outgoing call leg is closed.

```

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

```

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

```

11 Jun 22:11:24/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Stop
(Originate):
h323-call-origin      = 'originate'
h323-call-type        = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time       = '19:10:04.000 GMT Sun Jun 11 2006'
User-Name              = '16041234568'
Calling-Station-Id    = '16041234568'
Called-Station-Id     = '16046831234'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id                = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id       = '98dda488-69c74dd3@192.168.0.250'
h323-remote-address   = '70.68.128.186'
h323-ivr-out          = 'DURATION:7200'
h323-ivr-out          = 'PortaBilling_Seed:144514807'
h323-voice-quality    = '0'
Acct-Terminate-Cause  = 'User-Request'
h323-disconnect-time  = '19:11:23.000 GMT Sun Jun 11 2006'
h323-connect-time     = '19:10:09.000 GMT Sun Jun 11 2006'
Acct-Session-Time     = '74'
h323-disconnect-cause = '0'
alert-timepoint       = '19:10:08.000 GMT Sun Jun 11 2006'
Acct-Status-Type      = 'Stop'

```

The other party confirms call disconnection.

```

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

```

The call is finished.



```

h323-remote-address = '216.231.44.168'
h323-session-protocol = 'sipv2'
h323-ivr-out = 'PortaBilling_Routing:SIP'
h323-ivr-out = 'PortaBilling_AuthMethod:INVITE'
h323-ivr-out = 'PortaBilling_Notify:NAT'
h323-ivr-out = 'PortaBilling_Seed:144514807'
NAS-Port = '5060'
Jun 11 12:09:44: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:09:44: H323/SIP call, use h323-conf-id, but remember call-id
Jun 11 12:09:44: Checking if this call comes through a VoIP from vendor connection
Jun 11 12:09:44: No VoIP from vendor connections were found
Jun 11 12:09:44: PrepareNexecute 'AccountAuth'
Jun 11 12:09:44: Found Account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000]

```

Account information is located in the database.

```

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

```

Password verification is successful.

```

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

```

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

```

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

```

The dialed number is an off-net destination.

```

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

```

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

```

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

```

```
16046831234@192.168.0.30, prio = 6, cost = 0.60000, 'X-Telecom' - to
remote GW
```

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

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

An authorization response is sent to PortaSIP.

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

Accounting for the failed outgoing call leg is received.

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

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

```

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

```

### Connection matched.

```

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

```

### A failed CDR is inserted for the vendor.

```

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

```

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

```

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

```

```
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current request
```

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

```
Jun 11 12:10:09: Fixing time with duration 0 seconds from connect time
Jun 11 12:10:09: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:10:09: Looking up vendor/connection
Jun 11 12:10:09: Trying to match connection for call
Jun 11 12:10:09: Looking for a connection VoIP/answer
Jun 11 12:10:09: VoIP, matching by the node IP '193.28.87.106' and User-Name '16041234568'
Jun 11 12:10:09: No VoIP from vendor connections were found
Jun 11 12:10:09: Connection to vendor not found
Jun 11 12:10:09: PrepareNexecute 'InsertActiveLeg'
```

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

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

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

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

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

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

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

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

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

```
Jun 11 12:11:24: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address      = '193.28.87.106'
User-Name           = '16041234568'
Called-Station-Id   = '6831234'
Calling-Station-Id  = '16041234568'
Acct-Status-Type    = 'Stop'
h323-call-origin    = 'answer'
h323-call-type       = 'VoIP'
h323-setup-time     = '19:09:44.000 GMT Sun Jun 11 2006'
h323-connect-time   = '19:10:09.000 GMT Sun Jun 11 2006'
h323-disconnect-time = '19:11:23.000 GMT Sun Jun 11 2006'
h323-disconnect-cause = '0'
h323-voice-quality  = '0'
h323-conf-id        = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id             = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Time   = '74'
Acct-Delay-Time     = '0'
h323-session-protocol = 'sipv2'
h323-remote-address = '216.231.44.168'
h323-ivr-out        = 'PortaBilling_Seed:144514807'
Acct-Terminate-Cause = 'User-Request'
h323-ivr-out        = 'PortaBilling_Session:unlock'
alert-timepoint     = '19:10:09.000 GMT Sun Jun 11 2006'
NAS-Port            = '5060'
Exec-Program-Log    = 'porta-billing.pl'
Jun 11 12:11:24: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:11:24: Found a call in cache with such id
Jun 11 12:11:24: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request
Jun 11 12:11:24: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:11:24: PrepareNexecute 'DeleteActiveLeg'
Jun 11 12:11:24: Force unlock requested by NAS
Jun 11 12:11:24: Scheduling 16041234568 for logout, call lifetime reduced to 15
Jun 11 12:11:24: Logging out account '16041234568'(103) from 'A9B6252E 3863D707
17A91CC0 744348CB'
Jun 11 12:11:24: Set lifetime with 15s to Sun Jun 11 12:11:39 2006
Jun 11 12:11:24: Looking up vendor/connection
Jun 11 12:11:24: Trying to match connection for call
Jun 11 12:11:24: Looking for a connection VoIP/answer
Jun 11 12:11:24: VoIP, matching by the node IP '193.28.87.106' and User-Name
'16041234568'
Jun 11 12:11:24: Connection to vendor not found
Jun 11 12:11:24: No connection from vendor
```

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

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

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

```
Jun 11 12:11:24: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address      = '193.28.87.106'
User-Name           = '16041234568'
Called-Station-Id   = '16046831234'
Calling-Station-Id  = '16041234568'
Acct-Status-Type    = 'Stop'
```

```

h323-call-origin      = 'originate'
h323-call-type       = 'VoIP'
h323-setup-time      = '19:10:04.000 GMT Sun Jun 11 2006'
h323-connect-time    = '19:10:09.000 GMT Sun Jun 11 2006'
h323-disconnect-time = '19:11:23.000 GMT Sun Jun 11 2006'
h323-disconnect-cause = '0'
h323-voice-quality   = '0'
h323-conf-id         = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id              = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id      = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Time    = '74'
Acct-Delay-Time      = '0'
h323-session-protocol = 'sipv2'
h323-remote-address  = '70.68.128.186'
h323-ivr-out         = 'DURATION:7200'
h323-ivr-out         = 'PortaBilling_Seed:144514807'
Acct-Terminate-Cause = 'User-Request'
alert-timepoint      = '19:10:08.000 GMT Sun Jun 11 2006'
NAS-Port             = '5060'
Exec-Program-Log     = 'porta-billing.pl'
Jun 11 12:11:24: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:11:24: Found a call in cache with such id
Jun 11 12:11:24: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request
Jun 11 12:11:24: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:11:24: PrepareNexecute 'DeleteActiveLeg'
Jun 11 12:11:24: End of the outgoing call for logged in account. Waiting another
outgoing call or hang up
Jun 11 12:11:24: Set lifetime with 15s to Sun Jun 11 12:11:39 2006
Jun 11 12:11:24: Looking up vendor/connection
Jun 11 12:11:24: Trying to match connection for call
Jun 11 12:11:24: Looking for a connection VoIP/originate
Jun 11 12:11:24: Outgoing VoIP, matching by the remote IP address '70.68.128.186'
(env 4)
Jun 11 12:11:24: Found connection 11 'Termination to Globalnet' to vendor
'GlobalNet'

```

This call leg crosses a connection to the vendor.

```

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

```

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

```

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

```

PortaBilling also calculates the termination costs for this call.

```

Jun 11 12:11:24: Charging account for the call
Jun 11 12:11:24: Inserting CDR

```

```

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

CDRs are inserted and balances are modified.

```

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

## Verify Call History for an Account

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

Accounts of Retail Customer 'SIP Accounts'

Account ID	Batch	Ctr #	SIP Status	Advanced search
123005	AMT		Logged on	Show Accounts
123006				
009				
123000				
123001				
123004				
123008				
16473672987				

xDR History For Account '16473672987'

From Date: 2010-01-10  
 To Date: Now  
 Service: Voice Calls  
 Show Unsuccessful Attempts:   
 Show xDRs

xDR History For Account '16473672987'

Account	16473672987	Credits/Refunds	0.00000 USD
From	2010-01-10 09:35:09	Payments	0.00000 USD
To	2010-01-11 09:35:11	Subscriptions Charged	0.00000 USD
Charged by	'SIP Product' product	Services Charged	0.17334 USD
Type	Debit	<b>Total</b>	<b>0.17334 USD</b>
Total Transactions	3		

Show Totals By Services

### Voice Calls

View	From	To	Country	Description	Date/Time	Charged Time, min:sec	Amount, USD	Refund	Play	
	16473672987	16473554625	North America		2010-01-11 03:46:12	0:10	0.08667			
	16473672987	12123752556	North America		2010-01-11 03:44:30	0:13	0.08667			
	16473672987	18667748646	North America		2010-01-11 01:40:13	0:03	0.02000			
<b>Subtotal</b>							<b>0:26</b>	<b>0.17334</b>		



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

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

### 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.

**List of possible Disconnect reasons:**

- Normal completed call
- Normal uncompleted call
- Call progress code
- Calling side error
- Called side error
- Network error

- 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 **Trace 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 [PortaBilling100 Web Reference Guide](#) PDF file for detailed instructions on how to navigate and operate the web interface, as well as detailed explanations of particular fields.

## Incoming DID Calls (from PSTN)

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

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

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

## Incoming DID Calls (from VoIP Vendor)

### Checklist

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



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

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

Create a new accessibility entry in your SIP product, using the [ ]  
PortaSIP node, tariff A and the access code INCOMING.

Create a vendor for the DID supplier. [ ]

Create a “VoIP from Vendor” connection for this vendor using [ ]  
tariff B.

**Account provisioning**

Check that the retail SIP service customer is already created [ ]  
according to the instructions in the *Basic SIP Service* section.

Create a new account for this customer, with the account ID [ ]  
identical to the SIP phone number (DID).

**Testing**

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

Set up PSTN-to-SIP Service



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

Create a 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.

The top screenshot shows the 'Tariff Management' interface. It includes a navigation bar with 'Add', 'Close', 'Off-Peak Periods', and 'xDR Re-rating' buttons. Below the navigation bar is a table with columns: Applied To, Service, Managed By, and Search. The table contains one entry with 'ANY' in all three columns. Below the table is a 'Rates' section with columns: Name, Currency, Applied To, Service, Managed By, Routing, Description, and Delete.

The bottom screenshot shows the 'Add Tariff' form. It has a navigation bar with 'Save', 'Save & Close', and 'Close' buttons. The form includes the following fields:  
 - Name: SIP Subscribers - Incoming  
 - Applied To: Customer  
 - Managed By: Administrator only  
 - Currency: USD - US Dollar  
 - Service: Voice Calls  
 - General Info section:  
 - Default Off-Peak Period: NOT DEFINED  
 - Destination Group Set: [dropdown]  
 - Free Seconds: 0  
 - Post Call Surcharge: 0 %  
 - Login Fee: 0  
 - Connect Fee: 0  
 - Round Charged Amount: XXXXX.XXXXX  
 - Default Formula: [dropdown]  
 - Short Description: SIP Subscribers - Incoming  
 - Description: What we charge our customers for incoming calls to their SIP phones.

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 in which currency pricing information is defined. All pricing information for a single tariff must be defined in the same currency.

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

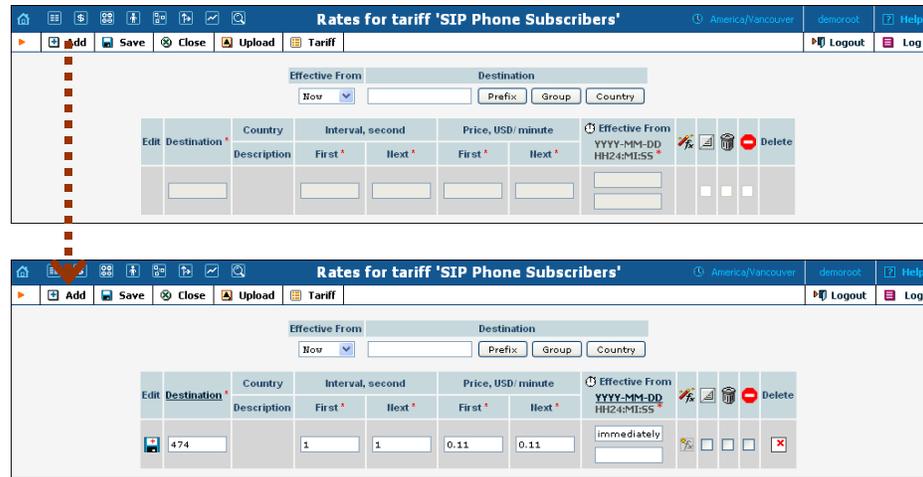
- **Applied To** – Choose **Customer** here, since this tariff will be applied to your subscribers.
  - **Managed By** – Choose “**Administrator Only**” here (this option is only visible after you select **Applied to: Customer** above).
  - **Service Type** – Choose **Voice Calls** here.
  - **Default 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.
  - **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.
  - **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.
  - **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.
  - **Short Description** – A short tariff description. This will be shown in the rate lookup on the admin interface and the self-care pages for your accounts and customers.
  - **Description** – An extended tariff description.
4. Click  **Save**.

## Enter Rates

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

### Managing rates online

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



1. On the Tariff Management page you will see a list of the available tariffs. Click the  **Rates** icon 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

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

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

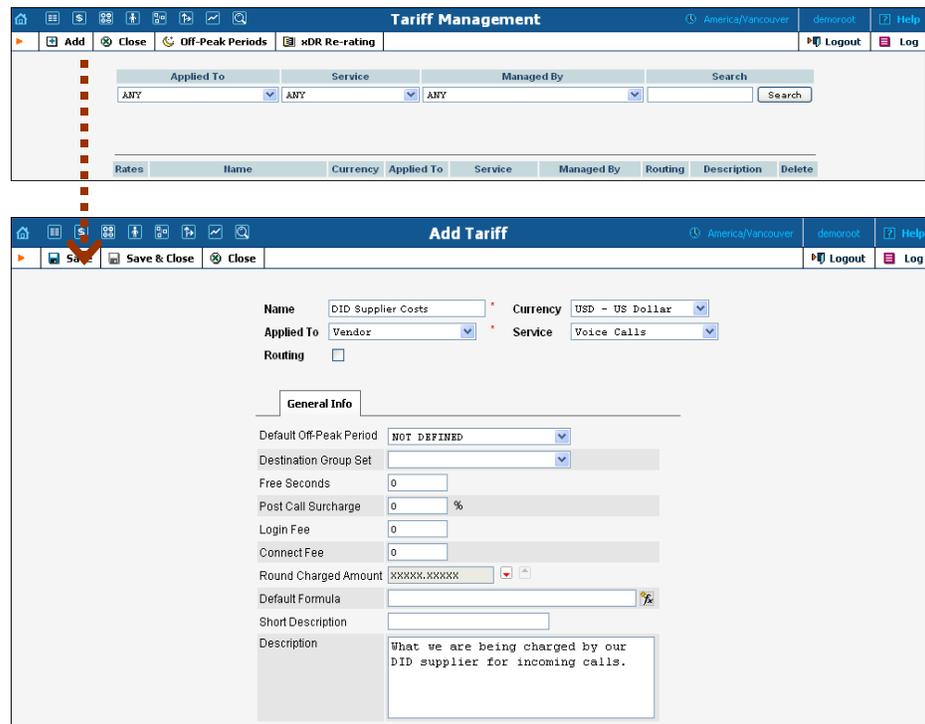
- The **Hidden, Forbidden** or **Discontinued** flags are optional.
4. Click the **Save** button in the toolbar, or the **+** icon on the left side of the row.
  5. Repeat these steps if you need to enter more rates.

### Uploading a rate list from a file

Please consult the instructions provided in the earlier chapters and [PortaBilling Templates Guide](#) for more information regarding this.

## Create a 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 previous section for a detailed description of its elements). In the **Applied To** select menu choose **Vendor** then clear the check-box **Routing**, since no routing is actually done for this vendor, i.e. the vendor will be sending calls to your network.
4. Click  **Save**.

## Enter Rates

Enter the rates applied to you by the DID provider.

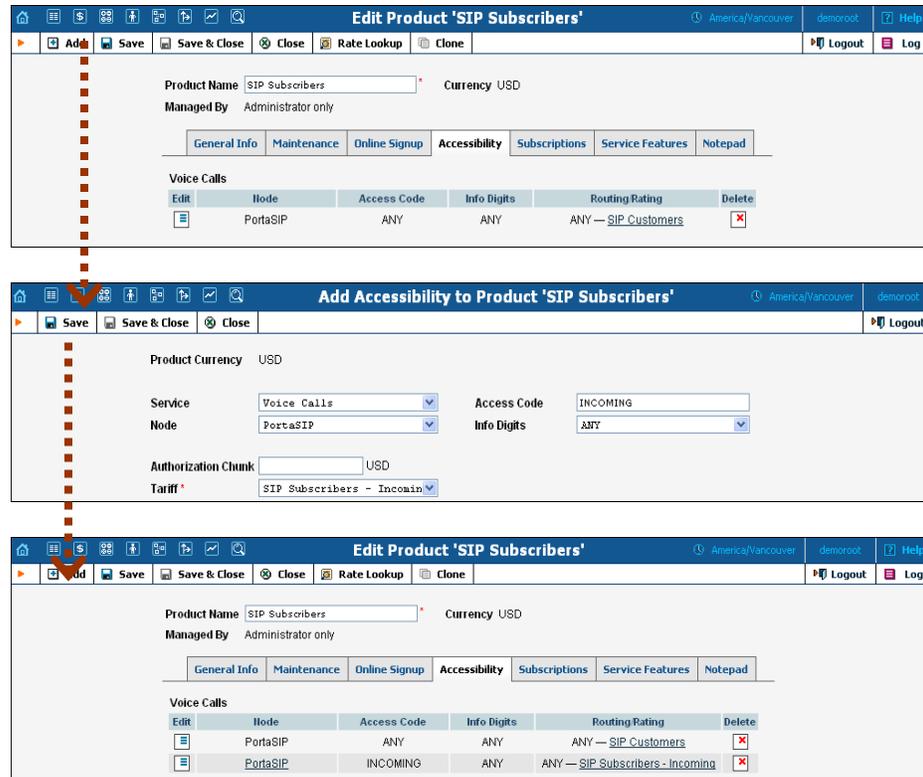


1. On the Tariff Management page you will see a list of the available tariffs. Click the  **Rates** icon previous to the name of the tariff. When you have entered Tariff Management for a particular tariff, click on **Rates** in the toolbar.
2. On the **Edit Rates** screen, click  **Add**.
3. Fill in the required information; please refer to the previous section for a detailed description of the available columns.
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.

## Modify Accessibility 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 Accessibility 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 **Default Tariff** menu, select the tariff which applies to your customers for receiving incoming calls.
9. Click **Save&Close** to save this accessibility entry.

## 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* chapter for a detailed description.
4. Click **Save**.
5. Click the Accounts tab.
6. Click **Add** to enter a record for the account which will be used by this vendor to send calls to your network.

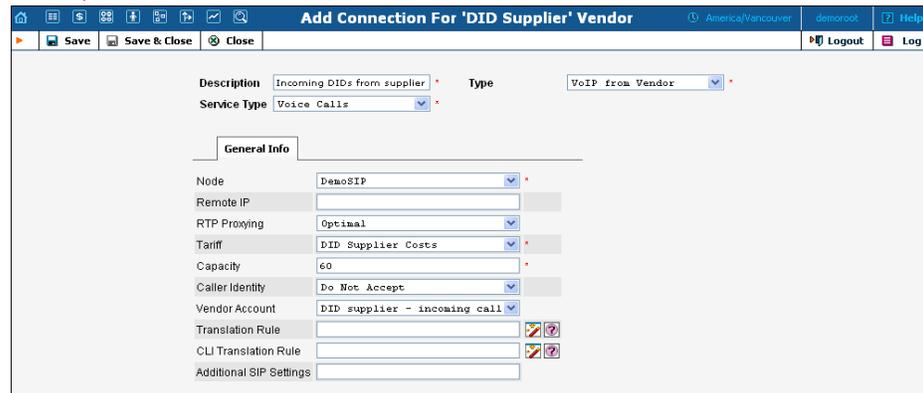
7. Enter the following values:
  - **Name** – A short name for this account information (visible when associating an account with a connection).
  - **Login** – Choose a username to be used by the vendor’s proxy or gateway to access your network. If the vendor does not support digest authentication, and you are using IP-based authentication, enter the IP address of the vendor’s gateway here and use **cisco** as the password.
  - **Password** – Choose a secure password (difficult to guess or crack) for the vendor’s account.
8. Click the **Save** button in the toolbar, or the icon on the left end of the row.
9. Click **Close** in order to return to the **Vendors** admin page.



## Define Connections

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

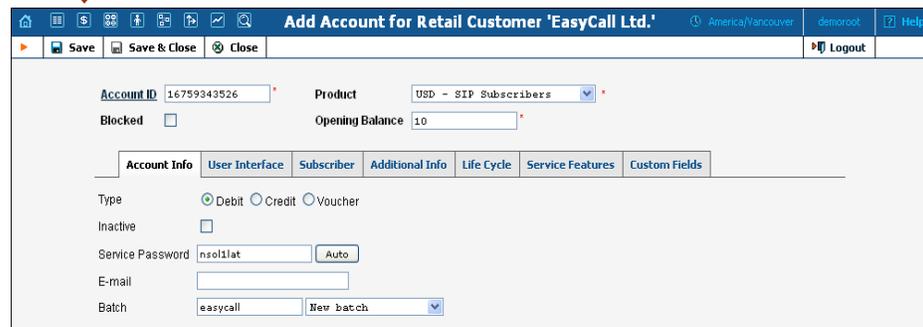
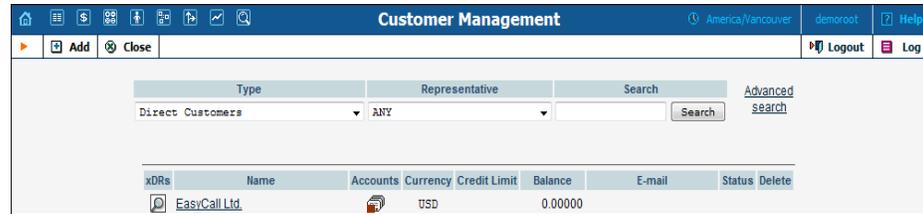
1. In the **Participants** section of the admin interface, choose **Vendors**.
2. Click on the **Connections** icon next to the vendor name.



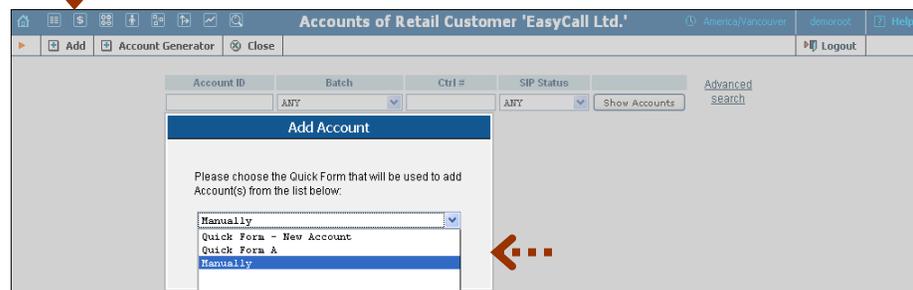
3. Press **Add** to add a new connection.
4. Fill in the connection information. Choose VoIP from Vendor as the **Type** of connection and your PortaSIP node as the **Node**, then select the tariff which defines your incoming DID costs for this vendor. In the **Vendor Account** select menu, choose the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

## Create Accounts

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



2. Next to the customer name, click the  icon (the one in the **Accounts** column) to go to account management for that customer.
3. Click on **Add**.
4. 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:

- **Account ID** – SIP ID, i.e. the phone number which will be used to login to the SIP server and receive incoming calls.
- **Product** – Choose the product you would like your account to have.
- **Blocked** – You may create the account as blocked, although this is rarely done with SIP service accounts.
- **Opening Balance** – The initial balance on the account.

#### Account Info tab

- **Type** – Select credit for postpaid, and debit for prepaid service.
- **Credit Limit** – For a credit account, specify the credit limit. If you leave this field empty, it means there is no credit limit for this account (but a customer credit limit may still apply).
- **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.
- **Email** – Enter the account owner's email address here. If he ever forgets his password for the web self-care pages, the password may be reset, and a new one will be sent to this email address. You can also just leave this field empty.
- **Batch** – A batch is a management unit for accounts. The batch name is alphanumeric. You can type a new name here, or use an existing name in order to generate more accounts for the same batch.

#### Additional Info tab

- **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

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

- **Login** – Account login to web self-care pages. This may be identical to account ID.
- **Password** – Password for the web self-care pages.

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

### Service Features tab

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

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

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

## Set up Basic SIP Service



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

## Modify a 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.

Rates	Name	Currency	Applied To	Service	Managed By	Routing	Description	Delete
	Carol reseller's tariff	USD	Reseller	Voice Calls	Administrator only			
	PID Supplier Costs	USD	Vendor	Voice Calls				
	GlobaNet Termination	USD	Vendor	Voice Calls		Yes		
	SIP Customers	USD	Customer	Voice Calls	Administrator only			
	SIP Phone Subscribers	USD	Customer	Voice Calls	Administrator only	Yes		
	SIP Subscribers - Incoming	USD	Customer	Voice Calls	Administrator only	Yes		

Edit	Destination	Country	Description	Peak/Off-Peak	Interval, second		Price, USD/minute		Effective From	Delete
					First	Next	First	Next		
				NOT DEFINED	1	1			Immediately	
					1	1				

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

Edit	Destination	Country	Description	Peak/Off-Peak	Interval, second		Price, USD/minute		Effective From	Delete
					First	Next	First	Next		
				NOT DEFINED	1	1			Immediately	
					1	1				

Edit	Destination	Country	Description	Peak/Off-Peak	Interval, second		Price, USD/minute		Effective From	Delete
					First	Next	First	Next		
	VOICEONNI			Night	1	1	0	0	Immediately	
					1	1				

Edit	Destination *	Country	Description	Peak/Off-Peak	Interval, second		Price, USD/ minute		Effective From YYYY-MM-DD HH24:MI:SS *	Delete
					First *	Next *	First *	Next *		
			NOT DEFINED							
	VOICEONNET	Not Applicable	SIP to SIP destination	Peak Off-Peak: Night	1 1	1 1	0.00000 0.00000	0.00000 0.00000	2010-01-12 03:56:15	
	VOICEVPN	Not Applicable	IP Centrex	Peak Off-Peak: Night	60 60	60 60	0.00000 0.00000	0.00000 0.00000	2010-01-12 03:45:53	

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

### Uploading a rate list from a file

Please consult the instructions provided in the earlier chapters and [PortaBilling Templates Guide](#) for more information regarding this.

### Tariff for SIP-to-SIP Call Costs

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

Applied To	Service	Managed By	Search
ANY	ANY	ANY	<input type="text"/> Search

Rates	Name	Currency	Applied To	Service	Managed By	Routing	Description	Delete
	Carol reseller's tariff	USD	Reseller	Voice Calls	Administrator only			
	DID Supplier Costs	USD	Vendor	Voice Calls				
	GlobaNet Termination	USD	Vendor	Voice Calls		Yes		
	SIP Customers	USD	Customer	Voice Calls	Administrator only			
	SIP Phone Subscribers	USD	Customer	Voice Calls	Administrator only	Yes		
	SIP Subscribers - Incoming	USD	Customer	Voice Calls	Administrator only	Yes		





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**.

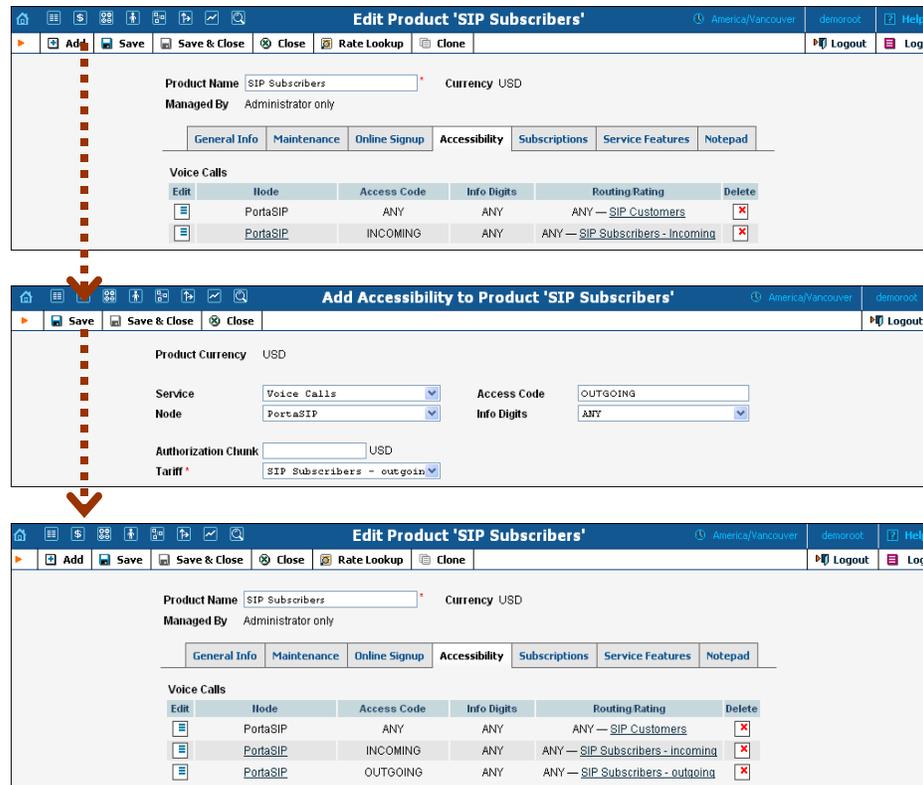


4. Fill in the required information; use VOICEONNET as the **Destination**. Please refer to the previous section for a detailed description of the available columns.
5. Click the  **Save** button in the toolbar, or the  icon on the left end of the row.
6. Repeat these steps if you need to enter more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

## Modify Accessibility 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.)

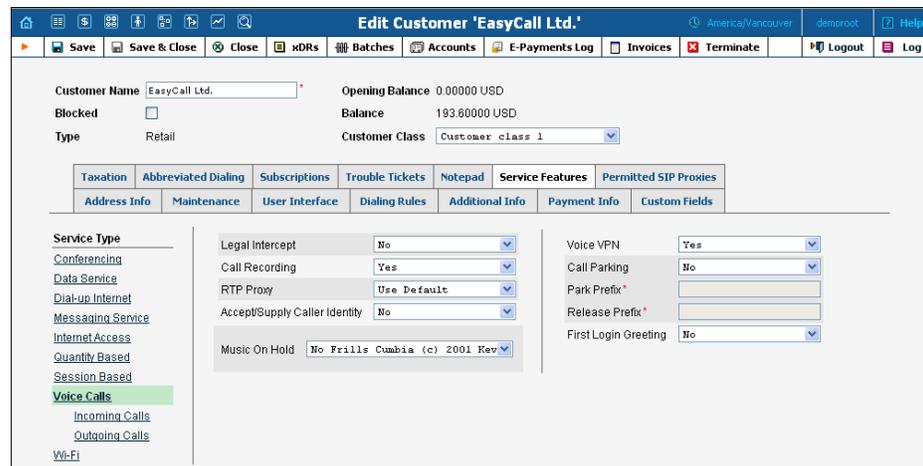
1. In the Rating section of the Admin-Index page, choose **Products**.
2. Click on the product name to enter the Edit Product screen.
3. After selecting the Accessibility tab, click the  **Add** icon.



4. Choose **Voice Calls** in the **Service** select menu.
5. In the **Node** select menu, select the **PortaSIP** node.
6. Type **OUTGOING** in the **Access Code** column.
7. **Info Digits** do not apply to this type of service (i.e. you do not differentiate between calls your customers receive from fixed or mobile networks). Therefore, just leave this field empty.
8. In the **Default Tariff** menu, select the tariff which applies to your customers' outgoing calls.
9. Click **Save&Close** to save this accessibility entry.

## Enable Voice VPN Feature for a Customer (Optional)

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



The screenshot shows the 'Edit Customer' page for 'EasyCall Ltd.' with the following details:

- Customer Name:** EasyCall Ltd.
- Opening Balance:** 0.00000 USD
- Balance:** 193.60000 USD
- Blocked:**
- Type:** Retail
- Customer Class:** Customer class 1

The **Service Features** tab is active, showing the following settings:

- Legal Intercept:** No
- Call Recording:** Yes
- RTP Proxy:** Use Default
- Accept/Supply Caller Identity:** No
- Music On Hold:** No Frills Cumbia (c) 2001 Kev
- Voice VPN:** Yes
- Call Parking:** No
- Park Prefix:** (empty)
- Release Prefix:** (empty)
- First Login Greeting:** No

The **Service Type** list on the left includes: Conferencing, Data Service, Dial-up Internet, Messaging Service, Internet Access, Quantity Based, Session Based, **Voice Calls** (highlighted), Incoming Calls, Outgoing Calls, and Wi-Fi.

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

# 4. 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
<b>Initial configuration</b>	
General and network configuration has already been done according to the instructions in previous chapters.	[ ]
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.	[ ]
<b>Rating configuration</b>	
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.	[ ]
<b>Account provisioning</b>	
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).	[ ]
<b>Testing</b>	
Program the parameters (phone #, password, SIP server address) into the SIP phone and make a test call.	[ ]

## 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**.

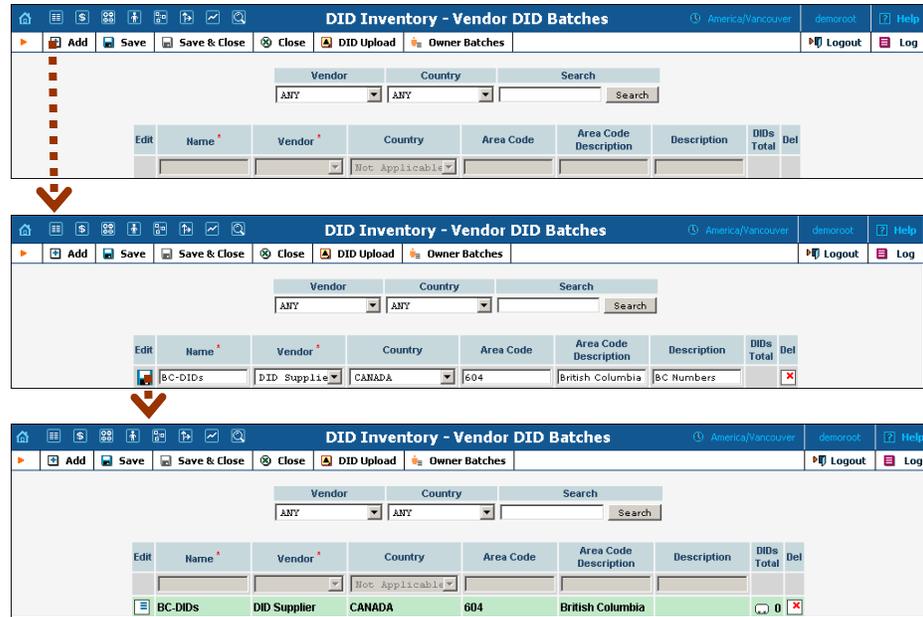
3. Fill in the **Add Vendor** form. Please refer to the instructions provided in the *Basic SIP Service* chapter for a detailed description.
4. Click **Save**.
5. If you need to enter a vendor-specific configuration (connections, vendor accounts), please refer to the instructions in the previous chapters.

## 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.





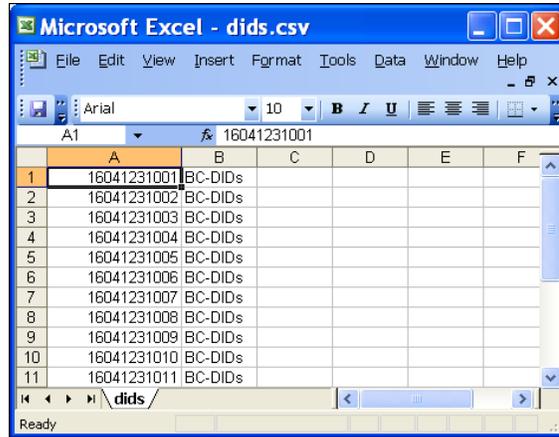
4. Click the  **Save** button in the toolbar, or the  icon on the left end of the row.
5. Repeat steps 1-4 if you need to create more DID batches.

## 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.

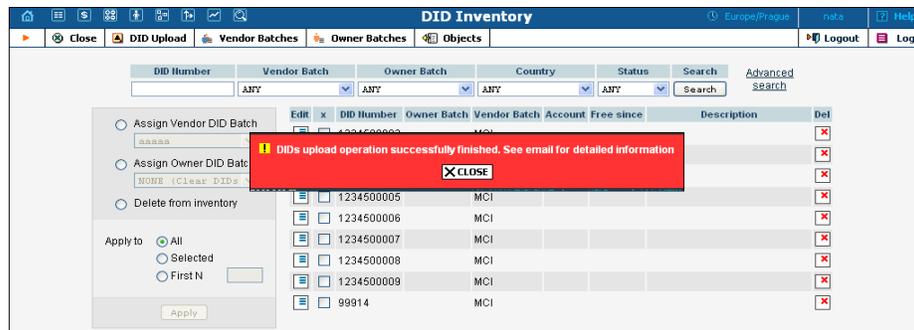
See the sample file below:



- In the DID Inventory page, click  **DID Upload** in the toolbar.
- Either enter the name of your file manually, or click **Browse...** and choose the file.



- Click  **Save & Close**.
- 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.

- In the **Networking** section of the admin interface, choose **DID Inventory**.

On the DID Inventory page, choose  **Owner Batches**.

- Press  **Add** to add a new batch.

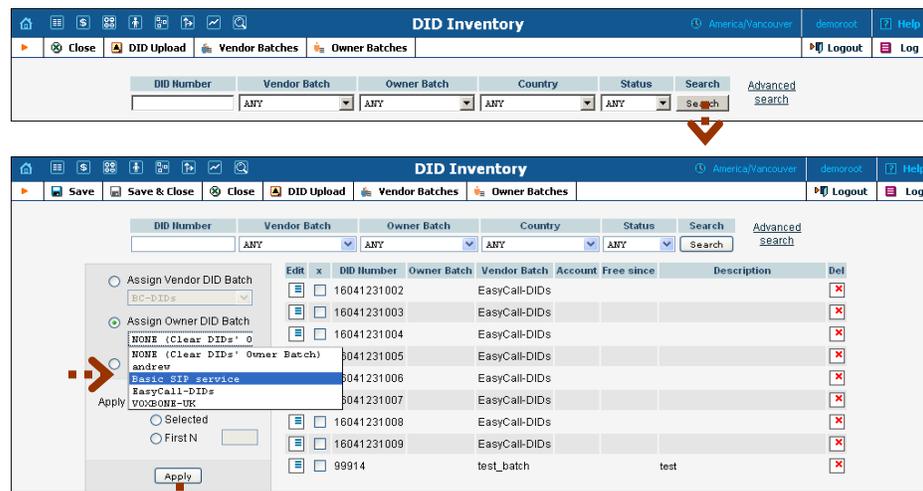


3. Click the  **Save** button in the toolbar, or the  icon on the left end of the row.
4. Repeat steps 1-4 if you need to create more owner DID batches.

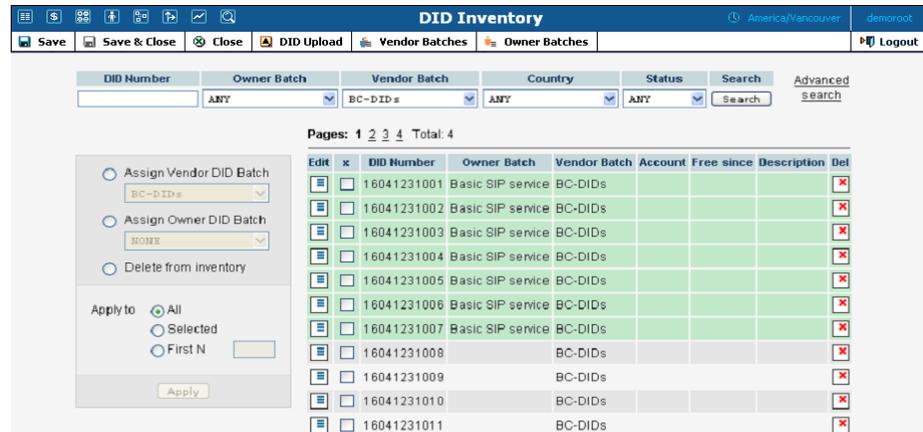
### Allocate DIDs to the Owner Batch

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**.

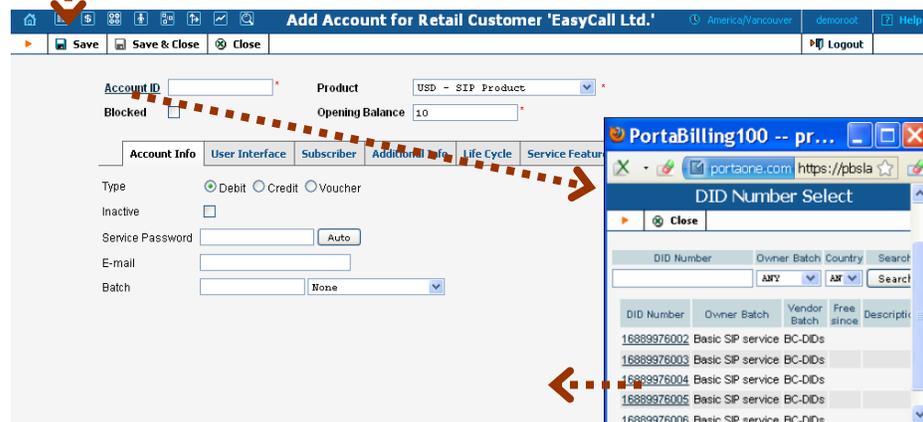
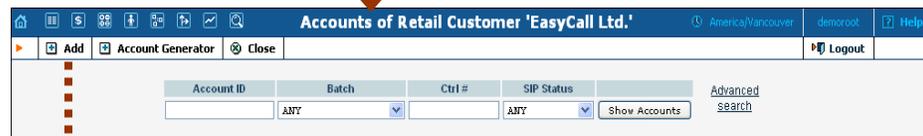
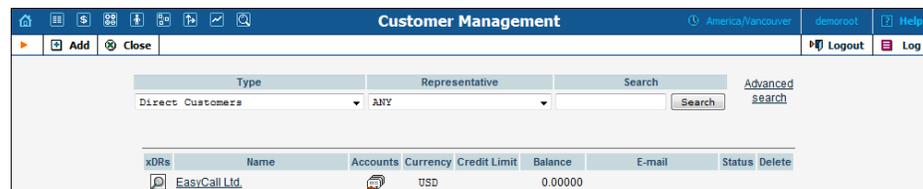


3. Select the appropriate DIDs or, alternatively, use the **First N** option to automatically process a large number of DIDs at once.
4. Choose the desired operation; in this case, choose **Assign Owner DID Batch** and select which owner batch these DIDs should be assigned to.
5. Click the **Apply** 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 (the one in the **Accounts** column) to go to account management for that customer.



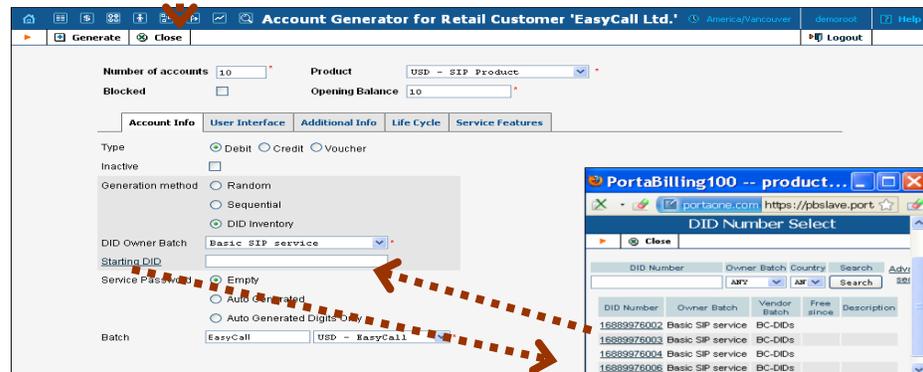
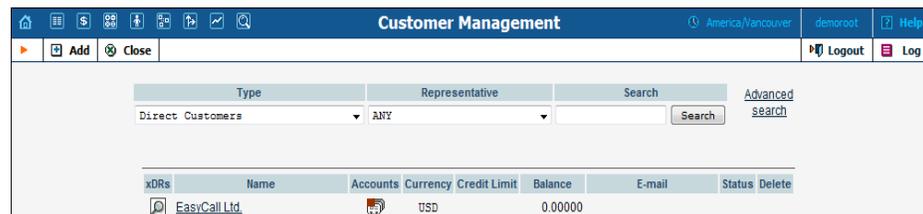
3. Click on **Add**.
4. In this example, we assume that you are creating the account manually, so choose **Manually** from the drop-down menu (do not use any of the Quick Form options).

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 PortaBilling100 web interface, without having to provision each unit individually.

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

# Setting up Auto-provisioning of IP Phones

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

## Checklist

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

Operation	Done
<b>General configuration</b>	
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)	[ ]
<b>IP phone inventory</b>	
Create a record for your IP phone in the IP phone inventory, making sure to enter the correct MAC address	[ ]
<b>Assigning a phone number to the IP phone</b>	
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	[ ]
<b>IP phone settings</b>	
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	[ ]
<b>Testing</b>	
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

The top screenshot shows the 'IP Phone Profiles' management window. It features a table with columns: Name, Type, Effective From, Managed By, Discontinued, Description, and Delete. Above the table are search filters for Effective From, Name, Type, and Managed By. An 'Add' button is located in the top left corner of the window.

The bottom screenshot shows the 'Add IP Phone Profile' form. The fields are:
 

- Name:** Sipura - Standard
- Managed By:** Administrator only
- Type:** Sipura 2000
- As Copy Of:** None
- Effective From:** Date: immediately
- Time:** HH24:MM:SS

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

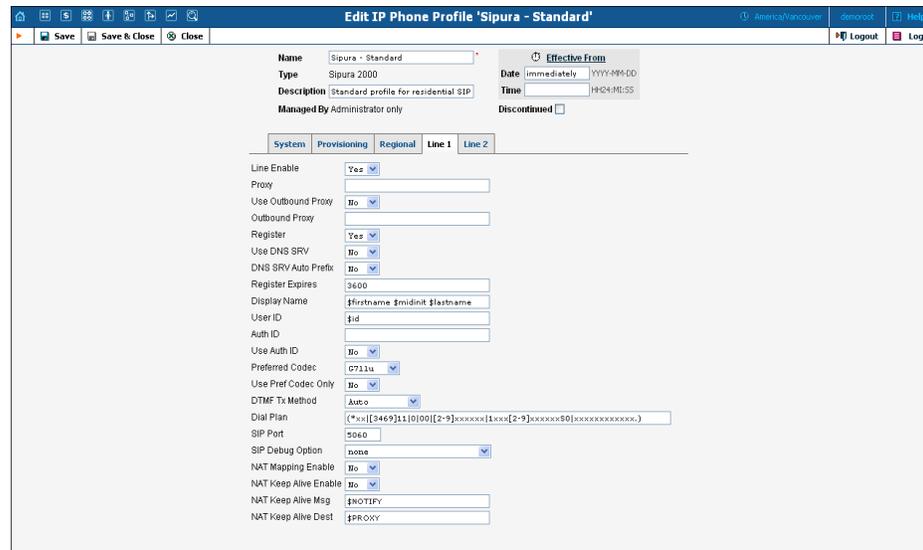
After clicking **Save** on the previous page, you will go to the Profile Edit page, where you can edit the generic device settings. These configuration parameters are dependent on the specific model of your IP device. The example below uses a Sipura-2000 device, but most of the settings should be the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

Field	Description
<b>Enable Web Server</b>	Enable/disable the built-in web server (for device monitoring)
<b>Enable Web Admin Access</b>	Enable/disable the built-in web server (for device administration and changing configuration parameters)
<b>Admin Password</b>	The password for administrator access
<b>User Password</b>	The password for user access

Field	Description
<b>Profile Rule</b>	Path to the profile configuration file. Replace PB_SLAVE_SERVER with the actual hostname or IP address of your PortaBilling100 slave server; for instance, the actual value in the field might be: [--

	<p>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.</p> <p>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).</p>
<p><b>GPP A, GPP B, GPP C</b></p>	<p>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.</p> <p>GPP A corresponds to \$A variable in the <b>Profile Rule</b> field, GPP B to \$B, etc.</p>

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
<b>Proxy</b>	SIP server for registration and placing outgoing calls.
<b>Outbound Proxy</b>	SIP Server for placing outgoing calls (if different from the SIP registration server).
<b>Use Outbound Proxy</b>	If set to “no”, the SIP server defined by the Proxy parameter will be used for all registrations and outgoing calls.

<b>Register</b>	Whether or not the IP phone should register with the SIP server; this is required to receive incoming calls.
<b>Register Expires</b>	Registration lifetime in seconds (the Expires value in a REGISTER request). The IP phone will periodically renew registration shortly before the current registration expires.
<b>Display Name</b>	The subscriber's display name, which appears in the caller-id for his outgoing calls; by default First Name + M Name + Last Name.
<b>User ID</b>	The phone number assigned to this port. The dynamic variable \$sid will be automatically replaced by the account's ID (phone number).
<b>Auth ID</b>	Authentication ID (if different from the phone number).
<b>Use Auth ID</b>	If set to "yes", the combination Auth ID and Password will be used for SIP authentication. Otherwise, User ID and Password are used.
<b>Preferred Codec</b>	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.
<b>Use Pref Codec Only</b>	Only use the preferred codec for all calls. The call will fail if the other endpoint does not support this codec.
<b>DTMF Tx Method</b>	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.
<b>Dial Plan</b>	Per-line dial plan script; see the product manual for a detailed description of the syntax.

## Create IP Phone Entry



The screenshot shows the 'IP Phone Inventory' page. At the top, there are navigation icons and a search bar. Below the search bar, there are 'Add' and 'Close' buttons. The main content is a table with the following data:

Name	Type	Profile	Managed By	MAC Address	Description	Ports free total	Inventory ID	Delete
My SPA-941	Linksys SPA-941	SPA-941	Administrator only	0:E:8:DB:F8:CE	My office Linksys SPA-941	2 2	KZ Linksys	
SPA 2000.117	Sipura 2000	Sipura 2000	Administrator only	0:1:2:C0:3:4:5		2 2	1	

Name	Type	Profile	Managed By	MAC Address	Description	Ports free	Ports total	Inventory ID	Delete
JB SPA-2000	Sipura 2000	Sipura - Standard	Administrator only	0:E0:8:AB:D7:A8	J.Brown SPA-2000	2	2	JB Sipura	X
My SPA-941	Linksys SPA-941	SPA-941	Administrator only	0:E0:8:DB:F8:CE	My office Linksys SPA-941	2	2	KZ Linksys	
SPA 2000.117	Sipura 2000	Sipura 2000	Administrator only	0:E1:2:DC:F4:B5		2	2	1	

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 `/home/porta-admin/apache/btdocs/` on the PortaBilling100 slave server, in subdirectories with a name identical to the environment ID (`i_env`): 1, 2, 3, and so on.

## IP Phone Device Configuration

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

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



Address <http://192.168.0.237/admin/advanced> Go Links

technology, inc. **Sipura Phone Adapter Configuration**

Info System SIP **Provisioning** Regional Phone Line 1 Line 2 User 1 User 2 User Login basic advanced

**Configuration Profile**

Provision Enable:  yes  Resync On Reset:  yes

Resync Random Delay: 2 Resync Periodic: 3600

Resync Error Retry Delay: 3600 Forced Resync Delay: 14400

Resync From SIP:  yes  Resync After Upgrade Attempt:  yes

Resync Trigger 1:

Resync Trigger 2:

Resync Fails On FNF:  yes

Profile Rule: [--key \$B]http://PB\_SLAVE\_SERVER/\$A/\$MA.cfg

Profile Rule B:

Profile Rule C:

Profile Rule D:

Log Resync Request Msg: \$PN \$MAC -- Requesting resync \$SCHEME://\$SERVIP:\$PORT\$PATH

Log Resync Success Msg: \$PN \$MAC -- Successful resync \$SCHEME://\$SERVIP:\$PORT\$PATH

Log Resync Failure Msg: \$PN \$MAC -- Resync failed: \$ERR

Report Rule:

**Firmware Upgrade**

Upgrade Enable:  yes  Upgrade Error Retry Delay: 3600

Downgrade Rev Limit:

Upgrade Rule:

Log Upgrade Request Msg: \$PN \$MAC -- Requesting upgrade \$SCHEME://\$SERVIP:\$PORT\$PATH

Log Upgrade Success Msg: \$PN \$MAC -- Successful upgrade \$SCHEME://\$SERVIP:\$PORT\$PATH -- \$ER

Log Upgrade Failure Msg: \$PN \$MAC -- Upgrade failed: \$ERR

**General Purpose Parameters**

GPP A: 1

GPP B:

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

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

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

## Advanced Provisioning Tips

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

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

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

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

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

You may change the default location for storing configuration files:

1. To change the http-root dir, edit the `DocumentRoot` parameter for port 80 in the Apache host's configuration file:

```
/usr/local/etc/apache/porta.htpd.conf:
    <VirtualHost _default_:80>
        DocumentRoot "/home/porta-admin/apache/htdocs/"
        Options ExecCGI
        DirectoryIndex index.pl
    </VirtualHost>
```

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

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

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


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:			

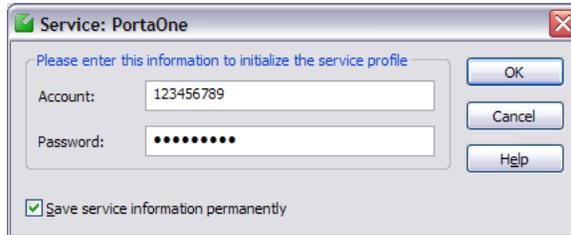
**Line 1 Status**

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:	

<b>Network Settings</b>			
SIP TOS/DiffServ Value:	<input type="text" value="0x68"/>	Network Jitter Level:	<input type="text" value="high"/>
RTP TOS/DiffServ Value:	<input type="text" value="0xb8"/>		
<b>SIP Settings</b>			
SIP Port:	<input type="text" value="5060"/>	SIP 100REL Enable:	<input type="text" value="no"/>
EXT SIP Port:	<input type="text"/>	Auth Resync-Reboot:	<input type="text" value="yes"/>
SIP Debug Option:	<input type="text" value="none"/>		
<b>Call Feature Settings</b>			
Blind Attn-Xfer Enable:	<input type="text" value="no"/>	MOH Server:	<input type="text"/>
Xfer When Hangup Conf:	<input type="text" value="yes"/>		
<b>Proxy and Registration</b>			
Proxy:	<input type="text" value="216.231.44.168"/>	Use Outbound Proxy:	<input type="text" value="no"/>
Outbound Proxy:	<input type="text"/>	Use OB Proxy In Dialog:	<input type="text" value="yes"/>
Register:	<input type="text" value="yes"/>	Make Call Without Reg:	<input type="text" value="no"/>
Register Expires:	<input type="text" value="3600"/>	Ans Call Without Reg:	<input type="text" value="no"/>
Use DNS SRV:	<input type="text" value="no"/>	DNS SRV Auto Prefix:	<input type="text" value="no"/>
Proxy Fallback Intvl:	<input type="text" value="3600"/>		
<b>Subscriber Information</b>			
Display Name:	<input type="text"/>	User ID:	<input type="text" value="1206001236"/>
Password:	<input type="text" value="*****"/>	Use Auth ID:	<input type="text" value="no"/>
Auth ID:	<input type="text"/>		
Mini Certificate:	<input type="text"/>		
SRTP Private Key:	<input type="text"/>		
<b>Supplementary Service Subscription</b>			
Call Waiting Serv:	<input type="text" value="yes"/>	Block CID Serv:	<input type="text" value="yes"/>
Block ANC Serv:	<input type="text" value="yes"/>	Dist Ring Serv:	<input type="text" value="yes"/>
Cfwd All Serv:	<input type="text" value="yes"/>	Cfwd Busy Serv:	<input type="text" value="yes"/>
Cfwd No Ans Serv:	<input type="text" value="yes"/>	Cfwd Sel Serv:	<input type="text" value="yes"/>
Cfwd Last Serv:	<input type="text" value="yes"/>	Block Last Serv:	<input type="text" value="yes"/>
Accept Last Serv:	<input type="text" value="yes"/>	DND Serv:	<input type="text" value="yes"/>
CID Serv:	<input type="text" value="yes"/>	CWCID Serv:	<input type="text" value="yes"/>
Call Return Serv:	<input type="text" value="yes"/>	Call Back Serv:	<input type="text" value="yes"/>
Three Way Call Serv:	<input type="text" value="yes"/>	Three Way Conf Serv:	<input type="text" value="yes"/>
Attn Transfer Serv:	<input type="text" value="yes"/>	Unattn Transfer Serv:	<input type="text" value="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.

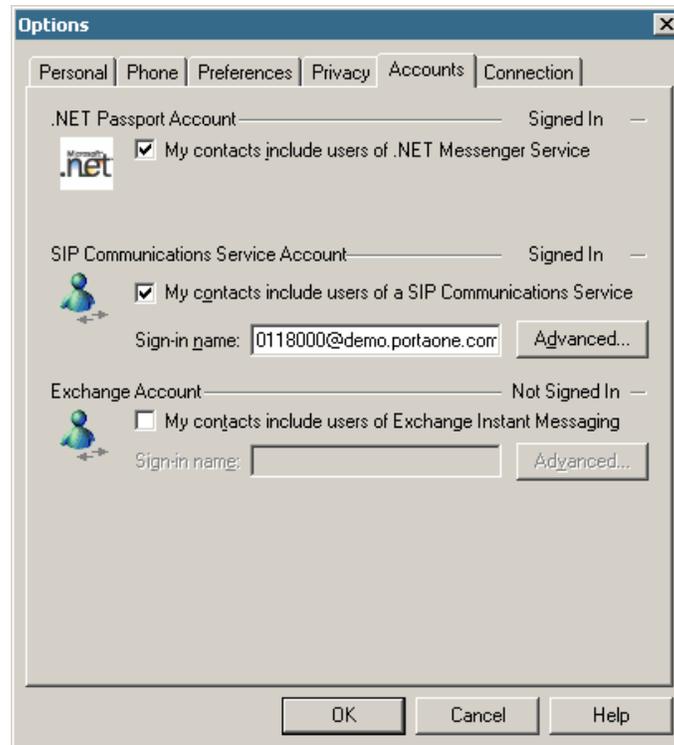
## APPENDIX C. Configuring Windows Messenger for Use as a SIP User Agent

The following instructions apply to Windows Messenger version 5.0.

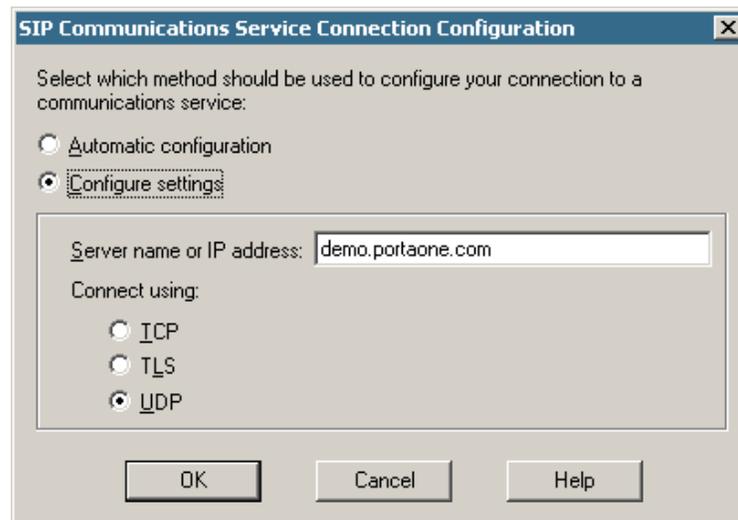
1. Start Windows Messenger, and select “Options...” from the “Tools” menu



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 or 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.



The screenshot shows a dialog box titled "Sign In to a SIP Communications Service". It contains the following fields and options:

- Instruction: "Enter your sign-in name, user name, and password to sign in to demo.portaone.com."
- Sign-in name: Text box containing "0118000@demo.portaone.com". Example: "someone@example.com".
- User name: Text box containing "0118000". Examples: "domain\username" and "someone@example.com".
- Password: Password field with six dots. A checkbox labeled "Save my password" is checked.
- Buttons: "OK" and "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.



The screenshot shows a dialog box titled "Start a Voice Conversation". It features two tabs: "My Contacts" and "Other". The "Other" tab is selected. The dialog contains the following elements:

- Instruction: "Enter the e-mail address of the person you want to contact."
- Text box: "Type the person's complete e-mail address:" with the value "1-604-521-5277".
- Dropdown menu: "Select the service that this person uses:" with "SIP Communications Service" selected.
- Buttons: "OK" and "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.

The screenshot shows the 'Add Tariff' form in a web browser. The form has a blue header with the title 'Add Tariff' and navigation buttons like 'Save', 'Save & Close', and 'Close'. Below the header, there are several input fields and dropdown menus. The 'Name' field contains 'DIDX Termination', 'Currency' is set to 'USD - US Dollar', 'Applied To' is 'Vendor', and 'Service' is 'Voice Calls'. The 'Routing' checkbox is unchecked. A 'General Info' section is expanded, showing fields for 'Default Off-Peak Period' (NOT DEFINED), 'Destination Group Set', 'Free Seconds' (0), 'Post Call Surcharge' (0%), 'Login Fee' (0), 'Connect Fee' (0), 'Round Charged Amount' (XXXXX.XXXXX), 'Default Formula', 'Short Description', and 'Description' (DIDX termination costs). The 'Description' field is a text area with the text 'DIDX termination costs'.

4. Click **Save**.

**Edit Tariff 'DIDX Termination'**

Name: DIDX Termination | Currency: USD  
 Applied To: Vendor | Service: Voice Calls

General Info | Web Upload & Download | Email Upload | Notepad

Default Off-Peak Period: NOT DEFINED  
 Destination Group Set: [Dropdown]  
 Free Seconds: 0  
 Post Call Surcharge: 0.00000 %  
 Login Fee: 0.00000 USD  
 Connect Fee: 0.00000 USD  
 Round Charged Amount: XXXXX.XXXXX  
 Default Formula: [Text]  
 Short Description: [Text]  
 Description: DIDX termination costs

**Rates for tariff 'DIDX Termination'**

Edit	Destination	Country	Interval, second		Price, USD/minute		Effective From	Delete
			First *	Next *	First *	Next *		

**Rates for tariff 'DIDX Termination'**

Edit	Destination	Country	Interval, second		Price, USD/minute		Effective From	Delete
			First *	Next *	First *	Next *		
	44		1	1	0.0011	0.0011	Immediately	

**Rates for tariff 'DIDX Termination'**

Edit	Destination	Country	Interval, second		Price, USD/minute		Effective From	Delete
			First *	Next *	First *	Next *		
	44	UNITED KINGDOM Proper	1	1	0.00110	0.00110	2010-01-12 06:31:20	

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.

Edit	Name *	Login *	Password	Delete
	DIDX - Incoming	67.15.180.14	cisco	X

Edit	Name *	Login *	Password	Delete
	DIDX - Incoming	67.15.180.14	*****	X

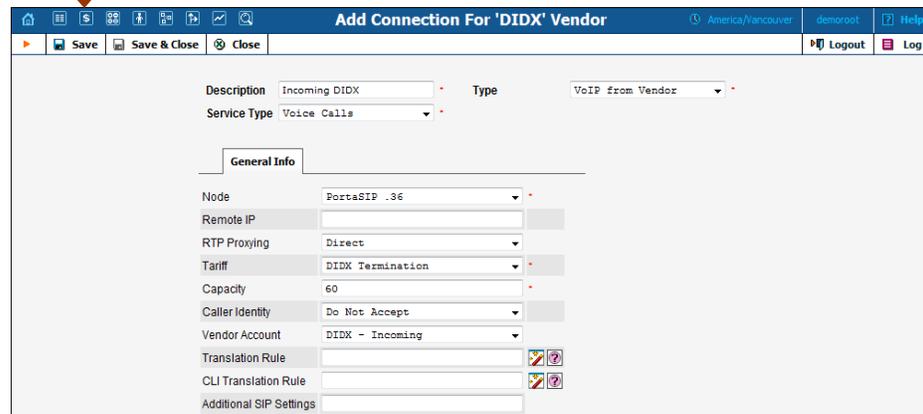
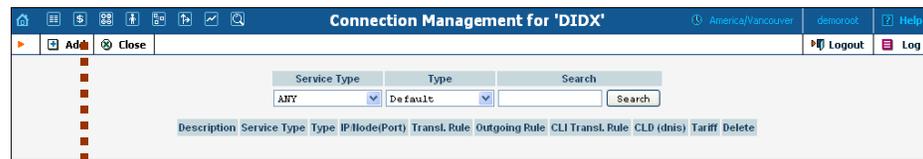
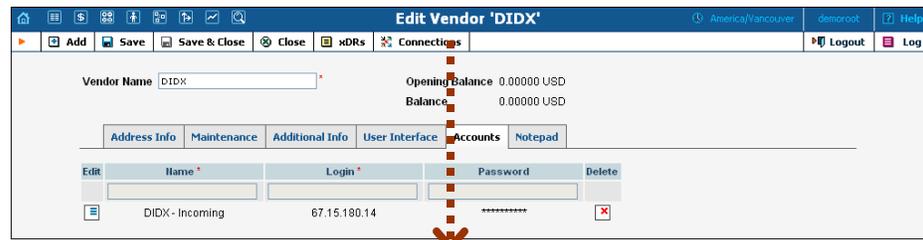
7. Enter the following values:
  - **Name** – A short name for this account description (visible when associating the account with a connection).

- **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. In the **Vendor Account** select menu, choose the account you previously created in the vendor configuration; this account will be used to authorize incoming calls from the vendor. In **RTP Proxying**, specify this

vendor’s capabilities for NAT traversal (if you are not sure whether this vendor can perform NAT traversal, select **On Nat**). **Description** and **Capacity** are mandatory for all connection types.

4. Click  **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:



Click On This Image on the web site to watch video instructions regarding that web page.

**Purchased DID's**

Tools: Bulk Ring to Setting Change, Request CSV

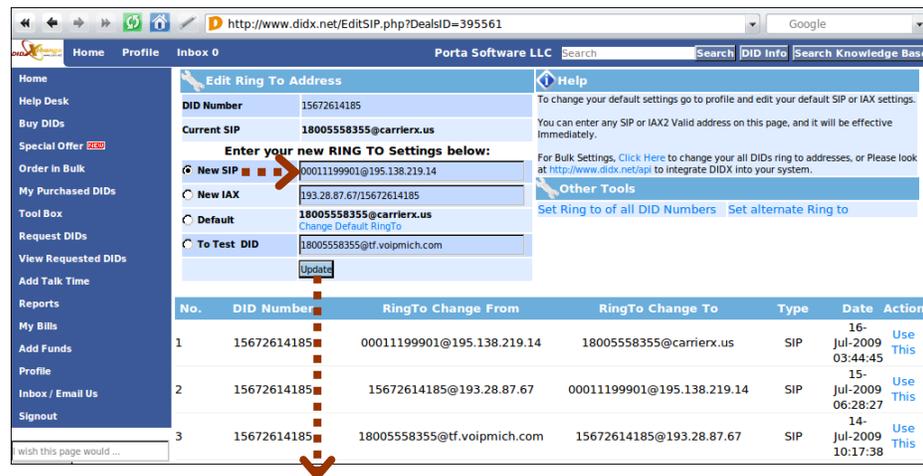
Search: DID Number, Vendor ID, Change Ringto

Search a DID number (or pattern) from your purchased DIDs by DID number prefix or vendor ID. e.g., 1567 or 3345687878

DID Number	Country	Area	Setup (\$)	Monthly (\$) Charges	Purchased Date	Ring To	Action	Logs	DID Test	Documents
1-567-2614185	USA	NW Ohio: Toledo	Free	Free	13-Jul-2009	18005558355@carrierx.us(SIP)	Remove DID	Call Logs	Test	
44-2-30265930	United Kingdom	London	Free	Free	13-Jul-2009	18005558355@tf.voipmich.com(SIP)	Remove DID	Call Logs	Test	

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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**:



http://www.didx.net/EditSIP.php?DealsID=395561

**Edit Ring To Address**

DID Number: 15672614185

Current SIP: 18005558355@carrierx.us

Enter your new RING TO Settings below:

**New SIP** 00011199901@195.138.219.14

**New IAX** 193.28.87.67/15672614185

**Default** 18005558355@carrierx.us

**To Test DID** 18005558355@tf.voipmich.com

Update

Help: To change your default settings go to profile and edit your default SIP or IAX settings. You can enter any SIP or IAX2 Valid address on this page, and it will be effective immediately. For Bulk Settings, Click Here to change your all DIDs ring to addresses, or Please look at http://www.didx.net/api to integrate DIDX into your system.

Other Tools: Set Ring to of all DID Numbers, Set alternate Ring to

No.	DID Number	RingTo Change From	RingTo Change To	Type	Date	Action
1	15672614185	00011199901@195.138.219.14	18005558355@carrierx.us	SIP	16-Jul-2009 03:44:45	Use This
2	15672614185	15672614185@193.28.87.67	00011199901@195.138.219.14	SIP	15-Jul-2009 06:28:27	Use This
3	15672614185	18005558355@tf.voipmich.com	15672614185@193.28.87.67	SIP	14-Jul-2009 10:17:38	Use This

6. Make a test call.