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**PortaSwitch Handbook: SIP Services, July 2006
V.1.12.8**

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Preface

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

Where to get the latest version of this guide

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

Conventions

This publication uses the following conventions:

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



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

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



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



Tips provide information that might help you solve a problem.

1. Setting up Standard SIP Services

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

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

Basic SIP service

Checklist

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

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

Initial Configuration of PortaSwitch

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

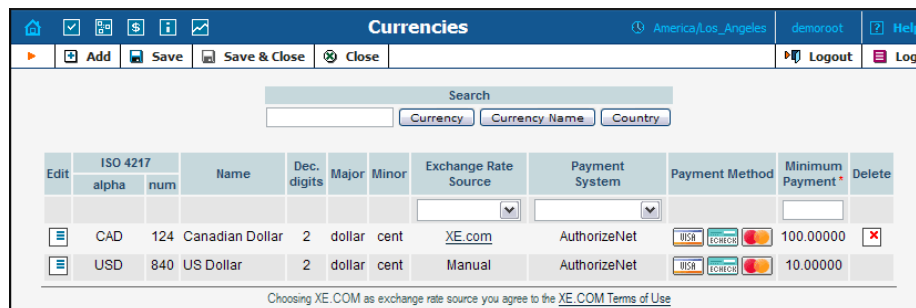


Visit **Company Info** on the main menu. Enter information about your company and set up your base currency. Naturally, this does not limit your operations to this currency only. However, on cost/revenue reports and the like different currencies will be converted to the one you specify here.

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	dollar	cent	XE.com	AuthorizeNet		100.00000	
	USD	840	US Dollar	2	dollar	cent	Manual	AuthorizeNet		10.00000	


Create Destinations

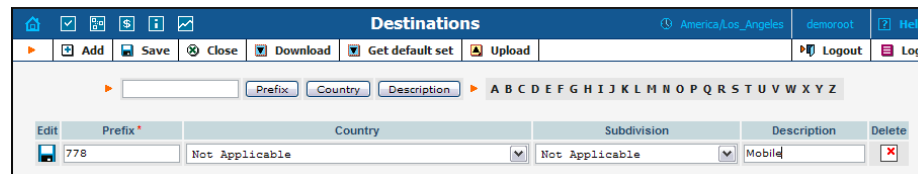
This step is only required if you have not previously defined the necessary destinations. There are two ways to insert a new destination into the system:



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

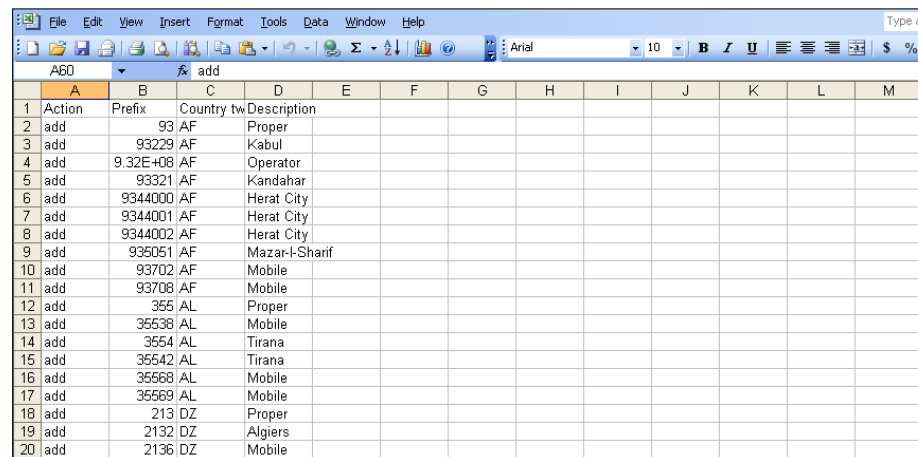


Edit	Prefix*	Country	Subdivision	Description	Delete
	778	Not Applicable	Not Applicable	Mobile	


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

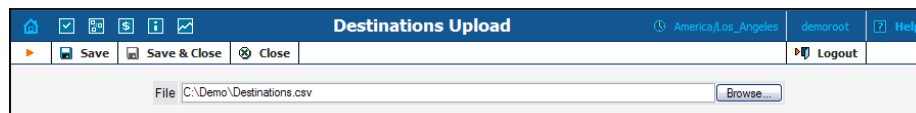
Uploading a set of destinations from a file:

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



Action	Prefix	Country tw	Description
add	93	AF	Proper
add	93229	AF	Kabul
add	9.32E+08	AF	Operator
add	93321	AF	Kandahar
add	9344000	AF	Herat City
add	9344001	AF	Herat City
add	9344002	AF	Herat City
add	935051	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
add	213	DZ	Proper
add	2132	DZ	Algiers
add	2136	DZ	Mobile

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



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

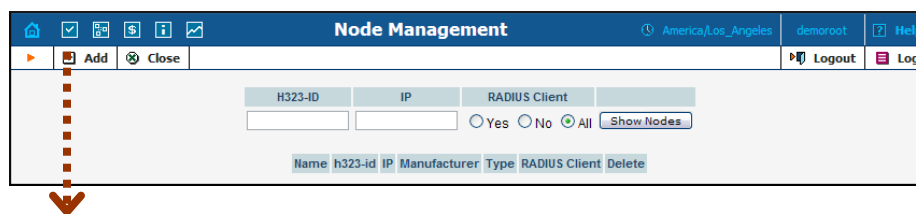
Destinations for SIP phones

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

Even if you have not obtained an official phone prefix, it is highly recommended not to assign IDs to your SIP user agents at random. Choose a non-existing prefix, e.g. **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, ...**

Create Nodes

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



1. In the management section of the Admin-Index page, choose **Nodes**.
2. In the Node management window, click the **Add** icon.
3. Fill in the New Node form:
 - **Node name** – a short descriptive name for your SIP server (this will be used in the select menus).
 - **H323-ID** – PortaSIP server’s hostname (recommended: `hostname.domainname`).
 - **VoIP Password** – if you plan to send calls from your SIP server to your Cisco gateways, where the default Cisco remote IP authentication script will be used, enter **cisco** here.
 - **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).
 - **Manufacturer** – select **PortaOne**.
 - **Type** – VoIP node type; select **PortaSIP**.
 - **Radius Client** – check this, since PortaSIP will need to communicate with the billing.
 - **Radius Key** – enter the radius shared secret here; this must be the same **key** which you entered during the PortaSIP installation.
 - **Radius Source IP** – see the **Node ID, NAS IP address and Radius source IP** section in the [PortaBilling Administrator Guide](#) for more information. Unless your PortaSIP server uses multiple network interfaces, the value here should be the same as the NAS-IP-Address.
4. Click **Save&Close**.

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

Create Tariff



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

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

1. In the Management section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, choose **Add**.
3. Fill in the **Add Tariff** form:
 - **Name** – A short name for the tariff object; this is the name you will then see in the select menus.
 - **Currency** – Indicates in which currency pricing information is defined. All pricing information for a single tariff must be defined in the same currency.

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

- **Type** – If this is a tariff that describes your vendor's termination costs, choose **Routing** here, as this tariff will be used not only to calculate termination costs, but also to route SIP calls. Otherwise, choose the **Ordinary** type.

- **Off-peak Period** – Defines the off-peak period. Click on the Off-peak period wizard icon () to summon the wizard, which will help you construct the correct period definition. Click **Help** for more information on period format definition. If you do not differentiate between peak and off-peak rates, just leave this field empty.
 - **Off-peak Description** – A description of the off-peak period, automatically filled in by the off-peak period wizard; thus you do not have to fill in this field.
 - **Destination Group Set** – If you wish to enter rates in the tariff not for every individual prefix, but for a whole group of prefixes at once, you should create a destination group set and destination groups beforehand. Leave this select menu empty for now.
 - **Free Seconds** – The number of free seconds allowed for each call. In order to claim free seconds, the length of the call must be at least one billing unit (first interval; see the ‘Enter Rates’ section above).
 - **Post-Call Surcharge** – Percentage of the amount charged for the call.
 - **Login Fee** – Amount to be charged immediately after the first user authentication (i.e. after the user enters his PIN).
 - **Connect Fee** – Amount to be charged for each connected call (call with a non-zero duration).
 - **Round Charged Amount** – Instead of calculating CDRs with a 5-decimal-place precision, round up CDR amount values (e.g. to cents, so that 1.16730 becomes 1.17). Set the rounding pattern to XXXX.XX000 (as shown on the picture) so every call will be rounded to the equal cent amount.
 - **Formula** – Default rating formula, which will be applied to every rate created in the tariff. If you leave this empty, the “old-style” rating will be used.
 - **Short Description** – A short tariff description. This will be shown in the rate lookup on the admin interface and the self-care pages for your accounts and customers.
 - **Description** – An extended tariff description.
4. Click  **Save**.
 5. Repeat steps 1-4 until you have entered all of the tariffs. You will need at least two tariffs – one, which you will use to charge your customers, and another, which describes your termination costs. Make sure you choose **Routing** in the **Type** select menu when creating tariffs for your vendors.



Enter Rates

Rates are per-destination prices. Please refer to the [System Concepts](#) chapter for more details on billing parameters.

Managing rates online

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

Edit	Destination *	Country	Description	Interval, sec.		Price, USD/min		Effective From		Delete
				First *	Next *	First *	Next *	YYYY-MM-DD *	HH24:MI:SS *	
	420			Peak	1	1	0.12	0.12	immediately	
				Off-Peak	1	1	0.11	0.11		

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

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

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

- **Huntstop** – do not try any routes with a lower preference.

Managing rates offline

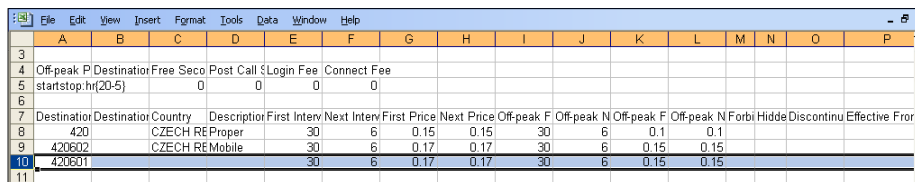
NOTE: Templates are available in PortaBilling, a powerful tool for uploading rates from custom format data files. However, in this particular example we assume that you will enter data using the PortaBilling default format.

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

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



A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	
1	Name	Currency	Description	Short Desc	Off-Peak	Description										
2	SIP Phone	USD	This tariff is SIP phone PERIOD: From 20:00 until 06:00													
3																
4	Off-peak P	Destination	Free Seco	Post Call	Login Fee	Connect Fee										
5	startstop:hr(20-5)		0	0	0	0										
6																
7	Destinator	Destination	Country	Description	First Interv	Next Interv	First Price	Next Price	Off-peak F	Off-peak N	Off-peak F	Off-peak N	Forbid	Hidden	Discontin	Effective From
8	420	CZECH RE	Proper		30	6	0.15	0.15	30	6	0.1	0.1				5/11/2004 3
9	420602	CZECH RE	Mobile		30	6	0.17	0.17	30	6	0.15	0.15				5/11/2004 3
10																
11																

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



	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P
3																
4	Off-peak P	Destination	Free Seco	Post Call S	Login Fee	Connect Fee										
5	startstop.hr(20-5)		0	0	0	0										
6																
7	Destination	Destination Country	Description	First Interv	Next Interv	First Price	Next Price	Off-peak F	Off-peak N	Off-peak F	Off-peak N	Forbid	Hidden	Discontin	Effective	Fror
8	420	CZECH RE	Proper	30	6	0.15	0.15	30	6	0.1	0.1					
9	420602	CZECH REMobile		30	6	0.17	0.17	30	6	0.15	0.15					
10	420601			30	6	0.17	0.17	30	6	0.15	0.15					
11																



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

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

Create All Required Tariffs



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


- A tariff for each account billing scheme. For example, if you plan to have two different SIP call products with different rates, you will need two separate tariffs. These tariffs should be created using the **Ordinary** type.
- Create a tariff with the termination costs for each termination partner you have, these tariffs will also include your routing preferences.
- If you have resellers, create the tariffs you will use to charge each of them. Do not create tariffs, which will be applied to subscribers of your resellers yet. Create customers first and then return to this step. Make sure that, when creating these subscriber tariffs, you choose the *Managed by NNN* in the **Type** menu, where *NNN* is the name of the corresponding resellers.

Create Product

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



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

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

- **Managed by** – If you want this product to be used for your reseller’s accounts, so the reseller himself can change the parameters of this tariff and create new accounts using this product, choose a customer name from the menu. Otherwise, choose **None** here.
 - **Breakage** – 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 later sections of this document you will find examples for setting up volume discounts).
 - **Info URL** – If you have an external server with a description of product features, enter the URL here (e.g. <http://www.myproduct.com>). Your customers will be able to go there from their self-care page.
 - **Description** – your comments about the intended use of this product.
3. Click  **Save**.
 4. Click on the **Accessibility** tab to edit this product’s accessibility.


Enter Node and Tariff into the product's accessibility list

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

1. When the Accessibility tab is selected, click on the  **Add** icon.
2. In the accessibility entry window, select the PortaSIP node and choose the appropriate tariff which applies to your customers when they make outgoing calls.
3. The **Access code** or **Info Digits** fields only make sense when a call originates from your customer in a public telephony network. Therefore, just leave this empty for the SIP service.
4. Click  **Save** to save this accessibility entry.

Create Vendors

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

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

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

Main form (top)

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

Additional info:

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

User Interface:

- o **Time zone** – the time zone that the vendor uses for his billing period (when sending you an invoice). Statistics will be split into periods in this time zone, so your statistics will match the vendor's.
4. Click **Save**.
 5. If you plan to terminate your calls to the vendor's SIP server, typically he would provide you with a username/password which will authorize you to send calls to his server. Enter this information as **Vendor account**.

Edit Vendor (America/Los_Angeles, demoroot, Help)

Vendor Name: GlobalNet | Opening Balance: 0.00000 USD | Balance: 0.00000 USD

Address Info | Maintenance | Additional Info | **User Interface** | Accounts | Notepad

Edit	Name *	Login *	Password	Delete
	GlobalNet-SIP	demoVendor	demo123	

Edit Vendor (America/Los_Angeles, demoroot, Help)

Vendor Name: GlobalNet | Opening Balance: 0.00000 USD | Balance: 0.00000 USD

Address Info | Maintenance | Additional Info | **User Interface** | Accounts | Notepad

Edit	Name *	Login *	Password	Delete
	GlobalNet-SIP	demoVendor	*****	

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

Define Connections

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

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

Vendor Management (America/Los_Angeles, demoroot, Help)

Search:

CDRs	Name	Connections	Currency	Balance	E-mail	Delete
	GlobalNet		USD	0.00000	info@globalnet.com	

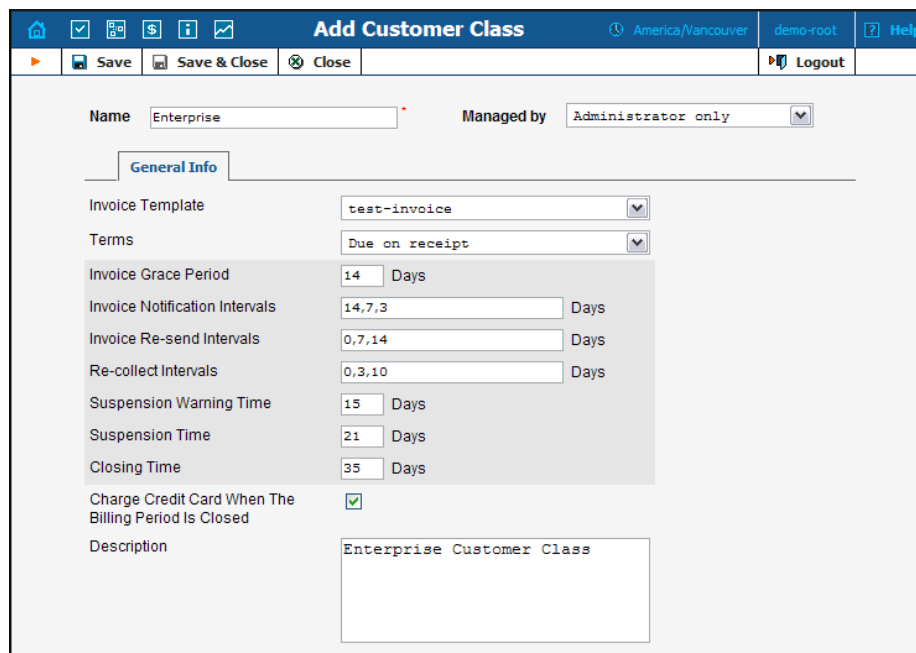
Edit	Load	Remote IP *	Tariff	RTP Proxying	Transl. Rule	Outgoing Rule	CLI Transl. Rule	Account	Capacity *	Delete
		70.68.128.186	Optimal					GlobalNet-SIP	3d	


- Choose the type of connection (**PSTN to Vendor**, **VoIP to Vendor**, etc.) by clicking on the corresponding tab.
- Press **Add** to add a new connection.
- 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.
- Click **Save**.
- Repeat steps 3-6 to add more connections to the same vendor, then click **Close** to exit to the **Vendor Management** screen.
- Repeat steps 2-7 to add connections for other vendors.

Create a 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.)


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



3. Fill in the customer class parameters:
 - **Name** – Short name for this customer class.
 - **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.
 - 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 product.
4. Click  **Save&Close**.

Create a Customer

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

1. In the Management section of the Admin-Index page, choose **Customers**.
2. On the Customer Management page, choose  **Add**.

- Fill in the **New Customer** form. Please note that there are several tabs with extra information available on the screen.

The most important fields are:

Main form (top)

- **Name** – short name for the customer object; this will be used on the web interface.
- **Currency** – the currency in which this customer will be billed.
- **Opening balance** – a starting balance for the customer; the default is zero.
- **Type** – is it a reseller or retail (direct) customer? (Normally, most of your customers would be retail customers. Only if a customer is reselling your services, and you are providing services and billing to his subscribers, would he be created as a reseller).
- **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 CDRs and other statistics will be sent to this address
- **Bcc** – Blind carbon copy in email; may be used for debug and archiving purposes.
- **Send Statistics** – If you choose “Summary only”, the statistics email will only include a brief call summary, without an attached details file. This is useful for large customers whose CDR file is too large to be sent via email.

Additional info tab

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

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 CDRs 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.

Call Features tab

This tab allows you to define global call parameters for a customer's IP Centrex environment. It also provides a convenient tool for managing parameters which are to be the same for a whole set of accounts (extensions) within the IP Centrex. 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.

- **Max number of simultaneous calls** – This allows you to limit the number of concurrent calls for accounts of this customer. This may be useful for preventing service abuse in the case of

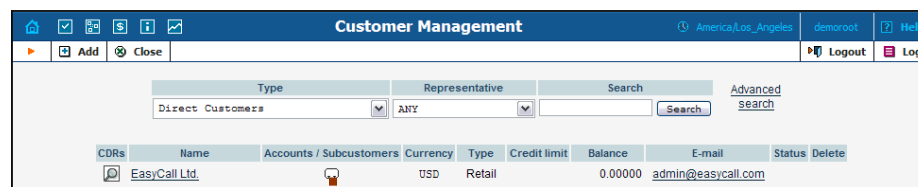
“free unlimited” calling plans. Leaving this field empty places no specific limit on the maximum number of calls.


- **VoiceVPN** – Enables the Voice VPN feature. Leave this option disabled for now. Later in this guide we will provide an example of setting up an advanced SIP service including VoiceVPN.
- **First login greeting** – If enabled, for each account of this customer PortaSwitch will call the phone back upon the first successful registration. When the user answers the phone, he will get a pre-recorded message confirming his service activation and providing information regarding your service.
- **Dialing rules** (on the right) – Here you should specify how the customer will be dialing phone numbers; you may also use one of the pre-defined examples.

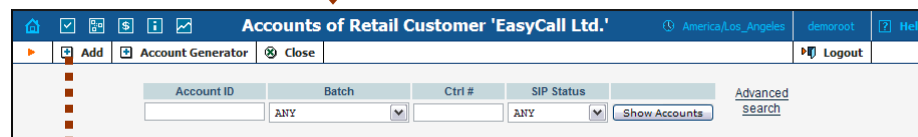
4. Click  **Save&Close**.

Create Accounts

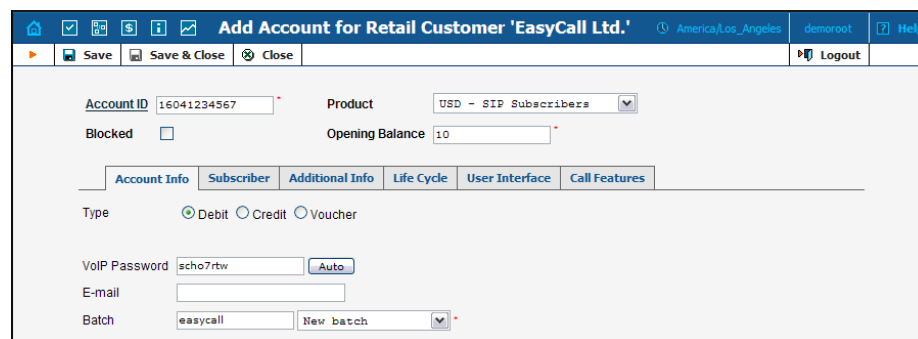
1. Go to the **Customers** screen (the one containing the list of customers). It should resemble the screenshot below.



CDRs	Name	Accounts / Subcustomers	Currency	Type	Credit limit	Balance	E-mail	Status	Delete
	EasyCall Ltd.		USD	Retail		0.00000	admin@easycall.com		



Account ID	Batch	Ctri #	SIP Status	Advanced search
	ANY		ANY	Show Accounts



Account ID: 16041234567 Product: USD - SIP Subscribers



Blocked: Opening Balance: 10

Type: Debit Credit Voucher

VoIP Password: scho7rtw Auto

E-mail:

Batch: easycall New batch

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

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

Account Info tab

- **Account type** – Account type; select credit for 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).
- **VoIP 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.
- **Lifetime** – Relative expiration date; since we are setting up a postpaid service, which should function for a long time, leave this field blank.

User Interface tab

- **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.

Call Features tab

- **Preferred IVR Language** – This is a custom attribute which is transferred to the IVR. Leave English here if you are unsure whether your IVR supports this function.
 - **Associated Number** – Redirect number. Leave this empty, since it is not used by PortaSIP.
 - **UM Enabled** – Check this box if the account has unified messaging services (e.g. voicemail) enabled.
 - **Follow Me Enabled** – Check this box if the account has the “Follow Me” feature enabled. If so, the account owner can define a list of numbers to which incoming calls to his UA will be redirected (for example his home phone, cell phone, and so on).
5. After clicking **Save&Close**, you will see a confirmation screen announcing that a new account has been created.
 6. Repeat steps 3-5 if the customer has more than one phone line (extension).

Set up Dialing Rules for the 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 America/Los_Angeles demoroot Help

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

Address Info Maintenance Additional Info Payment Info User Interface Periodic Payments Call Features **Abbreviated Dialing** Subscriptions Notepad

Max number of simultaneous calls:
 Set CLI to Account ID: No
 Set CLI To Centrix:
 Hide CLI: No
 Voice VPN: No
 Voice VPN Distinctive Ring: No
 Legal Intercept: No
 First Login Greeting: No
 Music On Hold: No Frills Cumbia (c) 2001 Kevin

Dialing rules wizard
 Load Sample: E.164
 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):
 Convert ANI (CLI) for incoming calls into this dialing format:

Dialing rules wizard Help

Save Save & Close Close

Your country code: 1
 Your area code(s): 604
 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): 1
 International dialing prefix (e.g. 011, 00, 0011): 011
 Emergency numbers (e.g. 911, 112): 911
 Exceptions (e.g. *98): 411
 Convert ANI (CLI) for incoming calls into this dialing format:

Sample settings

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

Clear Reset Load Sample

Check your self


To call 1234567 outside of your office, but within the same area you dial: 1234567
 To call long distance 604 1234567 (within your country) you dial: 1 604 1234567
 To call 420 2 12345678 internationally you dial: 011 420 2 12345678

Thus, when one of this customer's accounts tries to make a call to 90042021234567, 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 the Customer (optional)

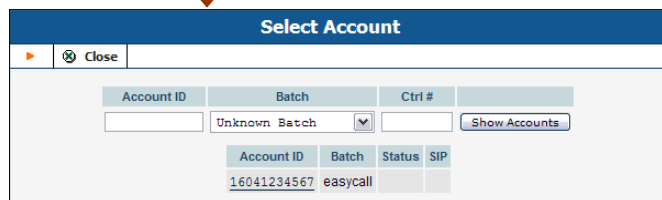
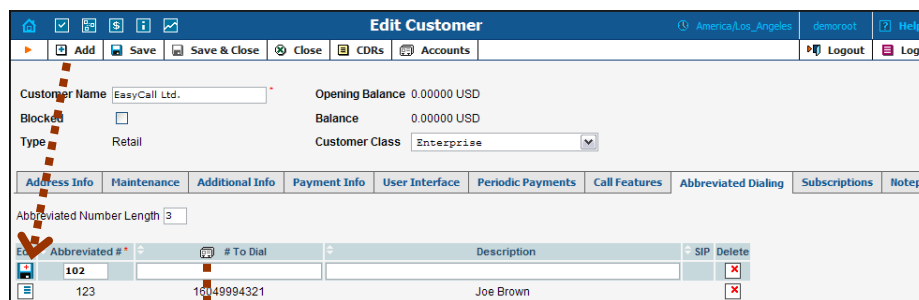
If your customer has multiple SIP accounts, and plans to make calls between them, it would be very inconvenient to have to dial a complete E.164 number each time. Therefore, you can create abbreviated dialing rules, so that it will suffice to dial, for example, 120 to reach a Jeff Smith from any SIP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

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

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

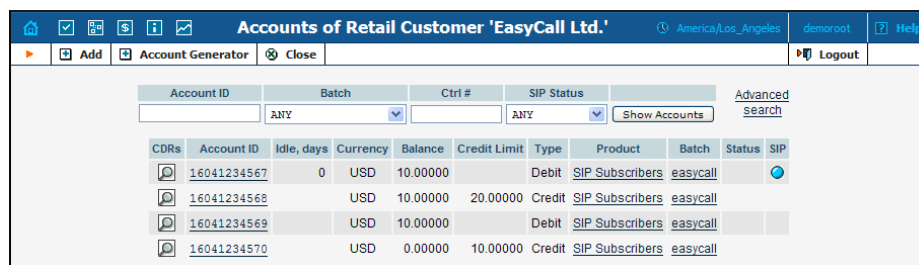
NOTE: If you enter an off-net PSTN number in **# To Dial**; it must be in the E.164 format, i.e. you cannot enter the number in the customer's dialing format.

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



Testing the Whole System

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



- 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 Info	Maintenance	Subscriber	Additional Info	Life Cycle	User Interface	Call Features	Subscriptions	Notepad
Customer	EasyCall Ltd.		Opening Balance	10.00000 USD				
Type	Debit		Refunds	0 USD				
VolIP Password	scho7rtw	Auto	Non call related charges	0 USD				
E-mail								
Batch	easycall							
Control number	1							

4. Try to make a call using one of the accounts

Check the Log Files

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

Using the SIP log viewer



You can also browse the SIP log file from the PortaBilling web interface using a SIP log viewer, which offers added convenience: for instance, automatically drawing call flow diagrams for you.

1. Choose the **SIP Log Viewer** item from the main menu.
2. Choose your PortaSIP node in the **PortaSIP node** select menu and select the time interval on the right side of the screen. Now you will see all call attempts in that interval. Press the **Generate** button to produce a call attempts list.

The screenshot shows the SIP Log Viewer interface. At the top, it displays the current node as '193.28.87.106 - PortaSIP'. Below this, there are several sections for configuring the search:

- Trace a call:** Includes fields for 'Call-ID' (with a warning 'but not H323-Conf-ID!'), 'Another Call-ID' (optional for callback calls), and a 'Search log file for' dropdown set to '--SELECT--'. There are checkboxes for 'include log file for the previous day in the search' and 'Output format' options: 'only text log' (selected), 'log with call diagram' (experimental, does not work for all calls).
- Show call attempts:** Includes a 'For last' dropdown set to '5 minutes' with a note: 'Use carefully for large interval, may produce huge list of calls.' and a 'Filter' checkbox for 'do not show selftest calls' which is checked.

A 'Generate' button is located at the bottom right of the search area, with a red dashed arrow pointing down to the next screenshot.

The screenshot shows the results of the search. It displays a table of call attempts in the current sip.log file on 193.28.87.106 PortaSIP node since 11 Jun 22:08:28 EEST (UTC +0300):

Setup Time (in SIP server TZ)	Caller's IP:UdpPort	CLI	CLD	Call-ID	Caller's User Agent
11 Jun 22:09:43	216.231.44.168:9062	16041234568	6831234	98dda488-69c74dd3@192.168.0.250	Sipura/SPA2000-3.1.5

A red dashed arrow points down from the 'Call-ID' column to the next screenshot.

```

siplogview version: 1.29, experimental mode.

PortaSIP node: 193.28.87.106
Call-ID: 98dda488-69c74dd3@192.168.0.250
H323-Conf-ID: A9B6252E 3863D707 17A91CC0 744348CB

PortaSIP UA server 216.231.44.168 sipura SPA2000-3.1.5
ser 193.28.87.106 PortaSIP
b2bua 193.28.87.106 PortaSIP
AAA PortaBilling
UA 193.28.187.3

22:09:43 0-> (A? 101/I) * INVITE ----->
22:09:43 1<- (A? 101/I) 100 trying -->
22:09:43 0-> (A? 101/I) * INVITE ----->
22:09:43 1<- (A? 101/I) 401 Unauthor -->
22:09:43 0-> (A? 101/A) ACK ----->
22:09:43 0-> (A? 102/I) * INVITE ----->
22:09:43 1<- (A? 102/I) 100 trying -->
22:09:43 0-> (A? 102/I) * INVITE ----->
22:09:43 1<- (A? 102/I) 100 Trying ----->
22:09:44 1<- Auth request accepted ----->

11 Jun 22:09:43/GLOBAL/ser[98218]: RECEIVED message from 216.231.44.168:9062:
INVITE sip:6831234@193.28.87.106 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-1035d24f
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f8700
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
Max-Forwards: 70
Contact: John Doe <sip:16041234568@192.168.0.250:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 428
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
    
```

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

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

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

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

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

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

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

Request is sent to B2BUA.

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

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

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

B2BUA receives this INVITE request.

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

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

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

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

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

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

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

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

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

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

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

```
11 Jun 22:09:43/GLOBAL/ser[98219]: RECEIVED message from 216.231.44.168:9062:
INVITE sip:6831234@193.28.87.106 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-1628d42b
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 102 INVITE
Max-Forwards: 70
Authorization: Digest
username="16041234568",realm="193.28.87.106",nonce="3051864d7d0c6578460cc4b0e28ad43b448c6a77",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=873d0427882f87o0;lr>
From: "John Doe" <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>;tag=b293b17e775f2bdf6d192e545261bb1d
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 102 INVITE
Server: Sippy
```

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

The remote SIP proxy requests authorization.

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

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

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

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

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

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

The called party's phone starts ringing.

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

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

The called party answers the call.

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

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

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

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

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

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

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

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

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

```
Max-Forwards: 70
Route: <sip:193.28.87.106;ftag=873d0427882f8700;lr>
Authorization: Digest
username="16041234568",realm="193.28.87.106",nonce="3051864d7d0c6578460cc4b0e28ad4
3b448c6a77",uri="sip:
193.28.87.106:5061",algorithm=MD5,response="4e3308a385aeff5159ad03f738f9bd31"
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 0
```

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

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

The outgoing call leg is closed.

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

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

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

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



The other party confirms call disconnection.

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

The call is finished.



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

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

PortaBilling receives the authorization request.

```
Jun 11 12:09:44: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address      = '193.28.87.106'
User-Name           = '16041234568'
Called-Station-Id   = '6831234'
Calling-Station-Id  = '16041234568'
h323-conf-id        = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id             = '98dda488-69c74dd3@192.168.0.250'
Digest-Attributes   = 'Realm = "193.28.87.106"'
Digest-Attributes   = 'Nonce = "3051864d7d0c6578460cc4b0e28ad43b448c6a77"'
Digest-Attributes   = 'Method = "INVITE"'
Digest-Attributes   = 'URI = "sip:6831234@193.28.87.106"'
Digest-Attributes   = 'Algorithm = "MD5"'
Digest-Attributes   = 'User-Name = "16041234568"'
Digest-Response     = '54e0b42337ace33edf36d004f1037ebd'
h323-remote-address = '216.231.44.168'
h323-session-protocol = 'sipv2'
h323-ivr-out        = 'PortaBilling_Routing:SIP'
h323-ivr-out        = 'PortaBilling_AuthMethod:INVITE'
h323-ivr-out        = 'PortaBilling_Notify:NAT'
h323-ivr-out        = 'PortaBilling_Seed:144514807'
NAS-Port            = '5060'
Jun 11 12:09:44: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:09:44: H323/SIP call, use h323-conf-id, but remember call-id
Jun 11 12:09:44: Checking if this call comes through a VoIP from vendor connection
Jun 11 12:09:44: No VoIP from vendor connections were found
Jun 11 12:09:44: PrepareNexecute 'AccountAuth'
Jun 11 12:09:44: Found Account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000]
```

Account information is located in the database.

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

Password verification is successful.

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

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

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

The dialed number is an off-net destination.

```

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

```

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

```

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

```

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

```

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

```

An authorization response is sent to PortaSIP.

```

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

```

Accounting for the failed outgoing call leg is received.

```

Jun 11 12:10:04: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address    = '193.28.87.106'
User-Name         = '16041234568'
Called-Station-Id = '16046831234'

```

```

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

```

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

```

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

```

Connection matched.

```

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

```

A failed CDR is inserted for the vendor.

```

Jun 11 12:10:04: Accounting response

```

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

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

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

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

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

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

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

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

```
Jun 11 12:10:09: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address      = '193.28.87.106'
User-Name           = '16041234568'
Called-Station-Id   = '16046831234'
Calling-Station-Id  = '16041234568'
Acct-Status-Type    = 'Start'
h323-call-origin    = 'originate'
h323-call-type      = 'VoIP'
h323-setup-time     = '19:10:04.000 GMT Sun Jun 11 2006'
h323-connect-time   = '19:10:09.000 GMT Sun Jun 11 2006'
h323-conf-id        = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id             = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id     = '98dda488-69c74dd3@192.168.0.250'
Acct-Delay-Time     = '0'
```

```

h323-session-protocol = 'sipv2'
h323-remote-address   = '70.68.128.186'
h323-ivr-out          = 'DURATION:7200'
h323-ivr-out          = 'PortaBilling_Seed:144514807'
alert-timepoint       = '19:10:08.000 GMT Sun Jun 11 2006'
NAS-Port              = '5060'
Exec-Program-Log      = 'porta-billing.pl'
Jun 11 12:10:09: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:10:09: Found a call in cache with such id
Jun 11 12:10:09: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request

```

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

```

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

```

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

```

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

```

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

```

Jun 11 12:11:24: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address        = '193.28.87.106'
User-Name             = '16041234568'
Called-Station-Id    = '6831234'
Calling-Station-Id   = '16041234568'
Acct-Status-Type     = 'Stop'
h323-call-origin     = 'answer'
h323-call-type       = 'VoIP'
h323-setup-time      = '19:09:44.000 GMT Sun Jun 11 2006'
h323-connect-time    = '19:10:09.000 GMT Sun Jun 11 2006'
h323-disconnect-time = '19:11:23.000 GMT Sun Jun 11 2006'
h323-disconnect-cause = '0'
h323-voice-quality   = '0'
h323-conf-id         = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id              = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-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: No VoIP from vendor connections were found
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 the Account

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

The screenshot shows the 'Accounts of Retail Customer 'EasyCall Ltd.' interface. The top navigation bar includes 'Add', 'Account Generator', 'Close', and 'Logout'. Below the navigation bar is a search and filter section with fields for 'Account ID', 'Batch' (set to 'ANY'), 'Ctrl #', and 'SIP Status' (set to 'ANY'), along with a 'Show Accounts' button and an 'Advanced search' link.

The main table displays account details:

CDRs	Account ID	Idle, days	Currency	Balance	Credit Limit	Type	Product	Batch	Status	SIP
	16041234567	3	USD	9.91000		Debit	SIP Subscribers	easycall		
	16041234568	0	USD	10.21000	20.00000	Credit	SIP Subscribers	easycall		
	16041234569	3	USD	9.98000		Debit	SIP Subscribers	easycall		
	16041234570		USD	0.00000	10.00000	Credit	SIP Subscribers	easycall		
	test123		USD	10.00000		Credit	SmartCall	test123		

A red arrow points from the 'test123' row to the 'Call history' window. The 'Call history' window has a 'Close' button and search filters for 'From Date' (2006-06-10) and 'To Date' (2006-06-11), both in YYYY-MM-DD format. There is a 'Show Incomplete Calls' checkbox (unchecked) and a 'Show CDRs' button. Another red arrow points from the 'Show CDRs' button to the detailed call history window.

The detailed 'Call history' window shows summary statistics for account 16041234568:

Account	16041234568	Balance	10.21000 USD
Total Charged Time, min:sec	2:48	Opening Balance	10.00000 USD
Total Charged	0.07000 USD	Charged by	'SIP Subscribers' product
Total Credits	0 USD	Type	Credit
Total Transactions	4	From	2006-06-10 00:00:00
		To	2006-06-12 00:00:00

Below the summary is a table of call records:

View	From	To	Country	Description	Date/Time	Charged time, min:sec	Amount, USD
	16041234568	16046831234	CANADA	British Columbia	2006-06-11 12:10:09	1:18	0.03000
	16041234568	16043102255	CANADA	British Columbia	2006-06-11 12:05:35	0:30	0.01000
	16041234568	16046282508	CANADA	British Columbia	2006-06-11 12:03:59	0:30	0.01000
	16041234568	16046282508	CANADA	British Columbia	2006-06-10 23:42:03	0:30	0.02000

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

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

Check the Call History

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

The screenshot shows the 'Call Trace' window with a search form and a table of call records. The search form includes fields for 'H323-conf-id', 'Destination %', 'From' (date and time), and 'To' (date and time). Below the search form is a table with columns: View, Error Report, CL(iani), CLD(dnis), Country, Description, Connect Time, Disconnect Time, Duration, Amount, Account, Customer, Vendor, and Disconnect Reason. The table contains six rows of call records. Below the table is a legend titled 'List of possible Disconnect reasons:' with color-coded boxes for: Normal completed call (green), Normal uncompleted call (cyan), Call progress code (yellow), Calling side error (orange), Called side error (red), and Network error (dark red).

View	Error Report	CL(iani)	CLD(dnis)	Country	Description	Connect Time	Disconnect Time	Duration, min:sec	Amount	Account	Customer	Vendor	Disconnect Reason
	E	16041234568	16046831234	CANADA	British Columbia	2006-06-11 12:10:09	2006-06-11 12:11:23	1:14	0.01234 USD	16041234568	EasyCall Ltd.	GlobalNet	Normal call clearing
	E	16041234568	16046831234	CANADA	British Columbia	2006-06-11 12:10:04	2006-06-11 12:10:04	0:00	0 USD	16041234568	EasyCall Ltd.	MCI Vendor	Recovery on timer expiry
	E	16041234568	16043102255	CANADA	British Columbia	2006-06-11 12:05:35	2006-06-11 12:05:55	0:20	0.00334 USD	16041234568	EasyCall Ltd.	GlobalNet	Normal call clearing
	E	16041234568	16043102255	CANADA	British Columbia	2006-06-11 12:05:30	2006-06-11 12:05:30	0:00	0 USD	16041234568	EasyCall Ltd.	MCI Vendor	Bearer service not implemented
	E	16041234568	16046282508	CANADA	British Columbia	2006-06-11 12:03:59	2006-06-11 12:04:05	0:06	0.00100 USD	16041234568	EasyCall Ltd.	GlobalNet	Normal call clearing
	E	16041234568	16046282508	CANADA	British Columbia	2006-06-11 12:03:54	2006-06-11 12:03:54	0:00	0 USD	16041234568	EasyCall Ltd.	MCI Vendor	Recovery on timer expiry

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

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

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

2. Setting up PSTN-to-SIP Services

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

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

Incoming DID calls (from PSTN)

Checklist

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



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

Operation	Done
Initial configuration	
General and network configuration has already been performed according to the instructions in the <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.	[]
Network configuration	
Create a node N for your PSTN gateway.	[]
Rating configuration	
Create a tariff A, which will apply to SIP subscribers for their incoming calls.	[]
Enter rates in tariff A for the destinations covering your incoming DID range, i.e. the rates you plan to charge your customers.	[]
Create a tariff B, which describes the costs of incoming DIDs from this DID provider (it should be of the Ordinary type).	[]
Enter rates in tariff B for the incoming DID range (zero cost).	[]
Create a new accessibility entry in your SIP product, using node N and tariff A.	[]
Create a vendor for DID suppliers.	[]
Create a “PSTN from Vendor” connection for this vendor on node N using tariff B.	[]
Account provisioning	
Check that the retail SIP service customer has already been created according to the instructions in the <i>Basic SIP service</i> section.	[]
Create a new account for this customer, with an account ID identical to the SIP phone number (DID).	[]
Testing	
Program the parameters (phone #, password, SIP server address) into the SIP phone and make a test call.	[]

Cisco gateway configuration guidelines

Obtain the PSTN2SIP script

This script serves to authorize incoming calls and provide the required information to the billing. It is available from the PortaOne web store.

Place the `psn2sip.tcl` script and all the required prompts on your TFTP server. For your convenience, the TFTP server is pre-configured in the directory `/tftpbboot` on the PortaBilling100 slave server.

Basic router configuration

The latest IOS and DSP telephony firmware is highly recommended. It is also recommended that the hostname be the same as `h323-id`.

```
hostname <h323_id>
ip domain name <default domain>
```

NOTE: VSA `h323-gw-id="hostname.domain"`

NTP

NOTE: It is very important to have reliable time services. Also make sure that the time zone abbreviation is one of the standard ones supported by PortaBilling.

```
ntp server <name/IP>
.....
ntp server <name/IP>
ntp master 5
clock timezone <your time zone> 1
clock summer-time <your summer time zone> recurring <your
rules>
```

AAA

```
aaa new-model
aaa authentication login h323 group radius
aaa authorization exec h323 group radius
aaa accounting connection h323 stop-only group radius
```

VoIP interface

```
interface <your interface to the world>
 h323-gateway voip interface
 h323-gateway voip h323-id <h323_id>
```

NOTE: If you want to use a virtual interface then add the line:

```
h323-gateway voip bind srcaddr <IP>
```

Outgoing SIP server

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

Enable gateway functionality

```
gateway
```

Enable gateway accounting

For older IOS versions:

```
gw-accounting h323 vsa
```

For newer IOS versions (12.2T or 12.3):

```
gw-accounting aaa
  acct-template callhistory-detail
```

NOTE: VSA does not work for all platforms.

Radius

IMPORTANT NOTE: Ports 1645/1646 are the traditional Radius ports used by many vendors, without obtaining an official IANA assignment. The official assignment is now ports 1812/1813, and users are encouraged to migrate to these new ports when possible.

Cisco notes:

- “radius-server” commands will be available only after issuing the “aaa new-model” command
- UDP port for RADIUS accounting server – the default is 1646 (see note above)
- UDP port for RADIUS authentication server – the default is 1645 (see note above)

Remember:

- The default ports for Cisco are 1645/1646
- The defaults in /etc/ services are 1812/1813

```
radius-server host <name/IP> auth-port 1812 acct-port 1813
radius-server key <key>
radius-server vsa send accounting
radius-server vsa send authentication
```

voice-card

controller

voice-port

This depends on your hardware configuration.

call application voice & dial-peers

```
call application voice pstn2sip tftp://your-server/pstn2sip.tcl
call application voice pstn2sip authenticate-by dnis
call application voice pstn2sip skip-password yes
call application voice pstn2sip authorize yes
call application voice pstn2sip dial-account-id yes
```

```
dial-peer voice 100 pots
  incoming called-number .T
  application pstn2sip
  voice-port 0:d
!
dial-peer voice 60 voip
  destination-pattern .T
  session protocol sipv2
  session target sip-server
!
```

The example above is for when you receive incoming calls with phone numbers which are already in E.164. If a number is received in a local format, you will have to use the translate feature in the PSTN2SIP script to convert it into E.164. For instance, if you receive a US phone number in NANP (area code + phone number), you should add the following command to the application configuration:

```
call application voice pstn2sip translate "/^/1/"
```

Then configure your gateway to send outgoing calls to the SIP server according to the instructions in the previous section.



Set up a 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 PSTN Gateway Node

Now you need to create a node which will represent your PSTN gateway. PortaBilling requires some key information about your network equipment, such as the IP address, h323-id, Radius shared secret, and so on.

The screenshot shows the 'Node Management' interface. At the top, there are navigation icons and the text 'Node Management'. Below that, there are buttons for 'Add', 'Close', 'Logout', and 'Log'. The main area contains a form with the following fields:

- H323-ID:
- IP:
- RADIUS Client: Yes No All

Below the form is a table with the following columns: Name, h323-id, IP, Manufacturer, Type, RADIUS Client, Delete.

Add Node

Node Name: NY-GW-01

Node info

h323-ID: ny-gw-01.mydomain.com

VoIP Password: cisco

NAS-IP-Address: 23.56.78.90

Hostname:

Domain:

Auth. Transl. Rule:

Manufacturer: Cisco

Type: VOIP-GW

RTP Proxying: Optimal

RADIUS Client:

RADIUS Key: secret123

RADIUS Source IP: 23.56.78.90

RADIUS Dictionary: Cisco

Node Management

Name	h323-id	IP	Manufacturer	Type	RADIUS Client	Delete
Mera	192.168.0.1	192.168.0.1	Mera	MVTS		
CT GW	begemot	193.28.87.3	Cisco	VOIP-GW		
Cisco GW	cisco	192.168.0.5	Cisco	VOIP-GW		
cisco-gw	cisco-gw.mydomain.com	192.168.0.99	Cisco	VOIP-GW		
NY-GW-01	ny-gw-01.mydomain.com	23.56.78.90	Cisco	VOIP-GW		

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

1. In the Management section of the Admin-Index page, choose **Nodes**.
2. In the Node Management window, click the **Add** icon.
3. Fill in the New Node form:
 - **Node name** – a short descriptive name for your gateway (this will be used in the select menus).
 - **h323-ID** (recommended: hostname.domainname)
 - **VoIP Password** – enter **cisco** here.
 - **NAS-IP-Address** – the IP address of the gateway.
 - **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).
 - **Manufacturer** – select **Cisco**.
 - **Type** – VoIP node type; select **VOIP-GW**.
 - **Radius Client** – check this, since gateway will need to communicate with the billing to authorize incoming calls.
 - **Radius Key** – enter the radius shared secret here; this must be the same **key** which you entered during the Cisco gateway configuration.
 - **Radius Source IP** – see the **Node ID, NAS IP address and Radius source IP** section in the [PortaBilling Administrator Guide](#) for more information; unless your PortaSIP server uses

multiple network interfaces, the value here should be the same as the NAS-IP-Address.

4. Click **Save&Close**.

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

Create 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' page. It has a header with navigation icons, a search bar, and a table with columns: Name, Currency, Type, Description, Rates, and Delete. A red dashed arrow points to the 'Add' button in the top left.

The bottom screenshot shows the 'Add Tariff' form. It has a header with navigation icons and a search bar. The form fields are:

- Name: SIP subscribers - incoming
- Currency: USD - US Dollar
- Type: Ordinary
- General Info tab is selected.
- Off-Peak Period: [empty]
- Off-Peak Description: [empty text area]
- Destination Group Set: [empty dropdown]
- Free Seconds: 0
- Post Call Surcharge: 0 %
- Login Fee: 0
- Connect Fee: 0
- Round Charged Amount: XXXXX.XX000
- Formula: [empty]
- Short Description: SIP subscribers - incoming
- Description: What we charge our customers for incoming calls to their SIP phones

1. In the Management section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, choose **Add**.
3. Fill in the **Add Tariff** form:
 - o **Name** – A short name for the tariff object; this is the name you will then see in the select menus.
 - o **Currency** – Indicates in which currency pricing information is defined. All pricing information for a single tariff must be defined in the same currency.

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

- **Type** – Choose the **Ordinary** type, since this tariff will be applied to your subscribers.
 - **Off-peak Period** – Defines the off-peak period. Click on the Off-peak period wizard icon () to summon the wizard, which will help you construct the correct period definition. Click **Help** for more information on period format definition. If you do not differentiate between peak and off-peak rates, just leave this field empty.
 - **Off-peak Description** – A description of the off-peak period, automatically filled in by the off-peak period wizard; thus you do not have to fill in this field.
 - **Destination Group Set** – If you wish to enter rates in the tariff not for every individual prefix, but for a whole group of prefixes at once, you should create a destination group set and destination groups beforehand. Leave this select menu empty for now.
 - **Free Seconds** – The number of free seconds allowed for each call. In order to claim free seconds, the length of the call must be at least one billing unit (first interval; see the ‘Enter Rates’ section above).
 - **Post-Call Surcharge** – Percentage of the amount charged for the call.
 - **Login Fee** – Amount to be charged immediately after the first user authentication (i.e. after the user enters his PIN).
 - **Connect Fee** – Amount to be charged for each connected call (call with a non-zero duration).
 - **Round Charged Amount** – Instead of calculating CDRs with a 5-decimal-place precision, round up CDR amount values (e.g. to cents, so that 1.16730 becomes 1.17). Set the rounding pattern to XXXX.XX000 (as shown on the picture) so every call will be rounded to the equal cent amount.
 - **Formula** – Default rating formula, which will be applied to every rate created in the tariff. If you leave this empty, the “old-style” rating will be used.
 - **Short Description** – A short tariff description. This will be shown in the rate lookup on the admin interface and the self-care pages for your accounts and customers.
 - **Description** – An extended tariff description.
4. Click  **Save**.

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.

The interface shows a web application for managing tariff rates. The top bar includes navigation icons, the title "'SIP subscribers - incoming' tariff rates", the user location "America/Los_Angeles", the user name "demoroot", and a "Help" link. Below the title bar is a toolbar with "Add", "Save", "Close", and "Tariff" buttons. The main content area has a form for "Effective From" (set to "Now") and "Destination" (with fields for "Prefix", "Group", and "Country"). Below this is a table with columns: "Edit", "Destination", "Country", "Interval, sec." (with sub-columns "First" and "Next"), "Price, USD min" (with sub-columns "First" and "Next"), "Effective From" (with sub-columns "YYYY-MM-DD" and "HH24:MI:SS"), and "Delete".

In the bottom screenshot, the "Add" button is highlighted with a red solid arrow. The table contains one entry:

Edit	Destination *	Country	Interval, sec.		Price, USD min		Effective From		Delete
	Description		First *	Next *	First *	Next *	YYYY-MM-DD	HH24:MI:SS *	
	1604123	CANADA SIP phones	30	6	0.01	0.01	Immediately		


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

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



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

- **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.

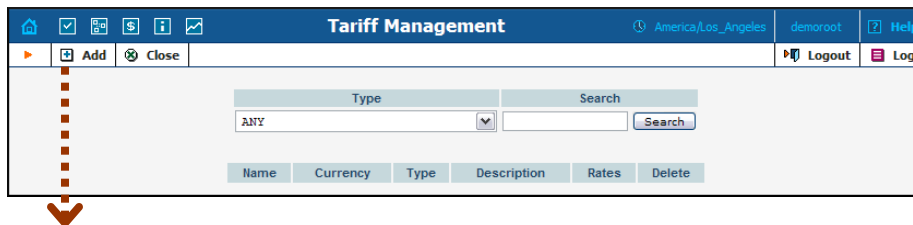
- **Formula**  – launches the wizard for creating a custom rating formula
- **Effective from** – If you want this rate to take effect sometime in the future, you can either type in a date manually, or use the calendar (click on the DD-MM-YYYY link).

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.

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.



Name	Currency	Type	Description	Rates	Delete
------	----------	------	-------------	-------	--------

Add Tariff

Name: DID supplier costs Currency: USD - US Dollar

Type: Ordinary

General Info

Off-Peak Period: []

Off-Peak Description: []

Destination Group Set: []

Free Seconds: 0

Post Call Surcharge: 0 %

Login Fee: 0

Connect Fee: 0

Round Charged Amount: XXXXX.XXXXX

Formula: []

Short Description: DID supplier

Description: What we are being charged by our DID supplier for incoming calls

1. In the Management section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, choose **Add**.
3. Fill in the **Add Tariff** form (please refer to the previous section for a detailed description of its elements). In the **Type** select menu choose **Ordinary**, 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.

'DID supplier costs' tariff rates

Effective From: Now Destination: []

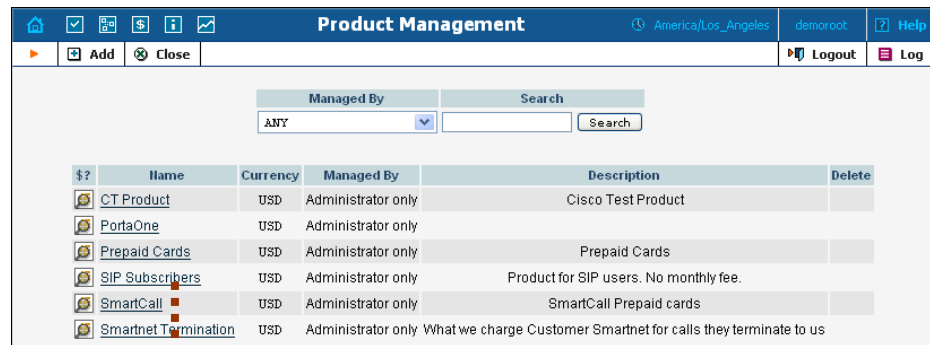
Edit	Destination *	Country	Interval, sec.		Price, USD/min		Effective From	Delete
			First *	Next *	First *	Next *		
	1604123	CANADA	60	60	0.005	0.005	immediately	

1. On the Tariff Management page you will see a list of the available tariffs. Click the **Rates** icon next to the name of the tariff. When you 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 the Accessibility for the Product

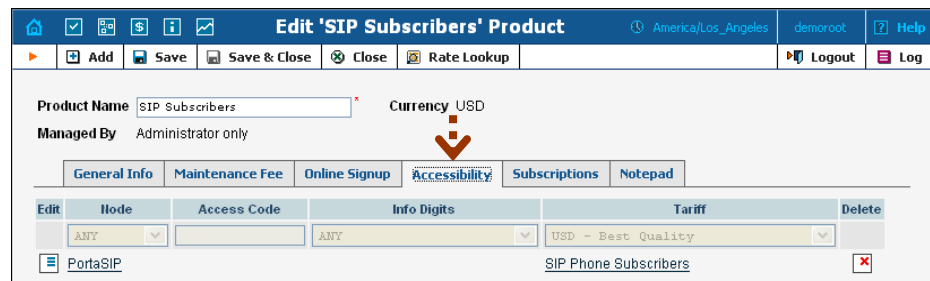
Now you must specify when the tariff you have created to charge your customers for incoming calls will actually be applied to them.



Product Management America/Los_Angeles demoroot Help

Managed By: ANY Search

Name	Currency	Managed By	Description	Delete
CT Product	USD	Administrator only	Cisco Test Product	
PortaOne	USD	Administrator only		
Prepaid Cards	USD	Administrator only	Prepaid Cards	
SIP Subscribers	USD	Administrator only	Product for SIP users. No monthly fee.	
SmartCall	USD	Administrator only	SmartCall Prepaid cards	
Smartnet Termination	USD	Administrator only	What we charge Customer Smartnet for calls they terminate to us	



Edit 'SIP Subscribers' Product America/Los_Angeles demoroot Help

Product Name: SIP Subscribers Currency: USD
Managed By: Administrator only

General Info Maintenance Fee Online Signup **Accessibility** Subscriptions Notepad

Edit	Icode	Access Code	Info Digits	Tariff	Delete
	ANY		ANY	USD - Best Quality	
	PortaSIP			SIP Phone Subscribers	X



Edit 'SIP Subscribers' Product America/Los_Angeles demoroot Help

Product Name: SIP Subscribers Currency: USD
Managed By: Administrator only

General Info Maintenance Fee Online Signup **Accessibility** Subscriptions Notepad

Edit	Icode	Access Code	Info Digits	Tariff	Delete
	NY-CM-01		ANY	USD - SIP subscribers - incoming	X
	PortaSIP			SIP Phone Subscribers	X

Edit	INode	Access Code	Info Digits	Tariff	Delete
	ANY		ANY	USD - Best Quality	
	NY-GW-01			SIP subscribers - incoming	X
	PortaSIP			SIP Phone Subscribers	X

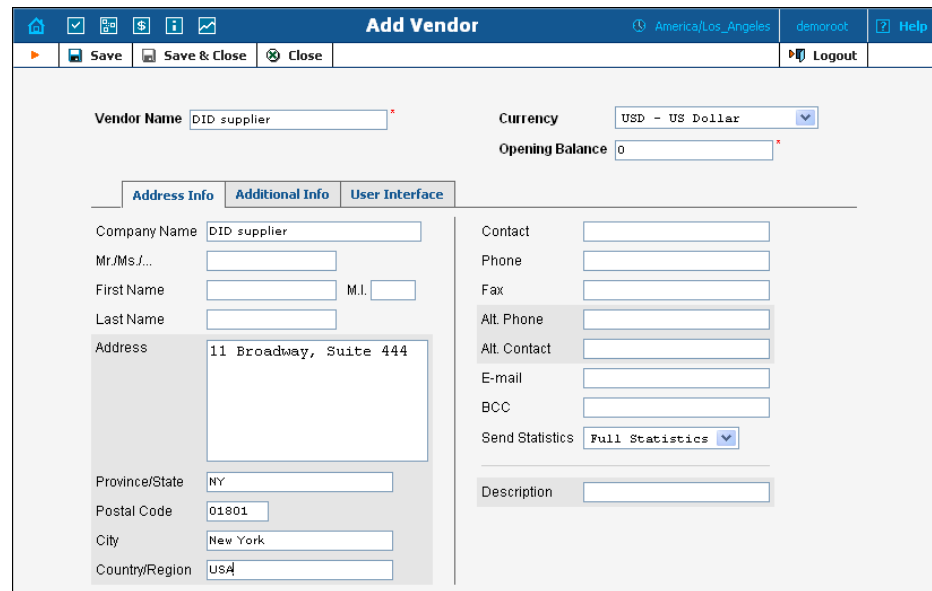
In the Management section of the Admin-Index page, choose **Products**.

1. Click on the product name to enter the Edit Product screen.
2. After selecting the Accessibility tab, click the **Add** icon.
3. In the Accessibility Entry window, select the PSTN gateway node and choose the tariff which applies to your customers for receiving incoming calls.
4. Normally, **Access Code** or **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 these empty.
5. Click **Save** 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 Management section of the admin interface, choose **Vendors**.
2. On the Vendor Management page, choose **Add**.



Add Vendor America/Los_Angeles demoroot Help

Save Save & Close Close Logout

Vendor Name DID supplier * Currency USD - US Dollar
Opening Balance 0 *

Address Info Additional Info User Interface

Company Name DID supplier Contact
Mr./Ms./...
First Name M.I.
Last Name
Address 11 Broadway, Suite 444
Province/State NY
Postal Code 01801
City New York
Country/Region USA
Phone
Fax
Alt. Phone
Alt. Contact
E-mail
BCC
Send Statistics Full Statistics
Description

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&Close** in order to save your data and return to the **Vendors** admin page.

Define Connections

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

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

CDRs	Name	Connections	Currency	Balance	E-mail	Delete
	Apollo-SIP		USD	12554.20369		
	Best VoIP		USD	0.00000		
	DID supplier		USD	0.00000		
	GlobalNet		USD	0.05102	info@globalnet.com	
	MCI Vendor		USD	18.58000		
	PortaVendor		USD	12513.82505		
	SPT Telecom		USD	0.00000	info@spt.cz	
	Test Vendor #1		USD	0.00000		
	X-Telecom		USD	0.00000	voip@x-telecom.com	

Vendor 'DID supplier' connections						
PSTN from Vendor						
Node	Port	CLD (dnis)	Info Digits	Transl. Rule	Capacity	Delete
NY-CW-01	ISDN 0%		AMY			
DID supplier cost	Incoming DID calls for SIP subscribers				24	

3. Choose the type of connection (**PSTN from Vendor**) by clicking on the corresponding tab.
4. Press **Add** to add a new connection.
5. Fill in the connection information. Choose your PSTN gateway as the **Node** and choose the tariff which defines your incoming DID costs for this vendor. If only one PSTN trunk is used for incoming DID calls, enter the port pattern in the **Port** column; otherwise just leave it empty. **Description** and **Capacity** are mandatory for all connection types.
6. Click **Save**.


Create Accounts

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

The first screenshot shows the 'Customer Management' screen. It has a table with columns: CDRs, Name, Accounts / Subcustomers, Currency, Type, Credit limit, Balance, E-mail, Status, and Delete. The table contains one row for 'EasyCall Ltd.' with a printer icon next to the name.

The second screenshot shows the 'Accounts of Retail Customer 'EasyCall Ltd.' screen. It has a table with columns: Account ID, Batch, Ctri #, and SIP Status. There is a 'Show Accounts' button and an 'Advanced Search' button.

The third screenshot shows the 'Add Account for Retail Customer 'EasyCall Ltd.' form. It has fields for: Account ID (16041234567), Product (USD - SIP Subscribers), Blocked (checkbox), Opening Balance (10), Type (radio buttons for Debit, Credit, Voucher), VoIP Password (scho7rtw), E-mail, and Batch (easycall). There are also buttons for Save, Save & Close, and Close.

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

- **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).

- **VoIP Password** – This password is used for SIP services as well. Together with the account ID, this password will be used to authenticate SIP server login.
- **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* chapter.

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

User Interface tab

- **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.

Call Features tab

- **Preferred Language** – This is a custom attribute which is transferred to the IVR. Leave English here if you are unsure whether your IVR supports this function.
- **Associated Number** – Redirect number. Leave this empty, since it is not used by PortaSIP.
- **UM Enabled** – Check this box if the account has unified messaging services (e.g. voicemail) enabled.

-
- **Follow Me Enabled** – Check this box if the account has the “Follow Me” feature enabled. If so, the account owner can define a list of numbers to which incoming calls to his UA will be redirected (for example his home phone, cell phone, and so on).
5. After clicking **Save&Close**, you will see a confirmation screen announcing that a new account has been created.

Incoming DID calls (from a VoIP vendor)

Checklist

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



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

Operation	Done
Initial configuration	
General and network configuration has already been performed according to the instructions in the <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.	[]
Rating configuration	
Create a tariff A, which will apply to SIP subscribers for their incoming calls.	[]
Enter rates in tariff A for the destinations covering your incoming DID range, i.e. the rates you plan to charge your customers.	[]
Create a tariff B, which describes the costs of incoming DIDs from this DID provider (it should be of the Ordinary type).	[]
Enter rates in tariff B for SIP destinations (zero cost).	[]
Create a new accessibility entry in your SIP product, using the PortaSIP node, tariff A and the access code INCOMING.	[]
Create a vendor for DID suppliers.	[]
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 <i>Basic SIP service</i> 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 a 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.

1. In the Management section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, choose **Add**.
3. Fill in the **Add Tariff** form:
 - o **Name** – A short name for the tariff object; this is the name you will then see in the select menus.
 - o **Currency** – Indicates in which currency pricing information is defined. All pricing information for a single tariff must be defined in the same currency.

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

- **Type** – Choose the **Ordinary** type, since this tariff will be applied to your subscribers.
 - **Off-peak Period** – Defines the off-peak period. Click on the Off-peak period wizard icon () to summon the wizard, which will help you construct the correct period definition. Click **Help** for more information on period format definition. If you do not differentiate between peak and off-peak rates, just leave this field empty.
 - **Off-peak Description** – A description of the off-peak period, automatically filled in by the off-peak period wizard; thus you do not have to fill in this field.
 - **Destination Group Set** – If you wish to enter rates in the tariff not for every individual prefix, but for a whole group of prefixes at once, you should create a destination group set and destination groups beforehand. Leave this select menu empty for now.
 - **Free Seconds** – The number of free seconds allowed for each call. In order to claim free seconds, the length of the call must be at least one billing unit (first interval; see the ‘Enter Rates’ section above).
 - **Post-Call Surcharge** – Percentage of the amount charged for the call.
 - **Login Fee** – Amount to be charged immediately after the first user authentication (i.e. after the user enters his PIN).
 - **Connect Fee** – Amount to be charged for each connected call (call with a non-zero duration).
 - **Round Charged Amount** – Instead of calculating CDRs with a 5-decimal-place precision, round up CDR amount values (e.g. to cents, so that 1.16730 becomes 1.17). Set the rounding pattern to XXXX.XX000 (as shown on the picture) so every call will be rounded to the equal cent amount.
 - **Formula** – Default rating formula, which will be applied to every rate created in the tariff. If you leave this empty, the “old-style” rating will be used.
 - **Short Description** – A short tariff description. This will be shown in the rate lookup on the admin interface and the self-care pages for your accounts and customers.
 - **Description** – An extended tariff description.
4. Click  **Save**.

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.

The top screenshot shows the 'Add' form for creating a new tariff rate. It includes fields for 'Effective From' (set to 'Now'), 'Destination' (with sub-fields for Prefix, Group, and Country), and a table for defining the rate structure.

Edit	Destination *	Country	Interval, sec.		Price, USD min		Effective From	Actions	Delete
			First *	Next *	First *	Next *			
							YYYY-MM-DD * HH24:MI:SS *		

The bottom screenshot shows the 'Edit Rates' screen for a specific tariff. It displays a table of existing rates for 'CANADA SIP phones'.

Edit	Destination *	Country	Interval, sec.		Price, USD min		Effective From	Actions	Delete
			First *	Next *	First *	Next *			
	1604123	CANADA SIP phones	30	6	0.01	0.01	Immediately		


6. On the Tariff Management page you will see a list of the available tariffs. Click the  **Rates** icon next to the name of the tariff. When you are in Tariff Management for a particular tariff, click on **Rates** in the toolbar.
7. On the **Edit Rates** screen, click  **Add**.
8. 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

- **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.

- **Formula**  – 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.
9. Click the  **Save** button in the toolbar, or the  icon on the left side of the row.
 10. 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.

The top screenshot shows the 'Tariff Management' page. It has a navigation bar with 'Add' and 'Close' buttons. Below the navigation bar is a search area with a 'Type' dropdown set to 'ANY' and a 'Search' button. Below the search area is a table with columns: Name, Currency, Type, Description, Rates, and Delete.

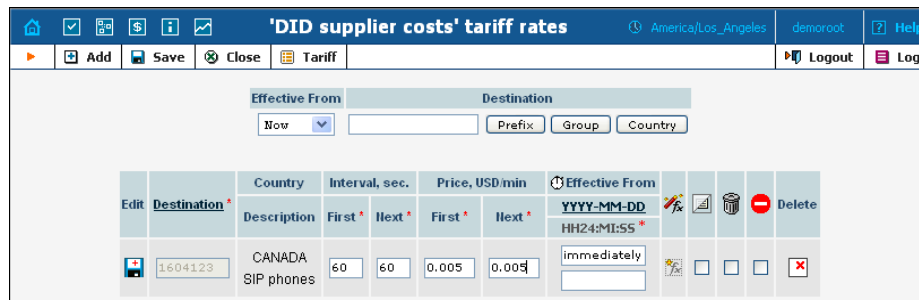
The bottom screenshot shows the 'Add Tariff' form. It has a navigation bar with 'Save', 'Save & Close', and 'Close' buttons. Below the navigation bar are the following fields:





- Name: DID supplier costs
- Currency: USD - US Dollar
- Type: Ordinary
- General Info tab
- Off-Peak Period: [empty]
- Off-Peak Description: [empty text area]
- Destination Group Set: [empty dropdown]
- Free Seconds: 0
- Post Call Surcharge: 0 %
- Login Fee: 0
- Connect Fee: 0
- Round Charged Amount: XXXXX.XXXXX
- Formula: [empty]
- Short Description: DID supplier
- Description: What we are being charged by our DID supplier for incoming calls

1. In the Management section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, choose **Add**.
3. Fill in the **Add Tariff** form (please refer to the previous section for a detailed description of its elements). In the **Type** select menu choose **Ordinary**, 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

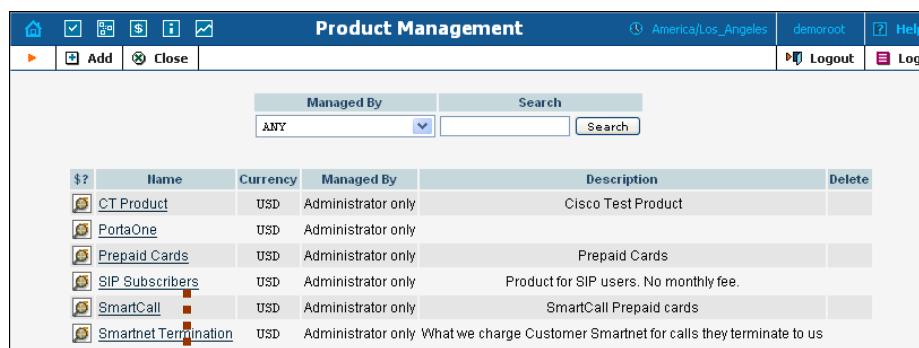
Enter the rates applied to you by the DID provider.



1. On the Tariff Management page you will see a list of the available tariffs. Click the  **Rates** icon next to the name of the tariff. When you have entered Tariff Management for a particular tariff, click on **Rates** in the toolbar.
2. On the **Edit Rates** screen, click  **Add**.
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 the Accessibility for the Product

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



Product Name: SIP Subscribers Currency: USD
Managed By: Administrator only

General Info Maintenance Fee Online Signup **Accessibility** Subscriptions Notepad

Edit	Node	Access Code	Info Digits	Tariff	Delete
	ANY		ANY	USD - Best Quality	
PortaSIP SIP Phone Subscribers					

Product Name: SIP Subscribers Currency: USD
Managed By: Administrator only

General Info Maintenance Fee Online Signup **Accessibility** Subscriptions Notepad

Edit	Node	Access Code	Info Digits	Tariff	Delete
	ANY		ANY	USD - Best Quality	
	PortaSIP	INCOMING		USD - SIP subscribers - incoming	
PortaSIP SIP Phone Subscribers					

Product Name: SIP Subscribers Currency: USD
Managed By: Administrator only

General Info Maintenance Fee Online Signup **Accessibility** Subscriptions Notepad

Edit	Node	Access Code	Info Digits	Tariff	Delete
	ANY		ANY	USD - Best Quality	
	PortaSIP	INCOMING		SIP subscribers - incoming	
PortaSIP SIP Phone Subscribers					

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

1. Click on the product name to enter the Edit Product screen.
2. After selecting the Accessibility tab, click the **Add** icon.
3. In the Accessibility Entry window, select the PortaSIP node and choose the tariff which applies to your customers for receiving incoming calls.
4. Type **INCOMING** in the **Access Code** column.
5. **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.
6. Click **Save** 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 Management section of the admin interface, choose **Vendors**.

2. On the Vendor Management page, choose **Add**.

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

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



- **Login** – Choose a username to be used by the vendor's proxy or gateway to access your network. If the vendor does not support digest authentication, and you are using IP-based authentication, enter the IP address of the vendor's gateway here and use **cisco** as the password.
 - **Password** – Choose a secure password (difficult to guess or crack) for the vendor's account.
8. Click the **Save** button in the toolbar, or the icon on the left end of the row.
 9. Click **Close** in order to return to the **Vendors** admin page.

Define Connections


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

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

CDRs	Name	Connections	Currency	Balance	E-mail	Delete
	Apollo-SIP		USD	12554.20369		
	Best VoIP		USD	0.00000		
	DID supplier		USD	0.00000		
	GlobalNet		USD	0.05102	info@globalnet.com	
	MCI Vendor		USD	18.58000		
	PortaVendor		USD	12513.82505		
	SPT Telecom		USD	0.00000	info@spt.cz	
	Test Vendor #1		USD	0.00000		
	X-Telecom		USD	0.00000	voip@x-telecom.com	

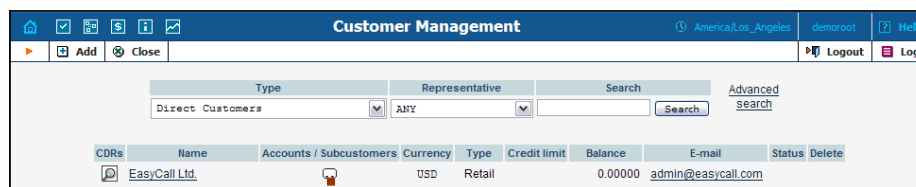
Vendor 'DID supplier' connections						
PSTN from Vendor						
VoIP from Vendor						
PSTN to Vendor						
VoIP to Vendor						
Edit	Load	Node	RTP Proxying	Transl. Rule	Account	Delete
		Tariff		Description *	Capacity *	
		PortaSIP	Optimal		DID supplier -	
		DID supplier cost:	Incoming DIDs from supplier	6d		


3. Choose the type of connection (**VoIP from Vendor**) by clicking on the corresponding tab.
4. Press **Add** to add a new connection.

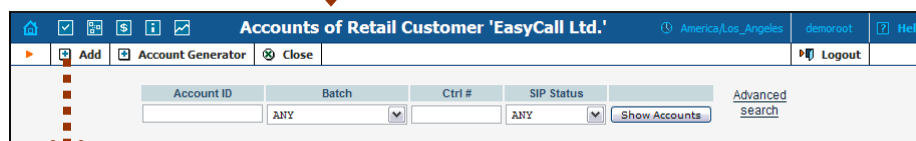
5. Fill in the connection information. Choose your PortaSIP node as the **Node** and choose the tariff which defines your incoming DID costs for this vendor. In the select menu in the **Account** column choose the account you have previously created in the vendor configuration, this account will be used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

Create Accounts

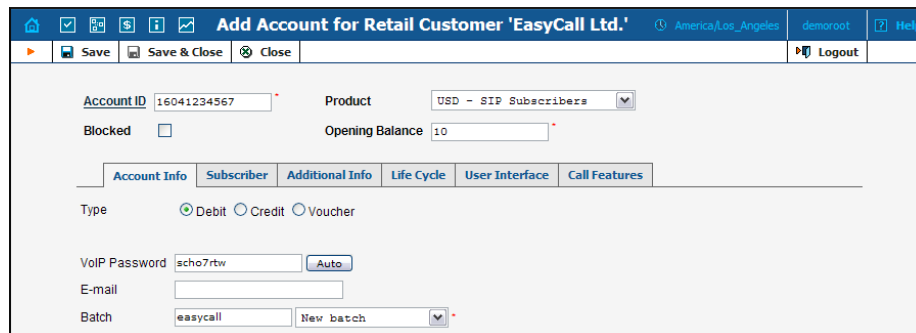
1. Go to the **Customers** screen (the one containing a list of customers). It should resemble the screenshot below:



CDRs	Name	Accounts / Subcustomers	Currency	Type	Credit limit	Balance	E-mail	Status	Delete
	EasyCall Ltd.		USD	Retail		0.00000	admin@easycall.com		



Account ID	Batch	Ctr #	SIP Status	Advanced search
	ANY		ANY	Show Accounts



Account ID: 16041234567 Product: USD - SIP Subscribers



Blocked: Opening Balance: 10

Type: Debit Credit Voucher

VoIP Password: scho7rtw Auto

E-mail: _____

Batch: easycall New batch

2. Next to the customer name, click the  icon (the one in the **Accounts** column) to go to account management for that customer.
3. Click on  **Add**.
4. Fill in the Add Account form:
 - o **Account ID** – SIP ID, i.e. the phone number which will be used to login to the SIP server and receive incoming calls.
 - o **Product** – Choose the product you would like your account to have.

- **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

- **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).
- **VoIP Password** – This password is used for SIP services as well. Together with the account ID, this password will be used to authenticate SIP server login.
- **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* chapter.

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

User Interface tab

- **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.

Call Features tab

- **Preferred IVR Language** – This is a custom attribute which is transferred to the IVR. Leave English here if you are unsure whether your IVR supports this function.
 - **Associated Number** – Redirect number. Leave this empty, since it is not used by PortaSIP.
 - **UM Enabled** – Check this box if the account has unified messaging services (e.g. voicemail) enabled.
 - **Follow Me Enabled** – Check this box if the account has the “Follow Me” feature enabled. If so, the account owner can define a list of numbers to which incoming calls to his UA will be redirected (for example his home phone, cell phone, and so on).
5. After clicking  **Save&Close**, you will see a confirmation screen announcing that a new account has been created.

3. Setting up Auto-provisioning of IP Phones

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

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

Setting up Auto-provisioning of IP Phones

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

Checklist

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

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

Create an IP Phone Profile

The top screenshot shows the 'IP Phone Profiles' management window. It has a search bar with 'Effective From' set to 'Now-->', 'Name' empty, 'Type' set to 'ANY', and 'Managed By' set to 'ANY'. A red dashed arrow points to the 'Add' icon in the top toolbar.

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', and Time: empty.

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

IP Phone Profile Settings

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

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

Field	Description
Profile Rule	<p>Path to the profile configuration file. Replace PB_SLAVE_SERVER with the actual hostname or IP address of your PortaBilling100 slave server; for instance, the actual value in the field might be: [--key \$B]http://demo.portaone.com/\$A/\$MA.cfg. The example on the screen assumes provisioning via the HTTP protocol; change http: to tftp: for TFTP provisioning.</p> <p>When an individual IP phone will make a request for a configuration file, \$MA will be automatically replaced by it's MAC-address, so leave it as is (otherwise if you put the actual MAC address in,</p>

	you cannot reuse the same profile for multiple phones).
GPP A, GPP B, GPP C	Dynamic variables. During config file generation, <code>\$_env</code> will be replaced by a unique environment ID, and <code>\$ascii_key</code> by the crypt key from the IP phone.

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

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

Field	Description
Proxy	SIP server for registration and placing outgoing calls.
Outbound Proxy	SIP Server for placing outgoing calls (if different from the SIP registration server).
Use Outbound Proxy	If set to “no”, the SIP server defined by the Proxy parameter will be used for all registrations and outgoing calls.
Register	Whether or not the IP phone should register with the SIP server; this is required to receive incoming calls.
Register Expires	Registration lifetime in seconds (the Expires value in a REGISTER request). The IP phone will periodically renew registration shortly before the current registration expires.

Display Name	The subscriber's display name, which appears in the caller-id for his outgoing calls; by default First Name + M Name + Last Name.
User ID	The phone number assigned to this port. The dynamic variable \$id will be automatically replaced by the account's ID (phone number).
Auth ID	Authentication ID (if different from the phone number).
Use Auth ID	If set to "yes", the combination Auth ID and Password will be used for SIP authentication. Otherwise, User ID and Password are used.
Preferred Codec	Select a preferred codec for all calls. However, the actual codec used in a call will still depend on the outcome of the codec negotiation protocol. Possible values: G711u, G711a, G726-16, G726-24, G726-32, G726-40, G729a, G723.
Use Pref Codec Only	Only use the preferred codec for all calls. The call will fail if the other endpoint does not support this codec.
DTMF Tx Method	Method for transmitting DTMF signals to the far end: Inband = Send DTMF using audio path; INFO = Use the SIP INFO method, AVT = Send DTMF as AVT events; Auto = Use Inband or AVT based on outcome of codec negotiation.
Dial Plan	Per-line dial plan script; see the product manual for a detailed description of the syntax.

Create an IP Phone Entry

The screenshot shows the 'IP Phone Inventory' page. At the top, there are navigation icons and the page title. Below the title, there are buttons for 'Add' and 'Close'. A search bar is present with a 'Type' dropdown set to 'ANY' and a 'Search' button. Below the search bar is a table with the following columns: Name, Type, Managed By, MAC Address, Description, Ports (sub-columns: free, total), Inventory ID, and Delete.

The screenshot shows the 'Add IP Phone' page. It features a form with the following fields: Name (Sipura), Description (Sipura Phone), Managed by (Administrator only), Type (Sipura 2000), Profile (Sipura - Standard), MAC Address (00:0E:08:AB:D7:A8), Ports (Total: 2, Free: 2), ASCII Key, and Inventory ID. There are also buttons for 'Save', 'Save & Close', and 'Close'.

Name	Type	Managed By	MAC Address	Description	Ports		Inventory ID	Delete
					free	total		
Sipura	Sipura 2000	Administrator only	0:E:8:AB:D7:A8	Sipura Phone	2	2		

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

Provisioning an Account on an IP Phone

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

Account ID: 16041234570 Product: USD - SIP Subscribers

Blocked: Balance: 0.00000 USD

User Agent: _____ Contact: _____

Account Info | Maintenance | Subscriber | **Additional Info** | Life Cycle | User Interface | Call Features | Subscriptions | Notepad

IP Phone: Sipura

IP Phone Port: 1

E-commerce Enabled:

Discount Plan: Product default

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

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

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

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

IP Phone Device Configuration

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

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



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: Resync Periodic:

Resync Error Retry Delay: Forced Resync Delay:

Resync From SIP: yes Resync After Upgrade Attempt: yes

Resync Trigger 1:

Resync Trigger 2:

Resync Fails On FNF: yes

Profile Rule:

Profile Rule B:

Profile Rule C:

Profile Rule D:

Log Resync Request Msg:

Log Resync Success Msg:

Log Resync Failure Msg:

Report Rule:

Firmware Upgrade

Upgrade Enable: yes Upgrade Error Retry Delay:

Downgrade Rev Limit:

Upgrade Rule:

Log Upgrade Request Msg:

Log Upgrade Success Msg:

Log Upgrade Failure Msg:

General Purpose Parameters

GPP A:

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.

4. Appendices

APPENDIX A. Clients' Cisco ATA 186 Configuration for PortaSIP

UID0	[CLIENT'S ACCOUNT ID (PHONE NUMBER) 1]
PWD0	[CLIENT'S PASSWORD FOR ACCOUNT ID 1]
UID1	[CLIENT'S ACCOUNT ID (PHONE NUMBER) 2]
PWD1	[CLIENT'S PASSWORD FOR ACCOUNT ID 2]
GkOrProxy	[IP ADDRESS OF SERVER RUNNING PORTASIP]
Gateway	0.0.0.0
GateWay2	0.0.0.0
UseLoginID	0
LoginID0	0
LoginID1	0
AltGK	0.0.0.0
AltGKTimeOut	0
GkTimeToLive	300
GkId	.
UseSIP	1
SIPRegInterval	180
MaxRedirect	5
SIPRegOn	1
NATIP	0.0.0.0
SIPPort	5060
MediaPort	[DIFFERENT FOR EACH CLIENT AS DESCRIBED IN THE SETUP GUIDELINES]
OutBoundProxy	0.0.0.0
NatServer	[IP ADDRESS OF SERVER RUNNING PORTASIP]
NatTimer	0x1e
LBRCodec	0
AudioMode	0x00150015
RxCodec	0
TxCodec	0
NumTxFrames	1
CallFeatures	0xffffffff
PaidFeatures	0xffffffff
CallerIdMethod	0xc0019e60
FeatureTimer	0
Polarity	0
ConnectMode	0xe0400
AuthMethod	0
TimeZone	[SEE CISCO ATA 186 DOCUMENTATION FOR ENTERING CORRECT VALUE]

NTPIP	192.43.244.18
AltNTPIP	131.188.3.222
DNS1IP	0.0.0.0
DNS2IP	0.0.0.0
UDPTOS	0xb8
SigTimer	0x64
OpFlags	0x62
VLANSettings	0x2b
NPrintf	0.0.0.0
TraceFlags	0

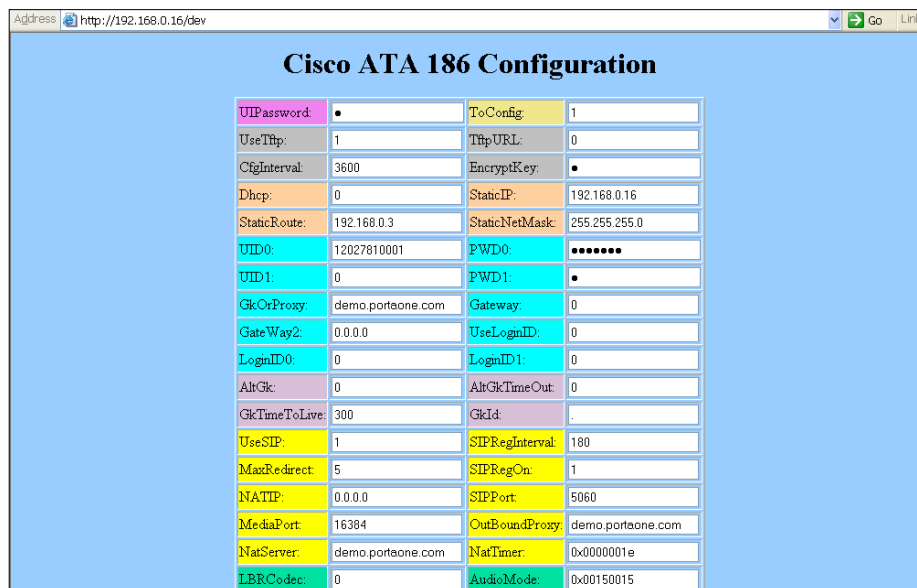
The manufacturer's default values are assumed for all options not listed here.

APPENDIX B. Configure Cisco ATA Using ATA Expert

Cisco ATA could be configured from the web interface accessible at <http://<ata-IP-address>/dev>. However, this web interface is designed to be used by experts, and parameter values must be entered in the protocol-specific format (e.g. 0x00150015). You may find more information at:

http://www.cisco.com/en/US/products/hw/gatecont/ps514/products_configuration_example09186a00800c3a50.shtml

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




Cisco ATA 186 Configuration	
UIPassword	ToConfig
UseTtp	TtpURL
CfgInterval	EncrypKey
Dhcp	StaticIP
StaticRoute	StaticNetMask
UID0	PWD0
UID1	PWD1
GkOrProxy	Gateway
GateWay2	UseLoginID
LoginID0	LoginID1
AltGk	AltGkTimeOut
GkTimeToLive	GkId
UseSIP	SIPRegInterval
MaxRedirect	SIPRegOn
NATIP	SIPPort
MediaPort	OutBoundProxy
NatServer	NatTimer
LBRCodec	AudioMode

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

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

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

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

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

APPENDIX C. 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
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Sipura Phone Adapter Configuration

[Info](#) | [System](#) | [SIP](#) | [Provisioning](#) | [Regional](#) | [Line 1](#) | [Line 2](#) | [User 1](#) | [User 2](#) | [User Login](#) | [basic](#) | [advanced](#)

System Information

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

Product Information

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

System Status

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

Line 1 Status

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

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

APPENDIX D. 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.

Service: PortaOne

Please enter this information to initialize the service profile

Account: 123456789

Password:

Save service information permanently

OK Cancel Help

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.

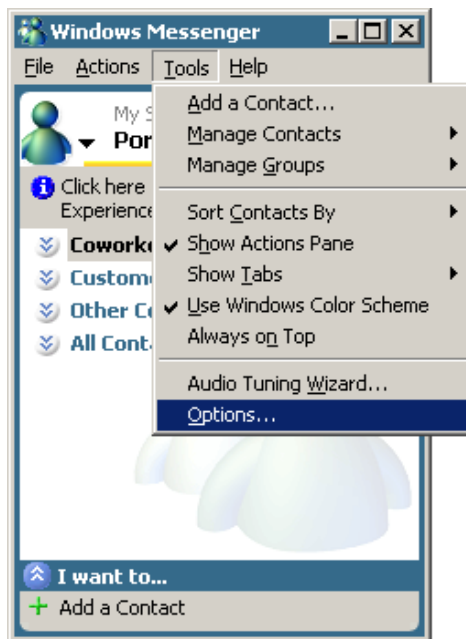


3. Right click on the softphone and press “Login...” to change or make corrections to the account/password.

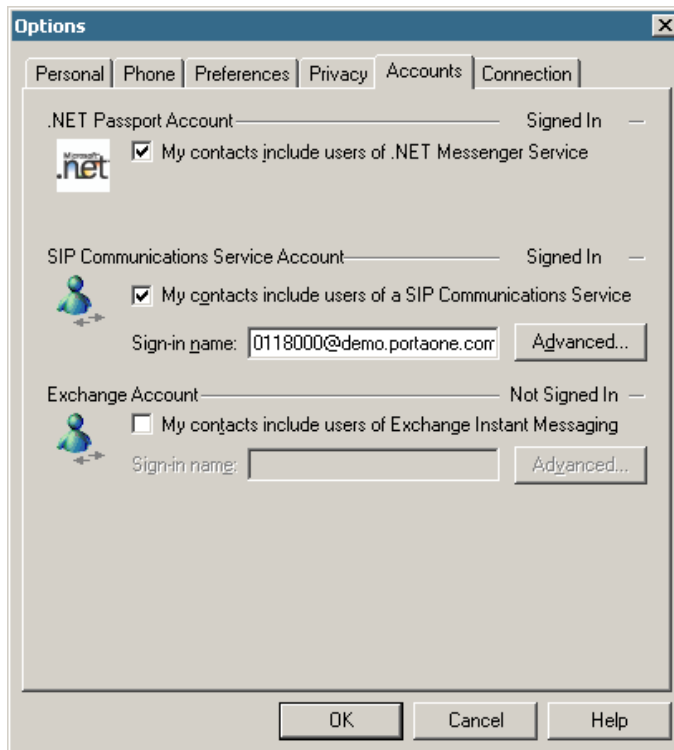
APPENDIX E. Configuring Windows Messenger for Use as a SIP User Agent

The following instructions apply to Windows Messenger version 5.0.

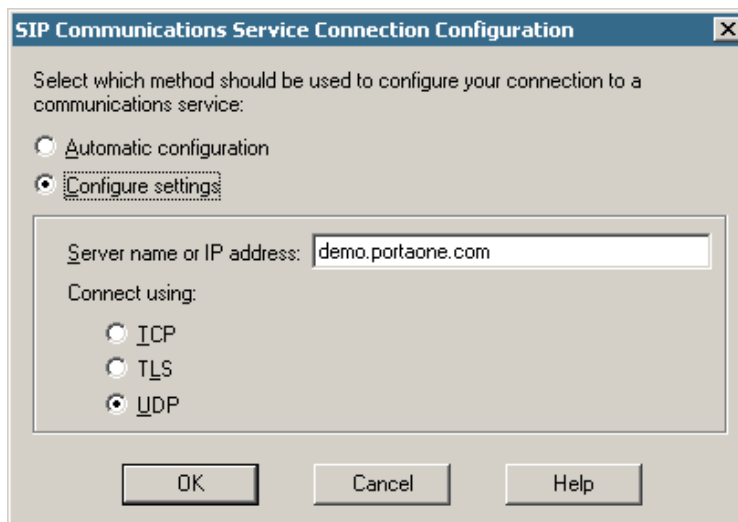
1. Start Windows Messenger, and select “Options...” from the “Tools” menu



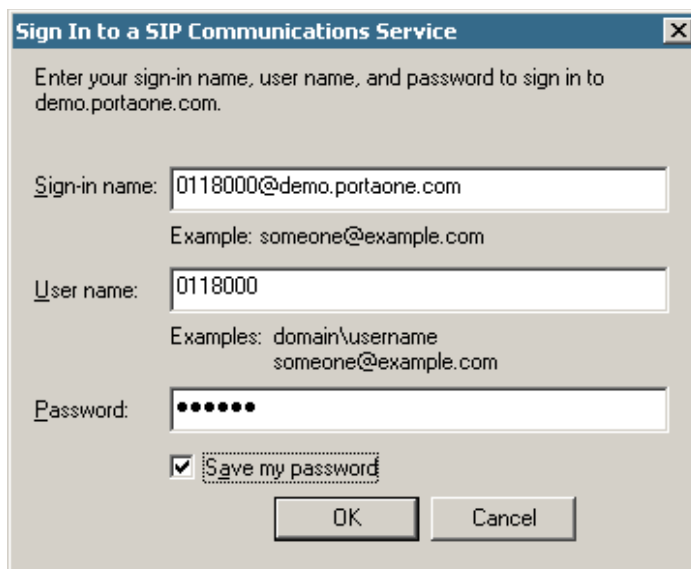
2. Check the “My contacts include users of a SIP Communication Service” check box. Enter your “Sign-in name” as shown, in the form *username@address*, where *username* is the name of the appropriate account in PB and *address* is either the IP address of the PortaSIP server or its name in DNS. Then click the “Advanced...” button.



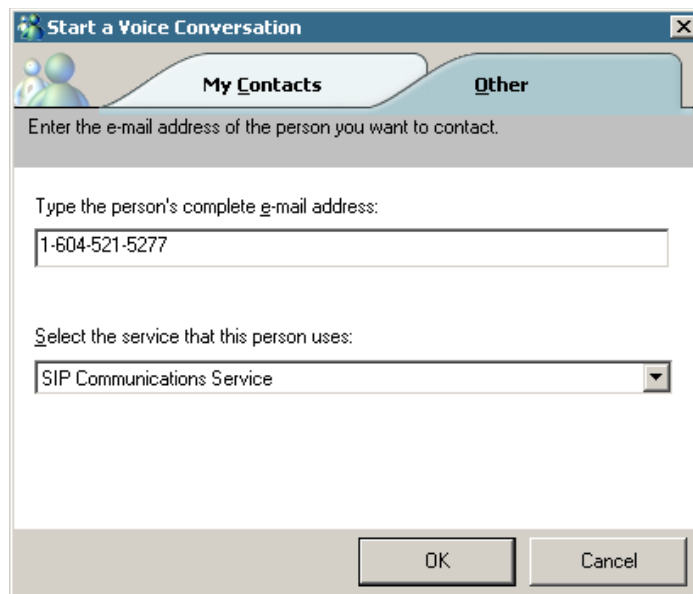
- Click the “Configure settings” radio button and enter the “Server name or IP address” using either the IP address of the PortaSIP server or its name in DNS. Make sure that the “UDP” radio button is selected, then click OK.



- 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.



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



APPENDIX F. Auto-provisioned IP Phones and Adapters

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

- Cisco ATA 186
- Grandstream HT-486
- Grandstream HT-488
- Grandstream HT-496
- Linksys PAP2
- Linksys WRT54GP2
- Sipura 1001
- Sipura 2000
- Sipura 2002
- Sipura 3000

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.