PORTA ONE

Porta █ Switch ®

Handbook:

IP Centrex and SIP Trunking Services
Maintenance Release 23

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IP Centrex and SIP Trunking Services, June 2011
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Preface

This document provides PortaBilling® users with step-by-step instructions for configuring IP Centrex and SIP Trunking services.

Where to get the latest version of this guide

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

Conventions

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

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

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

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

Tips provide information that might help you solve a problem.

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1. IP Centrex

This chapter demonstrates how to configure IP Centrex services so that your customers will be able to communicate within the IP Centrex environment. It will also assist you in creating phone lines, adding extensions and huntgroups, and finally configuring different call forwarding scenarios.

Please refer to the *IP Centrex Features* section in the **PortaSIP Administrator Guide** to read more about hosted IP PBX features available in PortaSwitch®.

Checklist

Print the following page and use it to check off the operations you have completed while performing configuration 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
Initial configuration	
Make sure configuration of general and network settings has already been performed according to the instructions in the *Basic SIP Service* section in the **PortaSwitch Handbook: Residential VoIP Services**	[ ]
Check that basic SIP service configuration has already been performed according to the instructions in the *Basic SIP Service* section in the **PortaSwitch Handbook: Residential VoIP Services**	[ ]
Rating configuration (Vendor)	
Make sure a "VoIP to Vendor" connection with the Remote Gateway ID field containing the **SIP-UA** string is created for the internal Vendor	[ ]
Rating configuration (Customer)	
Add VOICEONNETRX and INCOMING destinations	[ ]
Create a tariff (referred to below as **tariff A**) which will be used to calculate the charges for outgoing calls from the customer's IP Centrex environment	[ ]
Enter rates in **tariff A**, covering the destinations where the accounts can call. A zero cost rate for the special VOICEONNETRX destination should also be added to this tariff	[ ]
Create a tariff (referred to below as **tariff B**), which will be used to calculate the charges for incoming calls	[ ]
Enter a zero cost rate for the special INCOMING destination in **tariff B**. This rate will be used to calculate the charges for incoming calls	[ ]
Create an IP Centrex product	[ ]
Create a rating entry for this product, associating the **Voice Calls** service and **tariff A** with the PortaSIP node and ANY **Access Code**	[ ]
Create one more rating entry for this product, associating the **Voice Calls** service and **tariff B** with the PortaSIP node and INCOMING **Access Code**	[ ]
Account provisioning	
Create a retail customer who will use the IP Centrex service	[ ]
Upload phone numbers	[ ]
Create phone lines	[ ]
Add new extensions	[ ]
Create huntgroups	[ ]

Create Destinations

This step is only required if you have not previously defined the necessary destinations.

1. In the **Rating** section of Admin-Index, choose **Destinations**.
2. Click on the ⊞ **Add** button.
3. Enter a **VOICEONNETRX** destination in order for a special rating to be applied to all calls going between accounts of the same customer (extensions within the same IP Centrex environment).
4. Enter an **INCOMING** destination in order for a special rating to be applied to all incoming calls.

5. Click **Save**.

Create Customer Tariffs

To provide IP Centrex services successfully, you should create two customer tariffs. These tariffs will be used for separate charging of outgoing and incoming calls.

Follow the steps for creating customer tariffs below:

1. In the **Rating** section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, choose ⊞ **Add**.
3. Fill in the **Add Tariff** form:
   o **Name** – Type a short name for the tariff object; this is the name you will then see in the select menus.
   o **Currency** – Select the currency in which you will charge your customers.
   o **Applied To** – Choose **Customer** here.
   o **Managed By** – Select **Administrator Only** here.
   o **Service Type** – Choose **Voice Calls** here.

Please refer to the *Basic Residential VoIP Service* chapter in the **PortaSwitch Handbook: Residential VoIP Services** for a detailed description of the available tariff parameters.
4. Click 🖫 **Save**.
5. Repeat steps 1-4 until you have entered all of the tariffs.

Enter Rates

You should specify the rates for the two tariffs created in the previous section.

- Rates for the first tariff should cover the destinations where the accounts of one customer can call. These are used to calculate the charges for outgoing calls. A zero rate for the special VOICEONNETRX destination should also be added to this
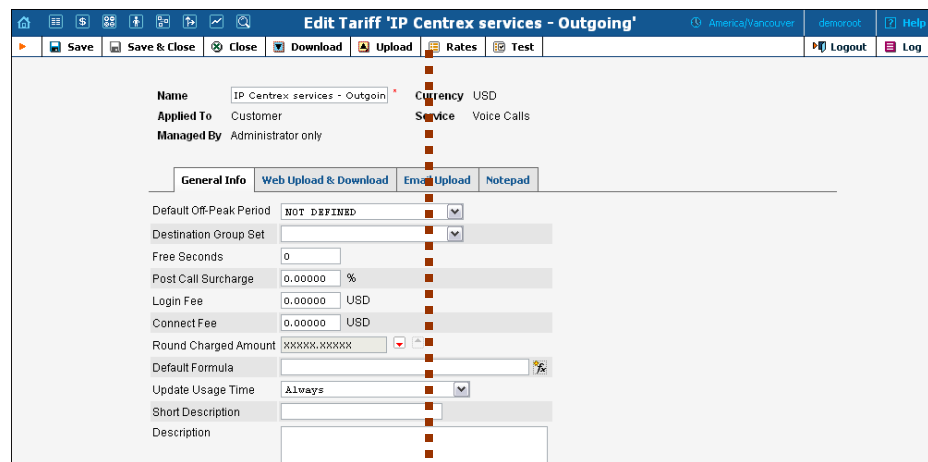
tariff. This rate will be used to produce zero charges for outgoing calls between accounts within the same IP Centrex environment.

- For the second tariff, only a zero rate for the INCOMING destination should be added. This rate will be used to produce zero charges for all incoming calls.

Managing Rates Online

1. On the Tariff Management page you will see a list of the available tariffs. Click the ▦ **Rates** icon in front of the name of the tariff. When you are in Tariff Management for a particular tariff, click on **Rates** in the toolbar.
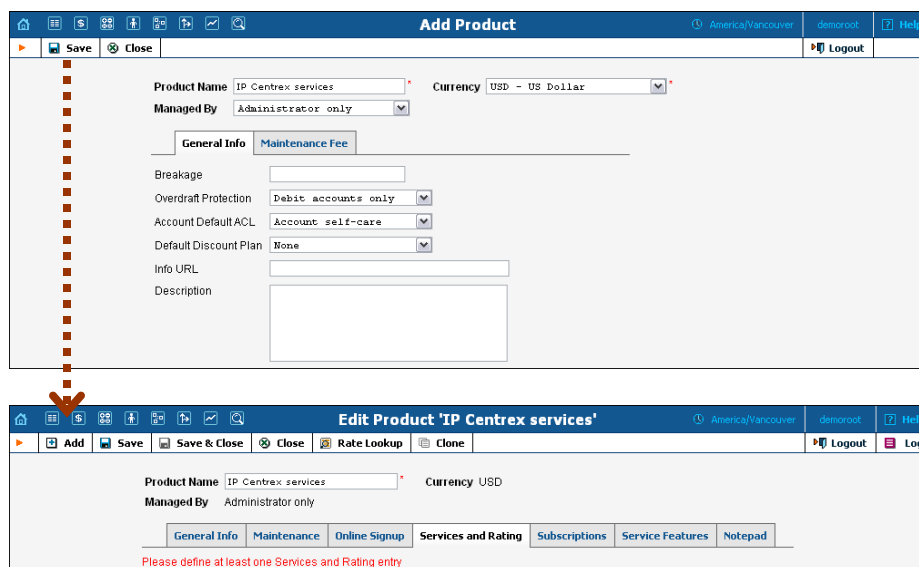
2. On the **Rates** screen, click ⊞ **Add**.
3. Insert VOICEONNETRX in the **Destination** field. Set the **First Price** and **Next Price** fields to zero if you are not going to charge for calls between extensions in the same IP Centrex environment. Leave the other fields as they are.
4. Click the 💾 **Save** button in the toolbar, or the 💾 icon on the left side of the row.
5. Go to the Tariff Management page and click the 📋 **Rates** icon in front of the **IP Centrex services – Incoming** tariff.
6. On the **Rates** screen, click ⊞ **Add**.
7. Insert INCOMING in the **Destination** field. Set the **First Price** and **Next Price** fields to zero if you are not going to charge customers for incoming calls. Leave the other fields as they are.
8. Click the 💾 icon on the left side of the row.

Create a Product

In the **Rating** section of the Admin-Index page, choose **Products**.
1. On the Product Management page, click the ⊞ **Add** icon.
2. Fill in the "Add Product" form:
    o **Product Name** – Type the product name.
    o **Currency** – Select the currency in which you will charge your IP Centrex customers.
    o **Managed by** – Choose **Administrator Only** here.

Please refer to the *Basic Residential VoIP Service* chapter in the **PortaSwitch Handbook: Residential VoIP Services** for a detailed description of the available fields.
3. Click ⊟ **Save**.
4. Click on the **Services and Rating** tab to edit this product's rating list.

Create Rating Entries for the IP Centrex Product

1. When the **Services and Rating** tab is selected, click on the ⊞ **Add** icon.
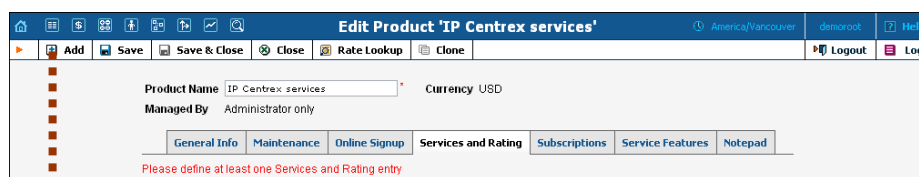
2. Fill in the required information:
    o **Service** – Choose "**Voice Calls**".
    o **Node** – Select the appropriate PortaSIP® node.
    o **Access Code** – Leave the **Access Code** field empty.
    o **Tariff** - Select the tariff which will be used to calculate the charges for outgoing calls.

3. Click 🖬 **Save&Close** to save this rating entry.
4. On the **Services and Rating** tab, click the 🔛 **Add** icon to add the second rating entry.

5. Fill in the required information:
   o **Service** – Choose "**Voice Calls**".
   o **Node** – Select the appropriate PortaSIP® node..
   o **Access Code** – Type INCOMING in this field.
   o **Tariff** - Select the tariff which will be used to calculate the charges for incoming calls.
6. Click 🖬 **Save&Close** to save this rating entry.

Create a Customer

⚠️   Create a customer according to the instructions in the *Basic Residential VoIP Service* chapter in **PortaSwitch Handbook: Residential VoIP Services**.

Upload Phone Numbers in DID Inventory

Perform the configuration of the DID Inventory according to the instructions provided in the *Managing Available Phone Numbers (DID Inventory)* chapter in **PortaSwitch Handbook: Residential VoIP Services**.

Create Phone Lines

A phone line is a PortaSwitch account provisioned on a port on an IP phone / ATA so that it allows the sending and receiving of calls via PortaSwitch.

1. In the **Participants** section of the Admin-Index page, choose **Customers.**
2. Next to the created customer, click on the 🗗 icon (the one in the **Accounts** column) to go to the account management for that customer.
3. Click on the ➕ **Add** button.
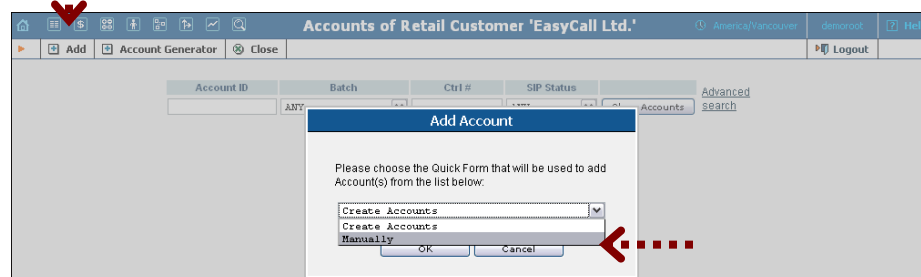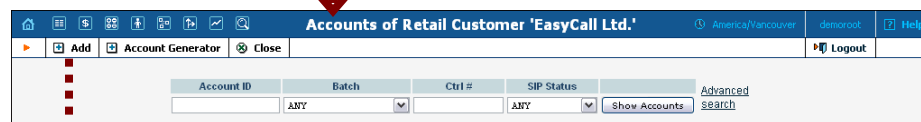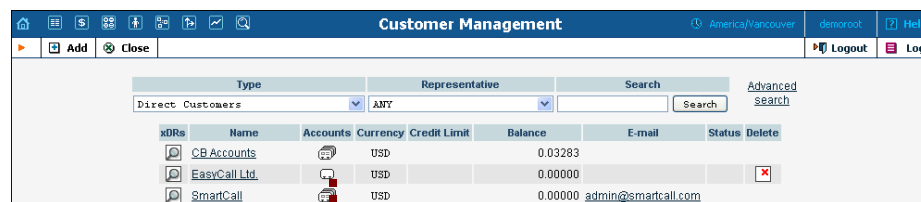
4. In this example, we assume that you are creating the account manually, so choose **Manually** from the drop-down menu (do not use any of the Quick Form options).
5. Fill in the "Add Account" form:
   o **Account ID** – Click on the Account ID link, and in the pop-up window choose one of the available DID numbers.
   o **Product** – Choose the product created in the previous step.
   o **Blocked** – Leave unchecked for IP Centrex service accounts.

o **Opening balance** – The initial account's balance.

Account Info tab

- o **Type** – Accounts for IP Centrex services are usually of the Credit type.
- o **Credit limit** – For a credit account, specify the credit limit. If you leave this field empty, it means there is no credit limit for this account (but a customer credit limit may still apply).
- o **Batch** – Accounts are grouped into batches. If a new batch name is provided in the text field, the created account will be placed in a new batch with the given name. If you leave this field empty, the created account will not be tied to any specific batch.

Please consult the **PortaBilling Web Reference Guide** for a detailed description of other available fields.
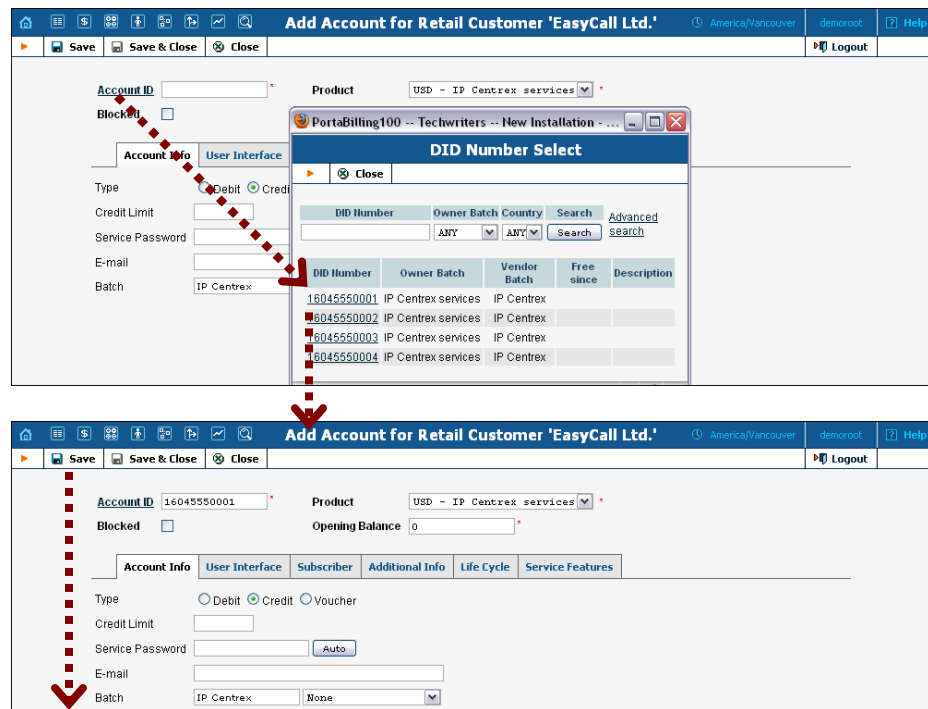
6. Click 🖫 **Save**. You will stay in Add Account mode, and can easily add another account.
7. Repeat steps 5-6 until you have created all the required accounts.

To create many accounts at once and allocate proper DID numbers to all of them, use the **Account Generator**. Please refer to the *Managing Available Phone Numbers (DID Inventory)* chapter in the **PortaSwitch Handbook: Residential VoIP Services** for detailed instructions on how to do this.
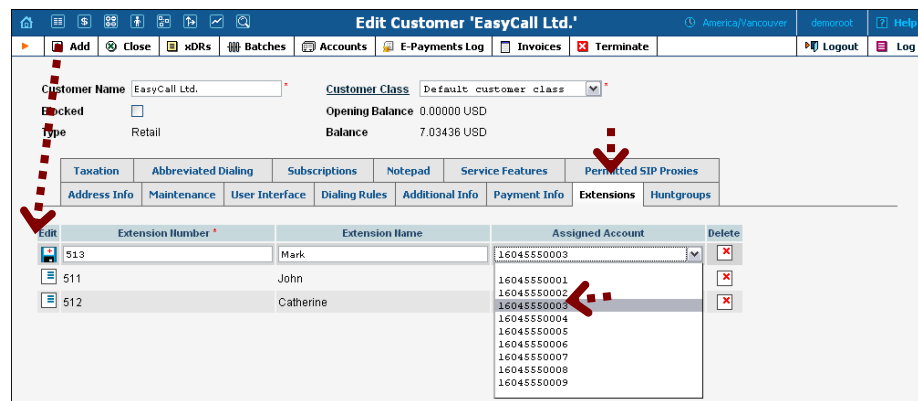
Add New Extensions

Once accounts have been created, they can be assigned to individual **extensions**. Proceed as follows to create new extensions:

1. In the **Participants** section of the Admin-Index page, choose **Customers**.
2. Click on the customer name to enter the Edit Customer screen.

3. Go to the **Extensions** tab.
4. Click ⊞ **Add** to add new extensions.

5. Enter the following information:
   - **Extension Number** – The number the end-user will dial on his phone (note that an extension number should contain only digits).
   - **Extension Name** – Logical name for this extension.
   - **Assigned Account** – The number the extension is associated with. Here you should specify the ID of one of the customer's accounts. Select the account from the drop-down list. (Note that each account from the list can be used only once. Thus, it is not possible to have several extensions associated with the same assigned account.)

6. Click the 🔲 **Save** icon on the left side of the row.

Add Huntgroups

Sometimes there is a need for a call to be delivered to several extensions at a time. You can easily do this using the huntgroup functionality.
1. While in Edit Customer mode, go to the **Huntgroups** tab.
2. Click ➕ **Add** to add a new huntgroup.

3. Enter the following information:
   - **Huntgroup Number** – The number that the end-user will dial on his phone to reach one or more assigned extensions.
   - **Huntgroup Name** – Logical name for this group of extensions.
   - **Assigned Extensions** – Choose one or several extensions that calls will be delivered to. Click the **Assigned Extensions** link to invoke the window with a list of extensions, and check the boxes next to the desired extensions. You may also leave this field empty for now and assign extensions later on.

- **Hunt Sequence** – Specify the order for delivering a call to the extensions.
  - o If **Order** is chosen, extensions will be called one by one from the first number to the last one.
  - o Choose **Random** if you want to use a random order.
  - o **Simultaneous** allows every extension from the list to be called simultaneously.
  - o If you choose **Least Used,** first the accounts assigned to the extensions will be sorted in descending order of their last use, and then the call will be delivered to these extensions accordingly.

5. Click the 💾 **Save** icon on the left side of the row.

Several huntgroups can be combined into one huntgroup. While in the window for selecting extensions, choose the **Huntgroups** tab and check the boxes next to the desired groups. Then click 🔲 **Save & Close**.

NOTE: Extension and huntgroup numbers must be different.

Call Forwarding

There are several call forwarding modes in PortaBilling:

- **Follow-Me**.
- **Advanced Forwarding**.
- **Forward to SIP URI**.
- **Simple Forwarding**.

You can choose the desired forwarding mode for an account from the **Forward Mode** drop-down list on the **Service Features** tab.

Follow-Me Forwarding Mode

Using follow-me you can forward calls to multiple destinations. You can also configure it so that each of these destinations will be active in its own time period. For example, let's suppose you want to redirect all your calls from 8 pm till 8 am to your cell phone (e.g. 35644890078) in case your IP phone is offline.

1. In the **Help Desk** section of the Admin-Index page, choose **Account Info** and search for the desired account.
2. Go to the **Service Features** tab to enable forwarding:
   o **Forward Mode** – Specify **Follow-Me**.
   o **Default Answering Mode** – Specify **Forward only**.

3. The additional **Follow-Me** tab now appears on the screen. Choose this tab.
4. Click ➕ **Add** to add the number on which you want to receive forwarded calls.

5. Enter the following information:
   - **Order** - Select the order for redirecting a call:
     o If you set **As Listed**, then every active number will be called, from the first number to the last one.
     o If you would like to call all numbers from the list simultaneously, use **Simultaneous**.
     o For random order, select **Random**.

   - **Name** – Type in a name for the follow-me number (the cell phone number in the example).
   - **Destination** – Enter a number for redirecting calls (e.g. 35644890078).
   - **Calling Party Display** – Choose how to display the caller's info during forwarding:
     o Select **Caller Number and Name** to see the phone number and the name of the original caller.
     o If you choose **Caller Number and Forwarder Name**, then the phone number of the caller and the forwarder's name will be displayed.

o To see the phone number and the name of the forwarder, select **Forwarder Number and Name**.

- **Active** – Define the period when the number is active and is allowed to receive a forwarded call. Click the **Active** link or wizard ![icon] icon to create a period. If you want the follow-me number to be active permanently, click the **Always** button while on the **Period Wizard** page. See the *How to …* chapter for more details.
- **Timeout** – Set the amount of seconds to wait before a call is forwarded to the next number in the list if the call is not answered.
- **Off** – If you wish to temporarily disable forwarding to this follow-me number, check this option.

6. Click the ![Save] **Save** icon to save the results of your work.
7. Repeat steps 5-7 until all the desired follow-me numbers have been added.

Advanced Forwarding Mode

This is similar to follow-me mode, but has a few extra options that allow you to route calls to SIP URI. For example, suppose you want to redirect your calls to SIP URI from 1 pm till 3 pm. To do this, follow the steps below:

1. In the **Help Desk** section of the Admin-Index page, choose **Account Info** and search for the desired account.
2. Go to the **Service Features** tab to enable forwarding:
   o **Forward Mode** – Specify **Advanced Forwarding**.
   o **Default Answering Mode** – Specify the method of processing incoming calls. Choose **Forward only** if you do not want the call to go to voicemail if not answered. Otherwise, choose **Forward then voicemail** here.
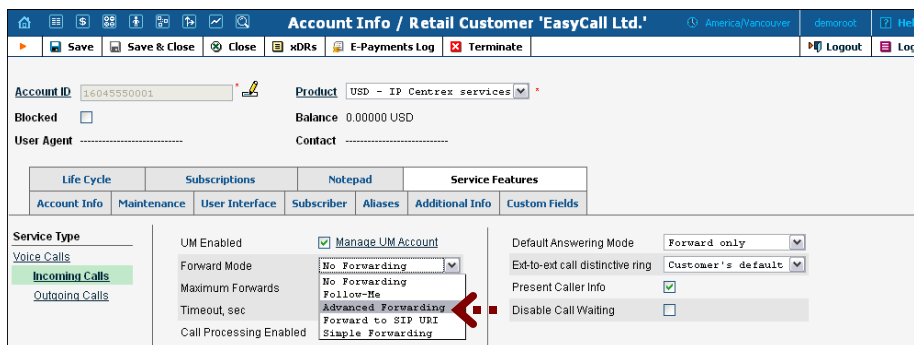
3. The additional **Follow Me** tab now appears on the screen. Choose this tab.
4. Click ➕ **Add** to add the number on which you want to receive forwarded calls.
5. Enter the information needed. The fields are very similar to those for adding a follow-me number as described above.
6. Below are additional fields that are not available when adding a follow-me number.
   - **SIP Proxy** – Select SIP proxies from the drop-down menu (SIP proxies available for a certain customer should be specified on the customer's **Permitted SIP Proxies** tab).
   - **Keep Original CLD** – Check this option to ensure that the originally dialed number is present in the call information when forwarding a call to another account.
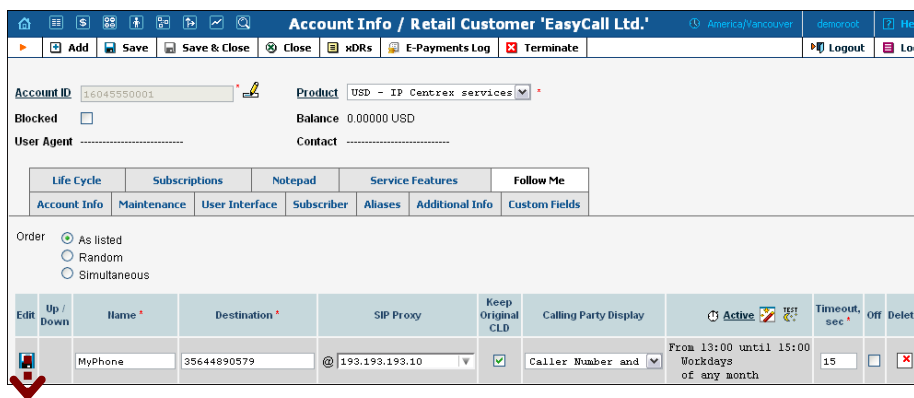
7. Click the 💾 **Save** icon to save the results of your work.

Simple Forwarding

The simplest type of forwarding is when you specify a single phone number to which all calls will be sent. For example, if you wish to redirect all your calls to 7779854213, use simple forwarding.

1. In the **Help Desk** section of the Admin-Index page, choose **Account Info** and search for the desired account.

2. Go to the **Service Features** tab to enable forwarding:
   o **Forward Mode** – Select **Simple Forwarding**.
   o **Default Answering Mode** – Specify **Forward only**.
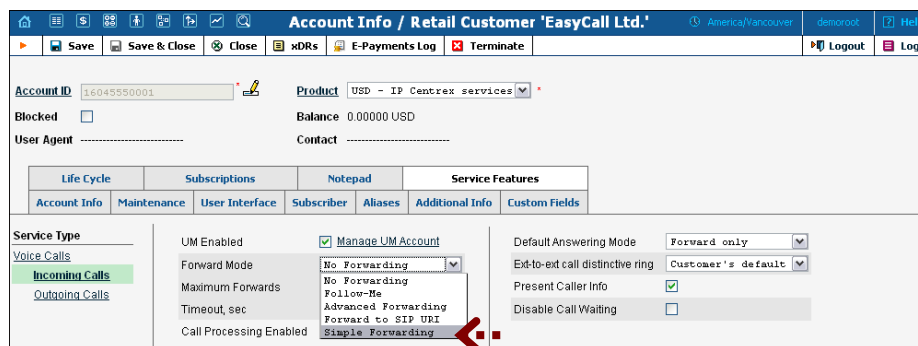
3. Go to the **Forward** tab.
4. Click ➕ **Add** to add the number on which you want to receive forwarded calls.
5. Enter the following information:
   - **Forward To** – The number you wish the calls to be forwarded to.
   - **Timeout** – Set how long the IP phone will ring before a call goes to voicemail (if **Forward then voicemail** is set in the **Default Answering Mode** field) or is disconnected.
   - **Keep Original CLD** – If this is checked, then the originally dialed number will be preserved during forwarding.
   - **Calling Party Display** – Choose one of the ways to display the caller's info during forwarding (a detailed description of each option is given above).
   - **Limit Simultaneous Calls To** – Specify the allowed number of concurrent calls here. When an account tries to place a call beyond the allowed limit, it will be rejected.
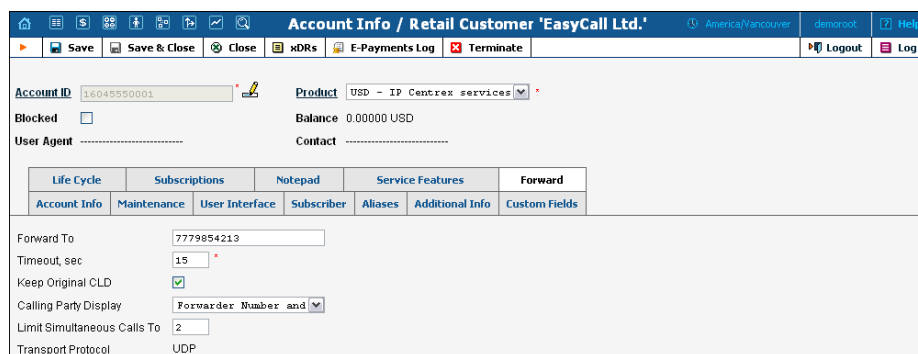
6. Click the 💾 **Save** icon.

Auto Attendant

Using the auto attendant feature, you can easily configure a voice menu.

1. In the **Help Desk** section of the Admin-Index page, choose **Account Info** and search for the desired account.
2. Go to the **Service Features** tab and click on **Incoming Calls**.
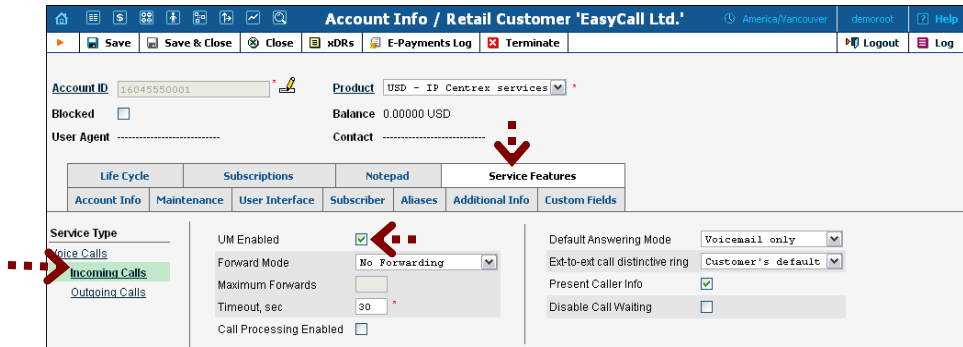
3. To access the unified messaging system for this account, check the box next to **UM Enabled**. To go directly to voicemail, select **Voicemail only** from the **Default Answering Mode** drop-down list. Then click 🖫 **Save**.
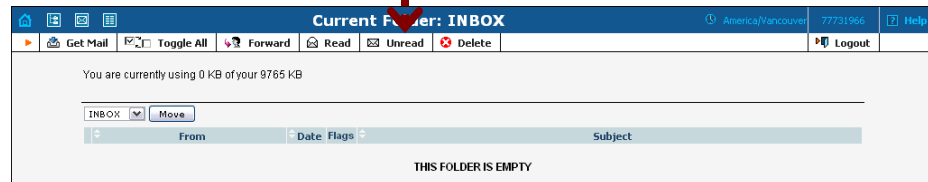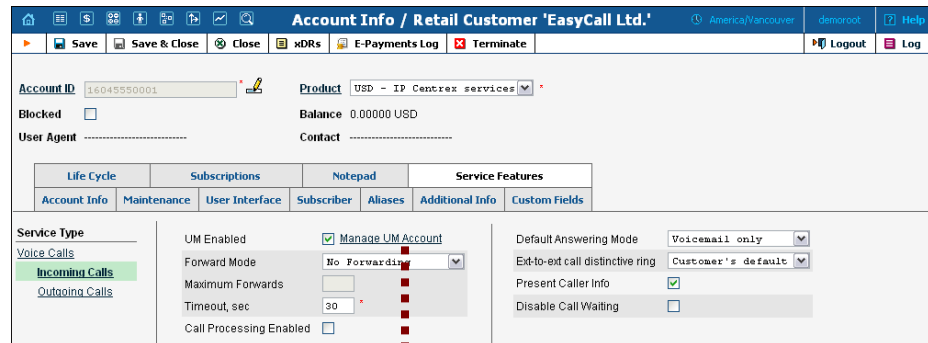
4. Once this is done, the **Manage UM Account** link appears. Click on this link to go to the account's UM administration page.
5. From the **Options** functional block, choose **UM Preferences**. On the **UM Preferences** page, go to the **Mode** tab and choose **Auto Attendant**. Then click **Save & Close**.

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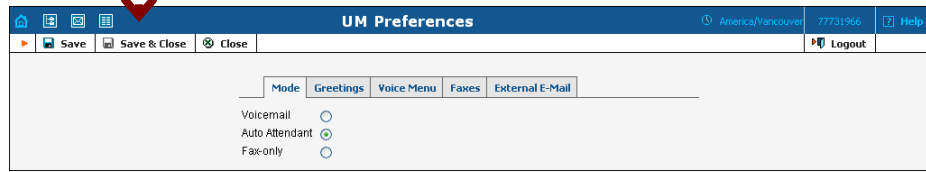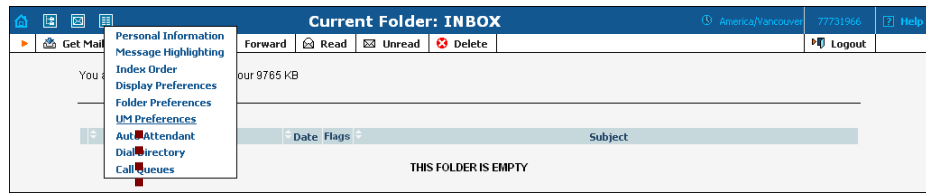
6. To go to the **Auto Attendant** page, choose **Auto Attendant** from the **Options** functional block.
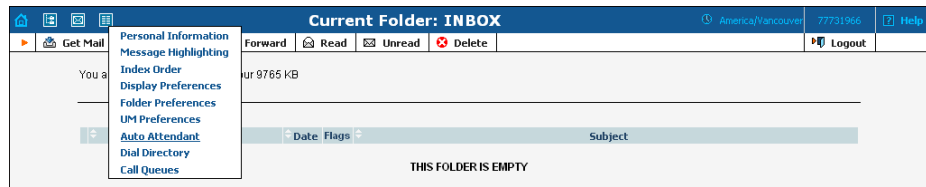
7. To create a new voice menu, click ➕ **Add** in the toolbar.

8. Enter the following information:
    o   **Name** – A logical name for the menu (e.g. Support for the support department).
    o   **Active** – Define the period when the menu is active. Click the wizard ✦ icon to create a period. If you want the menu to be active permanently, click the **Always** button next to it.
    o   **Intro** – The prompt played when the menu is entered for the first time (e.g. "Welcome to the PortaOne support department"). If you choose **None**, there will be no welcome message played. Choose **Custom** and click on **Browse** to select the needed file.
    o   **Timeout** – If no input is received and the **Standard** option is selected, then the **Default** prompt will be played. Choose **Custom** to use your own prompt.
    o   **Menu** – A menu prompt. Choose **None** if you would like to omit this step, otherwise use **Custom**. In this case, the user will hear all the available options (e.g. "Press 1 for operator, press 2 for support").
    o   **Unavailable** – If the **Unavailable** option is selected, the user will hear either the default prompt (if you choose the **Standard** prompt) or a custom one. For the latter case, choose **Custom**.
9. Choose the desired action for the specified events (e.g. you would like to set transfer as the second event and play the prompt before this action).
    1. Next to the second event in the **Action** column, choose **Transfer**.
    2. In the **Destination** column, specify the number you would like to transfer the call to.
    3. Check the **Play Before Action** box next to this row, since you would like to play the prompt before the transfer (e.g. "Thank you for contacting the support department"). You can either record the prompt (using

the ●, ▶ and ↩ icons) or upload an existing file (just click **Browse…** on the right and then choose the appropriate file).

4. Repeat steps 1-3 to define actions for the remaining events.

Please consult the *Auto-Attendant* chapter in the **PortaUM Administrator Guide** for a detailed description of all the parameters available here.

10. Click 🖫 **Save & Close**.

2. SIP Trunking

This chapter demonstrates how to set up SIP trunking so that

- A call to one of the DIDs that the customer has purchased is processed by PortaSwitch® and routed to the customer's external IP PBX
- Outgoing calls from the customer's IP PBX are processed and routed by PortaSwitch® to carriers

Please refer to the *IP Centrex Features* section in the **PortaSIP Administrator Guide** for information on how to activate and use the IP Centrex / hosted IP PBX features available in PortaSwitch®.

SIP Trunking Setup

Checklist

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

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

Operation	Done
Initial configuration	
Make sure configuration of general and network settings has already been performed according to the instructions in the *Basic SIP Service* section in the **PortaSwitch Handbook: Residential VoIP Services**.	[ ]
Check that basic SIP service configuration has already been performed according to the instructions in the *Basic SIP Service* section in the **PortaSwitch Handbook: Residential VoIP Services**, and you are able to make SIP-to-SIP calls.	[ ]
Confirm that the DID supplier vendor and its "VoIP from Vendor" connection has been configured according to the instructions in the *Incoming Calls from PSTN* chapter in the **PortaSwitch Handbook: Residential VoIP Services**.	[ ]
Rating configuration (Vendor)	
Create a tariff (referred to as **tariff A** later on) that describes your termination costs for calls forwarded to SIP-URI (make sure it is NOT a **Routing** type!).	[ ]
Enter a zero cost rate for the wildcard destination (|) in **tariff A**.	[ ]
Create an internal (fake) vendor for keeping track of calls forwarded to SIP-URI (you can use the internal vendor created earlier).	[ ]
Create a "**VoIP to Vendor**" connection with the **Remote Gateway ID** field containing the "**SIP-URI**" string for this vendor using **tariff A**.	[ ]
Rating configuration (Customer)	
Create a tariff (referred to as **tariff B** later on) that will be used to calculate the charges for outgoing calls from customer's IP PBX.	[ ]
Enter rates in **tariff B** that cover the destinations where your customers can call.	[ ]
Create a tariff (referred to as **tariff C** later on) that will be used to calculate the charges for forwarded calls.	[ ]

Enter rates in **tariff C** that cover the destinations where your customers can forward their incoming calls to (here you would typically just copy your rates from **tariff B** which is specifically for outgoing calls.)   [   ]

Enter a zero cost rate for the special VOICEONNETRX destination in **tariff C**. This rate will be used to produce zero charges for call segments between PortaSwitch and a remote IP PBX.   [   ]

Create a tariff (referred to as **tariff D** later on) that will be used to calculate the charges for incoming calls to DIDs.   [   ]

Enter rates in **tariff D**, that cover the DIDs you offer to your customers.   [   ]

Create your SIP trunking product.   [   ]

Create a rating entry for this product that associates the service **Voice Calls** and **tariff B** with the PortaSIP **Node** and OUTGOING **Access Code**.   [   ]

Create a rating entry for this product that associates the service **Voice Calls** and **tariff C** with the PortaSIP **Node** and FOLLOWME **Access Code**.   [   ]

Create a rating entry for this product that associates the service **Voice Calls** and **tariff D** with PortaSIP **Node** and INCOMING **Access Code**.   [   ]

Account provisioning

Create a retail customer who will represent the IP PBX owner.   [   ]

Create an account that will be used to authorize outgoing calls from IP PBX.   [   ]

Create accounts for incoming DIDs (depending on IP PBX capabilities).   [   ]

Testing

Provision the IP PBX to place outgoing calls via PortaSwitch.   [   ]

Provision the IP PBX to receive incoming calls via PortaSwitch.   [   ]

Check the parameters (SIP proxy address and port and optionally, account ID and password) in the IP BPX and then make a test call.   [   ]

PortaBilling requires that every call be accounted for on the revenue side (customers) and on the cost side (vendors) and since in case of calls forwarded to a remote IP PBX no vendor (in the traditional sense) is involved – you will need to create a special internal vendor, which will be used to keep the CDRs for these calls. This vendor and the connection to the vendor is a required step in order for the **Forward to SIP URI** function to work.

This connection should specify a "**SIP-URI**" string in the **Remote Gateway ID** field and have a blank **Remote IP** field. Skip the vendor configuration part of the SIP Trunking section if you already have this connection.

Create Internal Vendor Tariff

You may use the tariff created earlier for SIP-to-SIP calls (according to the *Basic Residential VoIP Service* section in the **PortaSwitch Handbook: Residential VoIP Services**) – in this case just skip this section and go to the *Define Connection for Internal Vendor* step.
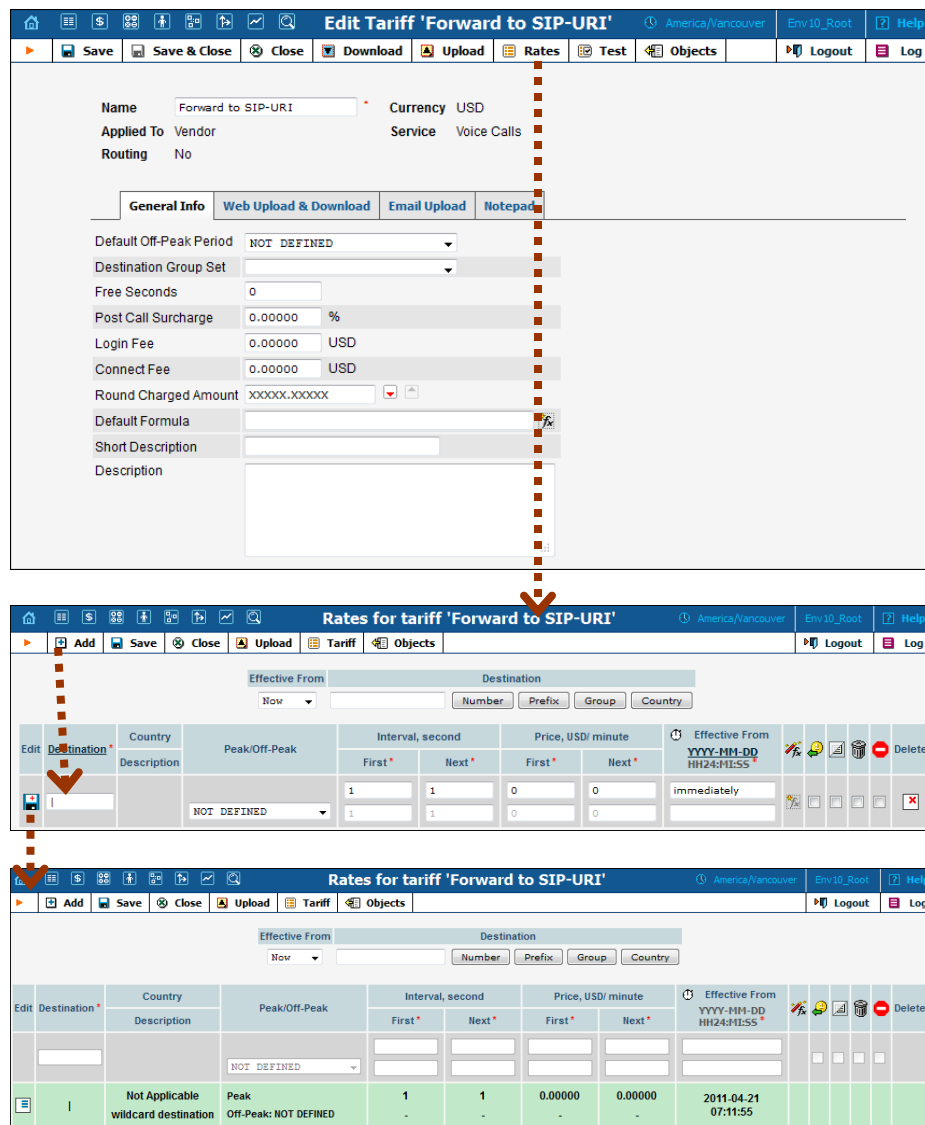

1. In the **Rating** section of the Admin-Index page, choose **Tariffs**.
2. On the Tariff Management page, choose ⊞ **Add**.
3. Fill in the **Add Tariff** form:

o **Name** – A short name for the tariff; this is the name you will then see in the select menus.
o **Currency** – Since this is an internal tariff, choose the same currency that you use as the base currency for this environment.
o **Applied To** – Choose "**Vendor**."
o **Routing** – Make sure the **Routing** check-box is turned off.
o **Service Type** – Choose **Voice Calls** here.
o **Short Description** – A short tariff description.
o **Description** – An extended tariff description.

4. Click 🖫 **Save**.

Enter Rates for Internal Vendor Tariff

1. Click the 🖽 **Rates** icon next to the name of the internal vendor tariff.

2. On the **Edit Rates** screen, click 🔲 **Add**.
3. Insert '**|**' (pipe) in the **Destination** field. Leave the other fields in the form as they are, because there is no need to define actual prices for calls forwarded to SIP-URI. This will create a special rate that matches any dialed phone number.
4. Click 🔲 **Save&Close**.

Define Connection for Internal Vendor

1. In the **Participants** section of the admin interface, choose **Vendors**.
2. Click on the **Connections** icon next to the internal vendor name.
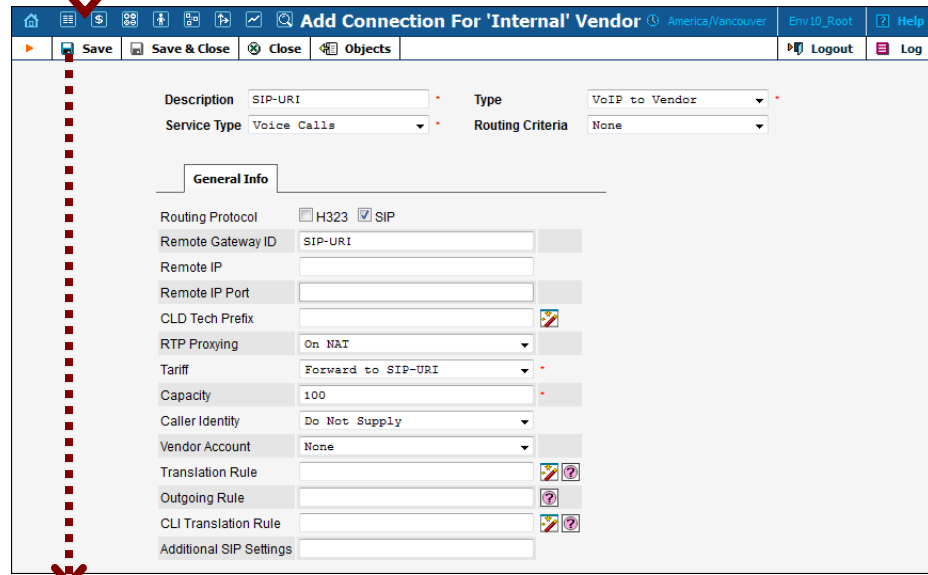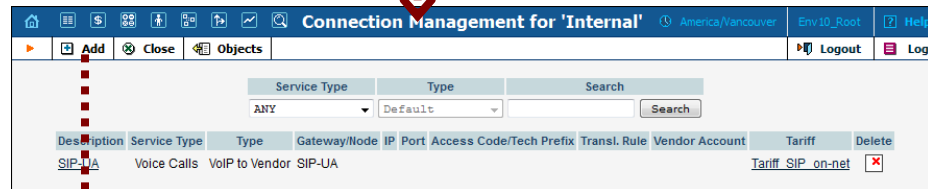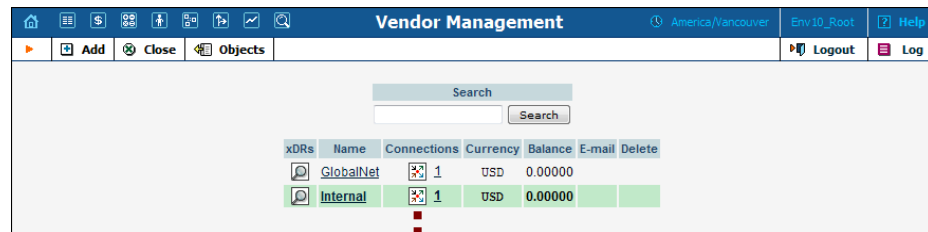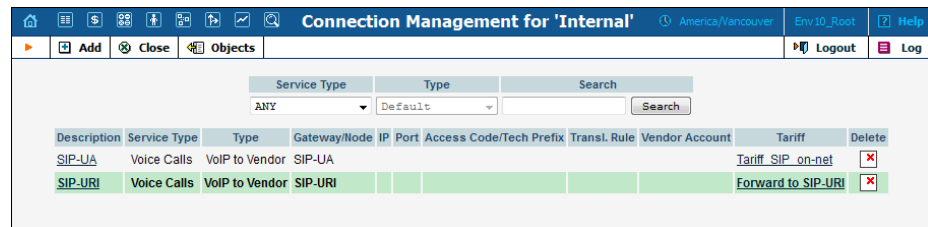
3. Press ➕ **Add** to add a new connection.
4. Fill in the connection information:
   o **Service Type** – Choose **Voice Calls** here.
   o Choose **VoIP to Vendor** as the **Type** of connection.
   o In the **Remote Gateway ID** field specify the "**SIP-URI**" string.
   o In the **Tariff** menu select the tariff created earlier that describes your termination costs for calls forwarded to the SIP-URI.
   o In **RTP Proxying,** select **On Nat**.
   o Specify **Description** and **Capacity** for this connection.
5. Click 💾 **Save**.

Create Customer Tariffs

You should create three separate tariffs for your IP PBX customer. These tariffs will be used for charging outgoing, forwarded and incoming calls.

Follow the steps for creating customer tariffs below:

1. In the **Rating** section of the Admin-Index page, choose **Tariffs**.
2. On the **Tariff Management** page, choose ➕ **Add**.

3. Fill in the "**Add Tariff**" form:
   o **Name** – A short name for the tariff object; this is the name you will then see in the select menus.
   o **Currency** – Indicates in which currency pricing information is defined. All pricing information for a single tariff must be defined in the same currency.

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

   o **Applied To** – Choose "**Customer"** as this tariff will be used to charge your customers.
   o **Managed By** – Choose "**Administrator Only**" here (this option is only visible after you select **Applied to: Customer** above).
   o **Service Type** – Choose **Voice Calls** here.
   o **Default Off-peak Period** – If you do not differentiate between peak and off-peak rates, just choose **Not defined**; otherwise, select one of the previously defined off-peak periods.
   o **Destination Group Set** – If you wish to enter rates in the tariff not for every individual prefix, but for a whole group of prefixes at once, you should create a destination group set and destination groups beforehand. Leave this select menu empty for now.
   o **Round Charged Amount** – Instead of calculating xDRs with a 5-decimal-place precision, round up xDR amount values (e.g. to cents, so that 1.16730 becomes 1.17). Set the rounding pattern to XXXX.XX000 (as shown on the picture) so every call will be rounded to the equal cent amount.
   o **Default Formula** – Default rating formula, which will be applied to every rate created in the tariff. If you leave this empty, the "old-style" rating will be used.

o   **Short Description** – A short tariff description. This will be
    shown in the rate lookup on the admin interface and the self-care
    pages for your accounts and customers.

o   **Description** – An extended tariff description.

4. Click 💾 **Save**.

Enter Rates

You should specify the rates for the three separate tariffs created in the
previous section:

- Rates for the first tariff (referred to as **tariff B** in the checklist)
  should cover the destinations that your customers can call. They
  are used to calculate the charges for outgoing calls from the
  customer's IP PBX.
- Rates for the second tariff (referred to as **tariff C** in the checklist)
  should cover the destinations that your customers can forward
  incoming calls to (typically you would just copy your rates from
  tariff B here for outgoing calls). A zero rate for a special
  VOICEONNETRX destination should also be added to this
  tariff. This rate will be used to produce zero charges for call
  segments between PortaSwitch and a remote IP PBX.
- Rates for the third tariff (referred to as **tariff D** in the checklist)
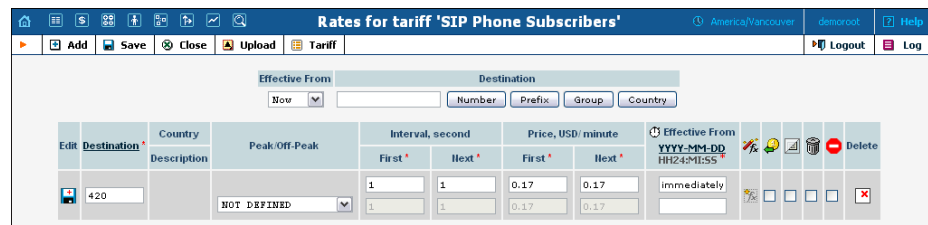 should cover the DIDs you offer to your customers.

If you charge the same rate for forwarded calls as you do for outgoing
calls and do not charge for calls made on your own network (a very
typical case), you may skip the step for creating tariff C and just use tariff
B for both outgoing and forwarded calls.

Rates can be managed online and offline. Use the section below to add
rates to all three tariffs.

Managing Rates Online

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

1. On the Tariff Management page you will see a list of the available tariffs. Click the ▦ **Rates** icon in front of the name of the tariff. When you are in Tariff Management for a particular tariff, click on **Rates** in the toolbar.
2. On the **Edit Rates** screen, click ➕ **Add**.
3. Fill in the required information:
   o **Destination** – A destination prefix may be entered directly, e.g. **420** for Czech Republic, or you can access the destinations directory by clicking the **Destination** link (in the column header). Here you can find the desired prefix by country name.

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

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

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

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

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

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

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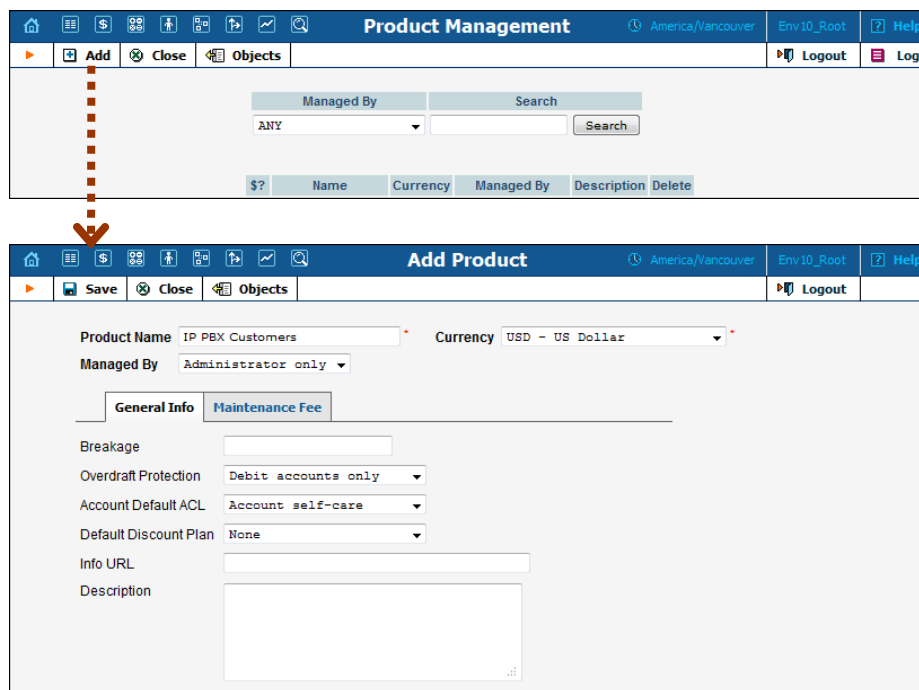5. Repeat these steps if you need to enter more rates.

Managing Rates Offline

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

Create Product

Create a SIP trunking product.

1. In the **Rating** section of the Admin-Index page, choose **Products**.
2. On the **Product Management** page, click the ⊞ **Add** icon.

3. Fill in the "**Add Product**" form:
    o **Product Name** – Product object name.
    o **Currency** – Product currency; only tariffs which have the same currency will be permitted in the rating list.
    o **Managed by** – If you want this product to be used for your reseller's accounts, so the reseller himself can change the parameters of this tariff and create new accounts using this
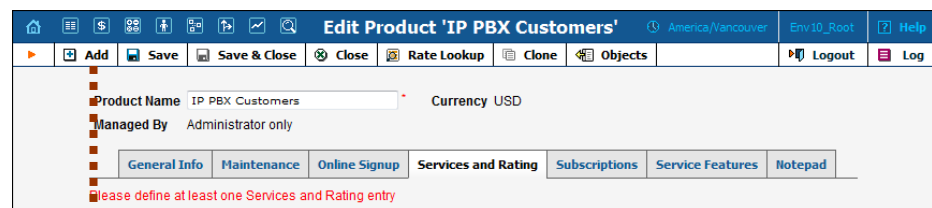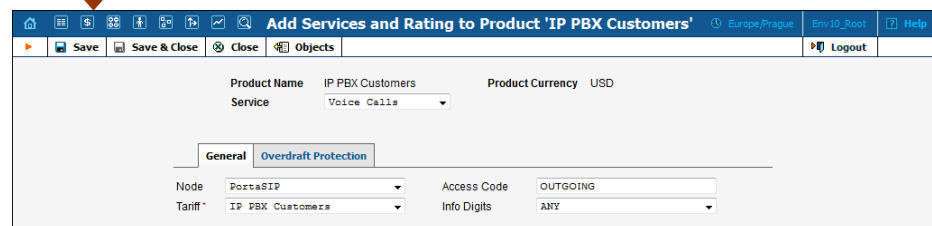
product, choose a customer name from the menu. Otherwise, choose **Administrator only** here.

- o **Breakage** – This parameter is typically used only for prepaid accounts, so leave it empty.
- o **Account Default ACL** – The access level assigned by default to new accounts created with this product. The ACL determines which operations may be performed by accounts on the self-care pages. The default value is "Account self-care" (pre-defined ACL), which allows all possible operations.
- o **Default Discount Plan** – Leave **None** as the selected entry if you do not plan to offer any discounted rates based on call volume. (In the **PortaSwitch Handbook: Converged Services** you can find examples for setting up volume discounts).
- o **Description** – your comments about the intended use of this product.

4. Click 💾 **Save**.

Create Rating Entries for the SIP Trunking Product

1. On the **Services and Rating** tab, click the ⊞ **Add** icon.

2. Fill in the required information:
   - o **Service** – Choose "**Voice Calls**."
   - o **Node** – Select your PortaSIP node.
   - o **Access Code** – type OUTGOING in this field.
   - o **Tariff** – Select the tariff that will be used to calculate the charges for outgoing calls (referred to as **tariff B** in the checklist).

3. Configure overdraft protection for this product on the **Overdraft Protection** tab. (Please consult the *Rating* section in the **PortaBilling**

Web Reference Guide for a description of the parameters available here.)

4. Click 🖫 **Save&Close** to save this rating entry.
5. On the **Services and Rating** tab, click the ➕ **Add** icon to add the second rating entry.

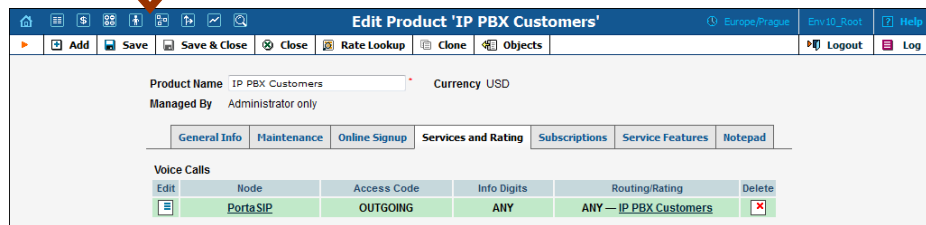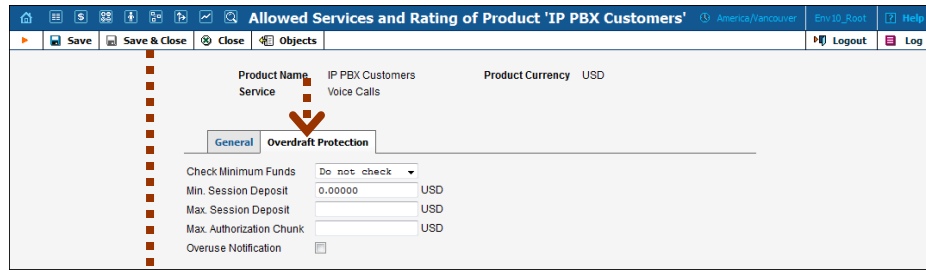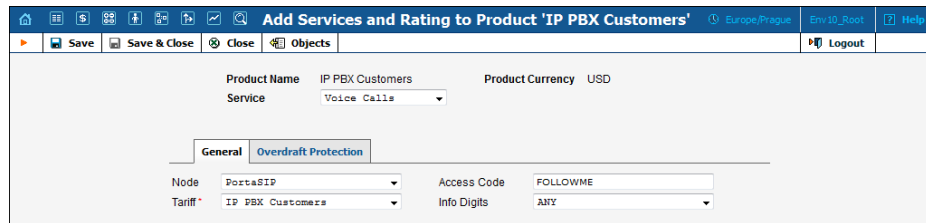6. Fill in the required information:
   o **Service** – Choose "**Voice Calls**."
   o **Node** – Select your PortaSIP node.
   o **Access Code** – Type FOLLOWME in this field.
   o **Tariff** – Select the tariff that will be used to calculate the charges for forwarded calls (referred to as **tariff C** in the checklist).

7. Configure overdraft protection for this product on the **Overdraft Protection** tab.
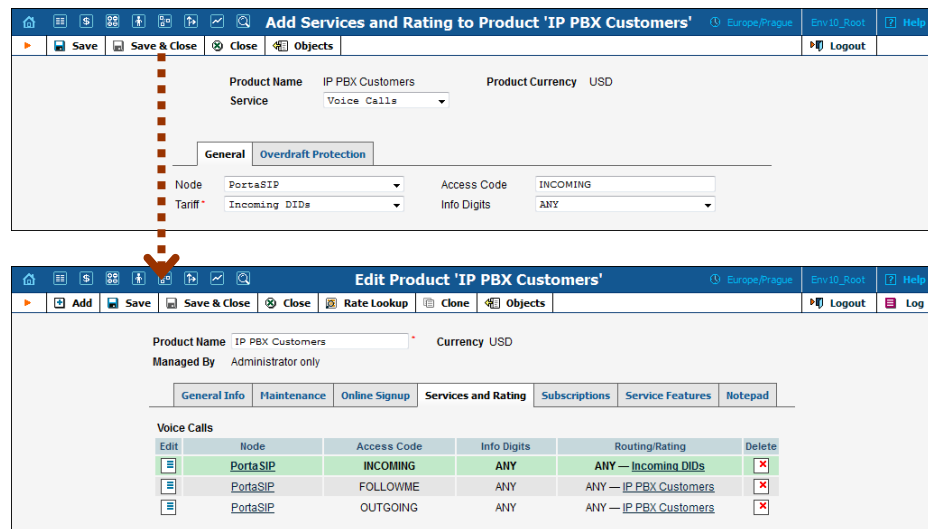8. Click 🖫 **Save&Close** to save this rating entry.
9. On the **Services and Rating** tab, click the ➕ **Add** icon to add the third rating entry.

10. Fill in the required information:
    o **Service** – Choose "**Voice Calls**."
    o **Node** – Select your PortaSIP node.
    o **Access Code** – Type INCOMING in this field.
    o **Tariff** – Select the tariff that will be used to calculate the charges
      for incoming calls (referred to as **tariff D** in the checklist).

11. Configure overdraft protection for this product on the **Overdraft Protection** tab.
12. Click ▤ **Save&Close** to save this rating entry.

Create a Customer

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

NOTE: If no Quick Forms have been created, you will go directly to the **Add Customer** page.
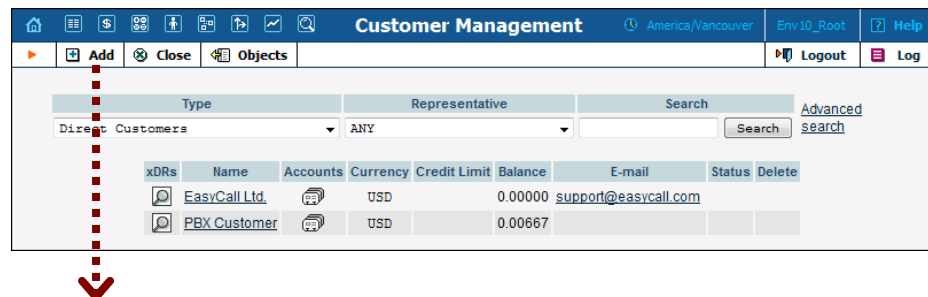
3. Fill in the **New Customer** form. Please note that there are several tabs with extra information available on the screen.

The most important fields are:

Main form (top)

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

Address Info tab

- o **Email** – An email address for the distribution of accounting information. After the billing period is over, a list of xDRs and other statistics will be sent to this address.
- o **Bcc** - Delivery to the specified email address of your account representative a copy of every outgoing email sent to the customer; this may be used for debug and archiving purposes.

Additional Info tab

- o **Billing period** – Frequency of invoice generation. For more details about the different available billing periods, see the **PortaBilling Administrator Guide**.
- o **Send Statistics** - **Summary only** – Deliver a call summary only and do not attach a details file; this could be useful in the case of

a large number of calls. Other options are **Full Statistics** (attach a complete list of xDRs) or **Do not send** (do not deliver call statistics to this customer via email at all).

Payment Info tab

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

o **Balance Warning Threshold** – the customer can be notified by email when his balance is dangerously close to the credit limit and service will soon be blocked. Here you can enter the value for such a warning threshold. This can be entered:

- as a percentage (e.g. 90%). The warning will be sent when the customer's balance exceeds that percentage of his credit limit. So, if the credit limit is USD 1000.00 and the threshold is 90%, a warning will be sent as soon as the balance exceeds USD 900.00. This is only applicable when the customer has a positive credit limit.

- as an absolute value. The warning will be sent as soon as the balance goes above the specified value.

User Interface tab

o **Time Zone** – the time zone in which customer will see his xDRs and also that which defines his billing period. For example, if you choose **America / New_York** here and the billing period is **Monthly**, it means the billing period will start on the first day of the month, 00:00 New York time.

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

Service Features tab

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

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

Please consult the **PortaBilling Web Reference Guide** for a description of other parameters available here. For now you may

leave these with their default values. If you change them later, these changes will automatically affect all accounts created under this Customer.

4.   Click 🖫 **Save&Close**.

Create Accounts

Account creation is described in the *Create Accounts* section in the **PortaSwitch Handbook: Residential VoIP Services**. Account configuration particularities that depend on the IP PBX features and preferences are described below.
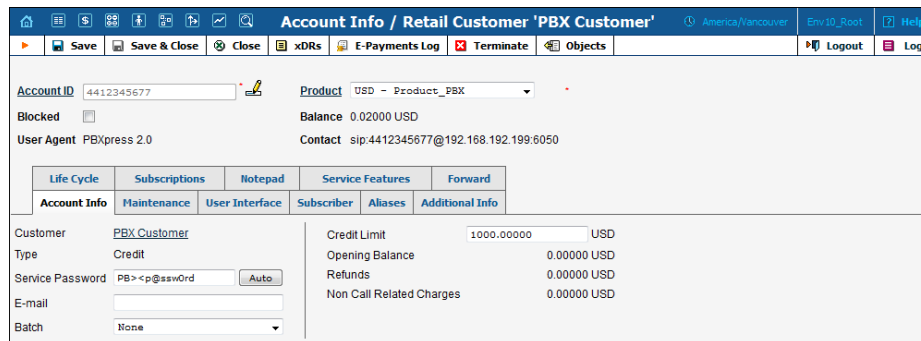
Accounts for Outgoing Calls from IP PBX

The configuration of outgoing calls depends on whether the IP PBX system is capable of digest authentication or not. Both types of IP PBX can interact with PortaSwitch and the difference in configuration is minor.

IP PBX Capable of Digest Authentication

If the IP PBX system is capable of digest authentication, you should create an account with the following options:
   o   **Type** – Choose **Credit** here.
   o   **Account ID** – Specify the account ID that will be used to authorize outgoing calls from IP PBX.
   o   **Service Password** – Specify a service password for this account.
   o   **Product** – Assign a previously created product to this account.
   o   **Opening Balance** – the initial balance on the account.

Note that you can limit the number of simultaneous calls for the specified customer (for more details, see the …*limit the number of concurrent calls for the appropriate customer?* section in the *How to …* chapter.)

IP PBX without Digest Authentication Capability

If the IP PBX system is not digest capable you should implement IP-based authentication to allow outgoing calls from IP PBX. In this case create an account with the following options:

- o **Type** – Choose **Credit** here.
- o **Account ID** – Specify account ID identical to the IP address of the PBX.
- o **Service Password** – Leave service password blank for this account.
- o **Product** – Assign a previously created product to this account.
- o **Opening Balance** – Type in the initial balance on the account.

Note that the system will automatically create a call handling rule for applying IP-based authorization for calls arriving from this IP address.

NOTE: If PBXpress is running on a public IP address, you may want to avoid engaging the SIP proxy in the SIP message exchange. This will result in a more than 20% decrease in call processing time. (Please consult the *Direct Incoming Calls to B2BUA* section in the **PortaSIP Admin Guide** for a detailed description.) In this case, specify a 5061 port in addition to the PortaSIP IP address.

Accounts for Incoming Calls to IP PBX

The configuration for incoming calls depends on the IP PBX features, i.e. whether IP PBX is capable of registering on a SIP server or not. Both types of IP PBX can interact with PortaSwitch and the difference in configuration is minor.

IP PBX Capable of Registering on a SIP Server

If the IP PBX system is capable of registering on a SIP server, you should create an account with the following options:

- o **Type** – Choose **Credit** here.
- o **Account ID** – Specify account ID (e.g. a DID number).
- o **Service Password** – Specify a service password for this account.
- o **Product** – Assign a previously created product to this account.
- o **Opening Balance** – Type the initial balance on the account.

NOTE: You may want to use the account that you already created for outgoing calls.

If you want to dedicate several DIDs to forwarding incoming calls to the IP PBX there are three comparable ways of doing this. They are described below.

Creating extra DIDs as accounts

You can create extra accounts (e.g. 12061234567, 18007778881 and 4412345678) with forwarding services on these accounts configured so that they always go to the main IP PBX account.

1. Next to the customer name, click on the ⊞ icon (the one in the **Accounts** column) to go to the **Accounts** menu for that customer.
2. Click on ⊞ **Add**.
3. Fill in the "Add account" form:
   - o **Account ID** – Specify an appropriate account ID (e.g. a DID number).
   - o **Service Password** – Specify a service password for this account.
   - o **Product** – Assign a previously created product to this account.
   - o **Opening Balance** – the initial balance on the account.

4. After clicking 🖫 **Save&Close** you will see a confirmation screen announcing that a new account has been created.
5. Click on the account ID that you have just created to modify it.
6. Click on the **Service Features** tab to enable forwarding:
   o **Default Answering Mode** – Specify **Forward only**.
   o **Forward Mode** – Specify **Simple Forwarding**.
7. The additional **Forward** tab now appears on the screen. Go to this tab:
   o Specify the ID of the main IP PBX account in the **Forward to** field.
   o Activate the **Keep Original CLD** option to store the called DID. The IP PBX can then correctly route the calls made to individual phone numbers to their appropriate extensions. (Please consult the *Forwarding with the original DNIS(CLD)* section in the **PortaSIP Admin Guide** for a detailed description.)

8. Click 🖫 **Save & Close**.
9. Repeat these steps if you need to create more accounts for forwarding calls to the main IP PBX account.

Creating a separate account for every IP PBX extension

If the IP PBX supports multiple SIP registrations you can create separate accounts in PortaBilling for every IP PBX extension.

Create extra DIDs as aliases

Please see the …*create extra DIDs as aliases?* section in *How to …* chapter.

IP PBX Incapable of Registering on PortaSIP

If the IP PBX system is not able to register on a SIP server you should configure a forwarding service by following the directions below.

1. Specify PBX IP addresses that you will forward calls to on the **Permitted SIP Proxies** tab for the specified customer.

2. Create an account for this customer as described in the *Create Accounts* section (**Account ID** can be a DID number that is used for forwarding calls to an external IP PBX).

3. Go to the Account's **Service Features** tab.
4. Select **Voice Calls** in the **Service Type** column.
5. Click on the **Incoming Calls** to enable forwarding:
   o In the **Default Answering Mode** menu choose **Forward Only**.
   o In the **Forward Mode** menu choose **Forward to SIP URI**.
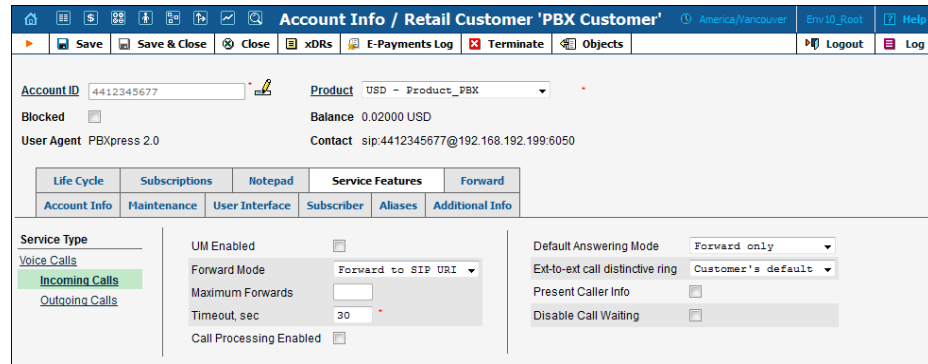
6. The additional **Forward** tab now appears on the screen. Go to this tab.
7. Specify an appropriate CLD and IP address for your IP PBX in the **SIP URI** fields.

NOTE: SIP URI values should contain one of the following: IP, IP:port, domain or domain:port. So, if your IP PBX uses a port different from 5060 for incoming calls, you should specify that port here.
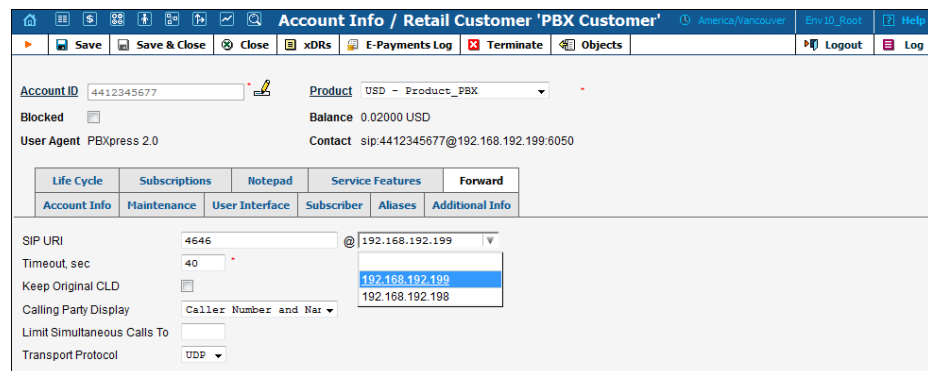
8. You should also activate the **Keep Original CLD** option to store the original DID. The IP PBX can then correctly route these calls to extensions. (Please consult the *Forwarding with the original DNIS (CLD)* section in the **PortaSIP Admin Guide** for a detailed description).
Please consult the **PortaBilling Web Reference Guide** for a description of other parameters available here. For now, you may leave these default values.
9. Click 💾 **Save**.
10. Repeat these steps if you need to create additional accounts for forwarding calls to IP PBX.

3. How to ...

...define a time period using the wizard?

Using the period definition wizard, you can easily define the time period when a number (e.g. a follow-me number) will be active. If you want the number to be active permanently, click the **Always** button at the bottom of the **Period Wizard** page. Otherwise, define the desired period. The following example illustrates the process of creating a period that starts at 8pm and lasts until 8am the next morning on workdays.
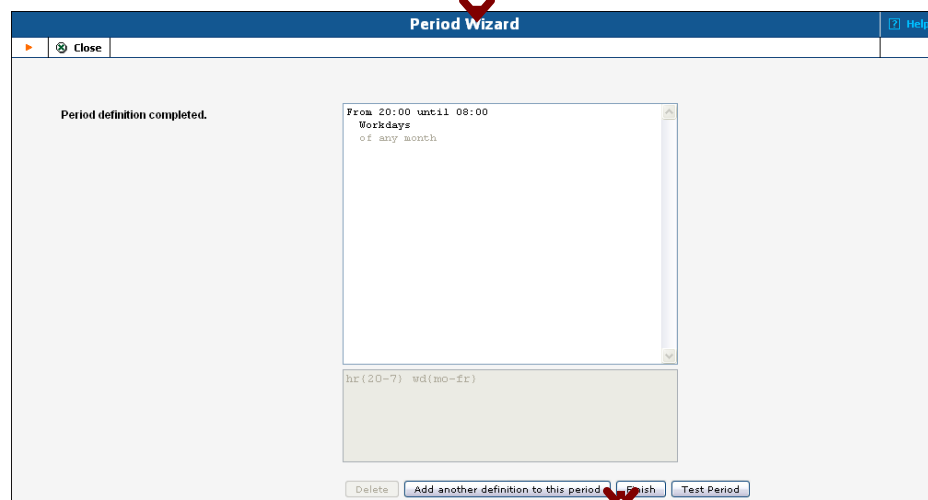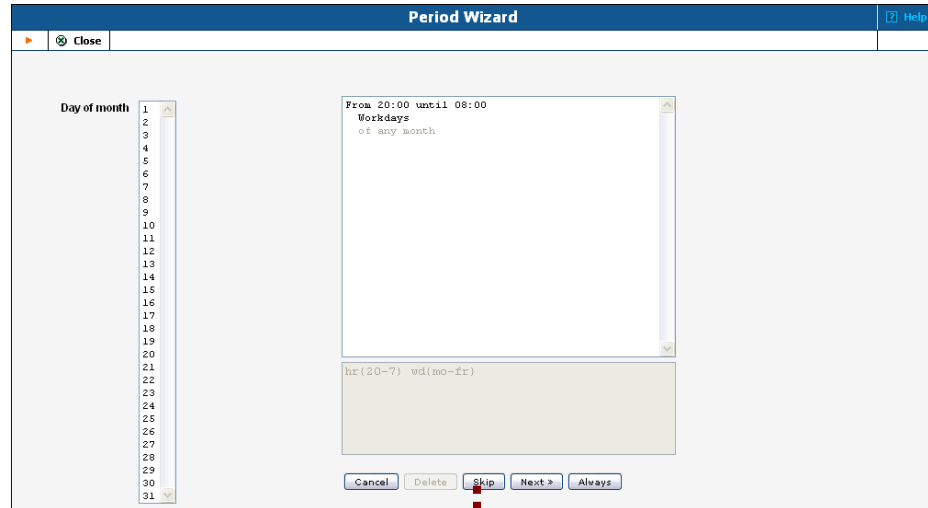
1. On the first page of the wizard, select 8 pm in the **From** box and 8 am in the **Until** box. Then click **Next**.

2. Select *Workdays* in the **Day of week** box and click **Next** to go to the next page.

3. Click **Skip** on the following two pages defining **Day of month** and **Month**.

4. Click **Finish** to complete the period definition. If you wish to set one more definition for this period, click **Add another definition to this period**.

...create extra DIDs as aliases?

For example, if you want to route DIDs 12061234567, 18007778881 and 4412345678 to the IP PBX, add them as aliases with the **Allow authentication / registration** parameter set to "**No**." (You may want to use the **Alias Generator** to create multiple aliases).

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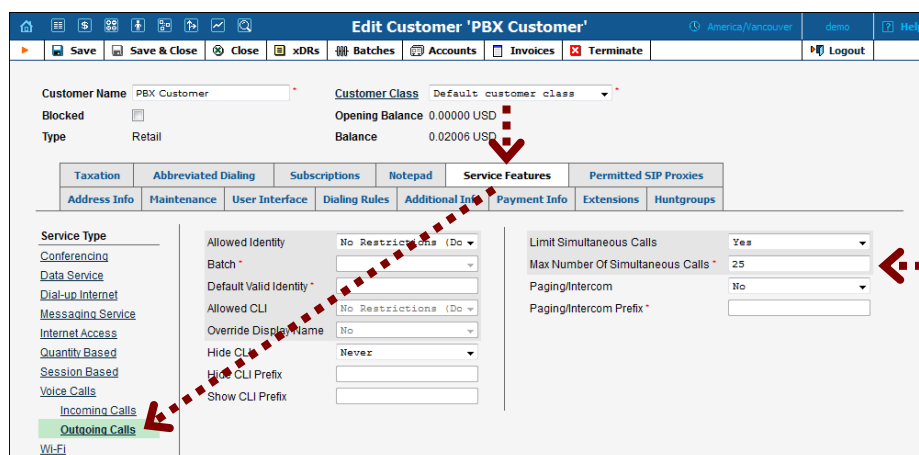

2. Click ⊞ **Add** to create a new alias for the current account.
3. Specify a DID number in the **Alias ID** field.
4. Click 🖫 **Save**.
5. Repeat steps 2-4 to add additional aliases.
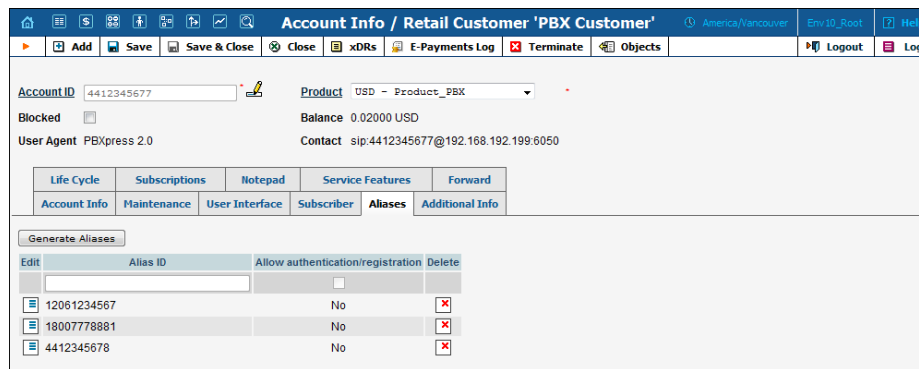
6. Click 🖫 **Save & Close**.

...limit the number of concurrent calls for the appropriate customer?

1. In the **Participants** section of the Admin-Index page, choose **Customers**.
2. On the **Customer Management** page, choose the specified customer.
3. Click on the **Service Features** tab.
4. Click on the **Voice Calls / Outgoing Calls** service type.
5. Define the **Max Number of Simultaneous Calls** parameter to allow up to a specific number of concurrent calls for the accounts of this customer.

6. Click 🖫 **Save & Close**.

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