PORTA ONE

Porta Switch™

PortaSwitch Handbook: Unified Communications

Maintenance Release 20

www.portaone.com

Copyright Notice & Disclaimers

PortaSwitch Handbook: Unified Communications
V.1.20.3, September 2009

Please address your comments and suggestions to: Sales Department, PortaOne, Inc. Suite #408, 2963 Glen Drive, Coquitlam BC V3B 2P7 Canada.

Table of Contents

Preface

This document shows how you can start offering unified messaging services with PortaUM and conferencing services with PortaBridge. Please refer to the **PortaUM Administrator Guide** for a general overview of the PortaUM system and an explanation of the PortaBridge architecture.

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 performing the action described in the paragraph.

Tips provide information that might help you solve a problem.

1. Setting up and Using UM Services

Checklist

Print this 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	
General configuration has already been done according to the instructions in **PortaSwitch Handbook: SIP Services**.	[ ]
Create destinations for UM services.	[ ]
Associate access numbers with applications.	[ ]
Network configuration	
Create a node for your PortaUM server.	[ ]
Rating configuration	
Add rates for UM destination to tariff A.	[ ]
Enter rates in tariff A for "|" wildcard destination.	[ ]
Create a new accessibility entry in your existing product A, using PortaUM node and tariff A.	[ ]
Create tariff B, which will define call routing to PortaUM. (Make sure this tariff has the **Routing** type and the **Voice Calls** service assigned!)	[ ]
Enter rates in tariff B for access numbers used for the conferencing service.	[ ]
Create a connection for your "internal" vendor (the one created in the basic SIP service configuration for handling SIP on-net calls) using tariff B.	[ ]
Account provisioning	
Enable the conferencing service for an existing account (with product A).	[ ]

Initial Configuration of PortaBilling

Before proceeding with UM setup, please perform initial system configuration according to the *Setting up Standard SIP Services* chapter in the **PortaSwitch Handbook: SIP Services**. At this point, all of your destinations, tariffs, products, customers and vendors should already be in the system.

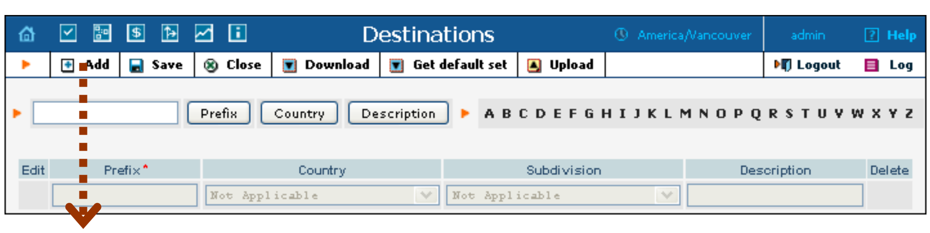
Create Special UM Destinations

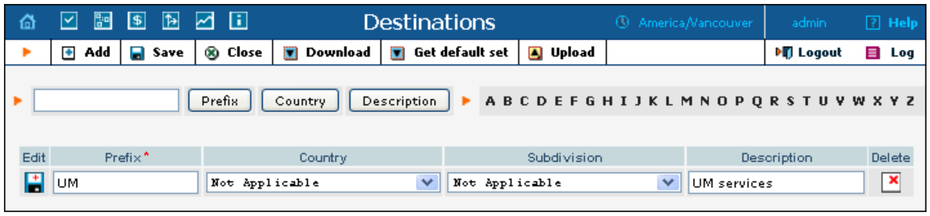
Your subscribers can access UM by two different methods:

- Dialing a special UM number from their SIP phone. Usually this is a non-E.164 number, so it will not overlap with any real phone number the customer might wish to call, e.g. ***98**. Another option for accessing UM is for the end user to dial his own phone number.
- Calling a certain number from the PSTN network. This should be a valid phone number allocated to you by a local telco and accessible to anyone on the PSTN network. If you do not have such a number, your customers will not be able to check their voicemail from a regular phone.

There is no need to create destinations which describe all UM numbers, because you are not going to charge your clients for accessing UM services. The only destinations you need to add to the system are a symbolic 'UM' destination and a wildcard destination with the '|' prefix. Instead of creating many rates for all UM access numbers in the vendor tariff, you can create a single rate using this destination wildcard.

1. In the Management section of Admin-Index, choose **Destinations**.
2. Click on the **Add** button.

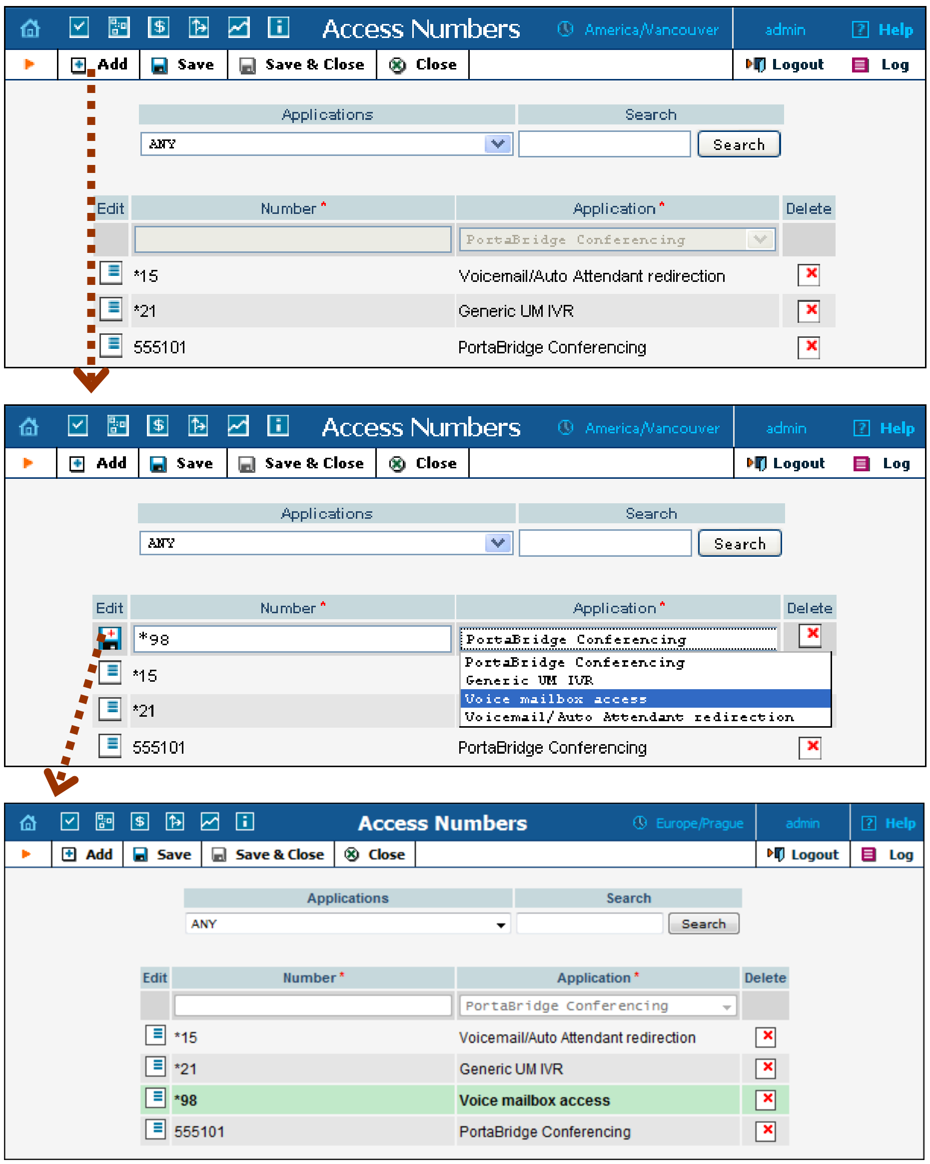




3. Enter UM for **Prefix** and leave **Not Applicable** for the country and country subdivision. Put a comment in the **Description** column that clearly identifies this as a special prefix assigned to unified messaging.
4. Click **Save.**
5. Repeat steps 2-4 to add the "|" wildcard destination.

Associate Access Numbers with Applications

You need to define which IVR applications will be launched if customers dial special access numbers from their IP phones or from the PSTN network.

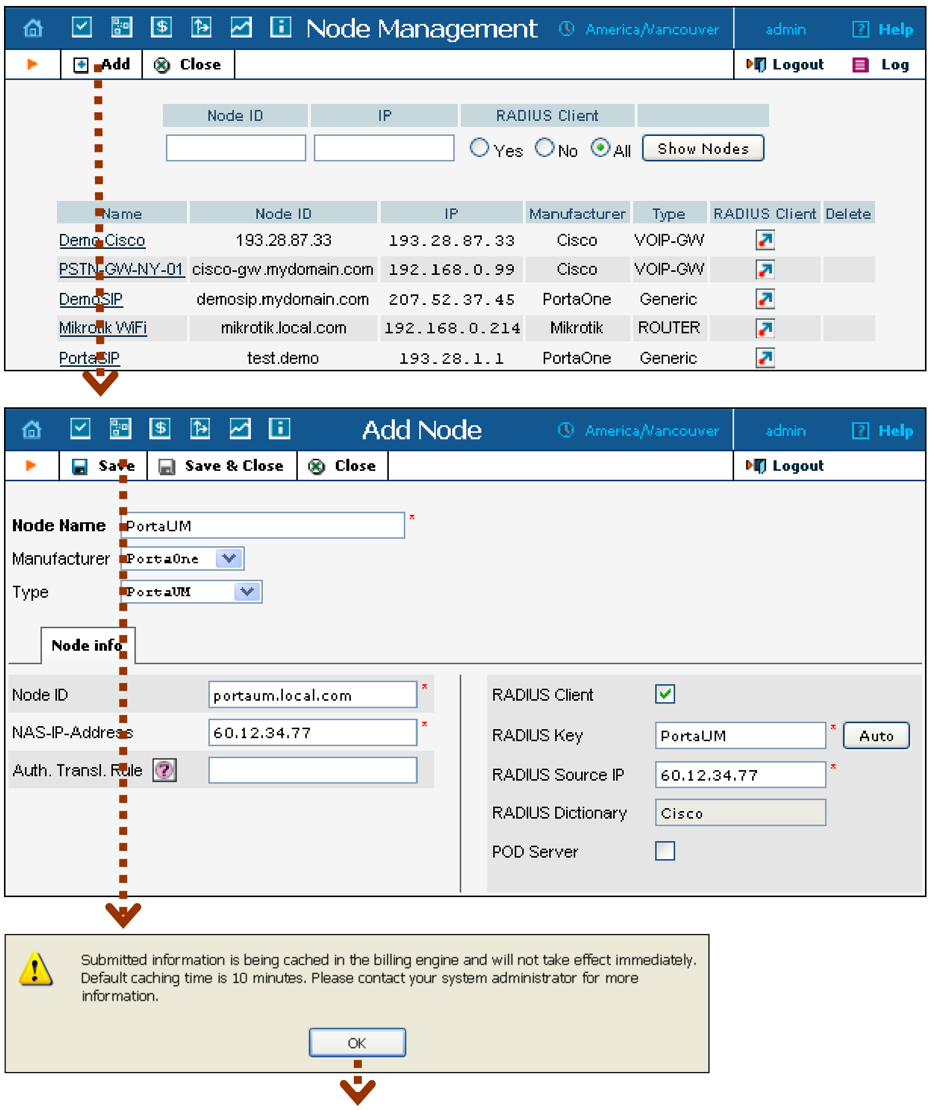


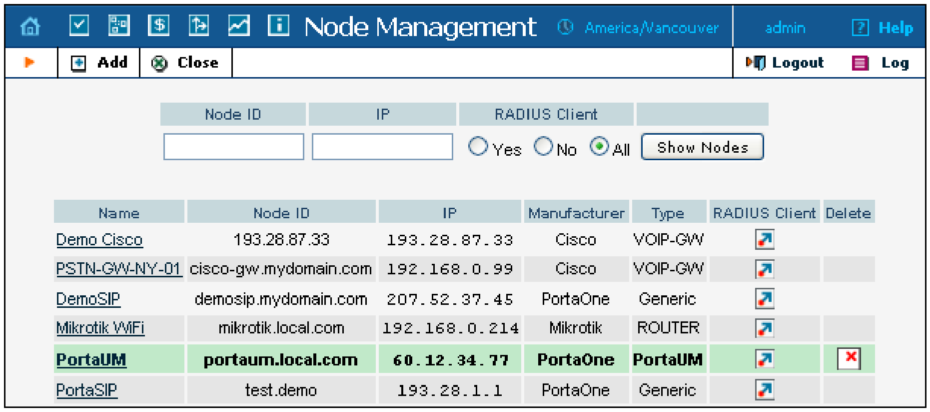
1. In the Routing section of Admin-Index, choose **Access Numbers**.
2. Click on the **Add** button.

3. Enter your UM number in the **Number** field. In the **Application** drop-down box, select the application you want to associate with this access number.
4. Click **Save**.

Create Nodes

Now you have to enter your PortaUM server and UM gateway (used as the platform for VXML) as nodes. PortaBilling requires some key information about your network equipment, such as IP address, Node ID, Radius shared secret, and so on.



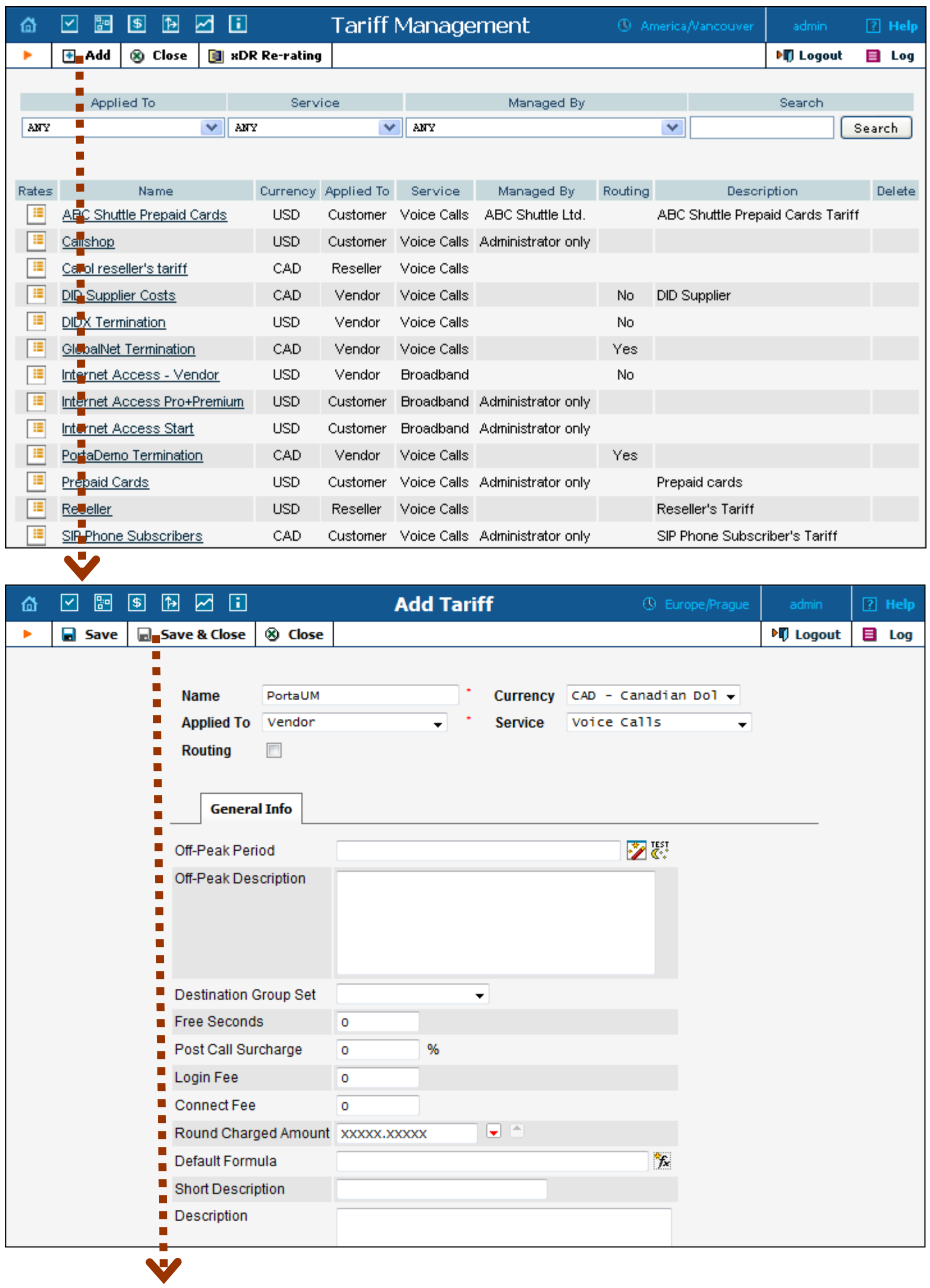


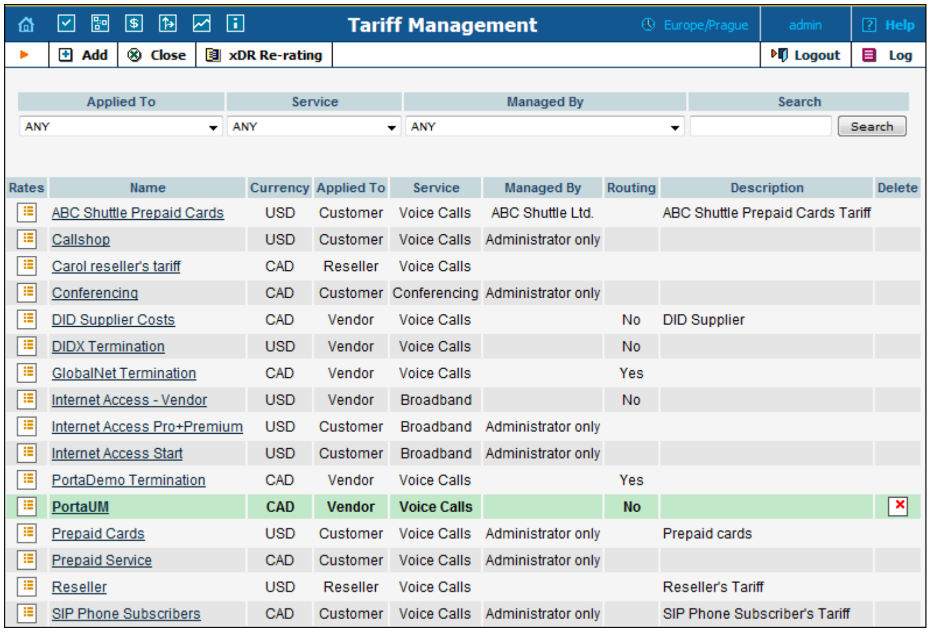
1. In the Networking section of the Admin-Index page, choose **Nodes.**
2. In the Node Management window, click **Add** icon.
3. Fill in the New Node form:
   - **Node Name** – A short descriptive name for your UM server or gateway (this will be used in the select menus).
   - **Manufacturer** – Select **PortaOne**.
   - **Type** – Node type; select **PortaUM**.
   - **Node ID** – The PortaUM server's hostname (recommended: `hostname.domainname`)
   - **NAS-IP-Address** – IP address of the UM gateway or PortaUM server.
   - **Auth. Translation rule** – You can just leave this empty.
   - **Radius Client** – Check this, since both PortaUM and the UM gateway will need to communicate with the billing.
   - **Radius Key** – Enter the radius shared secret here; must be the same **key** as you entered during PortaUM 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 PortaUM server uses multiple network interfaces, the value here should be the same as the NAS-IP-Address.
   - **POD Server** – Leave this unchecked for now.
4. Click **Save&Close.**

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

Create Tariff

You have already created all the necessary tariffs (for your customers and vendors) when setting up SIP services. Only one thing remains to be done, i.e. creating a special UM tariff which will be used for routing calls to the UM gateway.





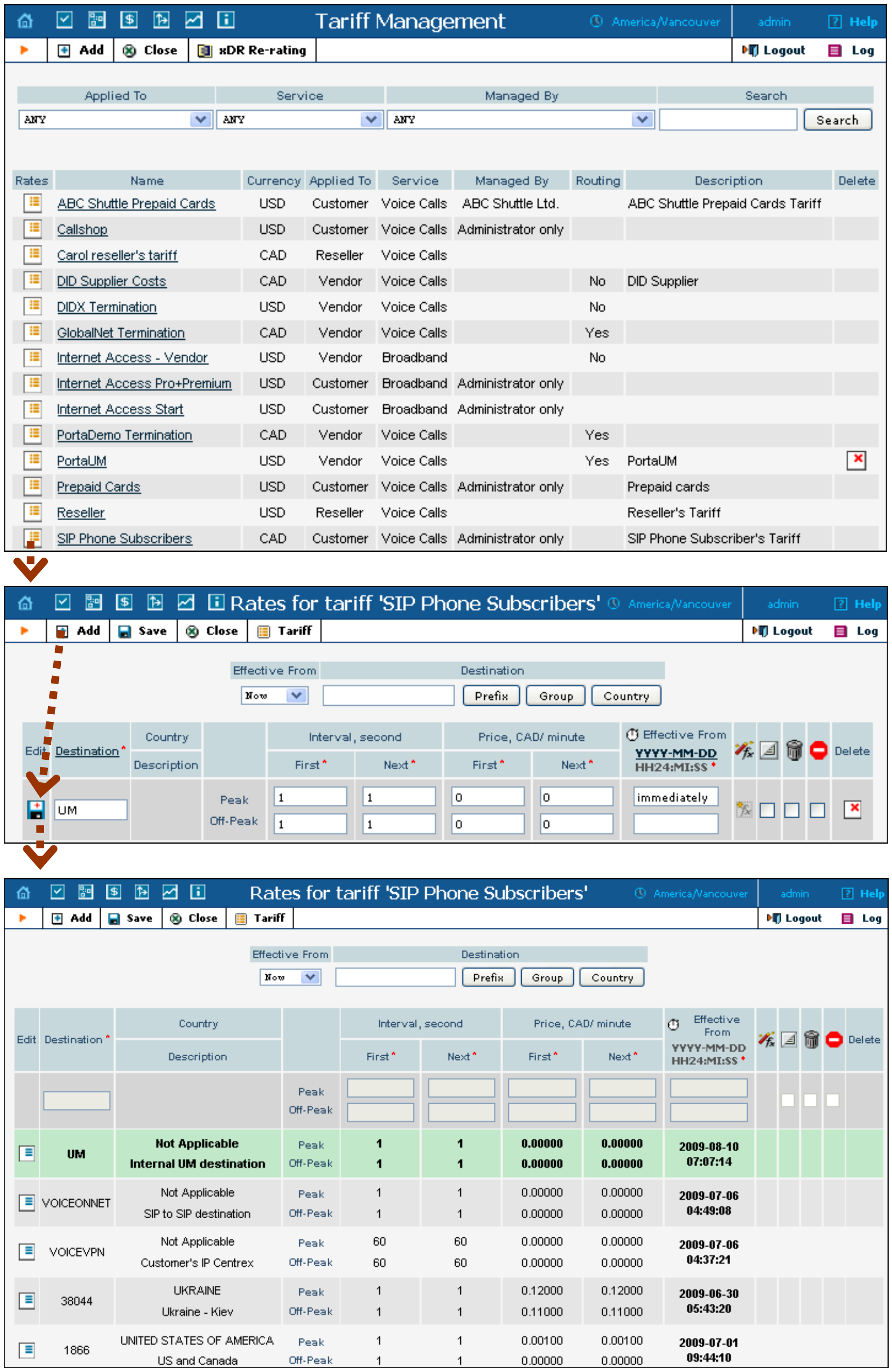
Rates	Name	Currency	Applied To	Service	Managed By	Routing	Description	Delete
	ABC Shuttle Prepaid Cards	USD	Customer	Voice Calls	ABC Shuttle Ltd.		ABC Shuttle Prepaid Cards Tariff	
	Callshop	USD	Customer	Voice Calls	Administrator only			
	Carol reseller's tariff	CAD	Reseller	Voice Calls				
	Conferencing	CAD	Customer	Conferencing	Administrator only			
	DID Supplier Costs	CAD	Vendor	Voice Calls		No	DID Supplier	
	DIDX Termination	USD	Vendor	Voice Calls		No		
	GlobalNet Termination	CAD	Vendor	Voice Calls		Yes		
	Internet Access - Vendor	USD	Vendor	Broadband		No		
	Internet Access Pro+Premium	USD	Customer	Broadband	Administrator only			
	Internet Access Start	USD	Customer	Broadband	Administrator only			
	PortaDemo Termination	CAD	Vendor	Voice Calls		Yes		
	PortaUM	CAD	Vendor	Voice Calls		No		
	Prepaid Cards	USD	Customer	Voice Calls	Administrator only		Prepaid cards	
	Prepaid Service	CAD	Customer	Voice Calls	Administrator only			
	Reseller	USD	Reseller	Voice Calls			Reseller's Tariff	
	SIP Phone Subscribers	CAD	Customer	Voice Calls	Administrator only		SIP Phone Subscriber's Tariff	

1. In the Billing section of the Admin-Index, choose **Tariffs.**
2. On the Tariff Management page, choose **Add.**
3. Fill in the New Tariff form. The most important fields are the following (leave the rest of the form empty):
   - **Name** – A short name for the tariff object. This is the name you will see later in the select menus.
   - **Currency** – Since this is a "fake" tariff, just choose the same currency as your **Base Currency**.
   - **Applied To** – Choose a **Vendor** here and uncheck the **Routing** check-box below this field.
   - **Service** – Choose **Voice Calls** here.
4. Click **Save&Close.**

Enter Rates

Modifying Tariffs for SIP Users

First of all, you should make sure that your customers are allowed to call voicemail. In order to do so, you should add a rate for calling the special voicemail number into the tariffs used for your SIP subscribers.

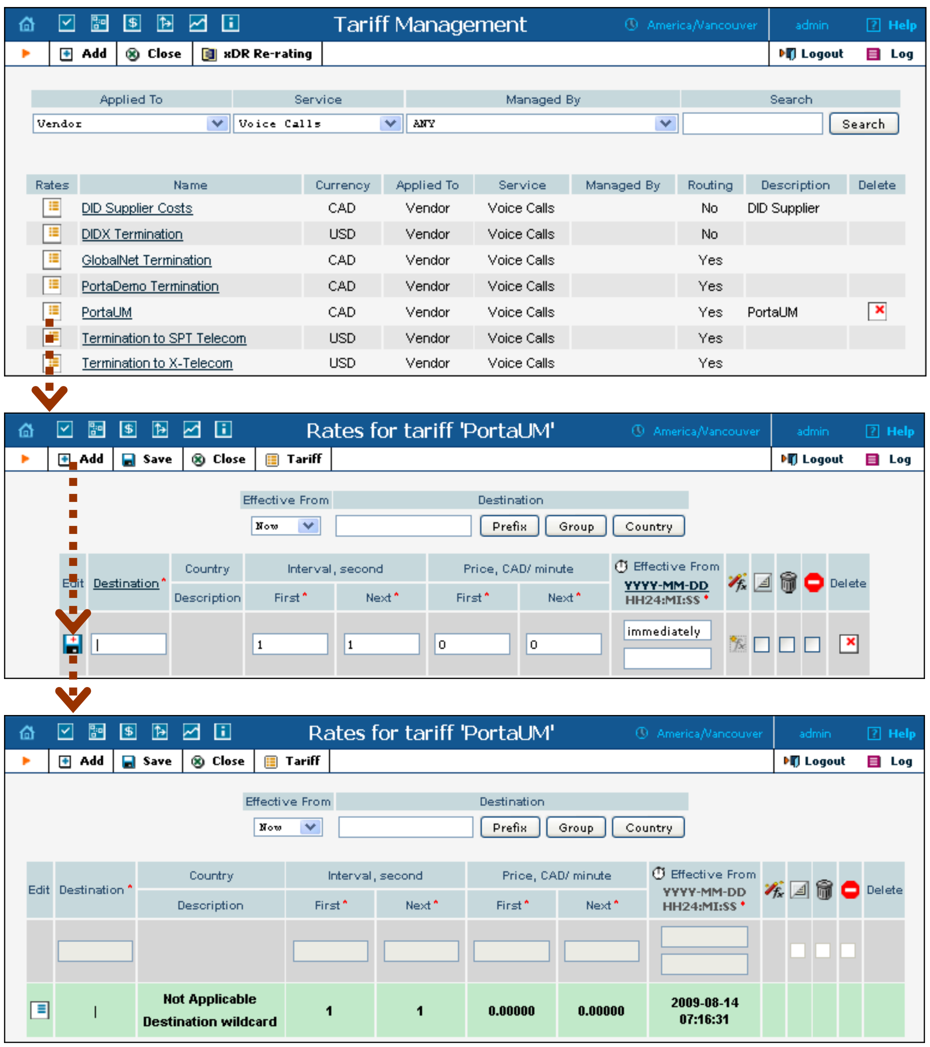


1. On the Tariff Management page you will see a list of the available tariffs. Click the **Rates** icon next to the name of a tariff. When you are in Tariff Management for a particular tariff, click on **Rates** in the toolbar.
2. On the **Edit Rates** screen, click **Add.**

3. Insert **UM** in the **Destination** field. Leave other fields in the form as they are, because there is no need to define prices for calling voicemail.
4. Click the **Save** button in the toolbar or the icon on the left side of the row.
5. Click **Close** to return to the Tariff Management screen.

Adding Rates for UM Vendor Tariff

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 UM vendor tariff.



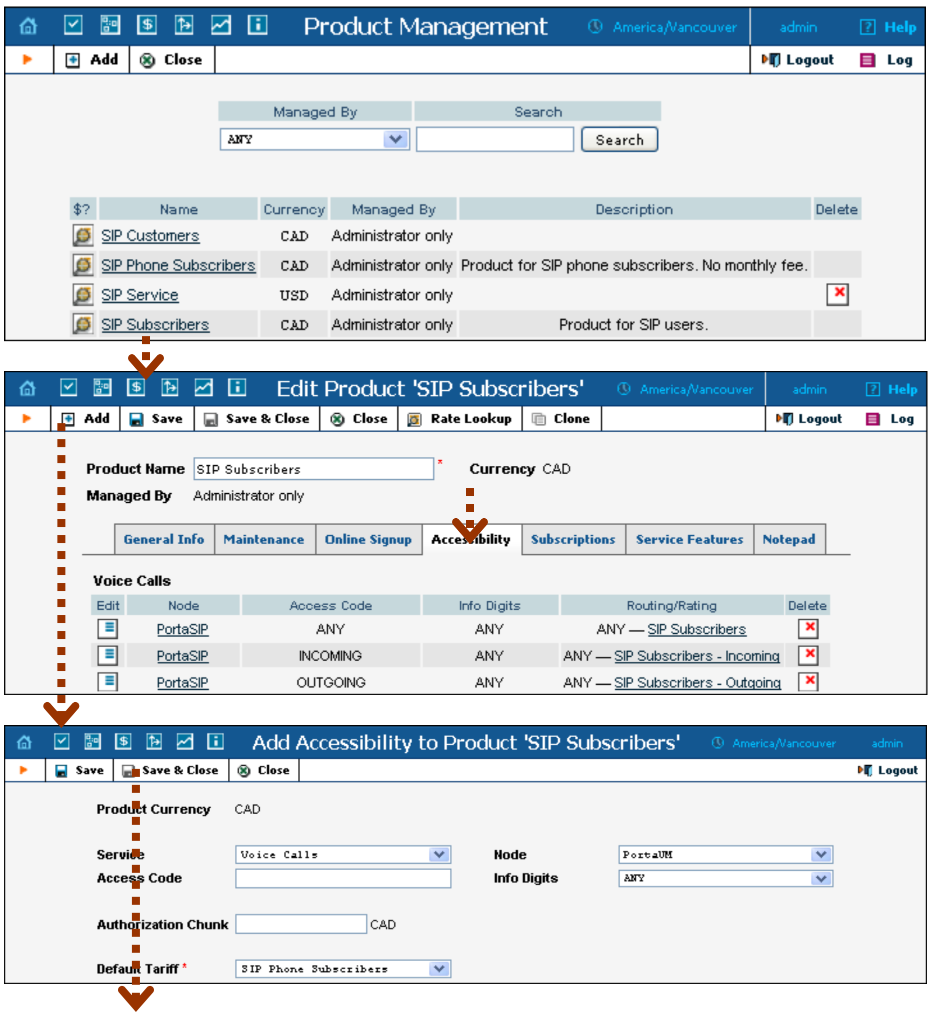
2. On the **Edit Rates** screen, click **Add**.
3. Fill in the required information:
   - **Destination** – Insert | here.
   - **Interval First** – Leave the default value.
   - **Interval Next** – Leave the default value.

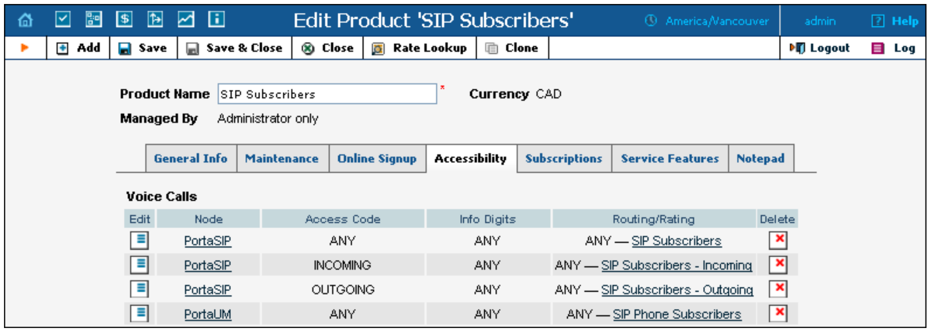
- **Price First** – Leave the default value 0.
- **Price Next** – Leave the default value 0.
- **Effective From** – Leave **immediately** in this field.
- The **Hidden**, **Forbidden** or **Discontinued** flags are optional.

4. Click **Save&Close**.

Modify Products

You already created all the necessary products for your customers when setting up SIP services. Only one thing remains to be done, i.e. ensuring that these products are allowed to use UM services. If a product's accessibility includes the PortaUM server, then accounts with this product are permitted to login into UM (via webmail, voicemail, or from their email client).



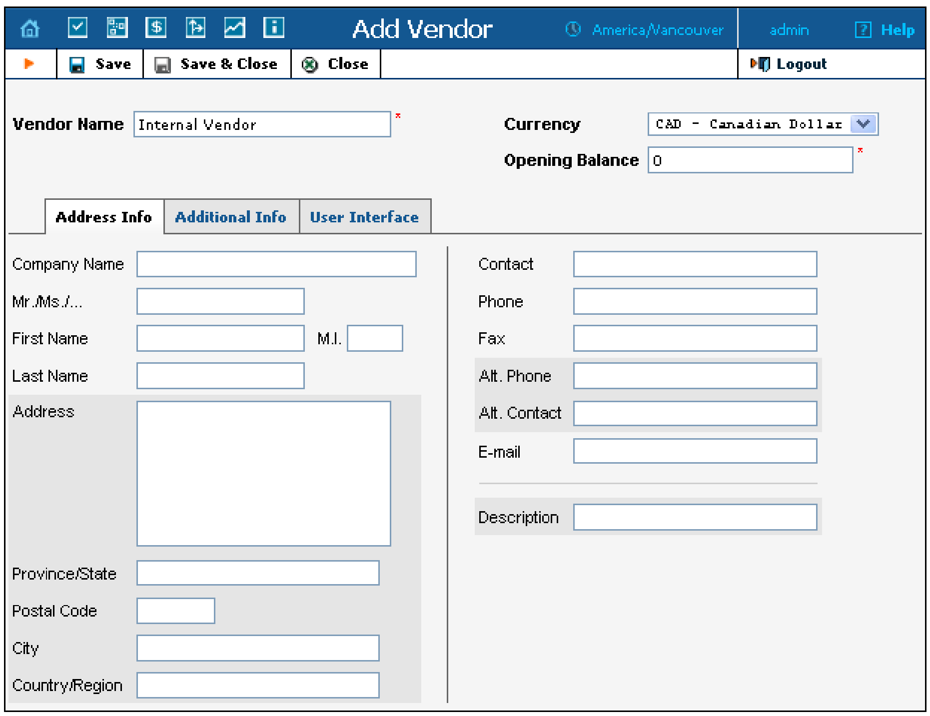


1. In the Billing section of the Admin-Index page, choose **Products**.
2. On the Product Management page, click on the name of the corresponding product.
3. Click on the **Accessibility** tab to edit this product's accessibility.
4. After the **Accessibility** tab has been selected, click on the **Add** icon.
5. Choose **Voice Calls** in the **Service** select menu.
6. In the **Node** field, select the node which represents your PortaUM server, and choose the appropriate tariff. (The actual tariff is not really important, since users will not be making any calls on the PortaUM server. It is best to choose the same tariff which is used to charge outgoing SIP calls.)
7. Click **Save** to save this accessibility entry.
8. Click **Close** to return to the Product Management page.
9. Repeat steps 2-8 to modify accessibility for all products which will include UM services.

Create an Internal Vendor

In order to route UM calls properly, and also to track statistics for such calls, it is advisable to create a special vendor and not mix it with any real vendors. Or, if you already have a vendor which represents your company (for example, for termination of SIP-to-SIP calls), then skip to **Define Connections**.

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



3. Fill in the New Vendor form. The most important fields are:

Main form (top):

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

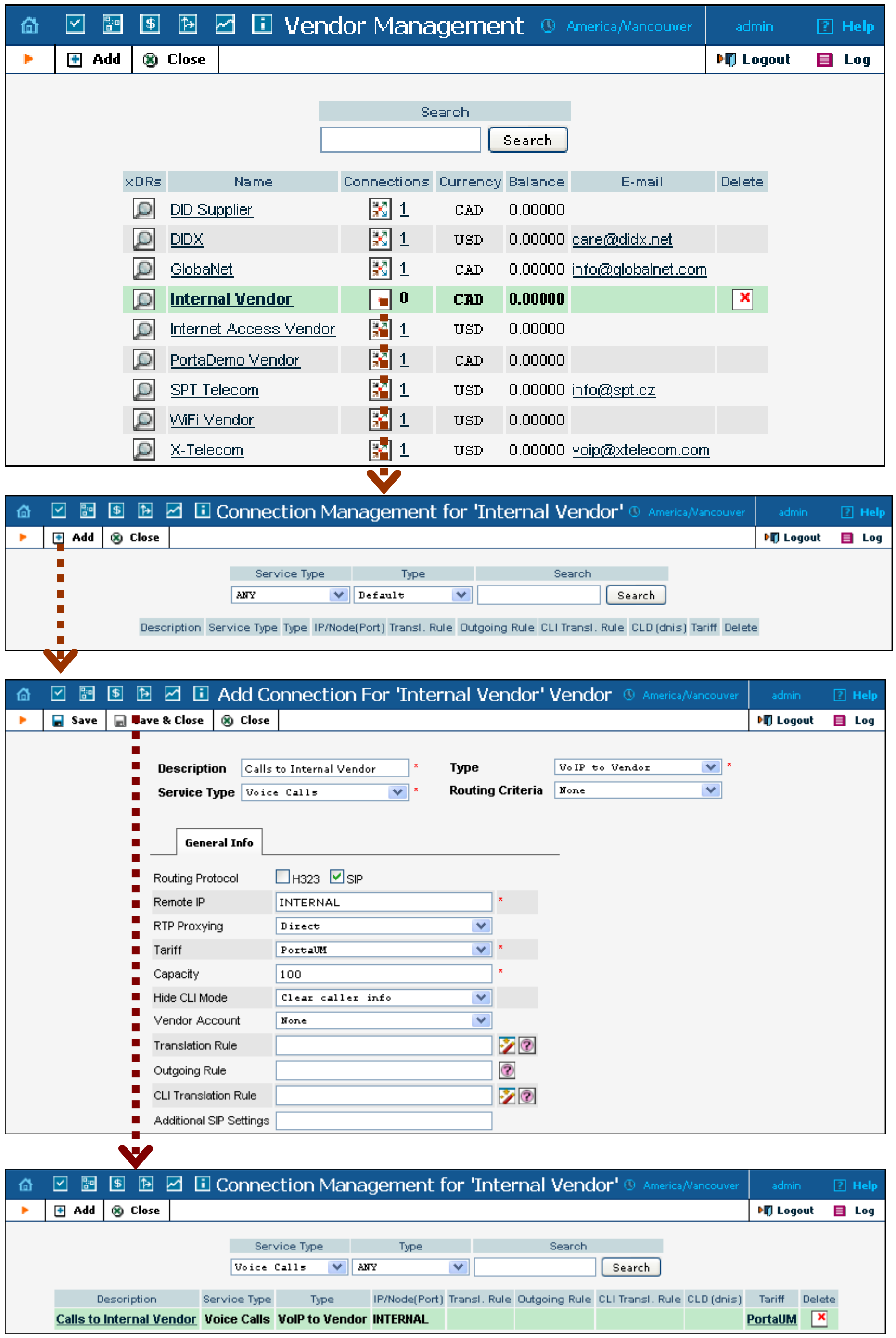
Additional info:

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

4. Click **Save&Close**.

Define Connections

1. In the Billing section of the admin interface, choose **Vendors**.
2. Click on the **Connections** icon next to the name of the vendor you created in the previous step.



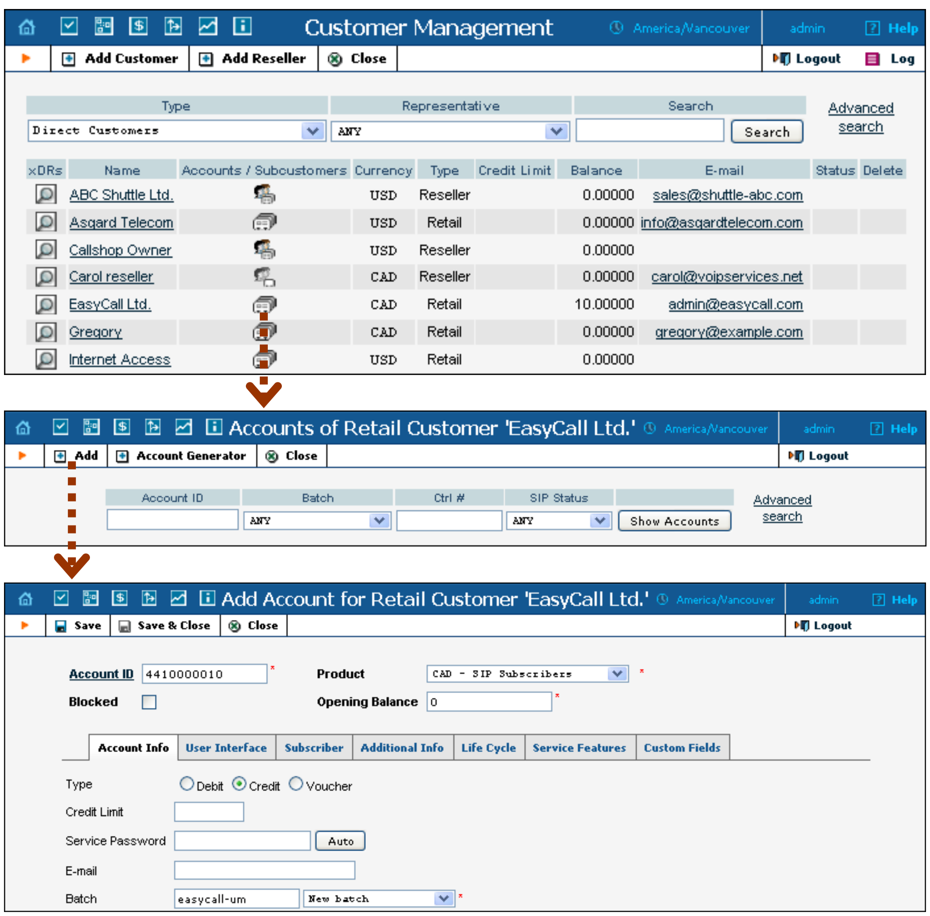
3. Press **Add** to add a new connection.

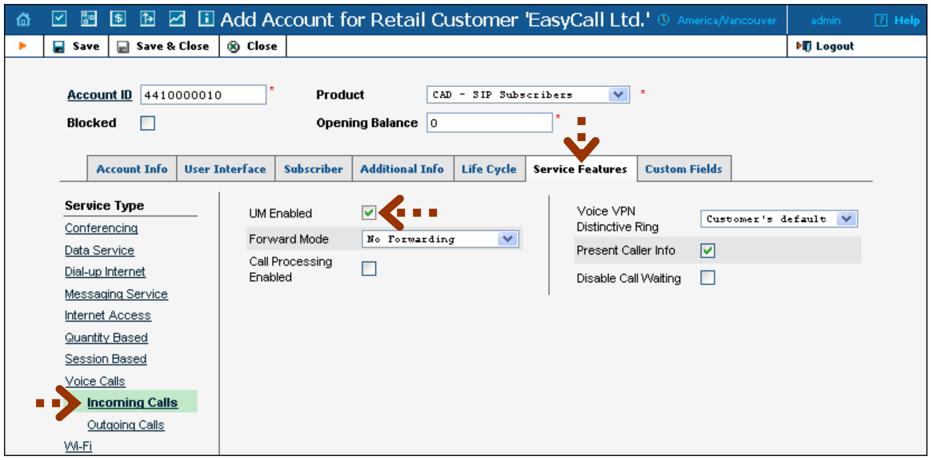
4. Choose the **VoIP to Vendor** connection type and **Voice Calls** service type by clicking on the corresponding drop-down menu.
5. Fill in the connection information. Type "INTERNAL" in the **Remote IP** field. Choose the PortaUM tariff you created earlier from the select menu in the **Tariff** column. Enter a comment in **Description**, and give the number of simultaneous calls your UM gateway can handle in **Capacity**.
6. Click **Save&Close**.
7. Click **Close** in order to exit to the Vendor Management screen.

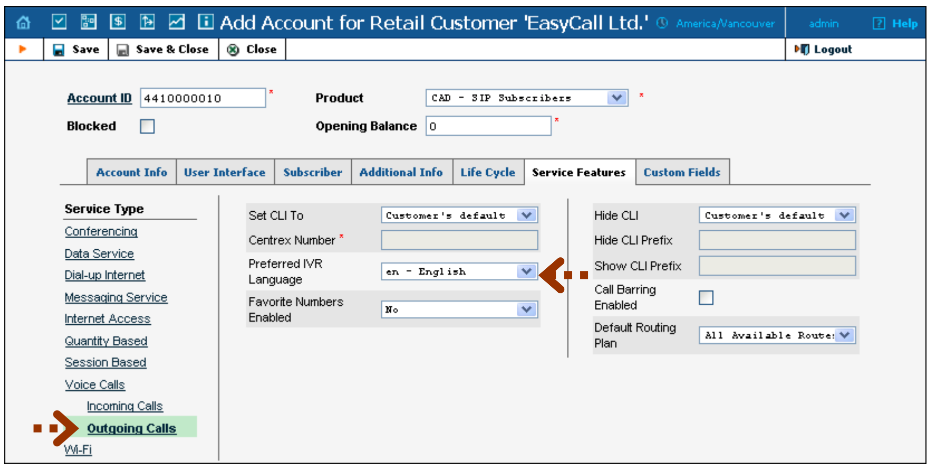
Create Accounts

Creating New Accounts

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







2. Next to the customer name, click on the icon (the one in the **Accounts** column). This will take you 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 log into the SIP server and receive incoming calls.
   - **Product** – Choose the product which you would like your account to have.
   - **Blocked** – You may create the account as blocked, but this is rarely done with SIP service accounts.
   - **Opening Balance** – The initial balance on the account.

Account Info

- **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, this means there is no credit limit for this account (but a customer credit limit may still apply).

- **Service Password** – The Account ID and this password will be used to authenticate SIP server login.
- **Email** – Enter the account owner's email address here. If he ever forgets his password for the web self-care, he will be able to reset it, and a new password will be sent to this email address. You can also leave this field empty.
- **Batch** – 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.

User Interface

- **Login** – Account login to web self-care pages; can be the same as account ID.
- **Password** – Password for web self-care pages.
- **Time Zone** – When an account owner (SIP services subscriber) accesses the web self-care pages to view a list of his calls, the time can be shown in the time zone most appropriate for him.
- **Web Interface Language** – The language to be used on the customer's self-care web interface.

Life Cycle

- **Activation Date** – Account activation date.
- **Expiration Date** – Account expiration date.
- **Life Time** – Relative expiration date; the account will expire on "first usage date" + "lifetime" days. If you do not want to use this feature, leave the field blank.

Service Features

In the **Service Type** column, select **Incoming Calls** under **Voice Calls**.

- **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.

Now select **Outgoing Calls**.

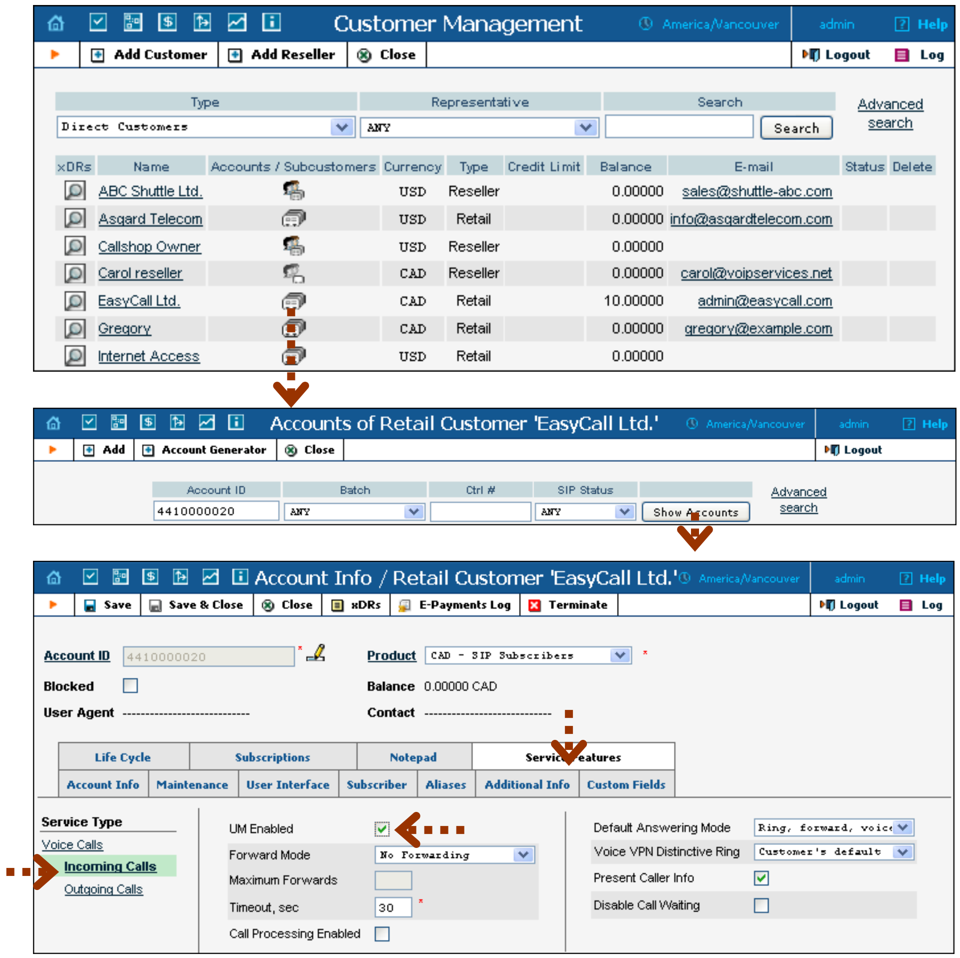
- **UM Enabled** – Check this box if the account has unified messaging services (e.g. voicemail) enabled.

5. Click **Save&Close**. You will see a confirmation screen saying that a new account has been created.

Enable UM Services for an Existing Account

1. If you are not already there, go to the **Customers** screen (the one containing a list of customers).

2. Next to the customer name, click on the icon (the one in the **Accounts / Subcustomers** column). This will take you to the account management for that customer.
3. Type the account ID in the **Account ID** field and click on the **Show Accounts** button.



4. Click on the **Service Features** tab.
5. In the **Service Type** column, select **Incoming Calls** under **Voice Calls**.
6. Click on the **UM Enabled** checkbox to activate UM services for this account.
7. Click **Save&Close.**

Setting up and Using Prepaid IVR on PortaUM

Sometimes an incoming call for a prepaid card service will arrive to your network via IP. For instance, say you bought an access number in a country where you do not own any network infrastructure; so instead of using E1 or T1 lines, a telco in that country will forward calls to you using SIP. In this case using a Cisco GW may be cumbersome, since you first have to convert the IP call to PSTN and then send it back via IP. Instead, you can run the built-in prepaid application on PortaUM which, combined with a PSTN access number delivered directly to PortaSIP via IP, allows you to offer a purely IP-based solution for prepaid calling card services.

This chapter assumes that you have already set up UM services as per the previous section, and have verified that they are working.

Configure Prepaid Application

First you need to configure general parameters for the prepaid application in **porta-um.conf**. These include, for instance, the special DNIS that the IVR runs on, the virtual PortaBilling environment where this application is used, languages, and prepaid card length. For a complete overview of the available options, please also consult the inline documentation at **porta-um.conf**.

Set up DNIS number

Set up the DNIS number for the virtual PortaBilling environment where you want to use this application. This uses the following format:

```
Ext<N>=<i_env>:<extension>:<ANI translation rule>
```

where N is an unique arbitrary number. For instance, if the ID of your environment is 1 and the DNIS will be 111222333, you should enter:

```
Ext1=1:111222333:
```

The option of configuring an ANI translation rule for this application will only be available if the parameter `AniAuth` is set to 'yes'.

Of course, you can have multiple entries in order to register the prepaid application on more than one internal DNIS mailbox. This allows you to run the application in multiple environments, on multiple access numbers, and/or with potentially different languages or ANI translation rules, according to your requirements. Just register these additional extensions as “Ext2”, “Ext3”, and so on.

Configure languages

Configure which languages should be used for the IVR prompts. This option uses the following format:

```
Languages<N>=<i_env>:dnis_number:<lang_1>/<lang_2>/<lang_3>
```

where N is an unique arbitrary number. For instance, if the ID of your virtual PortaBilling environment is 1, the DNIS is 111222333, and you would like to use Spanish voice prompts, you should enter:

```
Languages1=1:111222333:es:
```

where N is an unique arbitrary number. You can also configure more than one language for the same DNIS, as follows:

```
Languages1=1:111222333:en/ru:
```

In this case, the IVR’s first prompt will be the language selection prompt “Please select 1 for English, 2 for Russian”.

Note: Language IVR files (audio files) must first be uploaded before they can be used in any IVR applications. At the moment, only English prompts are provided by PortaOne, with instructions for adding other languages available upon request.

Configure additional options

Optionally, you can configure any of the following additional parameters.

Parameter	Description
`MaxCardLen`	Maximum length of card number; default: 11.
`MinCardLen`	Minimum length of card number; default: 11.
`AniAuth`	Is ANI authentication enabled? Default: no.
`ManualAuth`	Is manual authentication enabled? Default: yes.
`MaxLoginAttempts`	Maximum number of attempts to enter a card number; default: 3.
`MaxDialAttempts`	Maximum number of dial attempts within one session; default: 3.
`TollFreeAllowed`	If this parameter is set to ‘false’, the prepaid application will exit immediately if the account has insufficient funds. Default: false.

Configure PSTN access

Once you have configured the prepaid application on PortaUM, you need to give your customers access to it via PSTN. Of course, the DNIS_number that was configured in the first step can already be called from any SIP phone on your network. However, the purpose of this prepaid application will most likely be to give customers access to your network via a PSTN access number.

1. Please follow the instructions in our "PortaSIP Handbook: SIP Services", specifically section 2, entitled "Setting Up PSTN-to-SIP Services".
2. Create a service customer and an account with an 'INTERNAL' Account ID under this customer:

Since no SIP UA has been registered to this account, the call will go directly to PortaUM.

Restart UM services

Restart UM services so that the changes to porta-um.conf will take effect. To stop or start UM services, use the following commands:

```
sudo /usr/local/etc/rc.d/voice-mail.sh stop
sudo /usr/local/etc/rc.d/voice-mail.sh start
```

PortaBilling Configuration for Prepaid Card Service

You should perform the billing configuration according to the instructions in **PortaBilling100 Handbook: Prepaid Services**, the only exception being that you will use your PortaUM node instead of a gateway.

Using ANI Authentication for PIN-less Dialing

In order to support PIN-less dialing, the prepaid application can be configured with the parameter AniAuth set to 'yes'. If an incoming call matches one of your accounts by ANI, the customer will immediately be asked to enter a call destination, skipping the PIN number entry step.

Using an external ANI for authorization only

You may wish to allow customers to use their account not just by dialing from their registered SIP account (regular ANI authentication) but also from any other kinds of phones (cell phone, regular home phone). In order to successfully authenticate these ANIs, you need to create them as regular accounts under your customers.

However, to prevent PortaBilling from trying to terminate actual calls to these "ANI only" accounts, it is recommended that you adopt a naming convention that allows you to easily distinguish these accounts from your regular SIP accounts and to keep PortaBilling from routing to these accounts.

One suggested naming convention that does all of this is using the prefix 'ani' for your ANI authentication accounts (naturally, you can choose a different prefix as well). For instance, if you would like to allow your customer to use his cell phone with ANI 16041112222 for ANI authentication, you would create the account ani16041112222 under this customer.

Set up ANI translation rule

If you have adopted a naming convention for external numbers which are to be authenticated by ANI, you need to set up an ANI translation rule in order to successfully authenticate these accounts.

Assuming that the prefix you have chosen is 'ani', you will need to change your prepaid application configuration from:

```
Ext1=1:111222333:
```

to:

```
Ext1=1:111222333:s/^/ani/;
```

The prepaid application will use the incoming ANI, prefixing it with 'ani' and then trying to look up that account in the database.

Using UM Services

PortaUM allows Internet Telephony Service Providers (ITSPs) to offer their subscribers the ability to process emails and manage voicemail from within a web browser or using a favorite email client.

The PortaUM End-User Manual is available from the PortaOne website: www.portaone.com/support/documentation/

PortaUM users can also access their mailbox via a telephony interface, which allows them to listen to voice messages. UM options for accessing the mailbox can also be configured by phone.

2. Setting up and Using Conferencing Services

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

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.

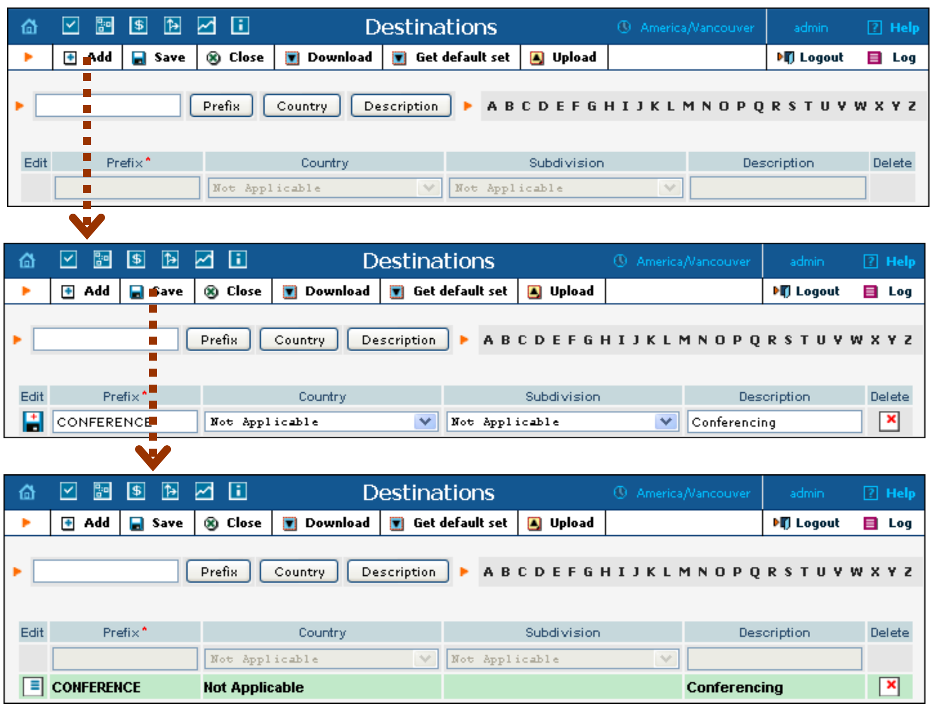
Operation	Done
General configuration	
General configuration has already been done according to the instructions in the **PortaSwitch Handbook: Basic SIP Services**.	[ ]
Create a symbolic CONFERENCE destination.	[ ]
Associate access numbers with applications.	[ ]
Network configuration	
Create a node for your PortaBridge server.	[ ]
Rating configuration	
Create tariff A, which will apply to conferencing services used by your customers (make sure this tariff has the **Conferencing** service assigned!).	[ ]
Enter rates in tariff A for access numbers used for the conferencing service.	[ ]
Specify rates for dialing into the conferencing bridge for your "normal" SIP accounts.	[ ]
Create a new accessibility entry in your existing product A, using the conferencing service, PortaBridge node and tariff A.	[ ]
Create tariff B, which will define call routing to PortaBridge. (Make sure this tariff has the **Routing** type and the **Conferencing** service assigned!)	[ ]
Enter rates in tariff B for access numbers used for the conferencing service.	[ ]
Create a connection for your "internal" vendor (the one created in the basic SIP service configuration for handling SIP on-net calls) using tariff B.	[ ]
Account provisioning	
Enable the conferencing service for an existing account (with product A).	[ ]
Testing	
Log in to the PortaBridge self-care interface as an account owner and create a new conference.	[ ]
Dial the conference access number from an IP phone and log in into the conference.	[ ]

General Configuration

This is identical to the procedure described in the *Setting up Standard SIP Services* chapter in the **PortaSwitch Handbook: SIP Services**. Please follow the instructions provided there.

Add a Symbolic Destination

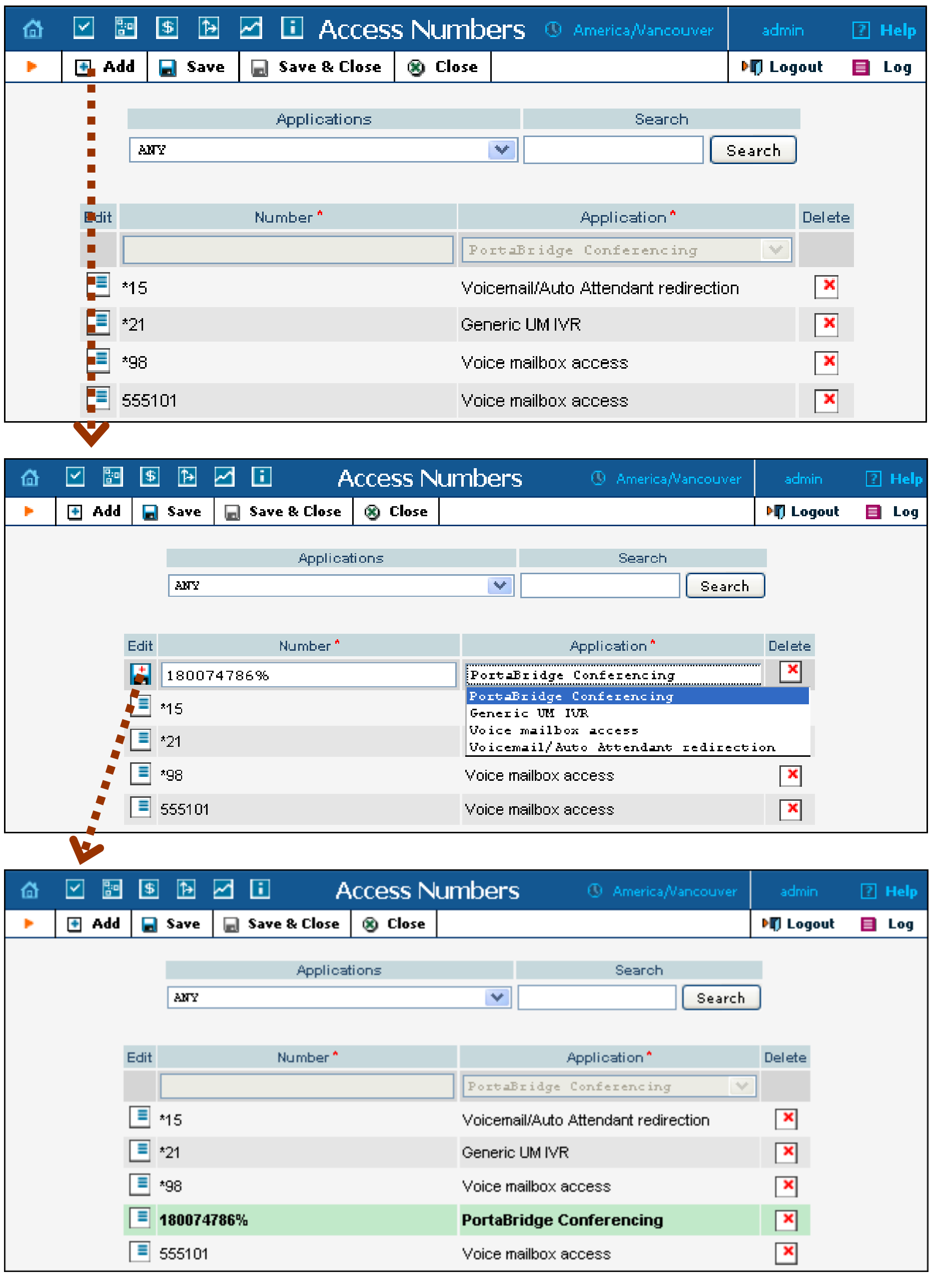
You will need to add a symbolic CONFERENCE destination.



1. In the Management section of Admin-Index, choose **Destinations**.
2. Click on the **Add** button.
3. Enter CONFERENCE for **Prefix**, and leave **Not Applicable** for the country and country subdivision. Put a comment in the **Description** column that clearly identifies this as a special prefix assigned to the conferencing service.
4. Click **Save.**

Associate Access Numbers with Applications

You need to define which application will be launched if your customers dial access numbers to the conferencing service.

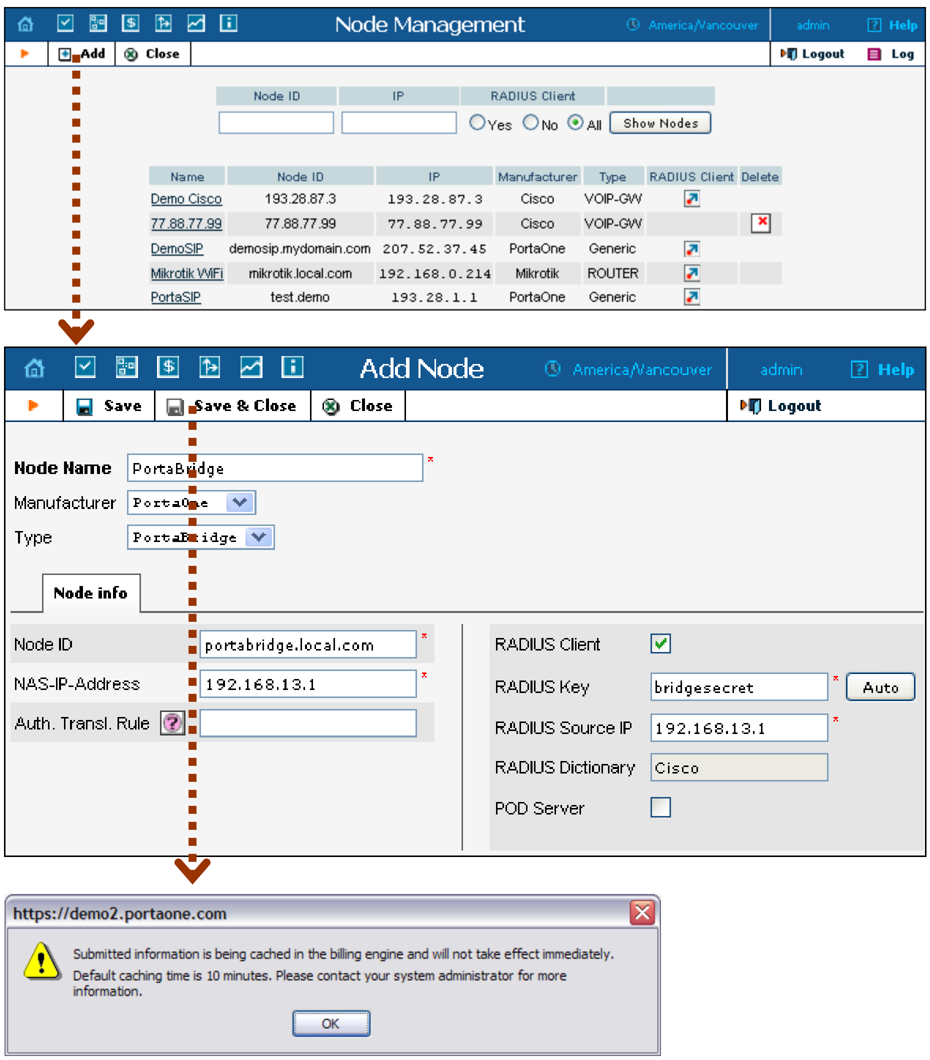


1. In the Routing section of Admin-Index, choose **Access Numbers**.

2. Click on the **Add** button.
3. In the **Number** field, enter your conferencing number. In the **Application** drop-down box, select **PortaBridge Conferencing**.
4. Click **Save**.

Create a PortaBridge Node

You now need to enter your PortaBridge server as a node. PortaBilling requires some key information about your network equipment, such as the IP address, Node ID, Radius shared secret, and so on.



1. In the Networking section of the Admin-Index page, choose **Nodes.**
2. In the Node Management window, click the **Add** icon.
3. Fill in the New Node form:

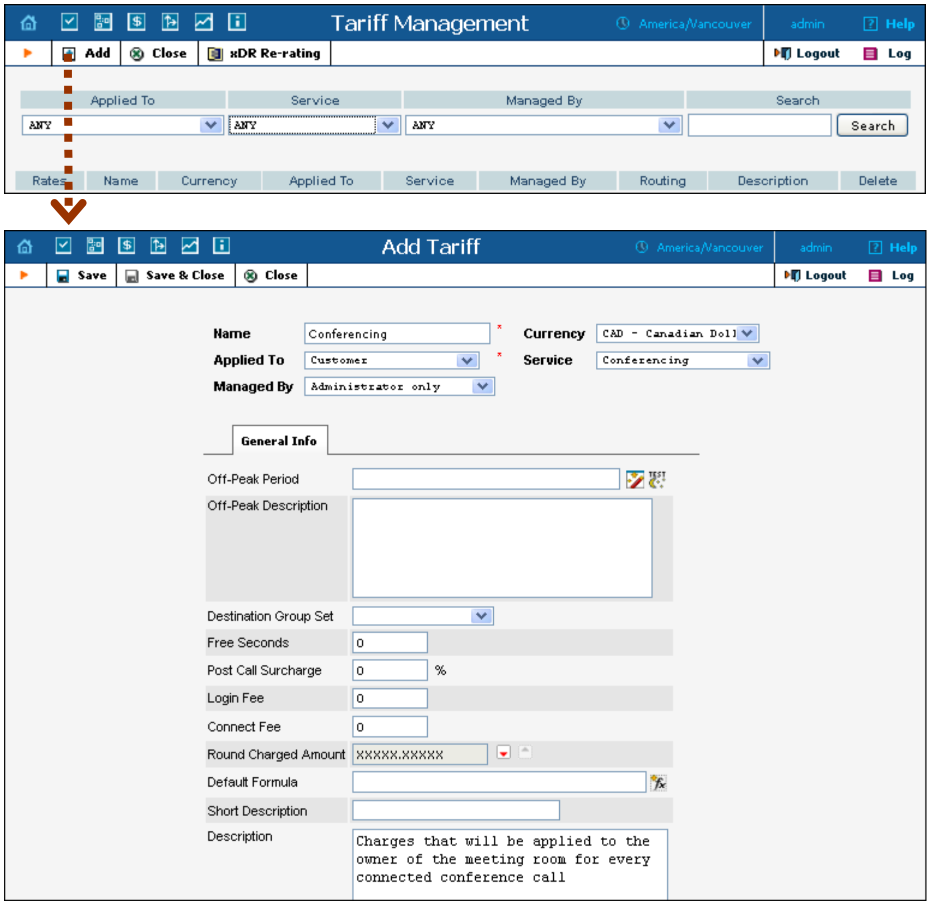
- **Node Name** – A short descriptive name for your PortaBridge server (this will be used in the select menus).
- **Manufacturer** – Select **PortaOne**.
- **Type** – VoIP node type; select **PortaBridge**.
- **Node ID** – The PortaBridge server's hostname (recommended: `hostname.domainname`).
- **NAS-IP-Address** – The IP address of the PortaBridge server.
- **Auth. Translation Rule** – Leave this blank.
- **Radius Client** – Check this, as **PortaBridge** will need to communicate with billing.
- **Radius Key** – Enter the Radius shared secret here; this must be the same key which you entered during PortaBridge 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 PortaBridge server uses multiple network interfaces, the value here should be the same as the NAS-IP-Address.
- **POD Server** – Leave this unchecked for now.

4. Click **Save&Close**.

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

Create a Conferencing Charge Tariff

This tariff will be used to charge your customers who own a conference (meeting room) for every incoming call to that conference.



1. In the Billing 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 will be used in the select menus).
    - **Currency** – Indicates in which currency pricing information is given. All pricing information for a single tariff must be defined in the same currency.

NOTE: If you plan to bundle the conferencing service with some existing product, this tariff must have the same currency as that product.

- **Applied To** – Choose the **Customer** in the **Applied To** field.
- **Managed By** – Choose **Administrator only** here (this option is only visible after you select **Applied To Customer** above).

- ○ **Service** – Choose **Conferencing** here.
- ○ The rest of the fields are identical to the tariff configuration for SIP services you performed earlier.

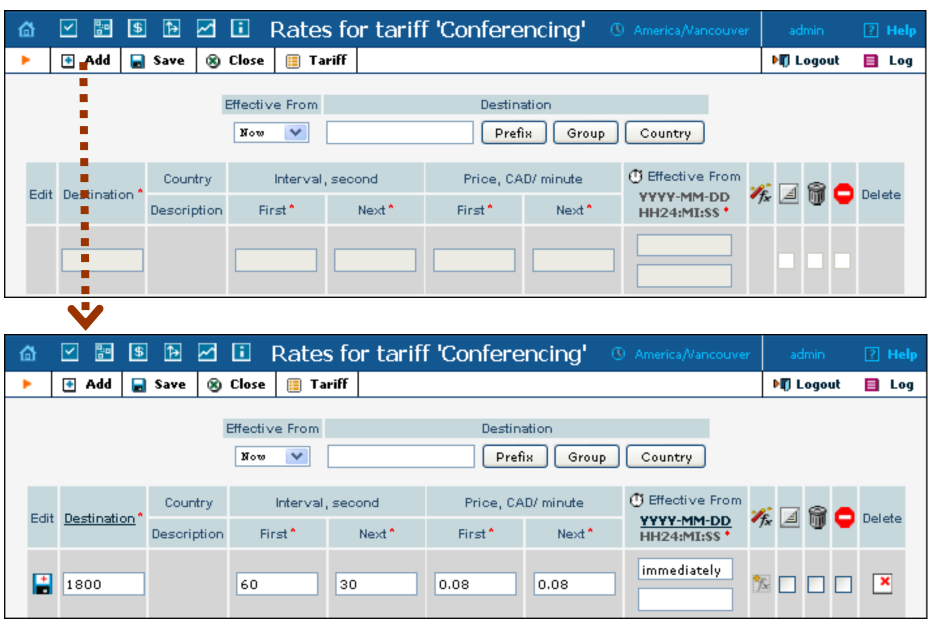
4. Click **Save**.

Enter Rates

Enter the rates you will charge your customers for conferencing service use. You may charge them a different rate based on access number, e.g. if someone calls a US local access number, the rate is $0.03/min, while it is $0.04/min to use a local access number in the UK, and $0.06/min to use a US toll-free access number.

Managing Rates Online

This is very convenient for maintaining existing rate tables and for reference purposes. For new price lists or major updates, the offline method is better.



1. On the Tariff Management page you will see a list of the available tariffs. Click the **Rates** icon next to the name of the tariff. When you are in Tariff Management for a particular tariff, click on **Rates** in the toolbar.
2. On the **Edit Rates** screen, click **Add**.
3. Fill in the required information:
   - ○ **Destination** – A destination prefix may be entered directly, e.g. **55** for Brazil, 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.

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

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

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

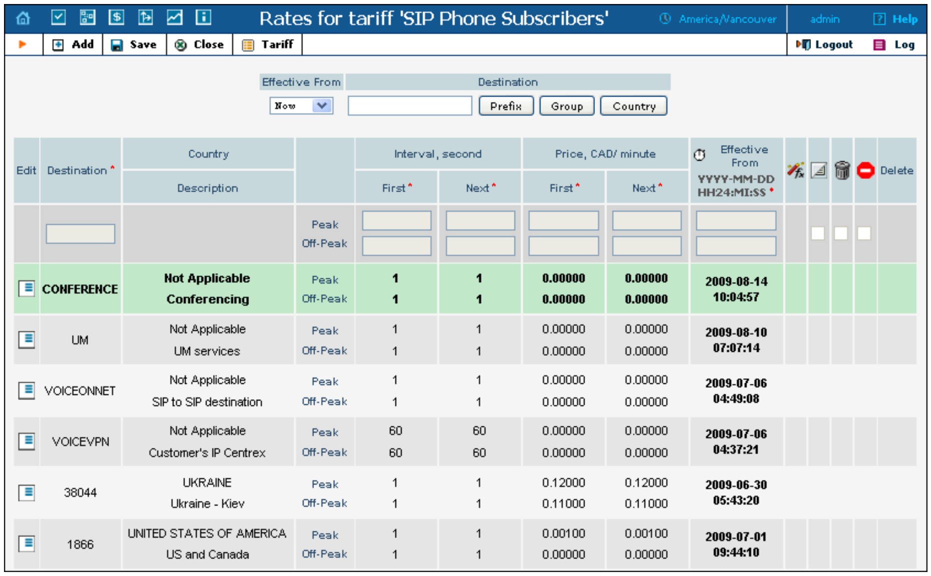
NOTE: When using the calendar, you can specify that the date you are entering is in a different time zone than your present one. PortaBilling will then automatically adjust the time.

- 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.

Modify Tariffs for SIP Users

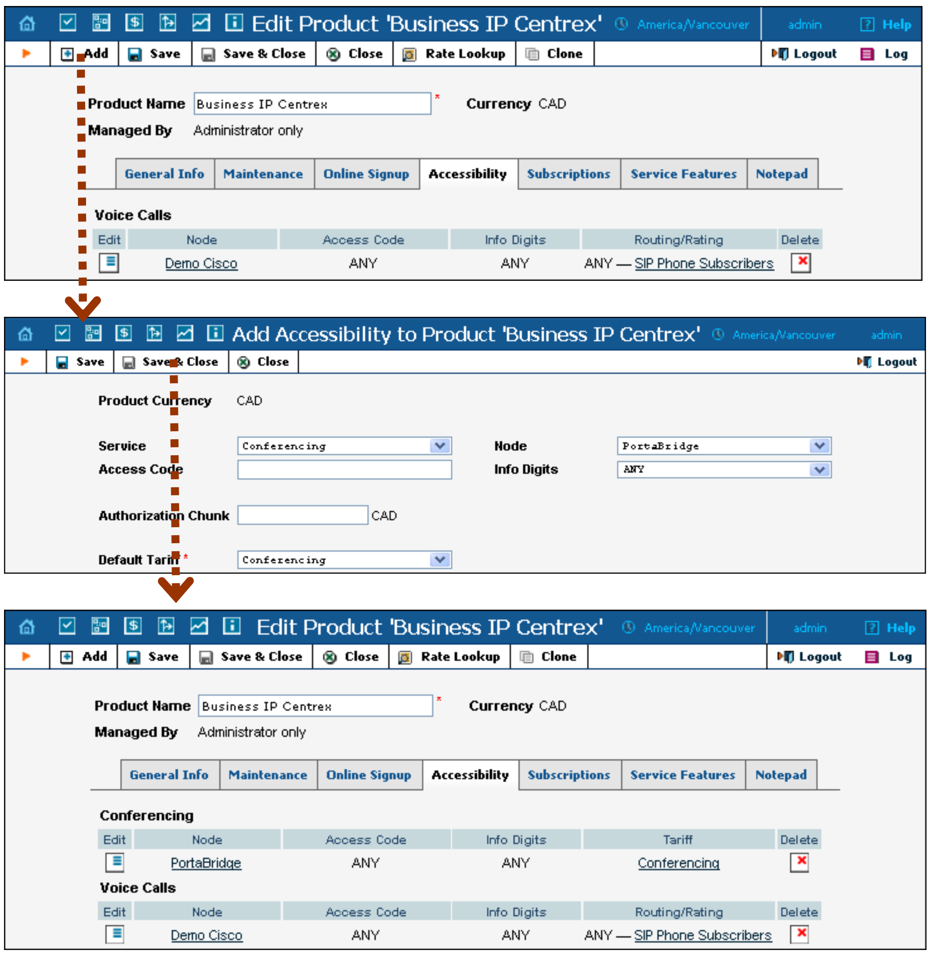
In order for customers to be able to call a conference access number from their IP phones, this needs to be allowed by their tariff plan. Add rates for the CONFERENCE destination you created earlier, as follows:



Edit	Destination *	Country / Description		Interval, second First *	Interval, second Next *	Price, CAD/ minute First *	Price, CAD/ minute Next *	Effective From YYYY-MM-DD HH24:MI:SS *	Delete
			Peak / Off-Peak						
	CONFERENCE	Not Applicable / Conferencing	Peak / Off-Peak	1 / 1	1 / 1	0.00000 / 0.00000	0.00000 / 0.00000	2009-08-14 10:04:57	
	UM	Not Applicable / UM services	Peak / Off-Peak	1 / 1	1 / 1	0.00000 / 0.00000	0.00000 / 0.00000	2009-08-10 07:07:14	
	VOICEONNET	Not Applicable / SIP to SIP destination	Peak / Off-Peak	1 / 1	1 / 1	0.00000 / 0.00000	0.00000 / 0.00000	2009-07-06 04:49:08	
	VOICEVPN	Not Applicable / Customer's IP Centrex	Peak / Off-Peak	60 / 60	60 / 60	0.00000 / 0.00000	0.00000 / 0.00000	2009-07-06 04:37:21	
	38044	UKRAINE / Ukraine - Kiev	Peak / Off-Peak	1 / 1	1 / 1	0.12000 / 0.11000	0.12000 / 0.11000	2009-06-30 05:43:20	
	1866	UNITED STATES OF AMERICA / US and Canada	Peak / Off-Peak	1 / 1	1 / 1	0.00100 / 0.00000	0.00100 / 0.00000	2009-07-01 09:44:10	

Add the Conferencing Service to Your Product

You need to ensure that your product is allowed to use the conferencing service. This can be done using the product's accessibility. 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.

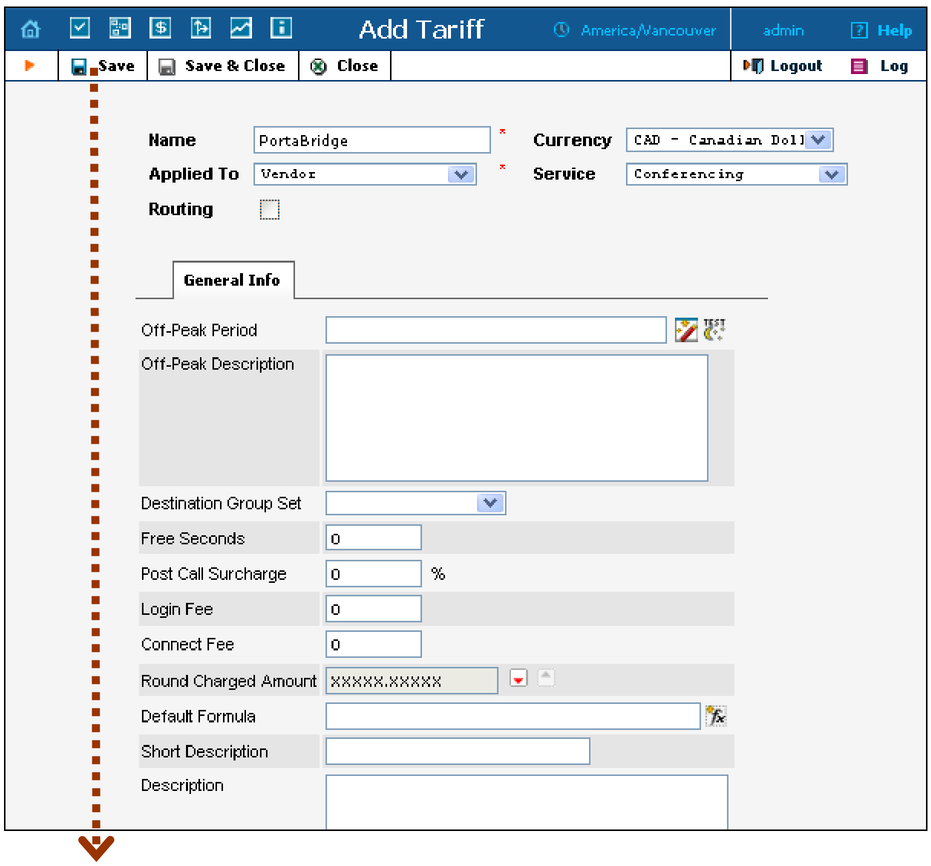


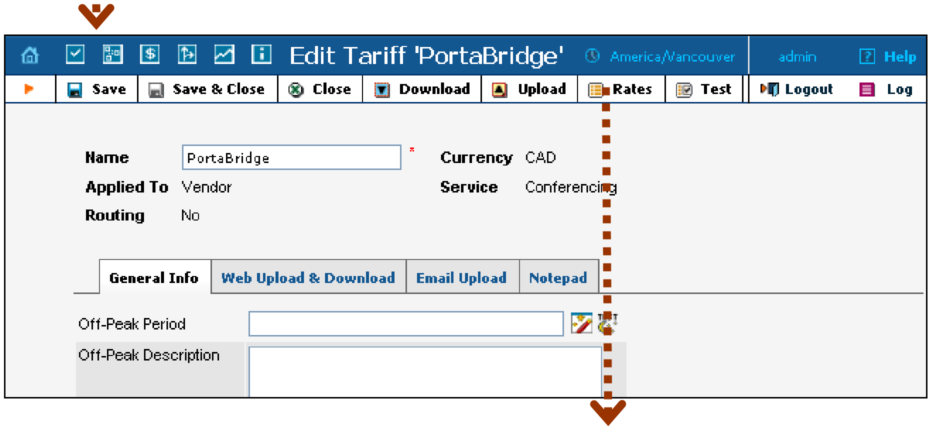
1. On the Product Management page, click the product name to enter the **Edit Product** form.
2. When the Accessibility tab is selected, click on the **Add** icon.
3. Choose **Conferencing** in the **Service** drop-down box.
4. Select the PortaBridge node and choose the conferencing tariff you created earlier.
5. The **Access Code** should be left empty for the conferencing service.

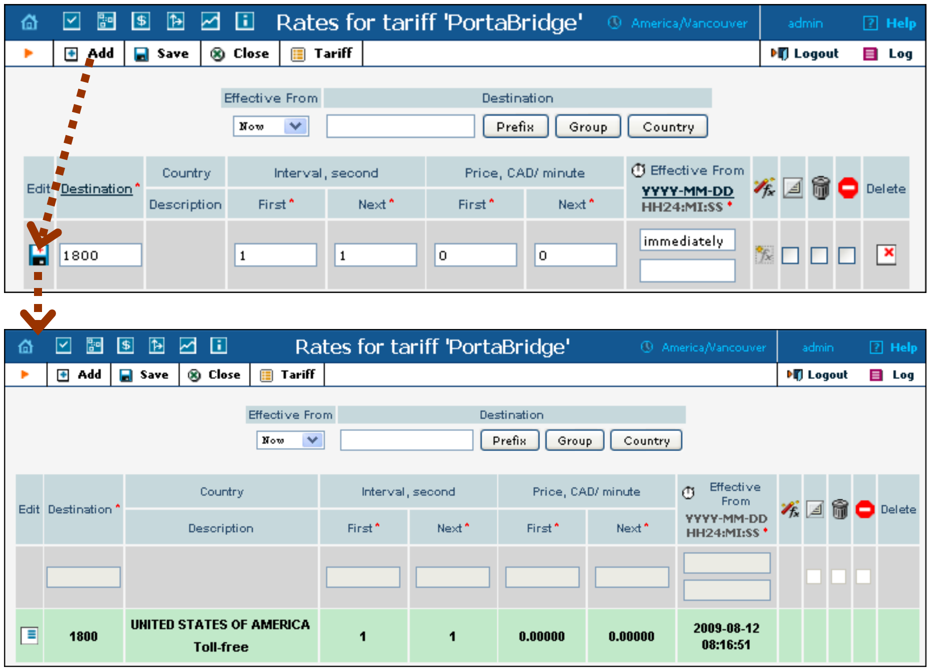
6. The **Info Digits** field is not applicable to the conferencing service; just leave it empty.
7. Click **Save** to save this accessibility entry.

Create a Call Routing Tariff and Add Rates

When a conference bridge is dialed from an IP phone, or an access number from PSTN, and the call arrives to your network, PortaSwitch has to recognize that it should be forwarded to PortaBridge. In order to do this properly, PortaSwitch must be able to determine routing in the same way as it does for any other call in the system. Therefore, there needs to be a routing tariff with a destination matching the dialed number. Later on, this tariff should also be assigned to a connection.





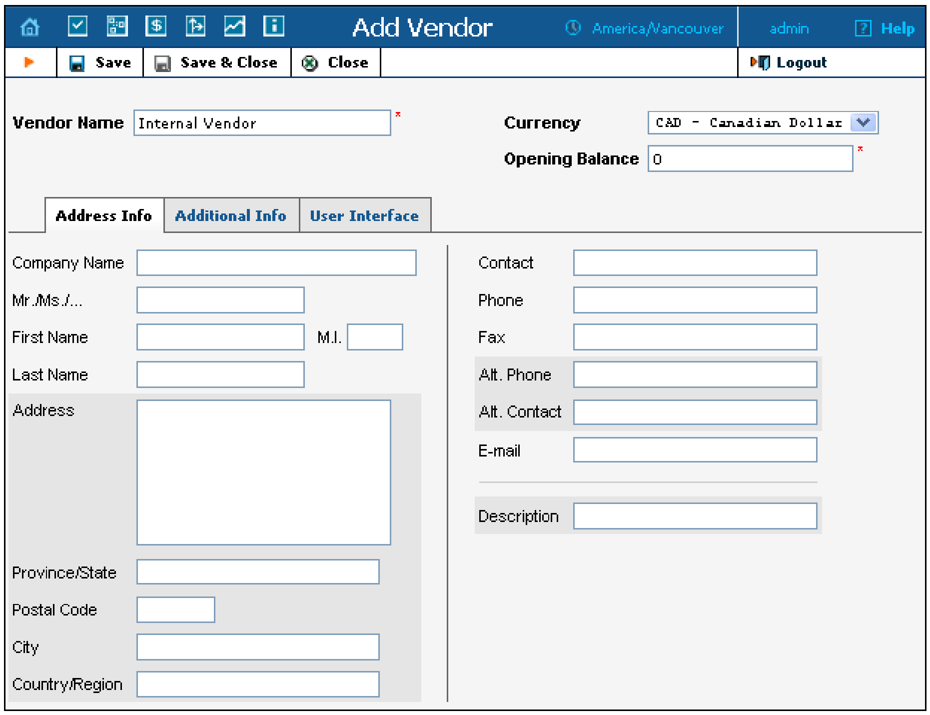


1. In the Billing section of the Admin-Index, choose **Tariffs.**
2. On the Tariff Management page, choose **Add.**
3. Fill in the New Tariff form. The most important fields are the following (leave the rest of the form empty):
   - **Name** – A short name for the tariff object. This is the name you will see later in the select menus.
   - **Currency** – Since this is a "fake" tariff, just choose the same currency as your **Base Currency**.
   - **Applied To** – Choose a **Vendor** here and uncheck the **Routing** check-box below this field.
   - **Service** – Choose **Conferencing** here.
4. Click **Save.**
5. On the **Edit Tariff** screen, click the **Rates** icon.
6. On the **Edit Rates** screen, click **Add**.
7. Fill in the required information:
   - **Destination** – Enter your conferencing destination here.
   - **Interval First** – Leave the default value.
   - **Interval Next** – Leave the default value.
   - **Price First** – Leave the default value0.
   - **Price Next** – Leave the default value0.
   - **Effective From** – Leave **immediately**.
   - The **Hidden**, **Forbidden** or **Discontinued** flags are optional.
8. Click **Save&Close**.

Create an Internal Vendor

In order to route conferencing calls properly, and also to track statistics for such calls, it is advisable to create a special vendor and not combine it with any real vendors.

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



3. Fill in the New Vendor form. The most important fields are:

Main form (top):

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

Additional info:

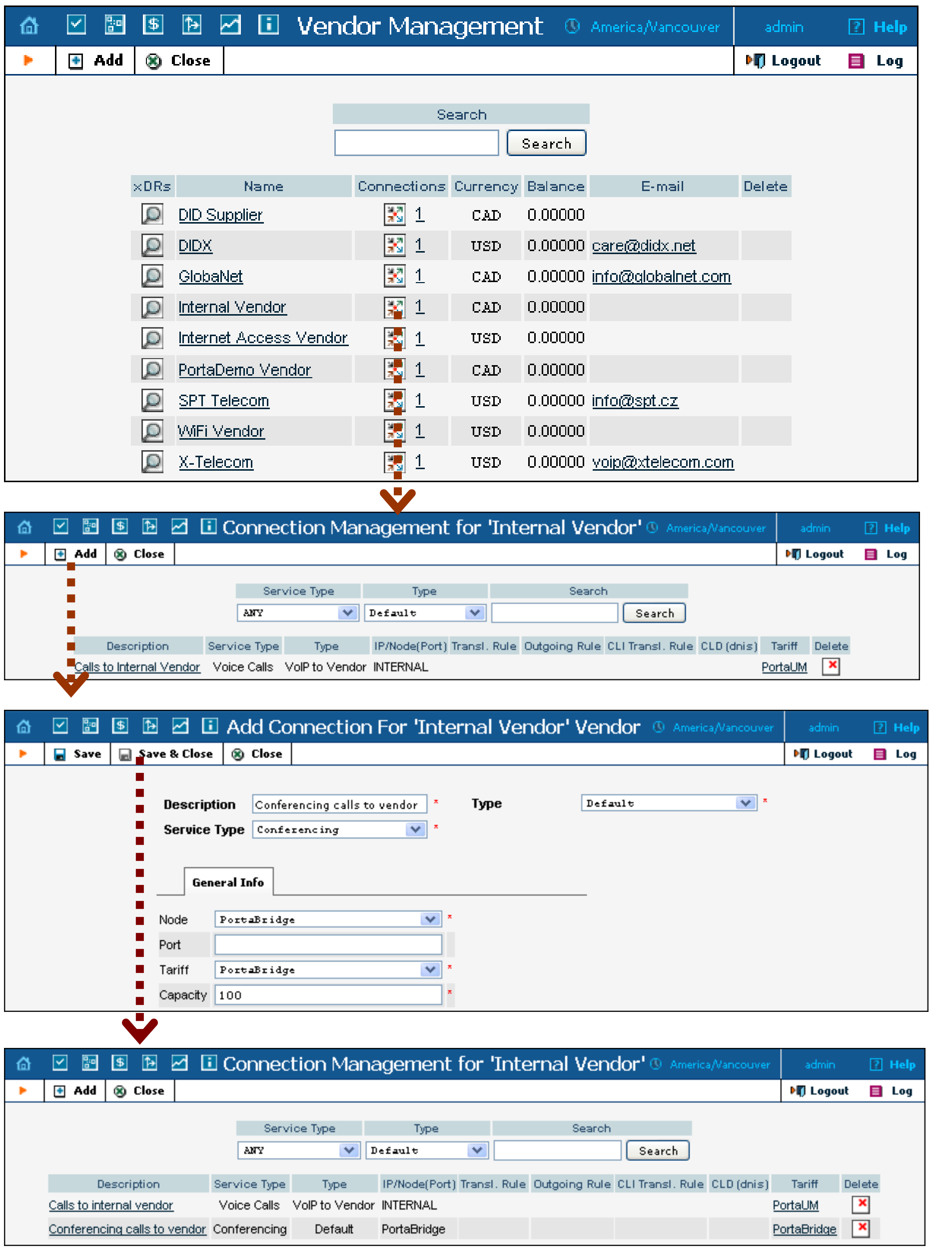
- **Billing period** – Split period for vendor statistics.

4. Click **Save&Close**.

Define Connections

In this step, you will add a new connection to the internal vendor. If you provide both unified messaging and conferencing services, you can use the same internal vendor when setting both services and, for each service type, add its own connection to this vendor.

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

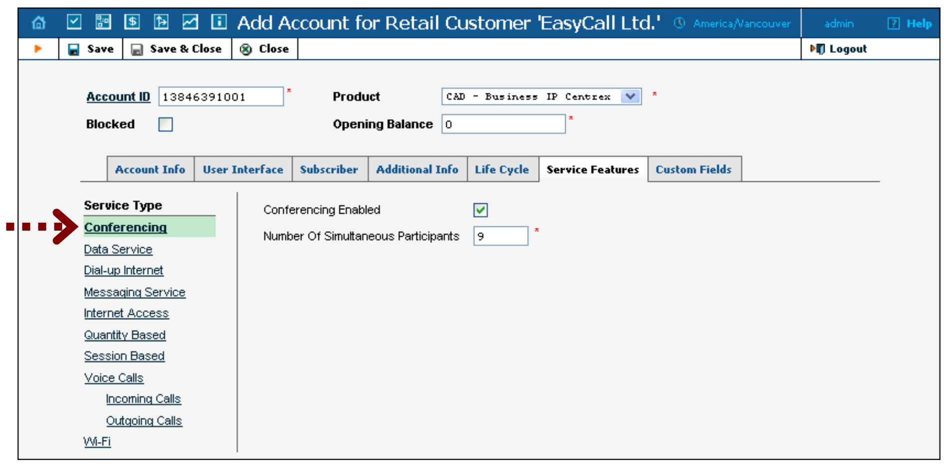


3. Press **Add** to add a new connection.

4. Enter a comment in the **Description** field. Choose the **Conferencing** service type by clicking on the corresponding drop-down menu; leave the **Default** type.
5. Fill in the connection information. Select the PortaBridge node and choose the vendor tariff you created earlier from the select menu in the **Tariff** column. Give the number of simultaneous calls the PortaBridge gateway can handle in **Capacity**. Leave the **Port** field blank.
6. Click **Save&Close**.
7. Click **Close** to exit to the Vendor Management screen.

Enable the Conferencing Service for an Account

1. Go to the **Customers** screen (the screen containing a list of customers). It should look like the screenshot below.
2. Next to the customer's name, click on the icon (in the **Accounts** column). This will take you to account management for that customer.
3. Type an ID in the **Account ID** field and click **Show Accounts**, so that the **Account Info** screen appears. (If you see a list with several accounts, click on the account ID link of the one you wish to edit).

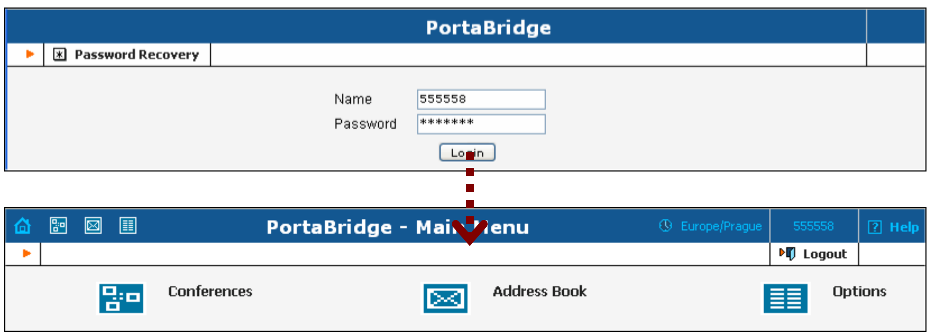


4. Select the **Service Features** tab, then choose the **Conferencing** service from the menu.
5. Tick the **Conferencing Enabled** checkbox and enter the maximum allowed number of participants in the field below.
6. Click **Save&Close** to save the account information.
7. Repeat steps 3-6 if you wish to enable the conferencing service for any other accounts.

Schedule a Conference

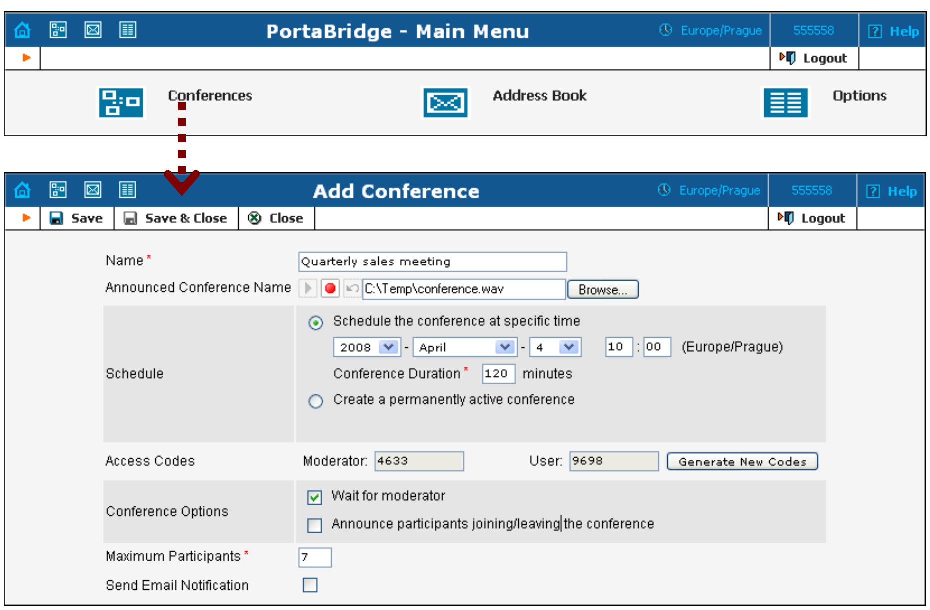
Log in to the PortaBridge Self-care Interface

1. Open **http://<your-porta-bridge-address>** in your browser (inserting the actual hostname or IP of your PortaBridge server).
2. Use the ID of your account with the conferencing service enabled as **Name**, and the **User Interface Password** as **Password**. After successfully logging in, the main menu of the PortaBridge self-care interface will appear.



Create a Conference

1. Click on the Conferences item in the main menu.



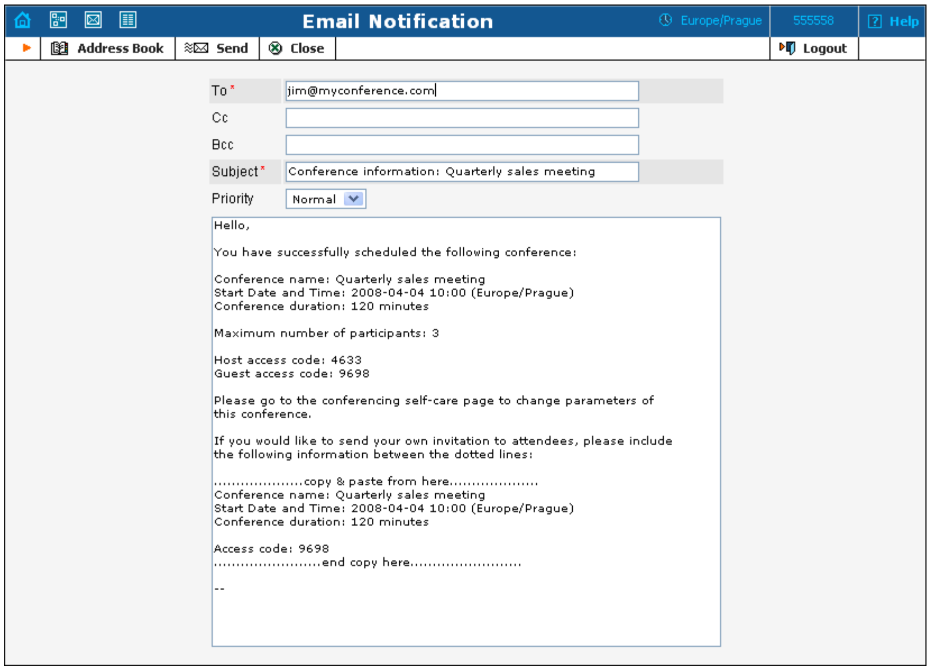
2. Fill in the **Add Conference** form; see the following table for a description of the available fields

Field	Description
Name	A description of your conference.
Announced Conference Name	Either upload a sound file, or record (using your PC's microphone) the name of the conference as it will be announced to people joining it.
Schedule	Specify the date and time when the conference will start.
Access Codes	Access codes are created automatically, but you can generate a new set of codes by clicking on the **Generate New Codes** button.
Conference Duration	In order to prevent service abuse, you must specify the maximum allowed conference duration (in minutes).
Maximum Session Time	If you are creating a "meeting room" (a permanently active conference), specify the maximum time that a single participant can stay in the conference. This is also done to prevent potential service abuse.
Conference Options: Wait for moderator	If activated, conference participants will not be able to communicate with each other until the host (moderator) arrives.
Conference Options: Announce participants	If activated, each participant will be asked to record his or her name initially. When he or she enters the conference, all the other participants will hear "… has joined the conference"; and when he or she leaves, the other participants will be informed of this as well.
Maximum Participants	You can limit the maximum allowed number of concurrent connections to the meeting room. Note that you may not specify a higher value here than the **Number of Simultaneous Participants** assigned by the PortaBilling administrator to your account (in the **Service Features** tab on your account info).
Send Email Notification	If selected, another screen will appear after you save the conference, allowing you to review and send yourself or other hosts an email containing conference access information.

3. Click **Save&Close** to save the conference.

Send a Note with the Conference Info

If you have chosen **Send Email Notification** on the **Add Conference** screen, you will now see the **Email Notification** screen. Edit the email text, if desired, and select who is to receive it (you can either type in the addresses or select them from your address book).



Click the **Send** button to send the message.

Test a Conference

1. Dial the conference access number from a phone and enter your access code. Do the same from the other phones for other participants.
2. When the conference is over, you can review the charges for the conferencing service on the PortaBilling web interface: