

Porta  Switch[®]



Handbook:

Unified Communications
Maintenance Release 23

Copyright Notice & Disclaimers

Copyright © 2000-2011 PortaOne, Inc. All rights reserved.

**PortaSwitch® Handbook: Unified Communications, June 2011
Maintenance Release 23
V.1.23.5**

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

Changes may be made periodically to the information in this publication. The changes will be incorporated in new editions of the guide. The software described in this document is furnished under a license agreement, and may be used or copied only in accordance with the terms thereof. It is against the law to copy the software on any other medium, except as specifically provided in the license agreement. The licensee may make one copy of the software for backup purposes. No part of this publication may be reproduced, stored in a retrieval system, or transmitted in any form or by any means, electronic, mechanical, photocopied, recorded or otherwise, without the prior written permission of PortaOne, Inc.

The software license and limited warranty for the accompanying products are set forth in the information packet supplied with the product, and are incorporated herein by this reference. If you cannot locate the software license, contact your PortaOne representative for a copy.

All product names mentioned in this manual are for identification purposes only, and are either trademarks or registered trademarks of their respective owners.

Table of Contents

Preface 4

1. Setting up and Using UM Services 5

 Checklist..... 6

 Initial Configuration of PortaBilling..... 7

 Create Special UM Destinations 7

 Associate Access Numbers with Applications..... 8

 Create PortaUM Node 9

 Add Rates into the Internal Vendor Tariff..... 10

 Define a Connection..... 11

 Modify Rates..... 12

 Modify Products 13

 Creating New Accounts 15

 Enable UM Services for an Existing Account 18

2. Setting up and Using Conferencing Services..... 19

 Checklist..... 20

 General Configuration..... 21

 Add a Symbolic Destination..... 21

 Create a PortaBridge Node 22

 Configure Access from PSTN..... 23

 Associate Access Numbers with Applications..... 23

 Create a Conferencing Charge Tariff 24

 Enter Rates 25

 Modify Tariffs for SIP Users..... 27

 Add the Conferencing Service to Your Product..... 27

 Define Connections..... 28

 Enable the Conferencing Service for an Account 29

 Make a Test..... 30

 Test a Conference..... 33

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 taking the action described in the paragraph.



Tips provide information that might help you solve a problem.

Trademarks and Copyrights

PortaBilling®, PortaSIP®, PortaUM® and PortaSwitch® are registered trademarks of PortaOne, Inc.

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	
Make sure general configuration has already been done according to the instructions in <i>Basic Residential VoIP Service</i> chapter in the PortaSwitch Handbook: Residential VoIP Services .	[]
Create destinations for UM services.	[]
Associate access numbers with applications.	[]
Network configuration	
Create a node for your PortaUM server.	[]
Rating configuration (Vendor)	
Enter a rate for the wildcard destination ‘ ’ in the internal tariff (used to calculate costs of on-net calls, referred to as tariff A later on).	[]
Create a connection for your “internal” vendor (the one created in the basic SIP service configuration for handling SIP on-net calls) using tariff A.	[]
Rating configuration (Customer)	
Add rates for the UM destination into the tariff used for charging your subscribers (referred to as tariff B later on).	[]
Create a new rating entry in the product for your subscribers using the PortaUM node and tariff B.	[]
Account provisioning	
Enable UM services for an existing account (with product A).	[]

Initial Configuration of PortaBilling

Before proceeding with UM setup, please perform initial system configuration according to the instructions in *Basic Residential VoIP Service* chapter in the **PortaSwitch Handbook: Residential VoIP 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 that describe all UM numbers, because you are not going to charge your clients for accessing UM services based on the access number. The only destination you need to add to the system for clients is a symbolic **UM** destination. Similarly, for vendor tariffs you may use a wildcard '|' (pipe) destination instead of creating many rate entries for all UM access numbers.

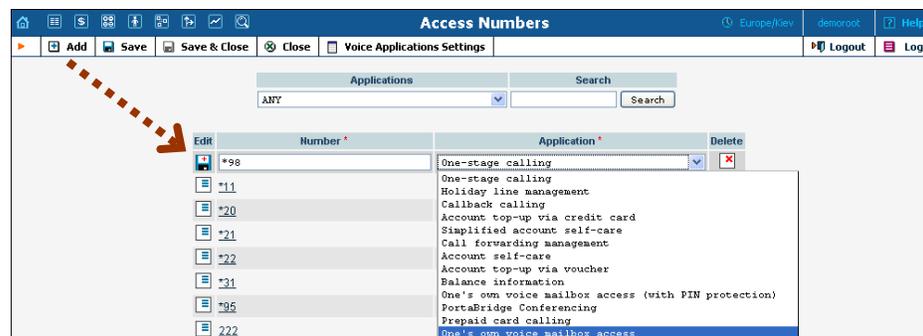
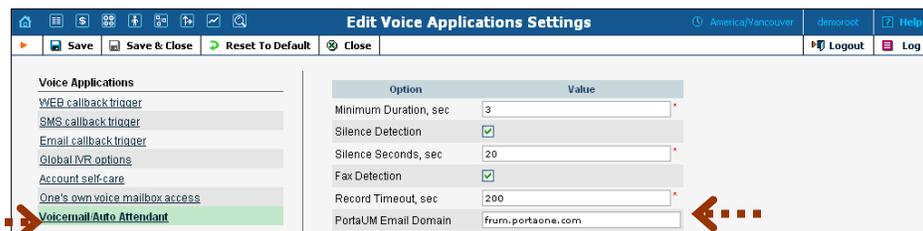
1. In the **Rating** section of Admin-Index, choose **Destinations**.
2. Click on the **Add** button.

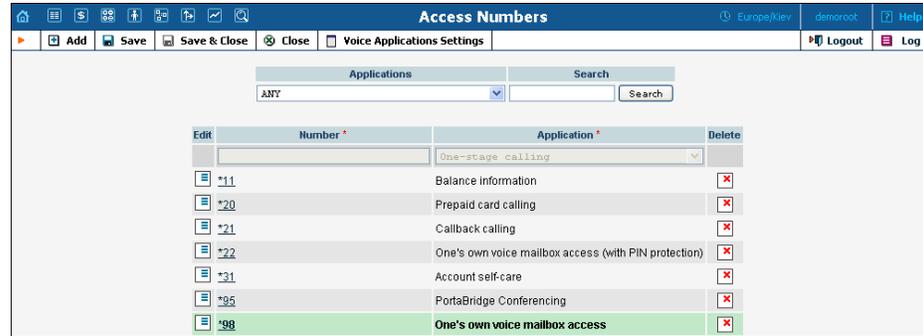
Edit	Prefix *	Country	Subdivision	Description	Delete
		Not Applicable	Not Applicable		
	UM	Not Applicable	Not Applicable	UM services	X

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.

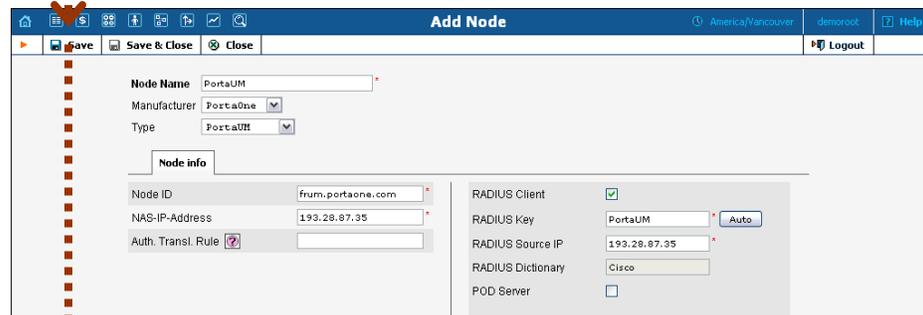
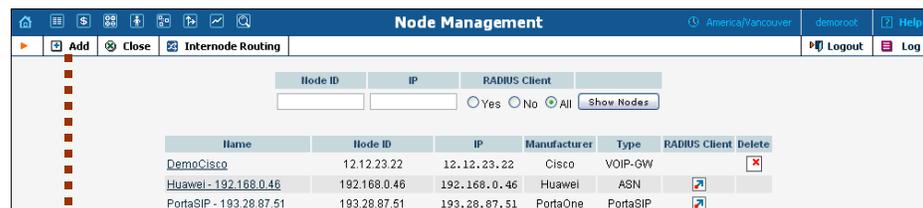




1. In the **Routing** section of Admin-Index, choose **Access Numbers**.
2. Click **Voice Applications Settings** in the toolbar.
3. Choose **Voicemail/Auto Attendant** from the menu.
4. Type the unique domain name (e.g. frum.portaone.com) in the **PortaUM Email Domain** field.
5. Click **Save&Close**.
6. Click on the **Add** button.
7. 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.
8. Click **Save**.

Create PortaUM Node

Now you have to enter your PortaUM® server (or in case if you are using Cisco gateway for IVRs – the Cisco gateway) as a node. PortaBilling® requires some key information about your network equipment, such as IP address, Node ID, Radius shared secret, and so on.



Name	Node ID	IP	Manufacturer	Type	RADIUS Client	Delete
DemoCisco	12.12.23.22	12.12.23.22	Cisco	VOIP-GW	<input type="checkbox"/>	
Huawei - 192.168.0.46	192.168.0.46	192.168.0.46	Huawei	ASN	<input type="checkbox"/>	
PortaSIP - 193.28.87.51	193.28.87.51	193.28.87.51	PortaOne	PortaSIP	<input type="checkbox"/>	
PortaUM	frum.portaone.com	193.28.87.35	PortaOne	PortaUM	<input checked="" type="checkbox"/>	

- In the **Networking** section of the Admin-Index page, choose **Nodes**.
- In the Node Management window, click **Add** icon.
- 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** – Make sure this check-box is enabled, 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.
- Click **Save&Close**.

Add Rates into the Internal Vendor Tariff

- On the Tariff Management page you will see a list of the available tariffs. Click the **Rates** icon next to the name of the internal vendor tariff (the one you have previously created for SIP-UA connection).

Tariff Management

Applied To: AMY Service: AMY Managed By: AMY

Rates	Name	Currency	Applied To	Service	Managed By	Routing	Description	Delete
<input type="checkbox"/>	Aardhoip	USD	Customer	Voice Calls	Administrator only			<input type="checkbox"/>
<input type="checkbox"/>	ABC Shuttle Prepaid Cards	USD	Customer	Voice Calls	Administrator only			<input type="checkbox"/>
<input type="checkbox"/>	Callshop	USD	Customer	Voice Calls	Administrator only			<input type="checkbox"/>
<input type="checkbox"/>	DID Supplier Costs	USD	Vendor	Voice Calls		No		<input type="checkbox"/>
<input type="checkbox"/>	SIP Phone Subscribers	USD	Customer	Voice Calls	Administrator only			<input type="checkbox"/>
<input type="checkbox"/>	SIP on-net (internal)	USD	Vendor	Voice Calls		No		<input type="checkbox"/>

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

Effective From: Now Destination: |

Edit	Destination *	Country	Description	Peak Off-Peak	Interval, second		Price, USD/minute		Effective From YYYY-MM-DD HH:MM:SS *	Delete
					First *	Next *	First *	Next *		
<input type="checkbox"/>				NOT DEFINED	1	1	0	0	immediately	<input type="checkbox"/>

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

Edit	Destination *	Country	Description	Peak Off-Peak	Interval, second		Price, USD/minute		Effective From YYYY-MM-DD HH:MM:SS *	Delete
					First *	Next *	First *	Next *		
<input type="checkbox"/>		Not Applicable	Default	Peak Off-Peak: NOT DEFINED	1	1	0.00000	0.00000	2011-01-11 16:50:01	<input type="checkbox"/>

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

Define a Connection

1. In the **Participants** section of the admin interface, choose **Vendors**.
2. Click on the **Connections** icon next to the name of the internal vendor (the one that holds SIP-UA connection).

Vendor Management

Search: _____

xDRs	Name	Connections	Currency	Balance	E-mail	Delete
<input type="checkbox"/>	Arbinet	2	USD	0.00000		<input type="checkbox"/>
<input type="checkbox"/>	Begemot	3	USD	0.25680		<input type="checkbox"/>
<input type="checkbox"/>	DID Supplier	2	USD	0.00000		<input type="checkbox"/>
<input type="checkbox"/>	DIDX	1	USD	0.00000		<input type="checkbox"/>
<input type="checkbox"/>	Internal Vendor	0	USD	0.00000		<input type="checkbox"/>

The first screenshot shows the 'Connection Management for Internal Vendor' interface. It features a table with columns: Description, Service Type, Type, Gateway/Node, IP, Port, Access Code/Tech Prefix, Transl. Rule, Vendor Account, Tariff, and Delete. A red dashed arrow points to the 'Add' button in the top left.

The second screenshot shows the 'Add Connection For Internal Vendor Vendor' form. The 'Description' field contains 'Calls to Internal Vendor'. The 'Service Type' is set to 'Voice Calls' and the 'Type' is 'VoIP to Vendor'. The 'General Info' section includes:

- Routing Protocol: H323 SIP
- Remote Gateway ID: INTERNAL
- Remote IP: [empty]
- Remote IP Port: [empty]
- CLD Tech Prefix: [empty]
- RTP Proxying: Direct
- Tariff: PortaUM
- Capacity: 100
- Caller Identity: Do Not Supply
- Vendor Account: None
- Translation Rule: [empty]
- Outgoing Rule: [empty]
- CLI Translation Rule: [empty]
- Additional SIP Settings: [empty]

 A red dashed arrow points to the 'Save & Close' button.

The third screenshot shows the updated 'Connection Management for Internal Vendor' table. The new connection is listed with the following details:

Description	Service Type	Type	Gateway/Node	IP	Port	Access Code/Tech Prefix	Transl. Rule	Vendor Account	Tariff	Delete
Calls to Internal Vendor	Voice Calls	VoIP to Vendor	INTERNAL						PortaUM	X

 A red dashed arrow points to the 'Add' button.

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 Gateway ID** 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.

Modify Rates

Modify Tariffs for SIP Users

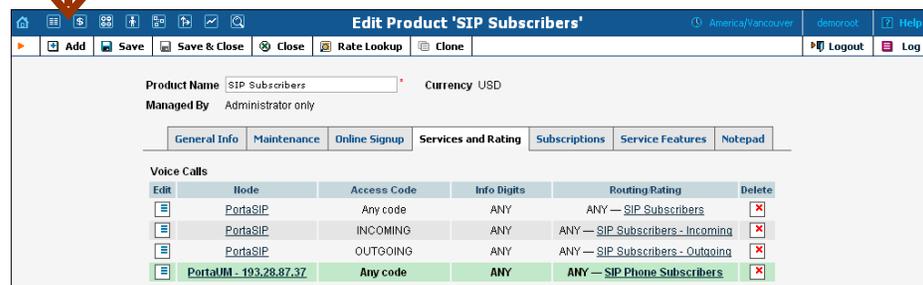
First of all, you should make sure that your customers are allowed to call voicemail. In order to do so, you should add a rate for the voicemail destination **UM** into the tariffs used for your SIP subscribers.



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.

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 rating list includes the PortaUM® server, then accounts with this product are permitted to login into UM (via webmail, voicemail, or from their email client).

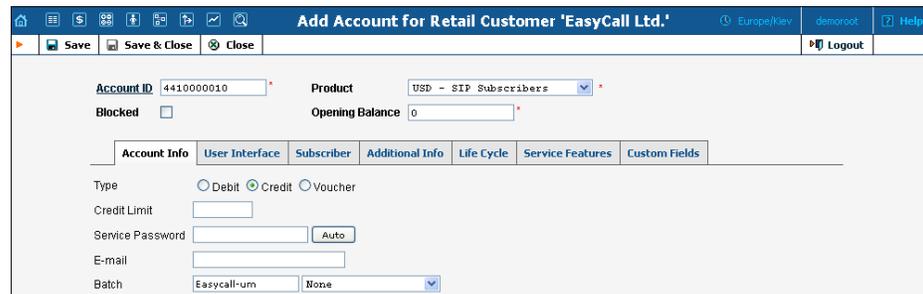
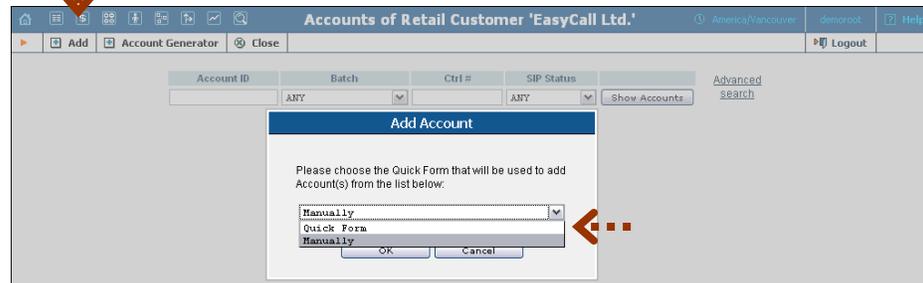


1. In the **Rating** section of the Admin-Index page, choose **Products**.
2. On the Product Management page, click on the name of the corresponding product.
3. Click on the **Services and Rating** tab to edit this product's rating list.
4. After the **Services and Rating** tab has been selected, click 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. Configure overdraft protection for this product on the **Overdraft Protection** tab. (Please consult the *Rating* section in the [PortaBilling Web Reference Guide](#) for a description of the parameters available here.)

8. Click **Save** to save this rating entry.
9. Click **Close** to return to the Product Management page.
10. Repeat steps 2-9 to modify rating lists for all products which will include UM services.

Creating New Accounts

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



Add Account for Retail Customer 'EasyCall Ltd.'

Account ID: 4410000010 Product: USD - SIP Subscribers

Blocked: Opening Balance: 0

Service Features:

- UM Enabled:
- Forward Mode: No Forwarding
- Call Processing Enabled:
- Exit-to-exit call distinctive ring: Customer's default
- Present Caller Info:
- Disable Call Waiting:

Service Type: Incoming Calls

Add Account for Retail Customer 'EasyCall Ltd.'

Account ID: 4410000010 Product: USD - SIP Subscribers

Blocked: Opening Balance: 0

Service Features:

- Allowed Identity: Customer's default
- Batch:
- Default Valid Identity:
- Allowed CLI: Use The Same Rule
- Override Display Name: No
- Display Name:
- Preferred IVR Language: en - English
- Favorite Numbers Enabled: No
- Hide CLI: Customer's default
- Hide CLI Prefix:
- Show CLI Prefix:
- Call Barring Enabled:
- Default Routing Plan: All Available Routes

Service Type: Outgoing Calls

- 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.
- Click on **Add** (In this example, we assume that you are creating the account manually, so if the Quick Form dialog pops up - choose the **Manually** option).
- 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.

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

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

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.

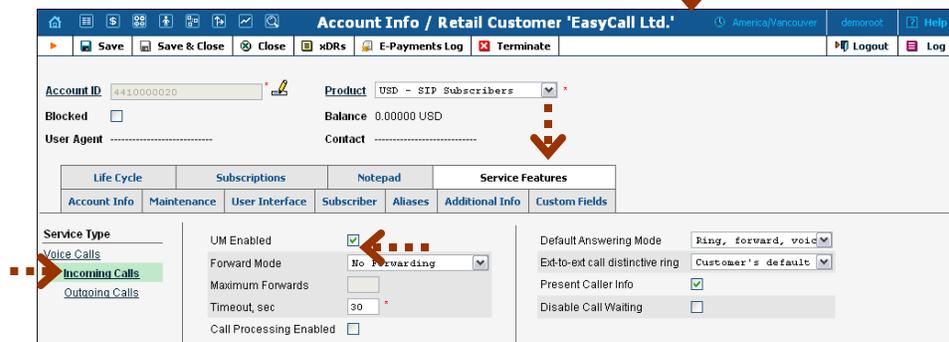
Service Features

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.
 - **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** 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. Verify that this account is assigned a product for which you have already modified the rating list to enable UM services.
5. Click on the **Service Features** tab.
6. In the **Service Type** column, select **Incoming Calls** under **Voice Calls**.
7. Click on the **UM Enabled** checkbox to activate UM services for this account.
8. Click **Save&Close**.

2. Setting up and Using Conferencing Services

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	
Make sure general configuration has already been done according to the instructions in <i>Basic Residential VoIP Service</i> chapter in the PortaSwitch Handbook: Residential VoIP Services .	[]
Create a symbolic CONFERENCE destination.	[]
Network configuration	
Create a node for your PortaBridge server.	[]
Configure the vendor that provides the IVR access number.	[]
Configure a “VoIP from Vendor” connection to that vendor that provides the IVR access number.	[]
Associate access numbers with applications.	[]
Rating configuration (Vendor)	
Enter a rate for the wild-card destination ‘ ’ in the internal tariff (used to calculate costs of on-net calls, referred to as tariff A later on).	[]
Create a connection for your “internal” vendor (the one created in the basic SIP service configuration for handling SIP on-net calls) using tariff A.	[]
Create a vendor tariff (referred to as tariff B later on) for Conferencing service.	[]
Enter a single rate for ‘ ’ wildcard destination in the tariff B.	[]
Create a conference service connection for your “internal” vendor (the one created in the basic SIP service configuration for handling SIP on-net calls) using tariff B.	[]
Rating configuration (Customer)	
Create a tariff (referred to as tariff C later on), which will apply to conferencing services used by your customers (make sure this tariff has the Conferencing service assigned!).	[]
Enter rates in tariff C for access numbers used for the conferencing service.	[]
Specify rates for dialing into the conferencing bridge for your “normal” SIP accounts.	[]
Create a new rating entry in your existing product A, using the conferencing service, PortaBridge node and tariff C.	[]
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 *Basic Residential VoIP Service* chapter in the [PortaSwitch Handbook: Residential VoIP Services](#). Please follow the instructions provided there.

Add a Symbolic Destination

You will need to add a symbolic CONFERENCE destination.

The screenshots illustrate the steps to add a symbolic destination in the Destinations management interface:

- The 'Add' button is clicked to open the form.
- The form fields are populated: Prefix is 'CONFERENCE', Country is 'Not Applicable', Subdivision is 'Not Applicable', and Description is 'Conferencing'.
- The 'Save' button is clicked to create the destination.

1. In the **Rating** 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**.

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.

The image shows two screenshots from a web-based management interface. The top screenshot is the 'Node Management' page, which includes a table of existing nodes. A red dashed arrow points from the 'Add' button in the top left to the 'Add Node' form in the bottom screenshot.

Name	Node ID	IP	Manufacturer	Type	RADIUS Client	Delete
DemoCisco	12.12.23.22	12.12.23.22	Cisco	VOIP-GW	<input type="checkbox"/>	
Huawei-192.168.0.46	192.168.0.46	192.168.0.46	Huawei	ASN	<input type="checkbox"/>	
PortaSIP-193.28.87.51	193.28.87.51	193.28.87.51	PortaOne	PortaSIP	<input type="checkbox"/>	

The bottom screenshot is the 'Add Node' form. It contains the following fields:

- Node Name:** PortaBridge
- Manufacturer:** PortaOne
- Type:** PortaBridge
- Node info section:**
 - Node ID:** portaum.bridge.com
 - NAS-IP-Address:** 192.168.13.1
 - Auth. Transl. Rule:** (blank)
 - RADIUS Client:**
 - RADIUS Key:** bridgesecret (with an 'Auto' button)
 - RADIUS Source IP:** 192.168.13.1
 - RADIUS Dictionary:** Cisco
 - POD Server:**

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** – Make sure this check-box is enabled, since **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**.

Configure Access from PSTN

Since the main purpose of this service is to give your customers access to the conferencing application from PSTN (by dialing your access number from their landline or mobile phone), you have to ensure that calls to that access number are delivered properly to your network and then handed over to PortaBridge.

Please refer to the instructions in the *Incoming Calls from PSTN* chapter of the **PortaSwitch Handbook: Residential VoIP Services** for complete details about how to set up the vendor and the connection.

Associate Access Numbers with Applications

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

The top screenshot shows the 'Access Numbers' configuration page. It features a table with columns for 'Number', 'Application', and 'Delete'. The table lists several access numbers and their corresponding applications:

Number *	Application *	Delete
	One-stage calling	
*11	Balance information	X
*20	Prepaid card calling	X
*21	Callback calling	X
*22	One's own voice mailbox access (with PIN protection)	X
*31	Account self-care	X
*95	PortaBridge Conferencing	X
*98	One's own voice mailbox access	X

The bottom screenshot shows the same table with a dropdown menu open for the number 180074. The dropdown menu lists the following applications:

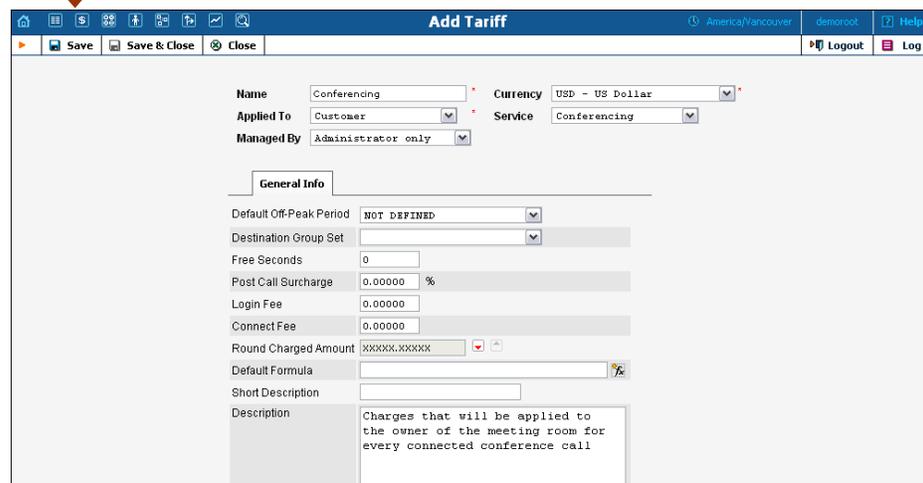
- Prepaid card calling
- One-stage calling
- Holiday line management
- Callback calling
- Account top-up via credit card
- Simplified account self-care
- Call forwarding management
- Account self-care
- Account top-up via voucher
- Balance information
- One's own voice mailbox access (with PIN protection)
- PortaBridge Conferencing**
- Prepaid card calling
- One's own voice mailbox access



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

3. Fill in the **Add Tariff** form:
 - **Name** – A short name for the tariff object (this will be used in the select menus).
 - **Currency** – Indicates the currency in which you charge your customers.

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.

The interface shows the 'Rates for tariff 'Conferencing'' page. The top screenshot highlights the 'Add' button. The bottom screenshot shows the following table of rates:

Edit	Destination *	Country	Description	Peak-Off-Peak	Interval, second		Price, USD/minute		Effective From YYYY-MM-DD HH24:MLSS *	Delete
					First *	Next *	First *	Next *		
	1800			NOT DEFINED	60	30	0.08	0.08	Immediately	
					1	1	0.08	0.08		

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 **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.

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	Peak Off-Peak	Interval, second		Price, USD/minute		Effective From YYYY-MM-DD HH:MM:SS *	Delete
					First *	Next *	First *	Next *		
			NOT DEFINED							
<input type="checkbox"/>	CONFERENCE	Not Applicable		Peak Off-Peak: Night	1 1	1 1	0.00000 0.00000	0.00000 0.00000	2011-01-13 13:40:16	
<input type="checkbox"/>	UM	Not Applicable Porta UM		Peak Off-Peak: Night	1 1	1 1	0.00000 0.00000	0.00000 0.00000	2011-01-13 13:40:43	

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

Top Screenshot: Edit Product 'Business IP Centrex'

Product Name: Business IP Centrex | Currency: USD
 Managed By: Administrator only

Tabs: General Info | Maintenance | Online Signup | **Services and Rating** | Subscriptions | Service Features | Notepad

Voice Calls Table:

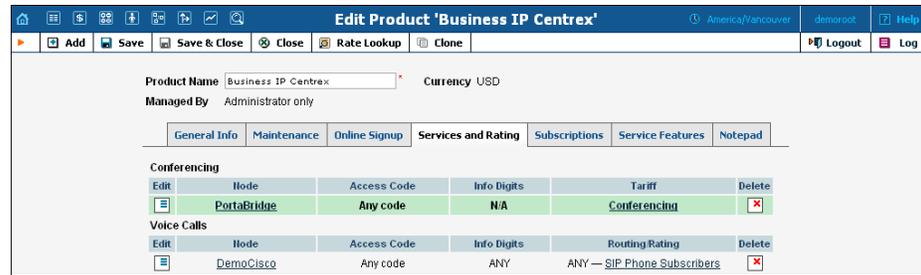
Edit	Node	Access Code	Info Digits	Routing Rating	Delete
<input type="checkbox"/>	DemoCisco	Any code	ANY	ANY — SIP Phone Subscribers	<input type="checkbox"/>

Bottom Screenshot: Add Services and Rating to Product 'Business IP Centrex'

Product Name: Business IP Centrex | Product Currency: USD
 Service: Conferencing

Tabs: General | **Overdraft Protection**

Node: PortaBridge | Access Code:
 Tariff: Conferencing

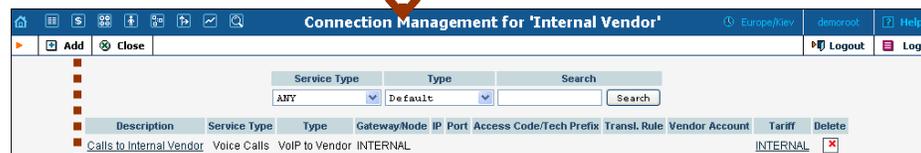


1. On the Product Management page, click the product name to enter the **Edit Product** form.
2. When the **Services and Rating** tab is selected, click on the **Add** icon.
3. Choose **Conferencing** in the **Service** drop-down box.
4. Select the PortaBridge node and choose the conferencing tariff you created earlier.
5. The **Access Code** should be left empty for the conferencing service.
6. 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.)
7. Click **Save** to save this rating entry.

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 **Participants** section of the admin interface, choose **Vendors**.
2. Click on the **Connections** icon next to the name of the internal vendor (the one that holds SIP-UA connection).



Add Connection For 'Internal Vendor' Vendor

Description: Conferencing calls to vendor * Type: Default *
 Service Type: Conferencing *
General Info
 Node: PortaBridge - 193.28.87.57 *
 Port: *
 Tariff: PortaBridge Conferencing *
 Capacity: 100 *

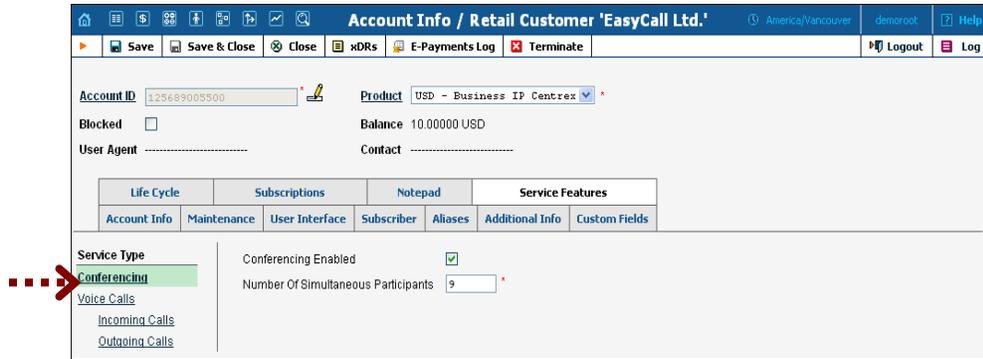
Connection Management for 'Internal Vendor'

Service Type		Type		Search						
ANY		Default		Search						
Description	Service Type	Type	Gateway/Node	IP	Port	Access Code/Tech Prefix	Transl. Rule	Vendor Account	Tariff	Delete
Calls to Internal Vendor	Voice Calls	VoIP to Vendor	INTERNAL						INTERNAL	X
Conferencing calls to vendor	Conferencing	Default	193.28.87.57	193.28.87.57					PortaBridge Conferencing	X

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).
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).



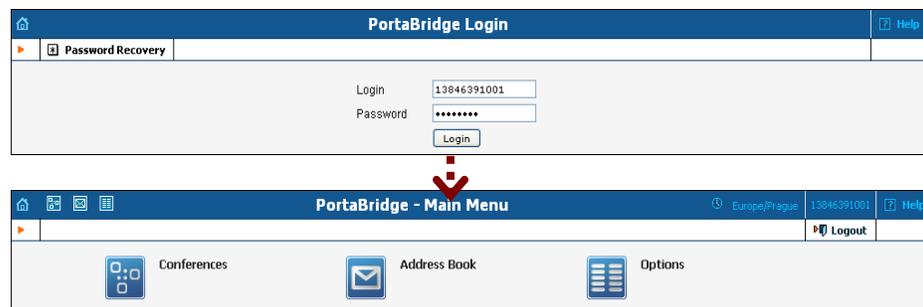
4. Select the **Service Features** tab, then choose the **Conferencing** service from the menu.
5. Tick the **Conferencing Enabled** checkbox and enter the maximum allowed number of participants in the field below.
6. Click **Save&Close** to save the account information.
7. Repeat steps 3-6 if you wish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

Make a Test

Schedule a Conference

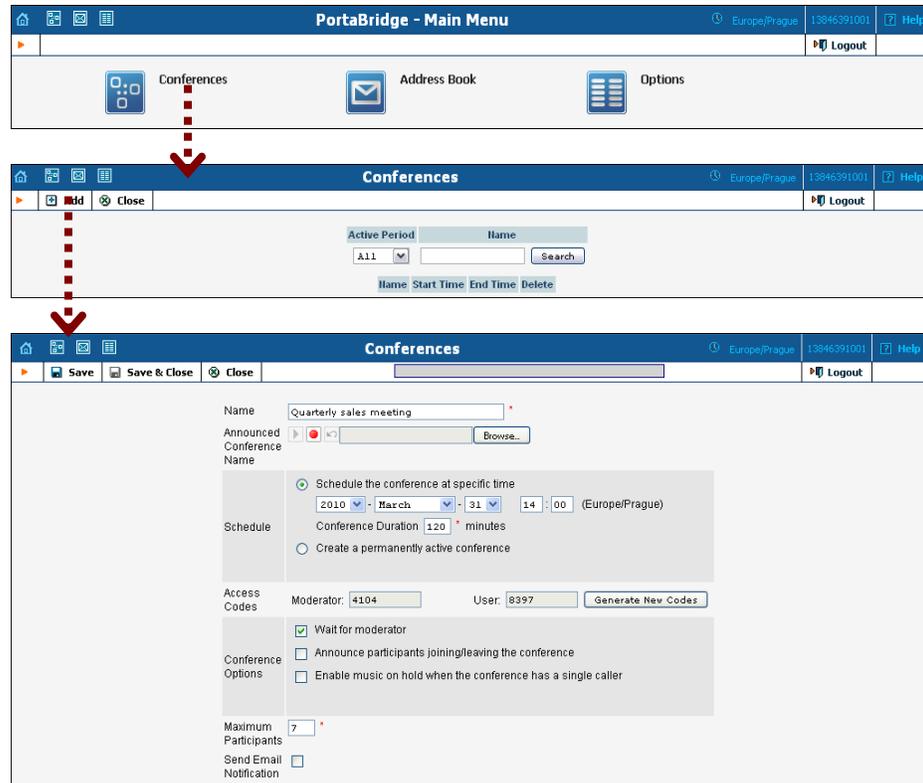
Log in to the PortaBridge self-care interface

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:

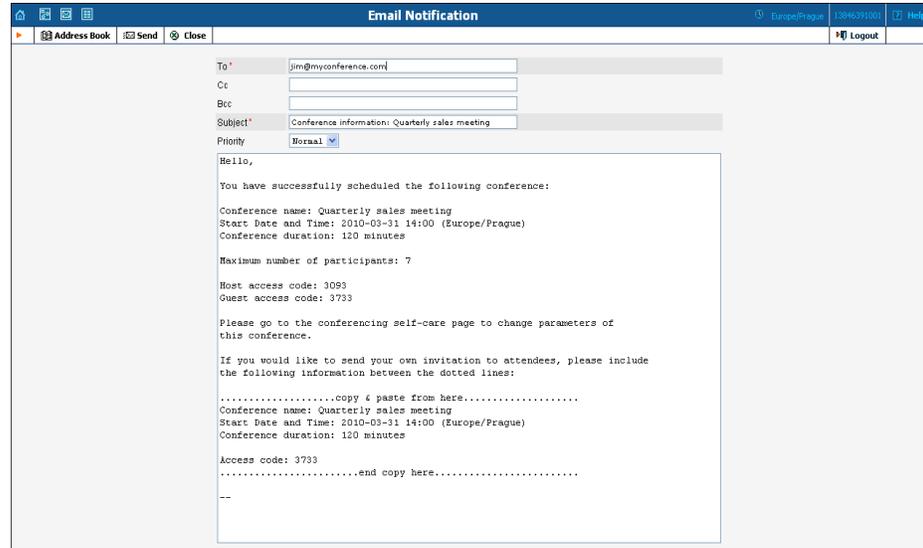
Field	Description
Name	A description of your conference.
Announced Conference Name	Either upload a sound file, or record (using your PC's microphone) the name of the conference as it will be announced to people joining it.
Schedule	Specify the date and time when the conference will start.
Access Codes	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.
Conference Options: Enable music while on hold when the conference has a single caller	If activated, the first conference participant will hear music while on hold until the second conference participant arrives and joins the conference.
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:

