PORTA ® ONE

PortaUM

Unified Messaging

User Guide

www.portaone.com

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PortaUM, March 2004

Please address your comments and suggestions to: Sales Department, PortaOne, Inc., Suite 400, 2963 Glen Drive, Coquitlam, BC, V3B 2P7, Canada

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Preface

This document provides a general overview of the PortaUM (Unified Messaging System), a platform for the delivery of enhanced business and residential communications services. PortaUM handles voice, fax, and regular e-mail messages as objects in a single mailbox that a user can access via a web interface or by telephone.

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 for 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/solutions/portaum/**.

Conventions

This publication uses the following conventions:

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

Caution means 'reader be careful'. You are capable of doing something that might result in program malfunction or loss of data.

NOTE: Means *reader take note*. Notes contain helpful suggestions or references to materials not contained in this manual

Timesaver means the described action saves time. You can save time by performing the action described in the paragraph.

Tips Means the following information might help you solve a problem

Hardware and software requirements

PortaUM requires a *dedicated* Cisco AS 5300/5350.

Cisco requirements

128M RAM, 64M flash, E1 or T1 voice ports, sufficient number of DSPs. IOS 12.3.5a (or other from the 12.3 branch).

AS5300 comes with 4 or 8 T1/E1's. You will not need more than 4 of them because of DSP resource limitation for AS5300.

For T1 configuration maximum voice resource will be 96 and for E1 -- 120.

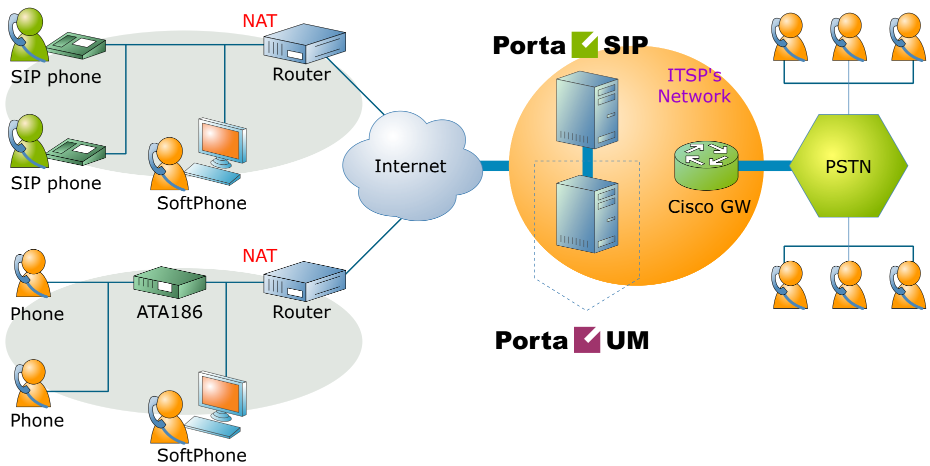
Client System Recommendations

- OS: Windows 95-XP, UNIX or Mac OS
- Browser: Internet Explorer 5 or higher, Netscape 6.2 or higher supporting DOM and with enabled JavaScript.
- Spreadsheet processor (MS Excel)
- Display Settings:
  - Min Screen Resolution: 1024 x 768
  - Color Palette: 16 bit color (minimum)

NOTE: To view downloaded CDR files in Windows please do the following: My Computer -> Control Panel -> Regional Settings -> Number -> List Separator type "," to match PortaBilling default list separator.

1. System concepts

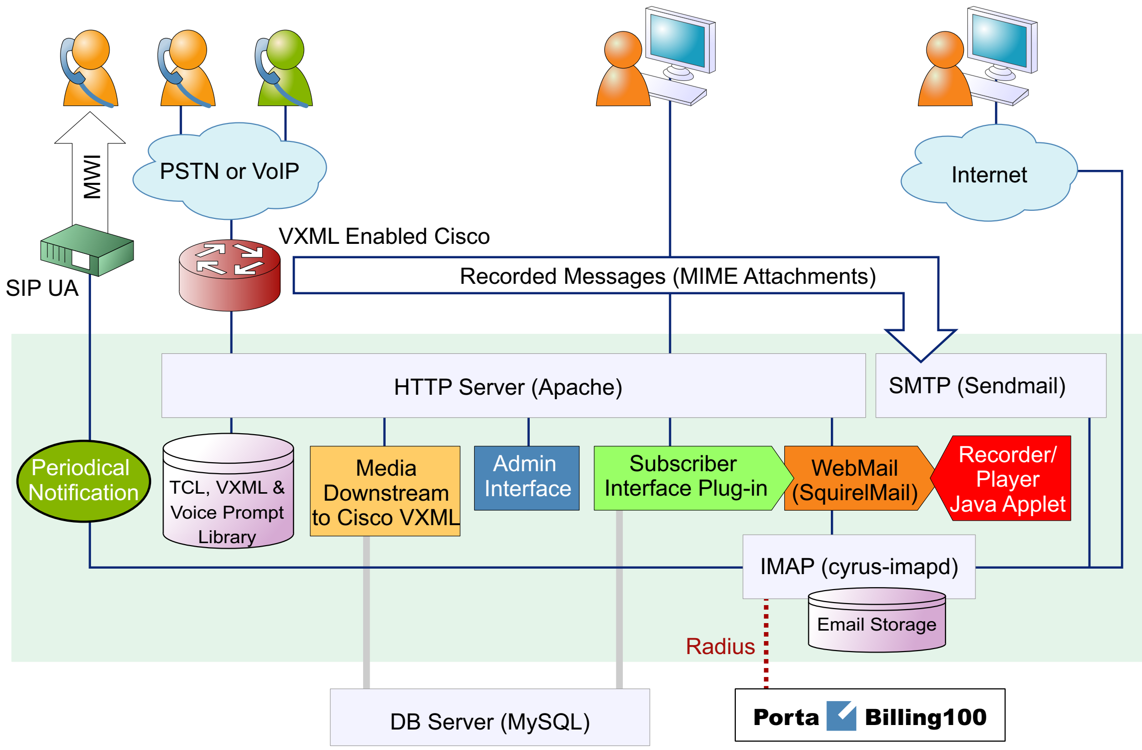
PortaUM's role in your VoIP network



PortaUM (Unified Messaging system) is one of the key components of PortaSwitch, a software-based Communication Service and Subscriber Management Platform directly addressing the needs of modern communication network providers seeking new revenue streams from tools that unify their voice, data, and fax traffic within a single network, and allow for the diversification of their current service offering.

PortaUM handles voice, fax, and regular e-mail messages as objects in a single mailbox that a user can access via a web interface or by telephone.

PortaUM components

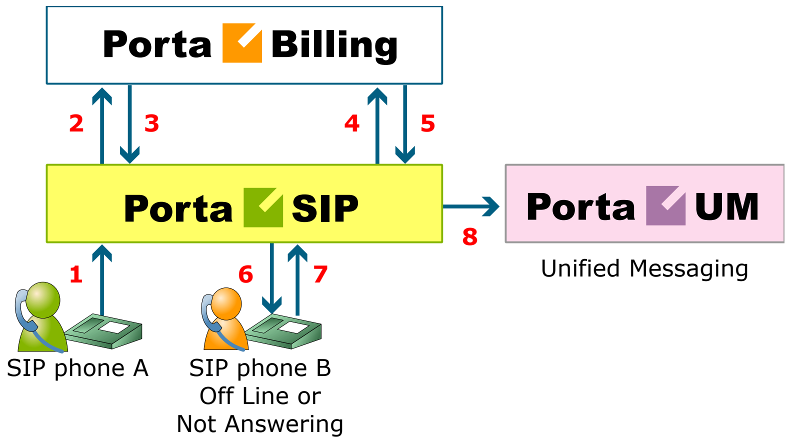


PortaSIP consists of the following main components:

- VXML platform and media server based on Cisco AS53XX. This component is used to record messages, receive faxes and to send them to the PortaUM machine for processing and storing. It also provides IVR (interactive voice response) to a user, allowing him to listen for his recorded messages and to manage his messages using his SIP phone.
- Web server. Web server is used to provide user with Web-bases assess to his mailbox. Also web server along with special CGI scripts used by VXML platform to retrieve recorded messages from PortaUM machine on demand.
- IMAP server. IMAP server is uses as a main storage for messages.
- Database server. Database server provides storage for user's custom settings and voice prompts.
- SMTP server. SMTP server is used to receive incoming messages from VXML platform and from other SMTP servers and also to send outgoing messages created by user using Web interface.
- Mail filter. Mail filter is used to convert audio and graphics attachments in messages received from VXML platform in formats compatible with Microsoft Windows operating system.

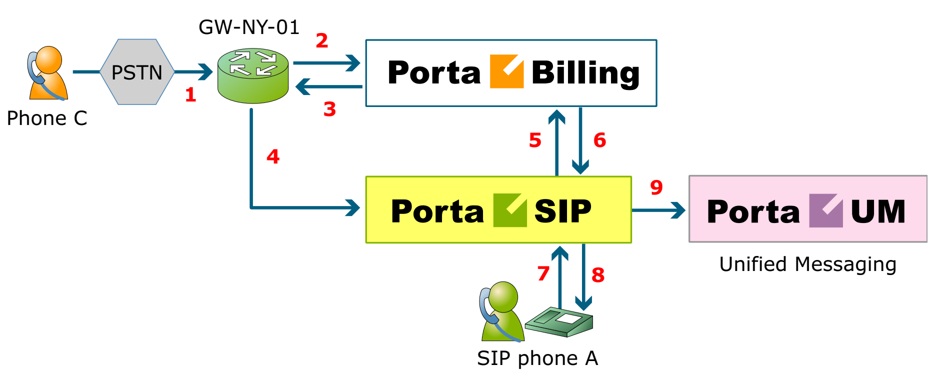
Call process

SIP UA ←→ SIP UA



- The user A dials 121 in attempt to teach B. His SIP user agent sends an INVITE request to the SIP server (1).
- SIP server performs number lookup in the billing (2). For example, if user A dialed 121, the billing will inform the SIP server, that the actual number is 12027810009, and this number actually belongs to B (3).
- SIP server will checks his registration database, but it appears that this account is not online at the moment. If B has unified messaging services enabled, then the call will be redirected to the voice mail system, and A can leave a message for him (8). The same thing would happen if B is online, but does not answer his phone (6), (7).

SIP UA → PSTN



This is another important aspect of the SIP telephony. Your subscribers not only want to make outgoing calls, they also want other people to be able to call them on their SIP regardless of where they are at the moment.

In order to do so, you will need to get range of phone numbers from your telecom operator, and make sure that calls made to these numbers on the PSTN network are routed to your gateway via the telephony interface.

- C wishes to call A. He dials A's phone number (and since C is in US, he dials it in the North American format, 2027810003).
- This call is routed through the telecom network to the gateway GW-NY-01. When the incoming call arrives on the gateway (**1**), it starts special TCL application to handle this call. This application does several things:
    - Converts phone number into the E.164 format, so 2027810003 will become 12027810003.
    - Performs authorization in the billing (**2**) – if A is allowed to receive incoming telephony calls from GW-NY-01, and in case if you charge for the incoming calls – what is the maximum allowed call time basing on A's current balance (**3**). Important issue is that authorization should happen without the password check, because application does not know valid password for the SIP account.
    - Starts outgoing call to 12027810003.
    - When the call will be established, starts the timer, and will disconnect the call when maximum call duration is exceeded.
    - Gateway is configured that it knows that the calls to 1202781….

      Numbers should be sent to the PortaSIP server, so it sends INVITE to the PortaSIP (**4**).
- PortaSIP receives INVITE, but without authorization information. So PortaSIP server performs authentication basing on the IP address (**5**), (**6**). And since this calls from our trusted node – gateway GW-NY-01, the call I authorized.
- SIP server will checks his registration database, but it appears that this account is not online at the moment. If B has unified messaging services enabled, then the call will be redirected to the voice mail system, and A can leave a message for him (**9**). The same thing would happen if B is online, but does not answer his phone (**7**), (**8**).

Supported services

Leaving voice messages to a PortaUM user

PortaUM allows your SIP customer to have an automatic answering machine wherever she is not online or does not answering within configurable time period. This service works both when the call comes from PSTN and when the call comes from another SIP customer. See section Call Process above for detailed call flows in this case.

Sending faxes to a PortaUM user

In addition to voice messages, your UM-enable SIP customer will be able to receive faxes to his "answering machine". When the call is forwarded to PortaUM it automatically detects start of fax transmission and switches into fax receiving mode.

Sending e-mail messages to a PortaUM user

PortaUM allows your SIP customer to have an ordinary e-mail account to which he can receive ordinary e-mails.

Retrieving own voice messages and e-mail messages using SIP phone

Your SIP customer will be able to retrieve his messages using SIP phone. For this, he have to dial special number, which transfers him into PortaUM IVR menu. This menu allows listening recorded messages. In addition, for SIP phones that support message waiting indicator (MWI), PortaUM automatically manages MWI status of SIP phone, so that the user is notified when he has new messages for him in PortaUM

Retrieving own voice messages, e-mail messages and faxes using Web browser

Your user is able to retrieve his messages using standard Web browser and to send ordinary e-mail messages. Advanced Java applet built into the interface allows user not only to listen for voice messages, but to compose new voice messages (microphone is required). It is also possible to access the system using any e-mail client supporting IMAP or POP3 protocols.

Managing personal PortaUM settings using SIP phone

In addition for retrieving his messages, your SIP user is able to manage his personal PortaUM settings, such as greetings, from his SIP phone using PortaUM IVR menu.

Managing personal PortaUM settings using Web browser

Your user is also able to manage his personal PortaUM settings by logging in into PortaUM Web interface. Advanced Java applet built into the interface allows user to record his personal greetings (microphone is required).

2. Setting up and using UM services

Setting up UM services

Initial configuration of the PortaBilling

Before continuing with the UM setup, please perform the initial system configuration according to the ***Setting up SIP services*** chapter in the ***PortaSIP User Guide***. At this point you should have all of your destinations, tariffs, products, customers and vendors already in the system.

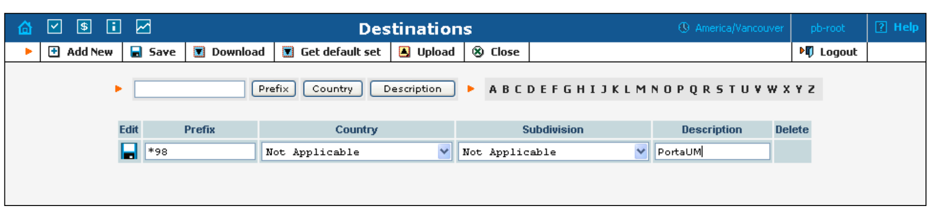
Create special UM destinations

Your subscribers can access UM by two different methods:

- Dialing special UM number from their SIP phone. Normally this is some non-E.164 number, so it will not overlap with any real phone number customer might want to call. For example ***98**.
- Calling a certain number from the PSTN network. It should a valid phone number, allocated to you by local telecom and accessible for anyone on the PSTN network. If you do not have such number, you customers will not be able to check their voicemail from just a regular phone.

In both cases such calls should be clearly identified as UM calls in your system. So you need to create destinations which describe these phone numbers.

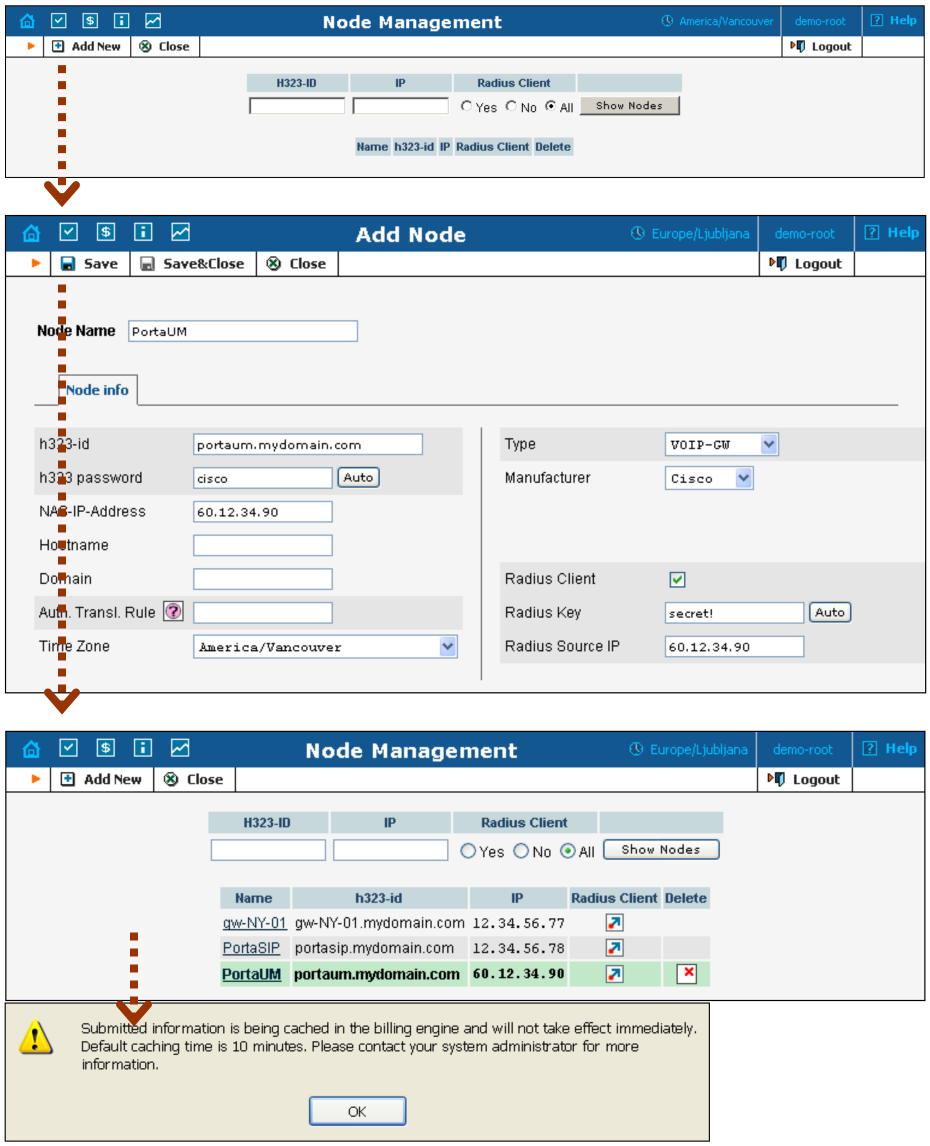
1. In Management section of Admin-Index choose **Destinations**
2. Click on the **+ Add New** button.
3. Put you on-net UM number in the **Prefix**, **N/A** for the country and country subdivision. Put the comment in the **Description** column that will clearly identify that this is a special prefix, assigned to UM.



4. Click **Save.**
5. Add destination for the PSTN UM number if you have it.

Create Nodes

Now you have to enter your PortaUM server and UM gateway (used as a platform for VXML) as nodes. PortaBilling requires some key information about your network equipment such as the IP address, h323-id, Radius shared secret, etc.



1. In the management section of the Admin-Index page choose **Nodes.**
2. In the Node management window click **Add new** icon.
3. Fill in the New Node form:
   - **Node name** – a short descriptive name for your UM server or gateway (will be used in the select menus).
   - **H323-ID** – h323-id (recommended `hostname.domainname`)

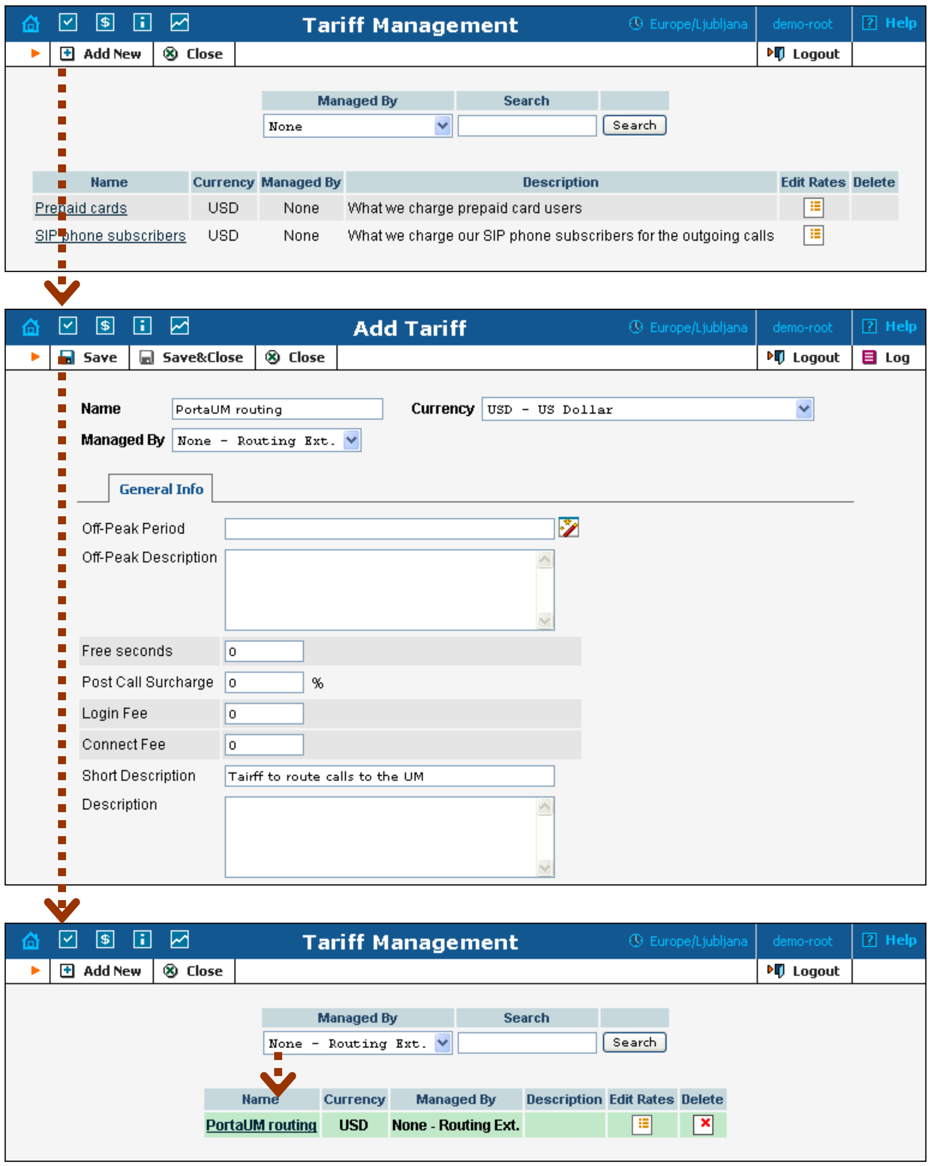
- **H323 Password** – If you plan to send calls from the UM gateway to your Cisco gateways, where the default Cisco remote IP authentication script will be used, put `cisco` here.
- **NAS-IP-Address** – IP address of the UM gateway or PortaUM server.
- **Auth. Translation rule** – you can just leave it empty.
- **Type** – node type, select **VOIP-GW**.
- **Manufacturer** - Select **Cisco (**even if the PortaUM is not manufactured by Cisco, it uses Cisco-compatible way to communicate with the billing).
- **Radius Client** – Check this, since both PortaUM and UM gateway will need to communicate with the billing.
- **Radius Key** – Put the radius shared secret here; must be the same **key** as you entered during the PortaUM installation.
- **Radius Source IP** – see ***Node ID, NAS IP address, and Radius source IP*** section in ***PortaBilling100 User Guide*** for more information. Unless your PortaUM server uses multiple network interfaces, value here should be the same as NAS-IP-Address

4. Click **Save&Close.**
5. Repeat steps 2-4 until to enter both .

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

Create Tariff

You have already created all necessary tariffs (for your customers and vendors) when you were setting up SIP services. Only one thing is remaining – to create the special UM tariff, which will be used for routing calls to the UM gateway.



1. In the Management section of the Admin-Index choose **Tariffs.**
2. On the Tariff Management page choose **Add New.**
3. Fill in the **New Tariff** form
   - **Name** – short name for tariff object. This is the name you will see later in the select menus.

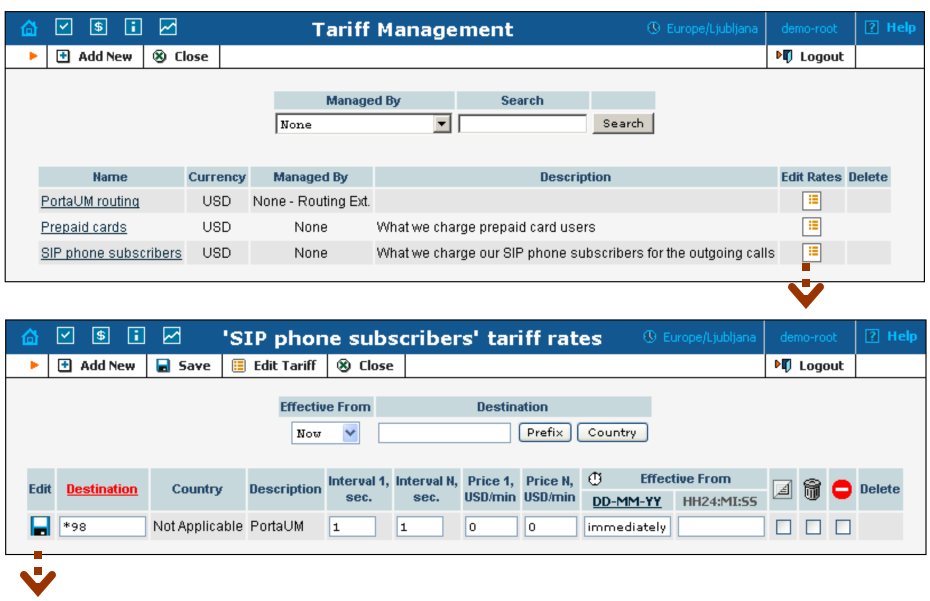
- **Managed by** – since you need to create a tariff for routing calls to the UM gateway, make sure you will choose **None – Routing Ext.** here.
- **Off-peak Period** – just leave this field is empty.
- **Off-Peak Description** – description of the off-peak period, automatically filled by the off-peak period wizard, you do not have to fill in this field
- **Currency** – since this is a "fake" tariff, just choose the same currency as your **Base currency**.
- **Free seconds, Post Call Surcharge, Login Fee, Connect Fee** – leave all these parameters empty.
- **Short Description** – short tariff description, put "Porta UM routing" or similar. This field is mandatory.
- **Description** – extended tariff description

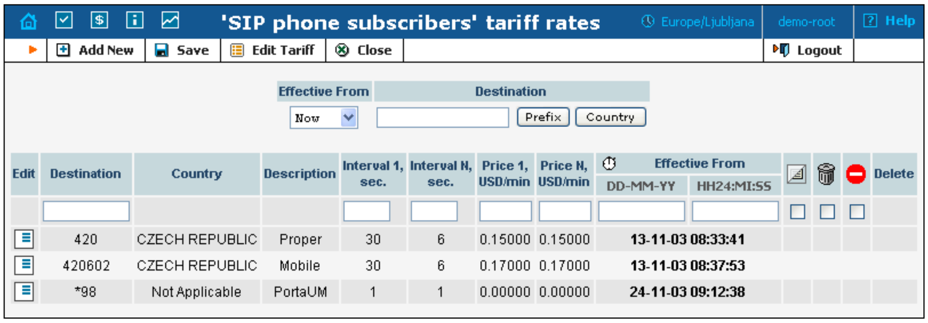
4. Click **Save&Close.**

Enter Rates

Modifying tariffs for SIP users

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





1. When you are on the Tariff management page and you see the list of the available tariffs, click the **Edit Rates** icon next to the name of the tariff. When you are in the Tariff management for a particular tariff – then click on the **Edit Rates** in the toolbar.
2. When in the **Edit Rates** screen click **Add New**
3. Fill in required information:
   - **Destination** – destination prefix for your voicemail, which you have created earlier.
   - **Interval 1** – first billing unit in seconds
   - **Interval N** – next billing unit in seconds
   - **Price 1** – per minute price for first interval. If you plan to provide calls to voicemail for free, put 0 here.
   - **Price N** – per minute price for next interval. If you plan to provide calls to voicemail for free, put 0 here.
   - **Off-peak Interval 1**– first billing unit in seconds for off -peak time
   - **Off-peak Interval N** – next billing unit in seconds for off-peak time
   - **Off-peak Price 1** – per minute price for first interval for off-peak time
   - **Off-peak Price N** – per minute price for next interval for off-peak time

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

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

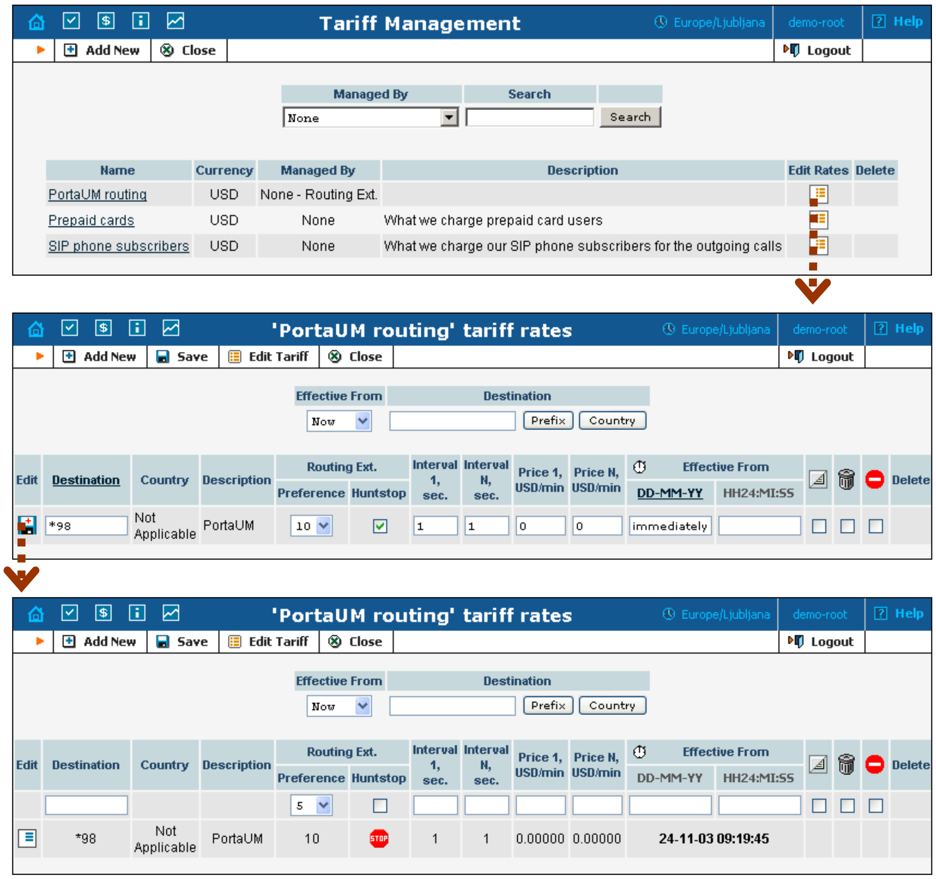
NOTE: When using calendar, you can specify that the date you entering is in the different time zone than your current one. PortaBilling will automatically adjust the time then.

   - **Hidden**, **Forbidden** or **Discontinued** flags are optional

4. Click **Save** button in the toolbar, or  icon on the left side of the row
5. Click **Close** button to return to the tariff management screen.
6. Repeat steps 1-5 and add rates for the UM prefix into all tariffs, where you would like you customers to use voicemail.

Rates for the UM routing tariff

1. When you are on the Tariff management page and you see the list of the available tariffs, click the **Edit Rates** icon next to the name of the UM routing tariff.

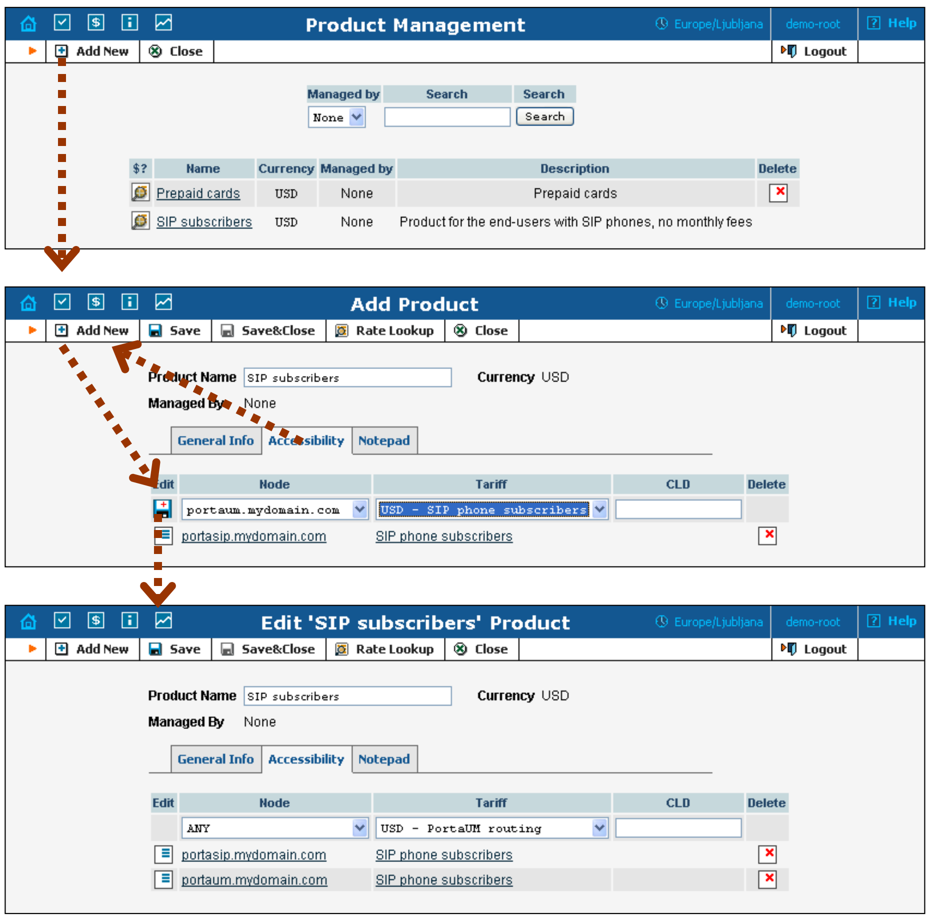


2. When in the **Edit Rates** screen click **Add New**
3. Fill in required information:
   - o **Destination** – destination prefix for your voicemail, which you have created earlier.
   - o **Preference** – put 10 in this field. This is the highest routing priority, since calls to voicemail should always be routed to the UM gateway.
   - o **Huntstop** – check the box, since we do not to use any other route for the UM except direct call to the UM gateway.
   - o **Interval 1, Interval N** – put 1 in both fields
   - o **Price 1, Price N** – put 0 in both fields (UM is our own gateway, so termination costs are zero).
   - o **Effective from** – leave **immediately** in the field.
   - o **Hidden**, **Forbidden** or **Discontinued** flags are optional
4. Click **Save** button in the toolbar, or icon on the left side of the row

5. Click **Close** button to return to the tariff management screen.

Modify Products

You have already created all necessary products for your customers when you were setting up SIP services. Only one thing is remaining – to make sure these product allow usage of the UM services. If some product's accessibility includes PortaUM server – it means that accounts with this product are permitted to login into the UM (via webmail, voicemail or from their email client).



1. In the Management 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. When Accessibility tab is selected, click on the **Add new** icon.
5. In accessibility entry window select the node which represents your Porta UM sever, and choose the appropriate tariff (the actual tariff is not really important, since on the PortaUM server users will not be

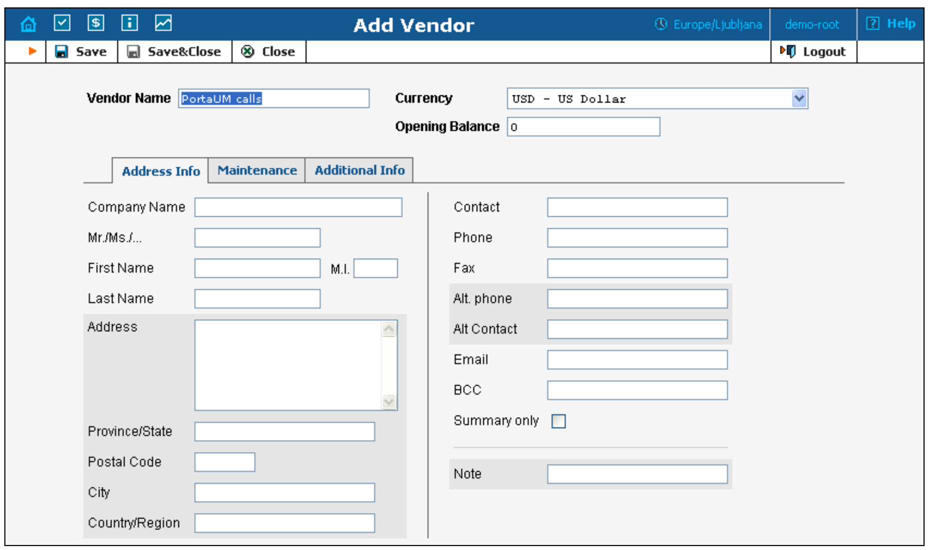
making any calls. The best is to choose the same tariff which is used to charge outgoing SIP calls)

6. Click **Save** to save this accessibility entry.
7. Click **Close** to return to the Product management page.
8. Repeat steps 2-7 to modify accessibility for all of the products which will include UM services.

Create UM vendor

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

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



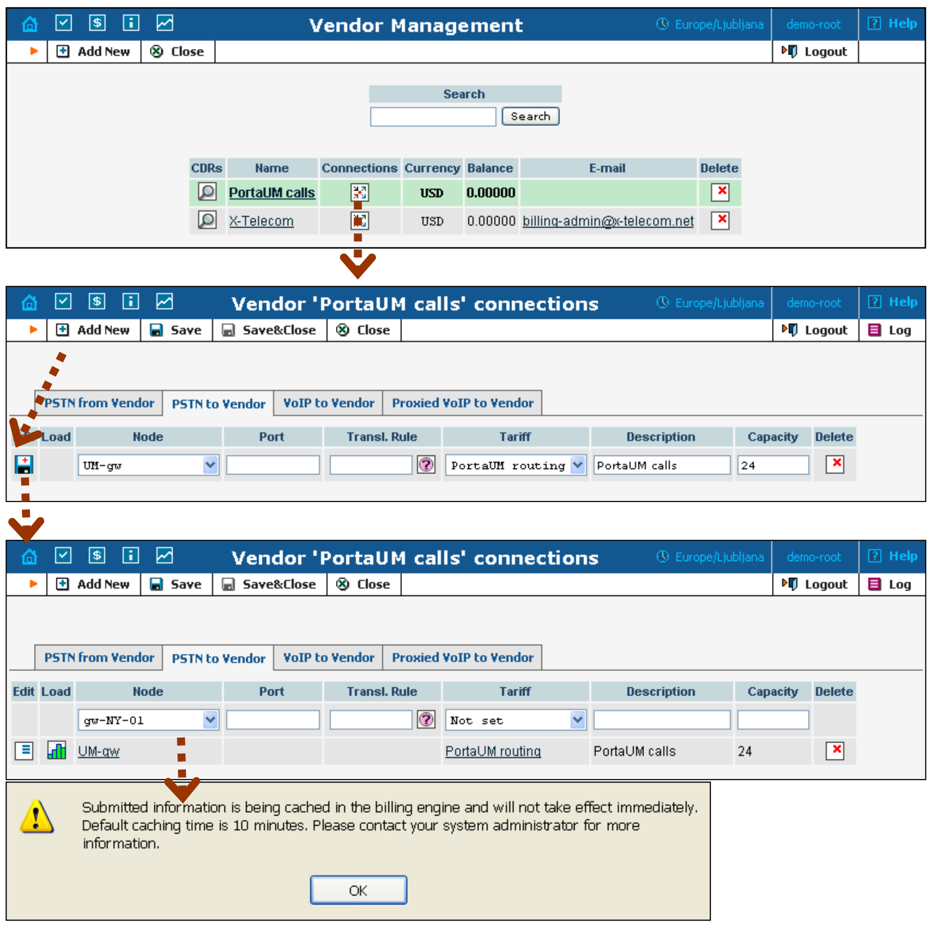
3. Fill in the **New Vendor** form. Please note that there are two tabs available on the screen. The most important fields are:
   Main form (top):
   - **Vendor Name** – short name for the Vendor object, will be used on the web interface
   - **Currency** – in which currency this vendor charges you.
   - **Opening balance** – starting balance for the vendor, default is zero

   Additional info:
   - **Billing period** – split period for vendor statistics.
   - **Time zone** – time zone, which vendor uses.
4. Click **Save&Close**.

Define connections

Vendors are your termination partners or providers of the incoming toll-free lines.

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

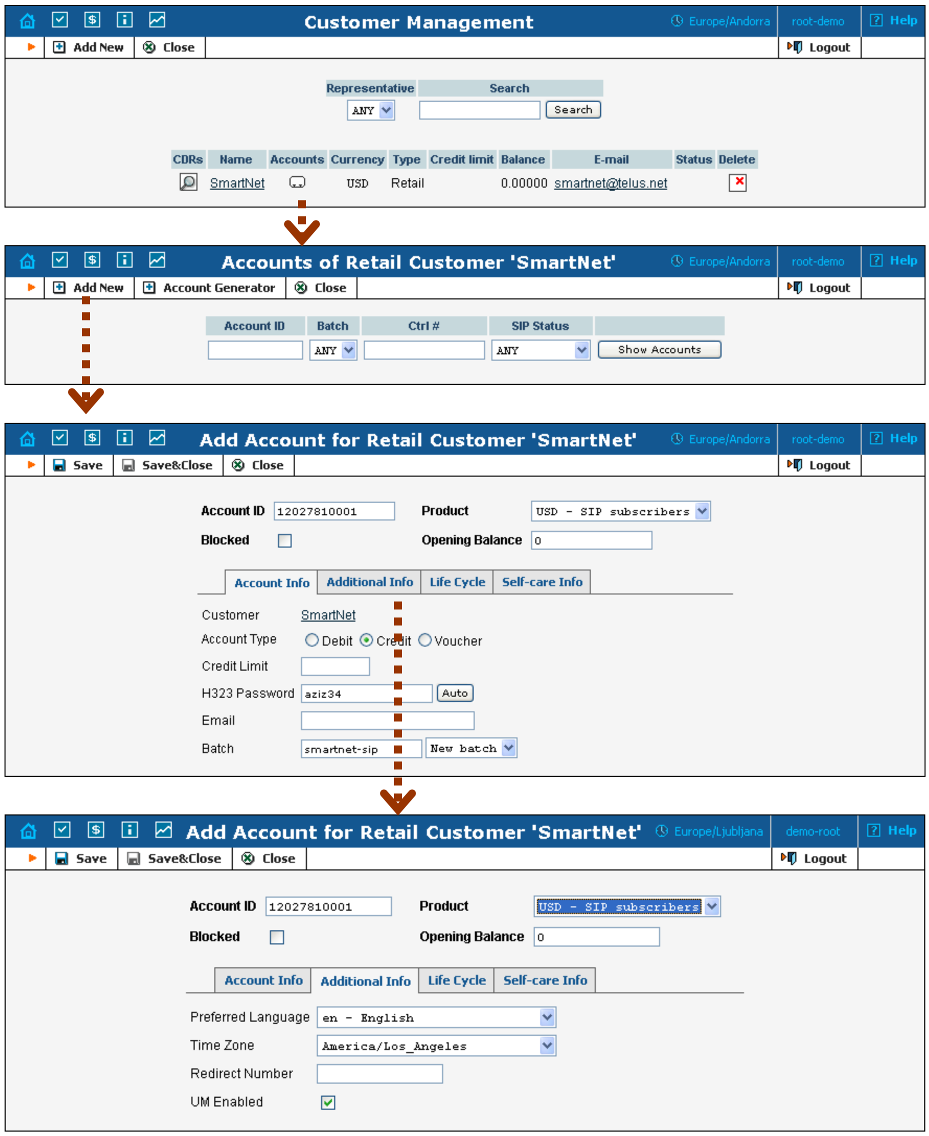


3. Choose type of the connection **PSTN to Vendor** by clicking on the corresponding tab.
4. Press **Add New** to add a new connection.
5. Fill in the connection information. Choose you UM gateway as a **Node**. Choose your Porta UM tariff you have created earlier in the select menu in the **Tariff** column. Put your comment in the **Description,** and **Capacity** should contain number of the simultaneous calls your UM gateway can handle.
6. Click **Save**.
7. Click **Close** in order to exit to the **Vendor Management** screen.

Create Accounts

Creating new accounts

1. If you are not yet in the **Customers** screen (the one which contains a list of customers) – enter there. It should look like the screenshot below.



2. Next to the customer name click on the icon (the one in the **Accounts** column), it will take you to the account management for that customer.
3. Click on the **Account New**.
4. Fill the "Add account" form:
   - **Account ID** – SIP ID – phone number, which will be used to login to the SIP server and to receive incoming calls.

- **Product** – choose the product which you would like your account to have.
- **Blocked** –. You may create your account as blocked, but it is rarely done with the SIP service accounts.
- **Opening balance** – the initial balance on the account.

Account info tab:

- **Account type** – account type, select credit for post-paid and debit for prepaid service.
- **Credit limit** – for the credit account specify the credit limit. If you leave this field empty, it means that there is no credit limit for this account (but customer credit limit may still apply).
- **H323 password** – this password is used for SIP services as well. Account ID and this password will be used to authenticate login to the SIP server.
- **Email** – you can enter the account owner's email address here. If he will ever forget his password to the web self-care he will be able to reset it, and the new password will be sent to this email address. You can just leave this field empty.
- **Batch** – batch is a management unit for accounts. Batch name is alphanumeric. You can type a new name here or use the existing name in order to generate more accounts for same batch.

Additional Info tab:

- **Preferred language** – this is a custom attribute, which is transferred to the IVR. Leave English there if you are not sure if your IVR supports it.
- **Time zone** - when account owner (pre-paid card user) will access web self-care pages to see a list of his calls, we can show time in the most appropriate time zone for him.
- **Redirect number** – redirect number (discussed in the **Advanced features** section), leave it empty.
- **UM enabled** – check if this account has unified messaging (e.g. voicemail) services enabled.

Life Cycle tab:

- **Activation date** – account activation date.
- **Expiration date** – account expiration date.
- **Life time** – relative expiration date, account will expire on "first usage date" + "life time" days. If you do not want to use this feature, leave the field blank

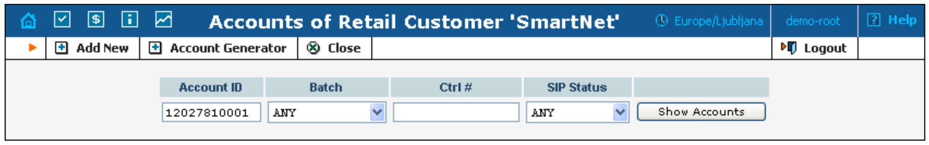
Self-care Info Cycle tab:

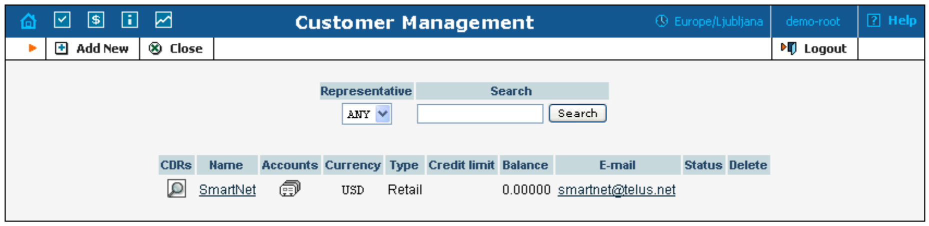
- **Login** – account login to the web self-care pages. Can be the same as account ID.
- **Password** – password for the web self-care pages.

5. Click **Save&Close**, you'll get confirmation screen saying that the new account has been created.

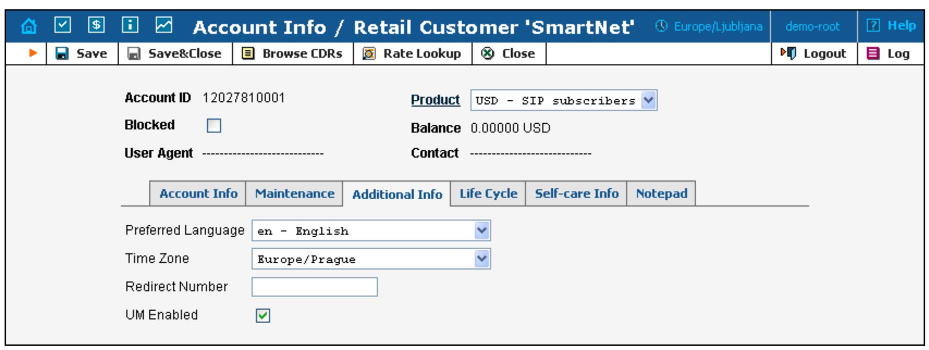
Enabling UM services for existing account

1. If you are not yet in the **Customers** screen (the one which contains a list of customers) – enter there. It should look like the screenshot below.





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



4. Click on the **Additional Info** tab.
5. Click on the **UM enabled** checkbox to activate UM services for this account.
6. Click **Save&Close.**

Using UM services

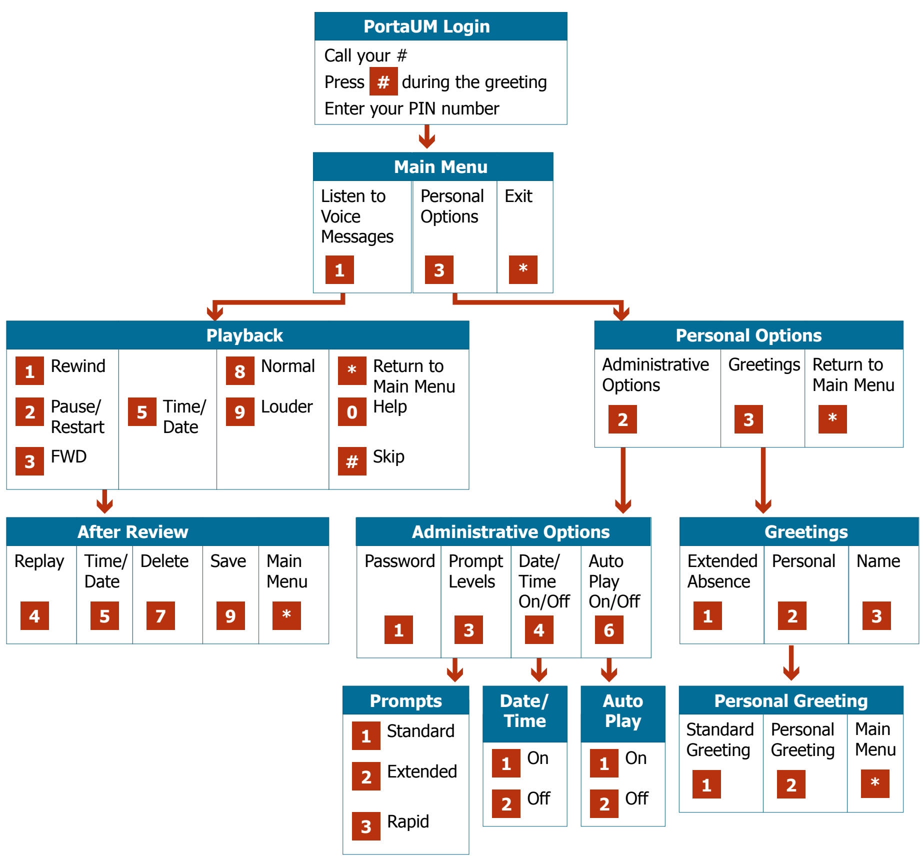
PortaUM allows Internet Telephony Service Providers to offer their subscribers the ability to process emails, manage voicemails, all from within a web browser, or a favorite email client.

The PortaUM End-User Manual is available for from PortaOne website: www.portaone.com/solutions/portaum/

The PortaUM user can also access the mailbox via a telephony interface which allows to listen to voice messages, and configure the UM options for accessing the mailbox via phone.

IVR Path Diagram

PortaUM mailbox can be accessed via phone by dialing *98.
Below is the IVR Path diagram to help users navigating and configuring the system.

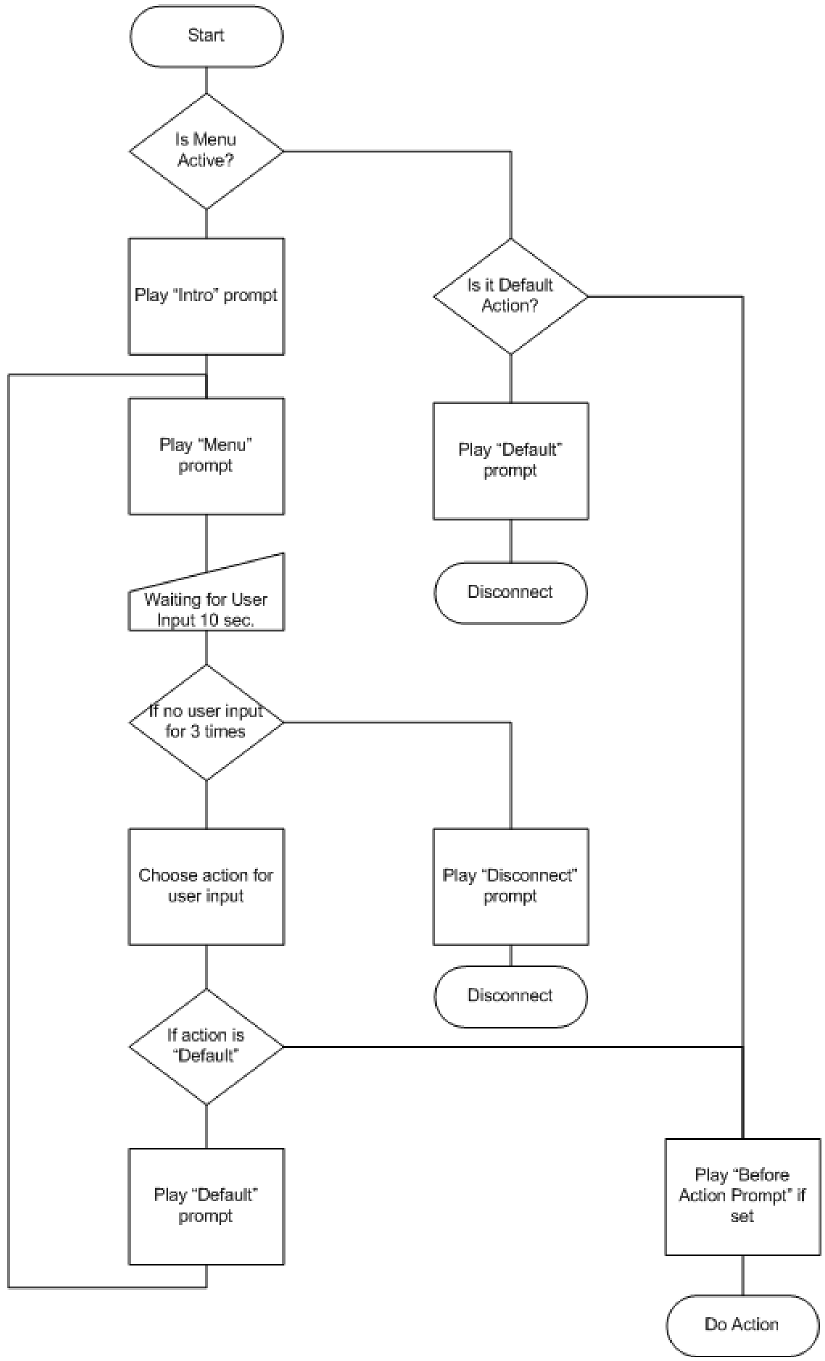


Auto Attendant

PortaUM Auto Attendant is a flexible utility designed to greet callers and transfer them either to an existing PortaSwitch accounts, or to your current phone system.

Basic concept

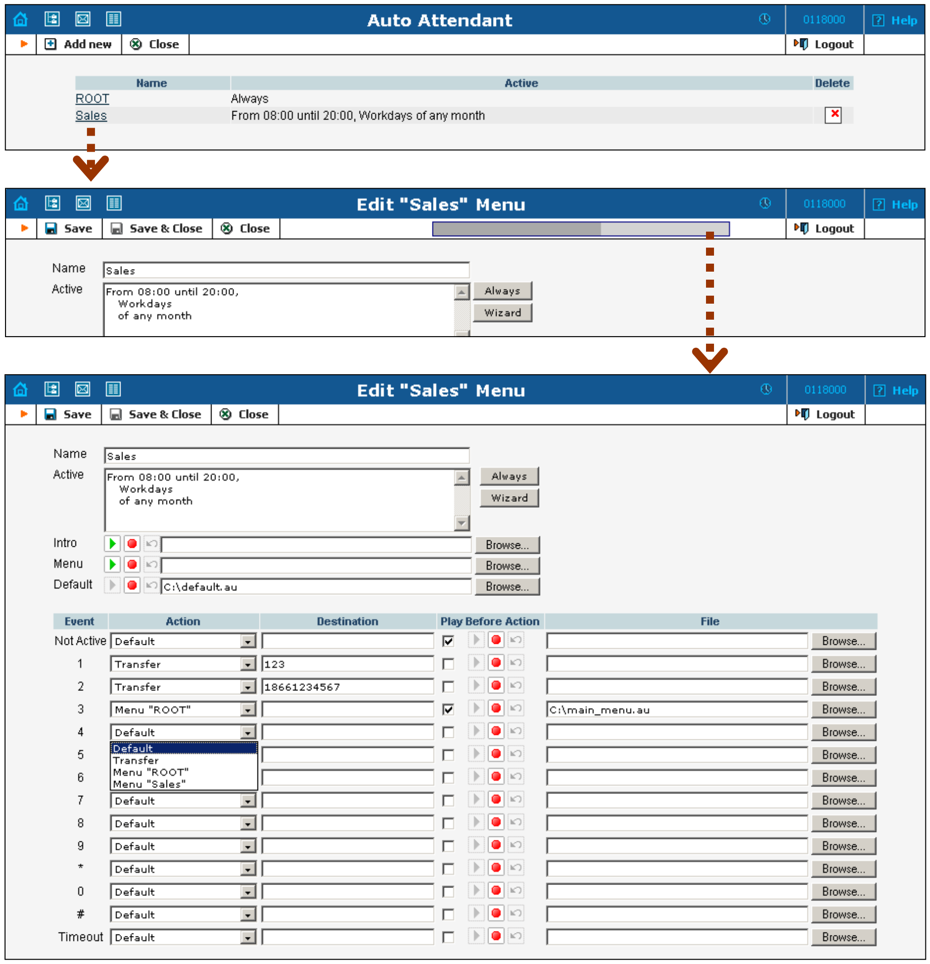
- PortaUM Auto Attendant (AA) is composed of a set of menus.
- All the menus are the same in every respect, except for the ROOT menu, which is always present and cannot be deleted, and whose name cannot be changed.
- When a caller dials the system, AA will answer the call with the Intro prompt from the ROOT menu.
- After this, the Menu prompt will be played, and AA will listen to the user input.
- The user input will trigger execution of the following available actions:
    - **Default** – plays the default prompt from the current menu
    - **Transfer** – transfers the call to a given telephone number or extension
    - **Menu** – starts interpreting (executing) the selected menu; the user can choose from any of the available menus
- The user may select whether the corresponding prompt is to be played prior to the action.
- A menu call flow chart is displayed in the diagram below:



Menu List mode

Auto Attendant can be selected from the Options menu. The main screen lists all the menus available in the system.

To modify one of the existing menus, select its name from the list.
To add a new menu, select **Add** from the action panel.



Menu Edit mode

After selecting one of the existing menus, please allow all the prompts to load in your browser; this may be viewed on the status indicator in the action pane.

The fields of the Menu Edit screen are explained below:

Name
A logical name for the menu, i.e. Sales for a sales department

Active
Time definition when the current menu is active. To set the menu as always active, select the Always button on the right. PortaUM also provides users with a Period Wizard, a flexible tool for defining a time period of any complexity.

Period Definition Wizard

Via a sequence of screens, the user may select the time interval, day of the week, day of the month, and month; multiple selection is allowed.

The following example shows the process of creating a period starting at 6pm every day and lasting until 6am the next morning. Another interval is used during weekends. We will also include some holidays, e.g. January 1 and December 24-26.

In the first screen, select 6pm in the **From** column and 6 am in the **Until** column. Now select the **Next** button. The two text areas on the right side of the screen provide the user with a display of the current period definition. The top text area displays a verbal definition of the period: `From 6:00pm until 6:00am`, while the bottom one contains the same information in a format which can be parsed by PortaBilling: `hr{6pm-5am}`. This sets up the first period; in order to continue, skip the following screens by pressing the **Skip** or **Next** button until the **Period definition completed** message is displayed. Click **Add** to create another period definition; the wizard returns to the first screen.

Now for weekends: by pressing the **Skip** or **Next** button, go to the second screen and select *Weekend*, or hold the <Ctrl> key and select *Saturday* and *Sunday* from the list. Now use the **Next** button, skipping forward until the **Period definition completed** message is displayed. Click **Add** to create another period definition.

To include January 1st in the period definition, skip to the Day of Month screen and select *1*. Now click the **Next** button. Select *January* and click **Next**, skipping forward until the **Period definition completed** message is displayed. Use the same approach to select the December 24-26 interval. Hold the <Ctrl> key to select multiple entries.

To review your work, look in the top text area. The following should be displayed:

```
From 6:00pm until 6:00am
   any day of any month
OR Sunday and Saturday
   of any month
OR 1
   of January
OR 24-26
   of December
```

If the definition is correct, select **Finish**.

Intro, Menu, Default
These three fields work similarly to the Voicemail recording feature.

- **Record**. Select to start recording your voice prompt. (You will need to connect a microphone to your computer sound card to use this feature.)
After the existing prompt is recorded over, the **Undo** icon becomes available, allowing rollback to the previous state. The blinking **Play** icon indicates that the existing prompt is being overwritten, but changes have not been saved yet.

- **Stop.** Select this to stop recording or playback of the recorded message.

- **Play.** Select this to play back the recorded prompt. When selected, this icon will turn into - **Pause**.

Each of the icons above may appear in grayscale, meaning it cannot be accessed because some other task is active.

To give your Auto Attendant a professional sound, we recommend using a professional speaker and a digital recording studio when recording voice prompts.

To upload a prompt, select the **Browse…** button on the right side. The native audio file format for the system is the following:

Type: NeXT/Sun (Java) file .au
Format: G.711 u-Law
Attributes: 8,000 Hz, 8-bit, Mono

PortaUM uses SOX - Sound eXchange, a universal sound sample translator for prompts uploaded into the native UM format.

Here's a short list of supported audio file formats:

Type	Description
.aiff	AIFF files used on Apple IIc/IIgs and SGI.
.au	SUN Microsystems AU files.
.gsm	GSM 06.10 Lossy Speech Compression
.mp3	MP3 Compressed Audio
.ogg	Ogg Vorbis Compressed Audio.
.raw	Raw files (no header).
.wav	Microsoft .WAV RIFF files.

Event Table

Column	Description
Event	**Not Active** – if the current menu is not active (see the

	active period definition above) **0-9, #, *** – user selection on telephone keypad **Timeout** – no selection received from user
Action	○ **Default** – play the default prompt from the current menu ○ **Transfer** – transfer the call to a given telephone number or extension ○ **Menu** – start interpreting (executing) the selected menu; the user can choose from any of the available menus
Play before action	Check this box if the corresponding prompt is to be played before the action is performed
File	File name and path