PORTA ONE

Porta UM
unified messaging

Administrator Guide
Maintenance Release 15

Copyright Notice & Disclaimers

Table of Contents

ccnessvia, aweb interefaceforby telephone.e

Hardware and Software Requirements

Should you decide to use a Cisco gateway for processing UM calls, please note that PortaUM requires a *dedicated* Cisco AS 5300/5350. No Cisco gateway is required if you wish to run the IVR embedded into PortaUM.

Cisco Requirements

128M RAM, 16M flash for AS5300 (32M for AS5350), E1 or T1 voice ports, sufficient number of DSPs, IOS 12.3.13 (or the latest available from the 12.3 branch).

AS5300 comes with 4 or 8 T1/E1s. You will not need more than 4 of them, due to the DSP resource limitation for AS5300.

For a T1 configuration, the maximum voice resource will be 96, and for E1 — 120.

Client System Recommendations

- OS: Windows 95-XP, UNIX or Mac OS
- Java Virtual Machine Support (JRE 1.4.2_01 or later)
- Browser: Internet Explorer 6.0 or higher, Netscape 7.1 / Mozilla 1.6 or higher, supporting DOM and with JavaScript enabled
- 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, proceed as follows in order to match the PortaBilling default list separator: My Computer -> Control Panel -> Regional Settings -> Number -> List Separator type ",".

Installation

PortaUM is provided via a jump-start installation CD. This CD contains installation media for FreeBSD (6.1-stable branch, with the latest security bug fixes), supplementary packages necessary for convenient system administration and maintenance, and PortaUM software packages.

PortaUM installation and configuration are automated and integrated within the main installation process. This allows you to install a completely functional PortaUM server from scratch in less than 15 minutes! In order to complete the installation process, your Cisco UM gateway must be configured according to the configuration guidelines.

For detailed installation instructions, please refer to the **PortaUM Installation Guide.**
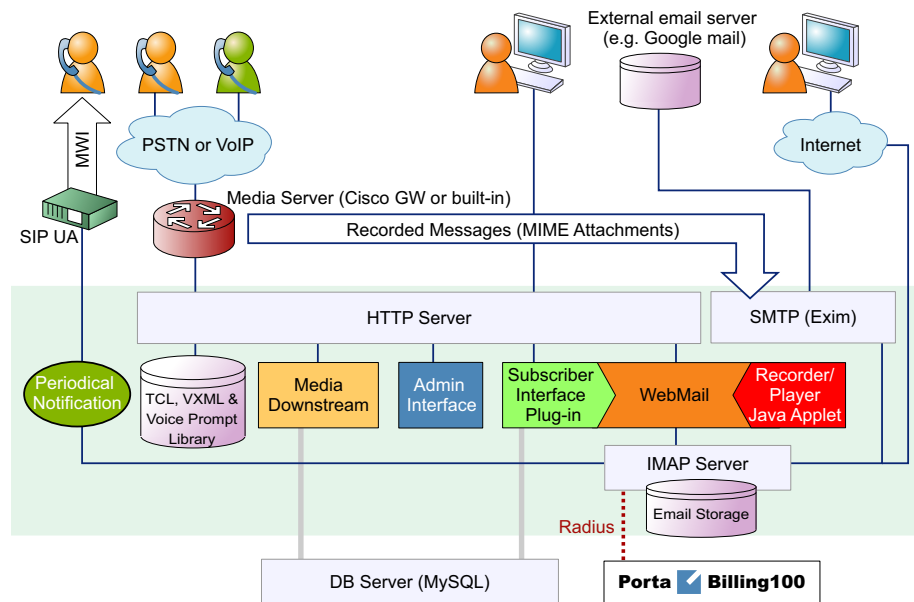
What's New in Maintenance Release 15?

This release includes several new features and improvements:
- Call queues
- Dial-by-name auto attendant menu
- Prepaid card IVR on PortaUM without using a Cisco gateway

1. System Concepts

PortaUM Components

PortaUM consists of the following main components

- Media server / VXML platform. This component is used to record messages and receive faxes, as well as send them to the PortaUM machine for processing and storing. It also provides users with IVR (interactive voice response), allowing them to listen to and manage recorded messages using a SIP phone. Starting with PortaUM Maintenance Release 11, this component can be implemented by:
    - using an external Cisco AS53XX gateway, or
    - using IVR tools built into the PortaUM server itself.
- A web server, used to provide the user with web-based access to his mailbox. The web server is also used, along with special CGI scripts, by the VXML platform to retrieve recorded messages from the PortaUM machine on demand.
- An IMAP server, used as the main storage for messages.
- A database server, providing storage for the user's custom settings and voice prompts.
- An SMTP server, used to receive incoming messages from the VXML platform and other SMTP servers, as well as to send outgoing messages created by the user on the web interface.
- A mail filter, used to convert audio and graphic attachments in messages received from the VXML platform in a format compatible with the Microsoft Windows operating system.

Conceptual Elements

Mailbox

A mailbox is a portion of disk space on the PortaUM server which has been allocated for storing messages. Messages can be voice messages, faxes, or generic emails. You may set a quota (maximum allowed disk space) for user mailboxes, to prevent a situation where some users store too much information in their mailboxes, thus creating problems because the disk space is unavailable for other customers.

Call Process

SIP UA <-> SIP UA
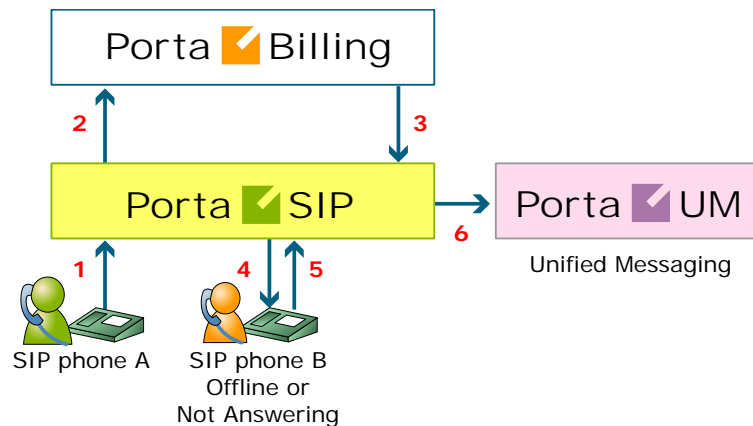
- User A dials 121 in an attempt to reach user B. A's SIP user agent sends an INVITE request to the SIP server (**1**).
- The SIP server performs authorization in the billing (**2**). For example, if A dialed 121, the billing will inform the SIP server that the actual number is 12027810009, and that this number belongs to B (**3**).
- The SIP server checks the registration database, but it appears that this account is not online at the moment. If B has Unified Messaging services enabled, the call will be redirected to the voicemail system, and A can leave a message for him (**6**). The same thing would happen if B were online, but did not answer his phone (**4**), (**5**).

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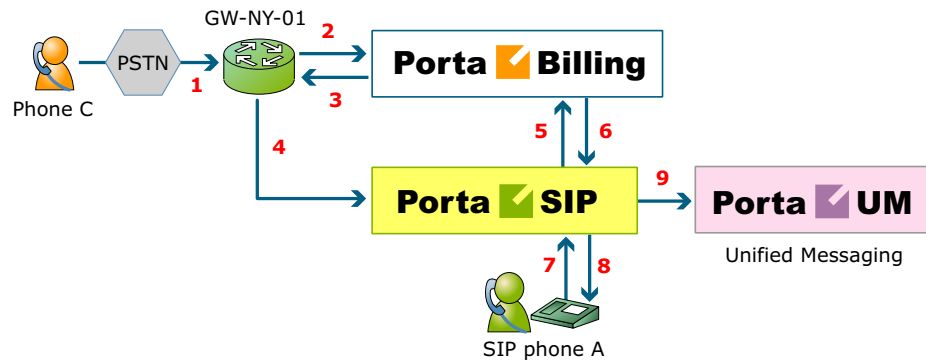
This is another important aspect of SIP telephony. Your subscribers not only want to make outgoing calls, they also want other people to be able to call them on their SIP phone regardless of where they are at the moment.

In order to allow this, you will need to secure a 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.

- User C wishes to call user A. He dials A's phone number; since C is in the USA, he dials it in North American format, i.e. 2027810003.
- The call is routed through the telecom network to gateway GW-NY-01. When the incoming call arrives on the gateway (1), it starts a special TCL application to handle this call. This application does several things:
  - Converts the phone number to E.164 format, so that 2027810003 becomes 12027810003.
  - Performs authorization in the billing (2), i.e. checks whether A is allowed to receive incoming telephony calls from GW-NY-01 and, if you charge for incoming calls, what is the maximum allowed call time based on A's current balance (3). One important point is that authorization must happen without a password check, since the application does not know the valid password for the SIP account.
  - Starts the outgoing call to 12027810003.
  - When the call is established, it starts the timer, and will disconnect the call if the maximum call duration is exceeded.
  - The gateway is configured so that it knows that calls to 1202781…. numbers should be sent to the PortaSIP server. Thus it sends an INVITE to the PortaSIP (4).
- PortaSIP receives the INVITE, but without authorization information. Thus the PortaSIP server performs authentication

based on the IP address (**5**), (**6**). Since this call comes from a trusted node, i.e. gateway GW-NY-01, it is authorized.

- The SIP server checks its registration database, but it appears that this account is not online at the moment. If A has Unified Messaging services enabled, the call will be redirected to the voicemail system, and C can leave a message for him (**9**). The same thing would happen if A were online, but did not answer his phone (**7**), (**8**).
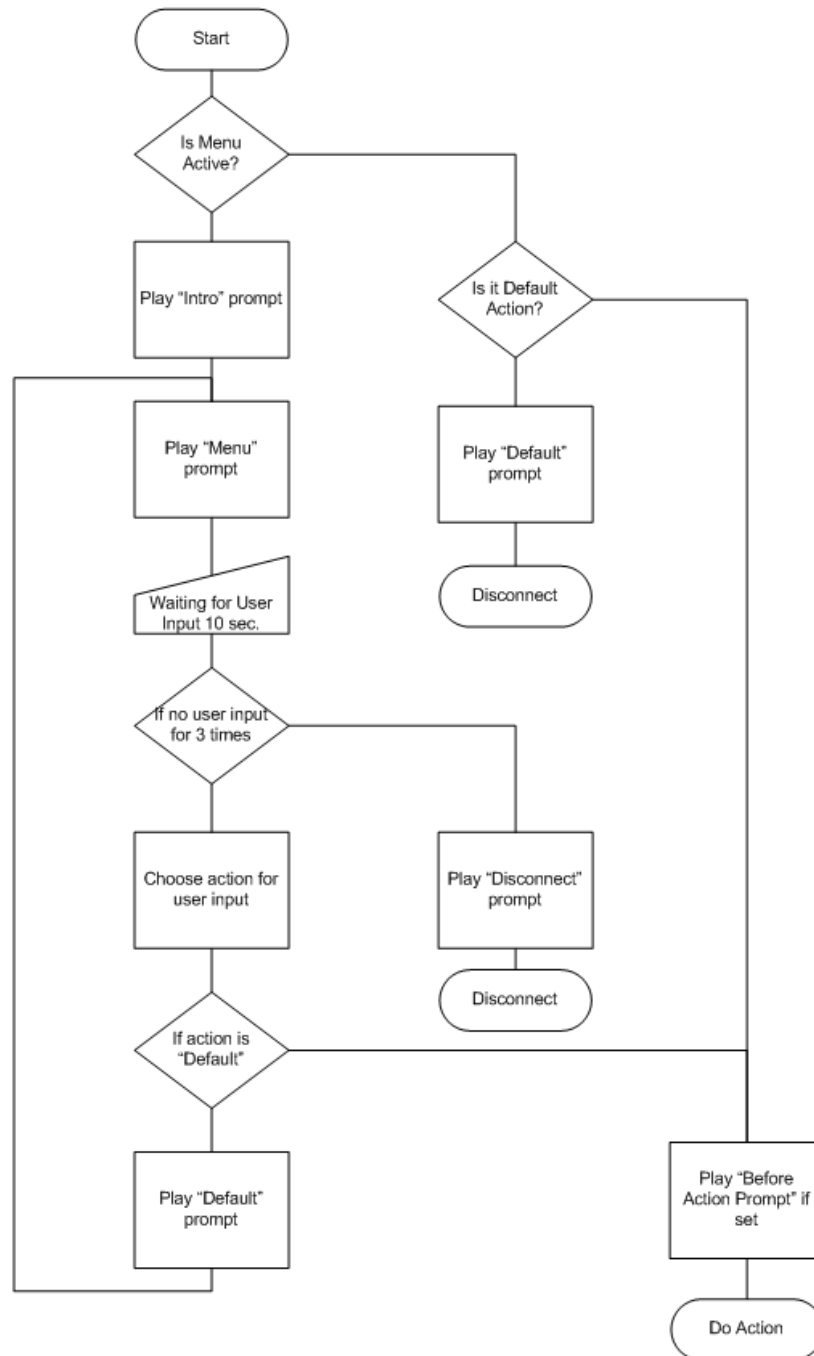
Auto Attendant

PortaUM's auto attendant is a flexible utility designed to greet callers and transfer them either to an existing PortaSwitch account, or to your current phone system.

Basic Concept

- PortaUM's auto attendant is composed of a set of menus.
- All the menus are identical 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, auto attendant will answer (connect) the call and proceed to the ROOT menu.
- If a user tries to access a menu which is not currently active, the action specified in the **Not Active** configuration parameter will be performed; for instance, the user may be automatically forwarded to an "after hours" menu.
- The **Intro** prompt (e.g. "Welcome to PortaOne, a VoIP solutions company!") is played when a user enters a menu for the first time.
- After this, the **Menu** prompt will be played, listing all the available options (e.g. "Press 1 for sales, press 2 for technical support"), and auto attendant will collect the digits dialed by the user on his phone touchpad.
- If no input is received (timeout), the **Default** prompt is played and the dialog reverts to the previous step (i.e. plays the **Menu** prompt and collects the user's input).
- The user's input will be matched with the corresponding menu items, and the action associated with this item will be performed. The following actions are possible:
  o **Default** – Plays the **Default** prompt from the current menu and returns to the "Play **Menu** prompt" step (this is the action used for all menu items where the initial value has not been modified).
  o **Transfer** – Transfers the call to a given telephone number or extension. The phone number should be entered in the same format as the customer would use to dial it from an IP phone

in his IP Centrex environment; for example, to transfer a call to extension 123, simply enter 123.

- o **Transfer to E.164 Number** – Transfers the call to a given number. The number should be specified in E.164 format: the country code, followed by the area code, and then the number (e.g. 16045551234 for Canada).
- o **Transfer to Extension** – Transfers the call to an extension number entered by the caller from his phone. To prevent abuse (e.g. someone attempting to enter a long-distance number in this way), you can specify the maximum allowed number of digits in an extension (**Max Size**).
- o **Transfer to Voicemail** – Switches to voicemail mode. This should be designated as an action for the "Fax" event, in order to allow storage of received faxes.
- o **Menu** – Transfers the user to the selected menu.
- o **Directory** – Launches this company's dial-by-name directory.
- o **Queue** – Transfers the call to the specified call queue.
- You may select whether the corresponding **Before Action** prompt is to be played prior to the action.
- A call menu flow chart is shown 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.
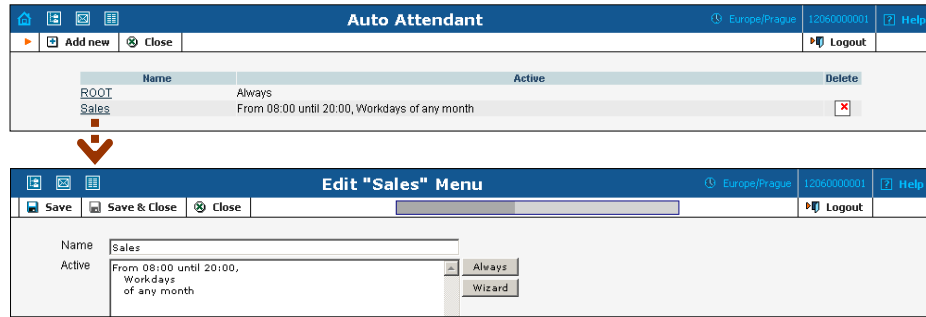
Menu Edit Mode

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.
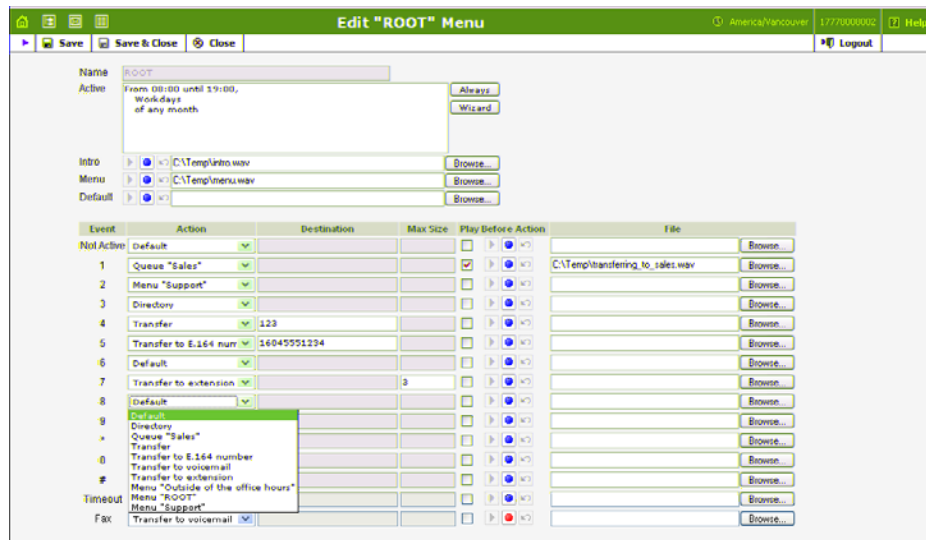
The fields of the Menu Edit screen are explained below:

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

Active
Time definition for 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 series of screens, the user may select a time interval, day of the week, day of the month, and month; multiple selections are allowed.

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

In the first screen, select 6 pm 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, and the wizard will return 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 the 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. Follow the same steps to select the December 24-26 interval. Hold the <Ctrl> key to select multiple entries.

To review your work, look at 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
You may define three separate prompts (see the previous section for an explanation of when each particular type of prompt is played). The recording controls on these three fields work similarly to the controls for recording a voicemail greeting.

🔴 - **Record**. Select to start recording your voice prompt. (You will need to connect a microphone to your computer's sound card to use this feature.)

After the existing prompt has been 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 native UM format.

Here is 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** – When 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. **Fax** – Fax CNG tone detected.
Action	See the description of available actions in *Basic Concept* section above.
Play before action	Check this box if the corresponding prompt is to be played before an action is performed.
File	File name and path for the prompt file.

Call Queues

This feature allows you to provide a "call center" functionality to your IP Centrex customers. When a large number of incoming calls from customers arrives to the auto attendant, PortaUM can forward these calls to the actual agents (customer service representatives) in a regulated fashion.

Every call queue contains several configuration parameters:
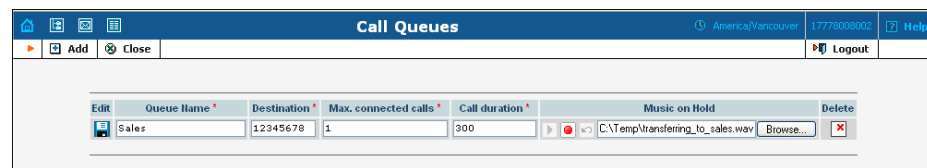
- **Destination** – A phone number where an outgoing call may be forwarded to. The fact that this is a single number does not, of course, mean that you can only have one agent answering calls. This is simply the number used to forward calls to PortaSIP. On the PortaSIP side, you can use the call forwarding feature to direct an incoming call to multiple IP phones. Or, alternatively, you may forward this call to an external IP PBX or a gateway with multiple FXS ports; then "hunting" for an available agent will be done on that side.
- **Max Connected Calls** – This defines the "bottleneck" of your queue, i.e. the maximum number of concurrently connected outgoing calls.
- **Call Duration** – The average expected processing time for each call (used to calculate the estimated waiting time).
- **Music on Hold** – A melody (or announcement) which is played to users waiting to be connected.

Each call queue contains a pool of incoming calls (users trying to get connected) and a number of connected outgoing calls (calls that have already been connected to agents). When a new incoming call arrives, it is

assigned a position in the queue. The caller will hear an announcement about his position in the queue and the estimated waiting time, which is calculated as (average call duration) / (maximum number of connected calls) * (total number of users before him in the queue). After that, the specified "music on hold" is played, and every minute the caller is updated as to his current position in the queue and the estimated waiting time.

If there are callers on hold and the number of connected outgoing calls is lower than the specified threshold, PortaUM will attempt to connect the first person in the queue. A call invitation is sent to the destination number in PortaSIP, and PortaUM waits until the call is answered by the other side. If the call is not connected on the first attempt (some representatives may not be available at the moment), PortaUM will make another attempt, then another one, and so on. This will continue until either all of the incoming calls are connected, or the maximum threshold for outgoing calls is reached. In the latter case, PortaUM will simply wait until one of the agents finishes serving his current customer, thereby disconnecting one call and making "room" for a new outgoing call.

Dial-by-name Directory

This is another element of the auto attendant IVR functionality. If a caller does not know the extension number of the person he is trying to reach, he may look up the called party using the first three letters of his surname.

Every UM account has its own dial-by-name directory; however, an initial list of extensions may be imported from the main list in PortaBilling to save time and effort. You can upload a voice prompt with the actual person's name for each extension. You may also exclude certain extensions from being accessible via dial-by-name (e.g. you do not want

tele-marketers to directly reach your CEO or CFO because their names are publicly accessible).

The dial-by-name directory can be assigned as an "action" item to any element in the ROOT menu or sub-menu. When a user reaches the dial-by-name dialog, he will be prompted to enter the three first letters in the called party's surname. Standard phone mapping is used, i.e. 2 is ABC, 3 is DEF, and so on. If no matching person is found, the user is informed of this, and may then re-enter the name or press * to exit. If more than one match is found (e.g. there are two persons with the "same" surname in the company, e.g. 276 will match both Brown and Asok), the user will hear a list of matching names and their extensions, and may then enter the correct extension.

Supported Services

Leaving voice messages for a PortaUM user

PortaUM gives your SIP customers an automatic answering machine whenever they are not online or do not answer within a certain (configurable) time period. This service works when a call comes from PSTN, as well as when it comes from another SIP customer. See the Call Process section above for detailed call flow descriptions.

Sending faxes to a PortaUM user

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

Sending email messages to a PortaUM user

PortaUM also allows your SIP customers to have regular email accounts, where they can receive regular emails.

Retrieving voice and email messages using a SIP phone

Your SIP customers will be able to retrieve their messages using a SIP phone. To do this, they must dial a special number, which transfers them to the PortaUM IVR menu. This menu allows them to listen to recorded messages. Additionally, for SIP phones that support a message waiting indicator (MWI), PortaUM can automatically manage the SIP phone's MWI status, so that the user is notified when he has new messages in PortaUM.

Retrieving voice messages, email messages and faxes using a web browser

Users can retrieve their messages and send regular email messages using a standard web browser. The advanced Java applet built into the interface allows users not only to listen to voice messages, but also to compose new voice messages (a microphone is required). It is also possible to access the system using any email client that supports the IMAP or POP3 protocols.

Forwarding messages to an external email server

When a new message arrives in the PortaUM message storage, a full copy of the message, or just a short notification, can be sent to the user's other email address (e.g. corporate mail server or private gmail account), depending on the user's preferences.

Managing personal PortaUM settings using a SIP phone

In addition to retrieving messages, your SIP customers can manage their personal PortaUM settings, such as greetings, from a SIP phone using the PortaUM IVR menu.

Managing personal PortaUM settings using a web browser

Users are also able to manage their personal PortaUM settings by logging into the PortaUM web interface. The advanced Java applet built into the interface allows users to record personal greetings (a microphone is required).

Managing your current balance

PortaUM users can listen to the current state of their accounts by using the PortaUM IVR menu. There is also an IVR menu enabling the use of vouchers to add funds to an account.

Prepaid card IVR

PortaUM includes an IVR system that allows you to offer a prepaid card service for incoming VoIP calls. This IVR supports all the PortaBilling extensions for the prepaid card application (access codes, real and announced call duration, and so on).

Auto attendant

Every user can program his own set of menus for auto attendant, from a simple message such as "I am on vacation right now, press 1 to connect to my assistant" to a complex menu system with various options for small or medium-sized businesses. Users can program the voice dialog from

their web browser, record voice prompts using a microphone on their PC or upload professionally recorded prompts, create multiple sub-menus and define an activity period for each of them, program their company's dial-by-name directory, construct multiple call queues, and so on.

2. Setting Up and Using UM Services

Setting Up UM Services

Initial Configuration of PortaBilling

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

Creating 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.

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. Note that in order for the Play Current Balance (***11**) and Recharge Voucher (***12**) features to work you are to add these two special destinations.

1. In the Management section of Admin-Index, choose **Destinations**.
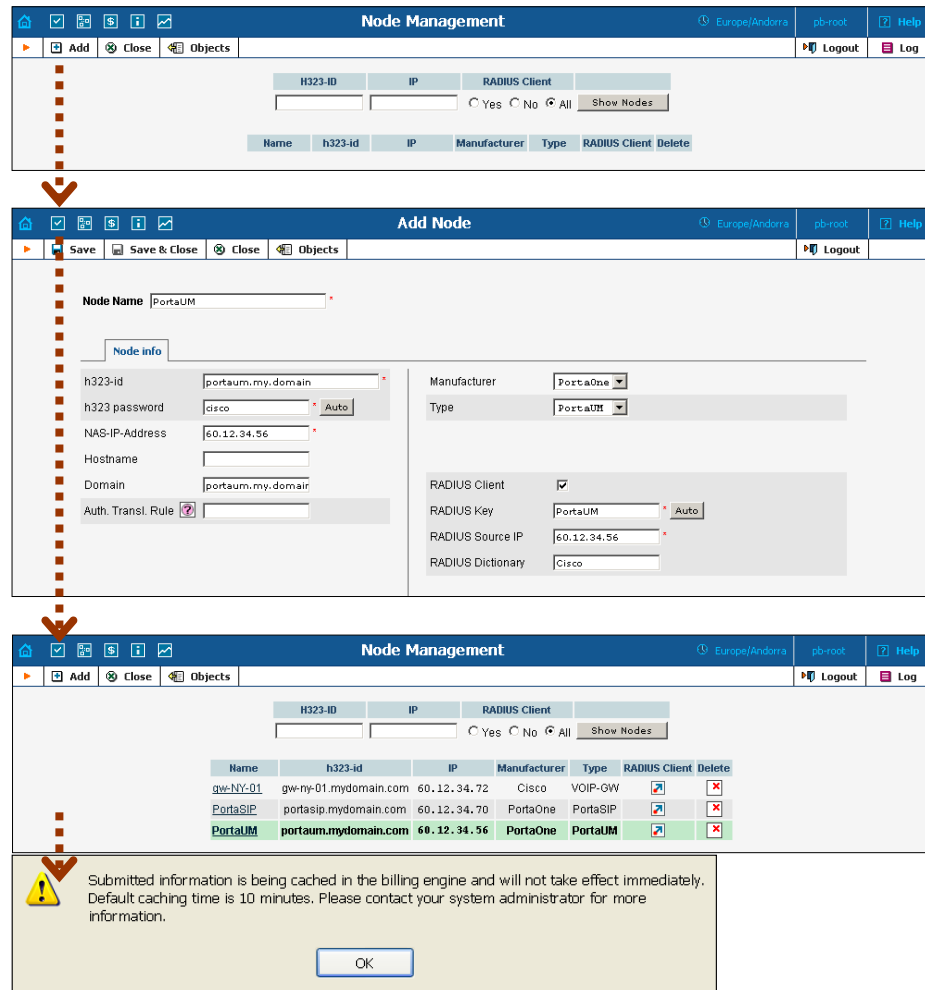2. Click on the ⊞ **Add** button.
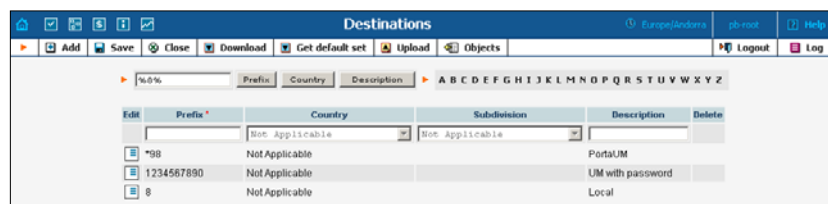
3. Enter your on-net UM number under **Prefix**, and **N/A** for the country and country subdivision. Put a comment in the **Description** column that clearly identifies this as a special prefix assigned to UM.
4. Click ▣ **Save.**

5. Add a destination for the PSTN UM number if you have one.
6. Add Play Current Balance destination (optional).
7. Add a destination for Recharge Voucher (optional).

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, h323-id, Radius shared secret, and so on.

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

- o **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, enter `cisco` here.
- o **NAS-IP-Address** – IP address of the UM gateway or PortaUM server.
- o **Auth. Translation rule** – You can just leave this empty.
- o **Manufacturer** – Select **PortaOne**.
- o **Type** – Node type; select **PortaUM**.
- o **Domain** – Enter the UM email domain, e.g. um.mydomain.com. This parameter is very important, since without it authentication requests from PortaUM cannot be processed correctly. ᴏᴏₒᴏᴏ
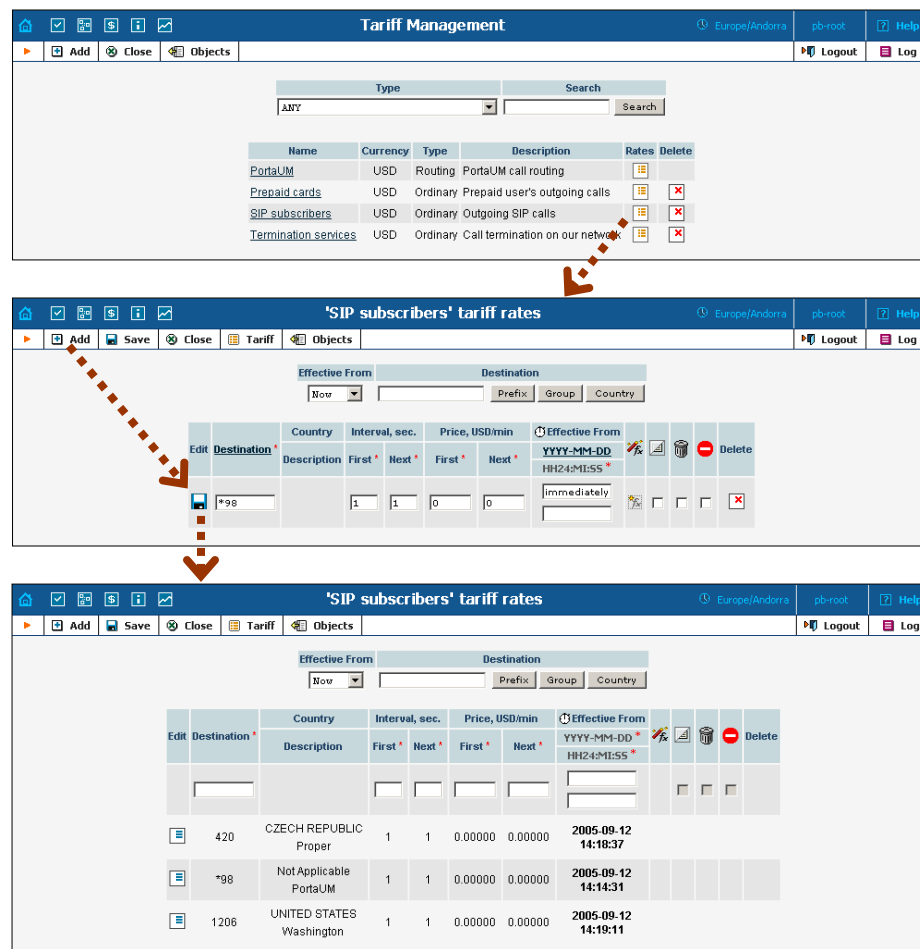
1. In the Management section of the Admin-Index, choose **Tariffs.**
2. On the Tariff Management page, choose ⊞ **Add.**
3. Fill in the New Tariff form:
   o **Name** – A short name for the tariff object. This is the name you will see later in the select menus.
   o **Currency** – Since this is a "fake" tariff, just choose the same currency as your **Base Currency**.
   o **Applied To** – Choose a **Vendor** here, then check the **Routing** check-box below.
   o **Service Type** – Choose **Voice Calls** here.
   o **Off-Peak Period** – Just leave this field is empty.
   o **Off-Peak Description** – A description of the off-peak period. Automatically provided by the off-peak period wizard, so you do not have to fill in this field.
   o **Destination Group Set** – Do not choose any value in the select menu.
   o **Free Seconds, Post Call Surcharge, Login Fee, Connect Fee, Formula** – Leave all these parameters empty.

- o **Short Description** – A short tariff description; enter "PortaUM routing" or something similar. This field is mandatory.
- o **Description** – An extended tariff description.
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 **Edit Rates** icon next to the name of a tariff. When you are in Tariff Management for a particular tariff, click on **Edit Rates** in the toolbar.
2. On the **Edit Rates** screen, click ➕ **Add.**
3. Fill in the required information:

- o **Destination** – The destination prefix for voicemail which you created earlier.
- o **Interval 1** – First billing unit in seconds.
- o **Interval N** – Next billing unit in seconds.
- o **Price 1** – Per minute price for the first interval. If you plan to provide free calls to voicemail, enter 0 here.
- o **Price N** – Per minute price for the next interval. If you plan to provide free calls to voicemail, enter 0 here.
- o **Off-peak Interval 1** – First billing unit in seconds for off-peak time.
- o **Off-peak Interval N** – Next billing unit in seconds for off-peak time.
- o **Off-peak Price 1** – Per minute price for first interval in off-peak time.
- o **Off-peak Price N** – Per minute price for next interval in off-peak time.

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

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

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

- o **Formula** – You can define a complex rating formula for calls to this destination. For UM calls you do not need complex routing, so just leave this empty.
- o 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. Click **Close** to return to the Tariff Management screen.
6. Repeat steps 1-5 to add rates for the UM prefix for all tariffs where you would like customers to have voicemail.

Rates for UM routing tariff

1. On the Tariff Management page you will see a list of the available tariffs. Click the **Edit Rates** icon next to the name of the UM routing tariff.

			Now ▾		Prefix	Group	Country				

Edit	Destination *	Country	Routing			Interval, sec.		Price, USD/min		⏱ Effective From					Delete
		Description	Route Category	Preference	Huntstop	First *	Next *	First *	Next *	YYYY-MM-DD HH24:MI:SS *					
💾	*98		Default ▾	5 ▾	☑	1	1	0	0	immediately		☐	☐	☐	❌

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).
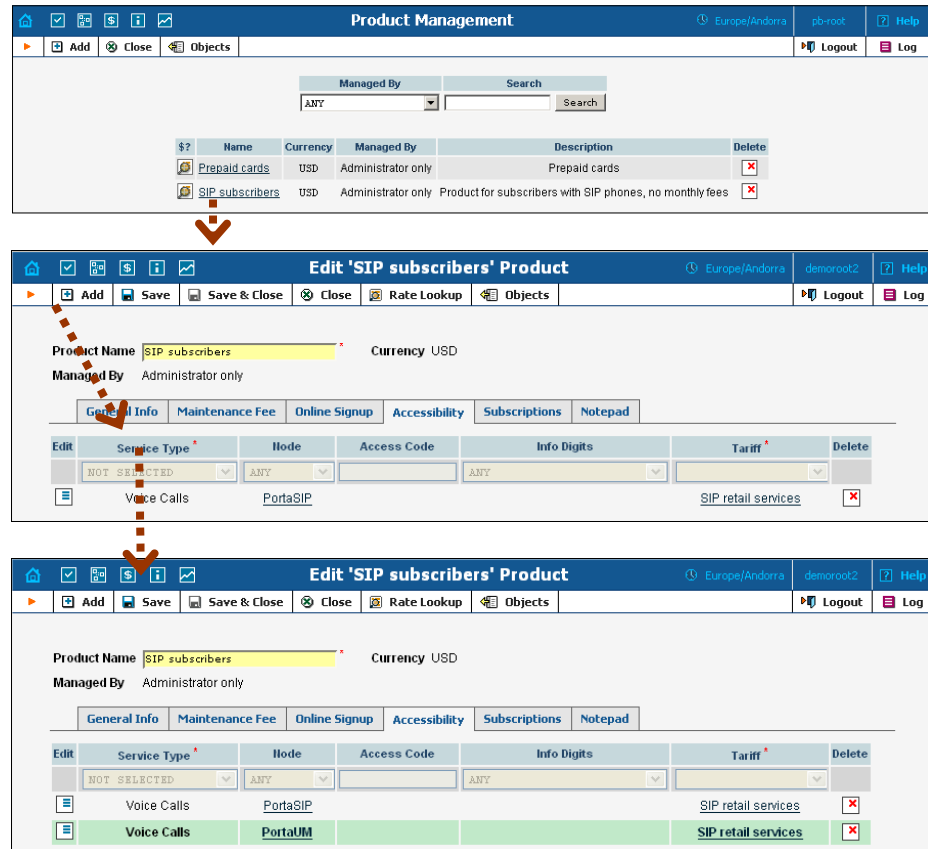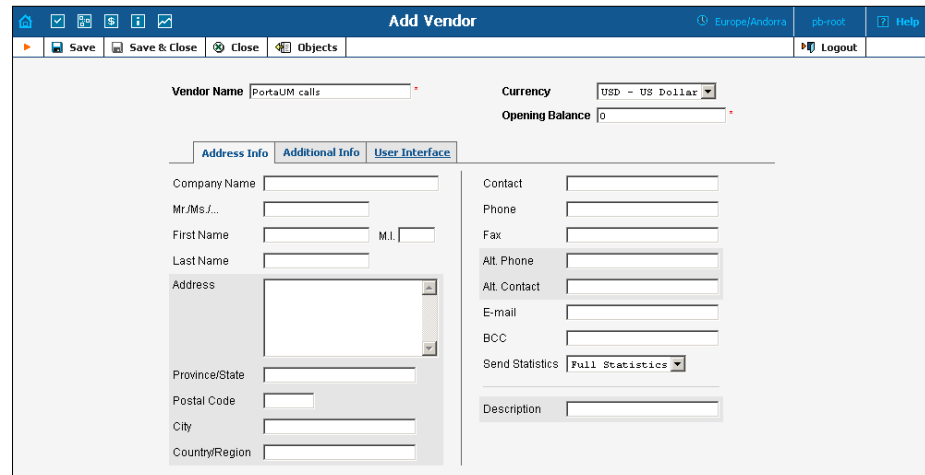
1. In the Management section of the Admin-Index page, choose **Products**.
2. On the Product Management page, click on the name of the corresponding product.
3. Click on the **Accessibility** tab to edit this product's accessibility.
4. After the **Accessibility** tab has been selected, click on the ⊞ **Add** icon.
5. Choose **Voice Calls** in the **Service Type** select menu.
6. In the accessibility entry window, select the node which represents your Porta UM 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 UM 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 Management section of the admin interface, choose **Vendors**.
2. On the Vendor Management page, choose ⊞ **Add**.

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):
    o **Vendor Name** – A short name for the vendor object which will be used on the web interface.
    o **Currency** – Enter the currency in which this vendor charges you.
    o **Opening balance** – Starting balance for the vendor; the default is zero.
    Additional info:
    o **Billing period** – Split period for vendor statistics.
4. Click ⊟ **Save&Close**.

Define Connections

Vendors are your termination partners or providers of 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 created in the previous step.
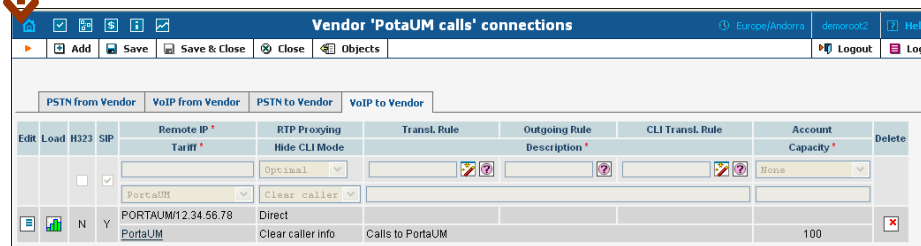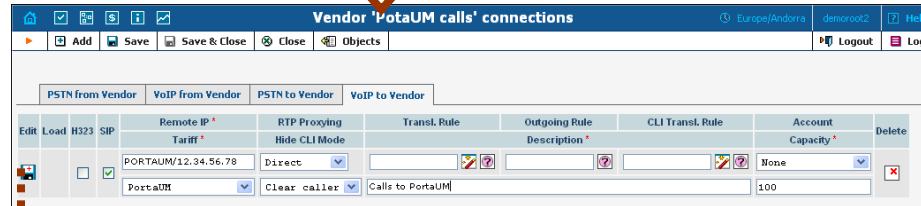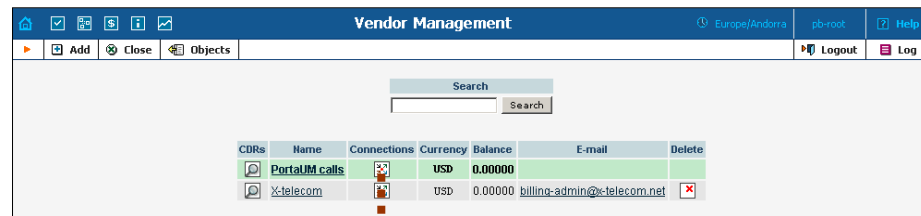
3. Choose the **VoIP to Vendor** connection type by clicking on the corresponding tab.
4. Press ⊞ **Add** to add a new connection.
5. Fill in the connection information. Type "PORTAUM/x.x.x.x" in the **Remote IP** field, where x.x.x.x is the address of the UM gateway (or the IP address of the PortaUM server itself, if you are using PortaUM's built-in media server). Choose the Porta UM 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**.
The format of the **Remote IP** field must be "PORTAUM/x.x.x.x" in order for UM to function properly; so if your PortaUM IP address is 193.28.87.37, you would fill in "PORTAUM/193.28.87.37".
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 already there, go to the **Customers** screen (the one containing a list of customers). It should look like the screenshot below.
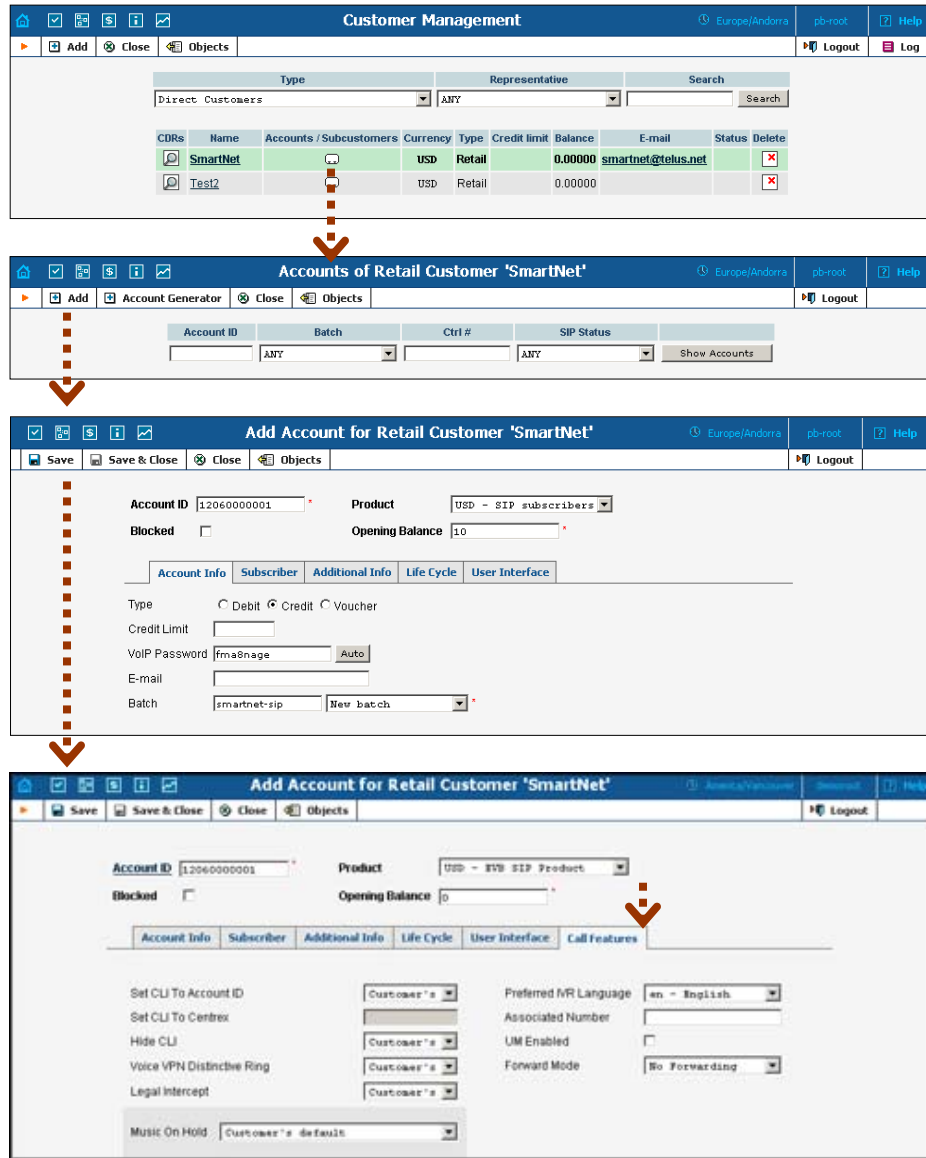
2. Next to the customer name, click on the ⊞ icon (the one in the **Accounts** column). This will take you to the account management for that customer.
3. Click on **Account New**.
4. Fill in the Add Account form:
   o **Account ID** – The SIP ID, i.e. the phone number, which will be used to log into the SIP server and receive incoming calls.
   o **Product** – Choose the product which you would like your account to have.
   o **Blocked** – You may create the account as blocked, but this is rarely done with SIP service accounts.
   o **Opening Balance** – The initial balance on the account.

Account Info tab

- o **Type** – Select credit for postpaid and debit for prepaid service.
- o **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).
- o **VoIP Password** – This password is used for SIP services as well. The Account ID and this password will be used to authenticate logins to the SIP server.
- o **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.
- o **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.

Call Features tab

- o **Preferred IVR Language** – This is a custom attribute which is transferred to the IVR, such as a debit card application. Since this attribute is not used by PortaUM, you may leave the default value (English).
- o **UM Enabled** – Check if this account has Unified Messaging (e.g. voicemail) services enabled.

Life Cycle tab

- o **Activation Date** – Account activation date.
- o **Expiration Date** – Account expiration date.
- o **Lifetime** – 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.

User Interface tab

- o **Login** – Account login to the web self-care pages. This can be the same as the Account ID.
- o **Password** – The password for the web self-care pages.
- o **Time Zone** – When an account owner (prepaid card user) accesses the web self-care pages to see a list of his calls, the time can be shown in the time zone most appropriate for him.
5. Click ▣ **Save&Close**. You will see a confirmation screen saying that a new account has been created.

Enabling 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). It should look like the screenshot below.
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.
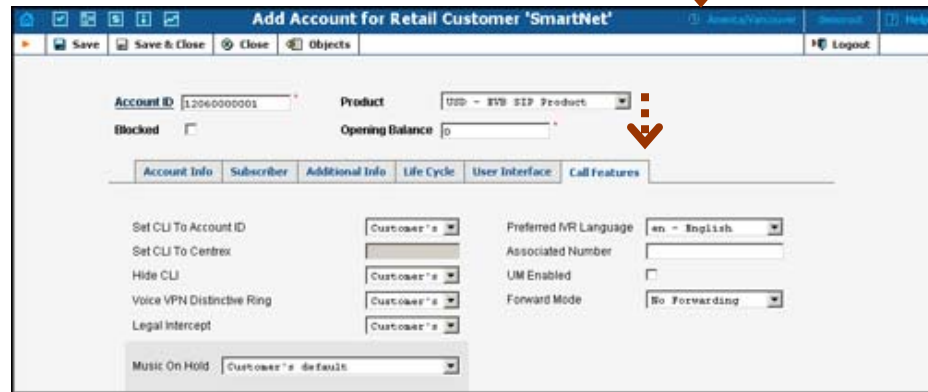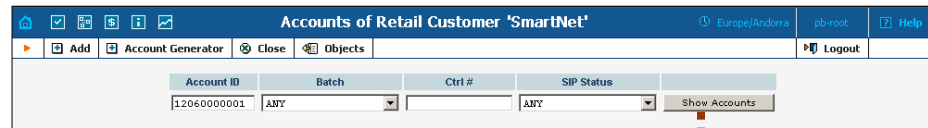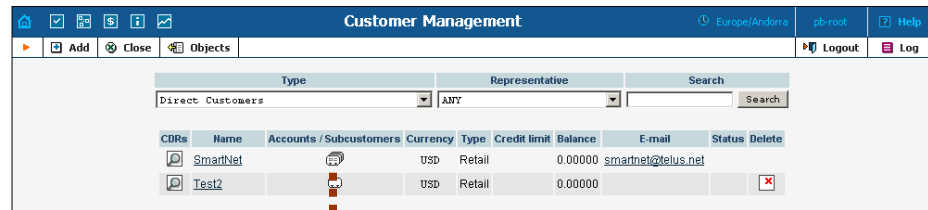
4. Click on the **Call Features** tab.
5. Click on the **UM Enabled** checkbox to activate UM services for this account.
6. Click ▣ **Save&Close.**

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. Enable UM for the PSTN access number account. As with any PSTN-to-SIP service, the account will have an account number identical to your PSTN access number or the DID forwarded by your vendor to PortaSIP. You must enable UM for this account to associate it with the internal "DNIS_mailbox" on the PortaUM you have already configured.
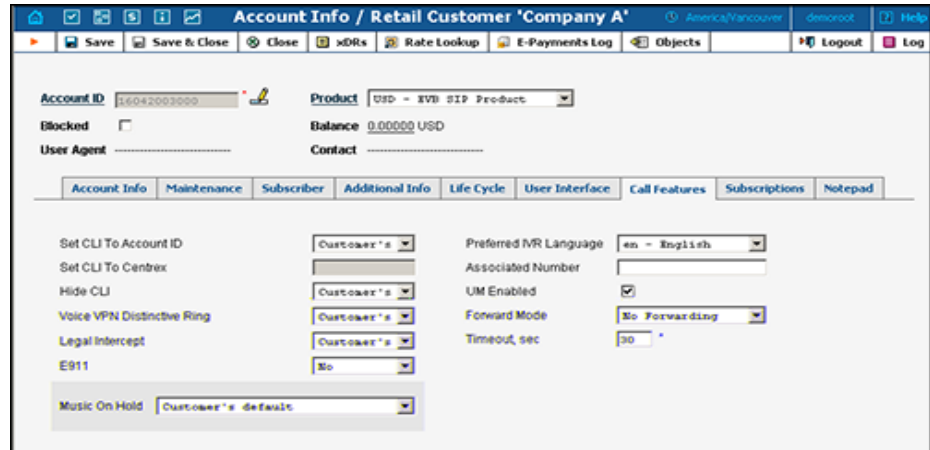
3. Configure forwarding to "Follow-me", set to "Always". This is necessary for internal routing to PortaUM.

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/solutions/portaum/

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.

IVR Path Diagram

The PortaUM mailbox can be accessed by phone by dialing *98.
Below is an IVR path diagram to help users in navigating and configuring the system.

3. Administration

PortaUM configuration

PortaUM provides a unified configuration tool. Even if a system consists of several components, using different technologies and configuration methods, you need only edit one simple configuration file. This master configuration file is then used by PortaOne configuration scripts to manage and provision other modules, e.g. IMAP, MWI, IVR applications, and so on.

porta-um.conf

This is the only file you need to edit in order to modify PortaUM parameters. Every row starting with a # is considered to be a comment; other lines will contain VAR=VALUE pairs, separated by an equal sign (=). This file is created automatically during installation. Thus, assuming you provided the correct parameters during installation, you will not have to change anything. However, you might want to examine certain defaults and change them to values more appropriate to your business.

Note: As always, if you are not certain what the various parameters mean and what they do, please contact our support team first for clarification, and let them assist you in making the appropriate changes.

Variable	Description
Global	
DefaultPassword	The default voicemail password for new accounts. Initially 777, but can later be changed by customers on their self-care interface.

DefaultOffse

	`um/SquirrelMail/brandpane.html`).
JavaPolicyInstallerURL	`http://www.portaone.com/resources/PortaUM_Client_1.0.exe`

RechargeVoucher	
MaxVoucherLen	Maximum length of voucher number; default: 11.
MinVoucherLen	Minimum length of voucher number; default: 11.

DebitCard	
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', then the Debit Card application will exit immediately if the account has insufficient funds. Default: false.
Ext\<N\>	Description of extensions used for the prepaid card application. These can contain several lines in the following format: `Ext<N>=<i_env>:<extension>:<ANI translation rule>` Where: N - arbitrary unique number i_env - ID of the environment extension - extension for the prepaid card application ANI translation rule – a regular expression (used only if AniAuth=yes) Example: `Ext1=1:1234567891:s/^8.*(.)$/888888888$1/;`

Voicemail		
MinServers	200	Daemon options for Cisco-less setup
MaxServers	300	
MinSpareServers	20	
MaxSpareServers	30	
MinDuration	Minimum length of voice message, in seconds. If the voice message is shorter or equal to the MinDuration value, the message will be canceled. The default is 3 seconds. A	

	value equal to '0' will disable this check.
InitialPause	Initial delay in seconds before first playout; default: no pause.
SilenceDetection	Enable silence detection during voicemail recording; default: on.
SilenceSeconds	How many seconds of silence will result in canceling of voicemail recording; default: 2.
FaxDetection	Enable fax detection during voicemail recording; default: on.
MailTemplate	Mail message template. Specifies the file that will be used as the body template for voicemail and fax messages. Messages will have no body if MailTemplate is not set. Default: no message body. Example: `MailTemplate=/home/porta-um/etc/mail-template.txt`
CleanupPeriodINBOX	Cleanup of user's Inbox folder. All messages older than the selected value (in days) will be moved to the Trash folder. A zero value disables cleanup. Default: 30.
CleanupPeriodTrash	Cleanup of user's Trash folder. All messages older than the selected value (in days) will be permanently deleted. A zero value disables cleanup. Default: 30.
DefaultPromptTimeout	Default timeout for all voice prompts, in milliseconds; default: 9000.
RecordTimeout	Default timeout for recording voicemails and greetings, in milliseconds; default: 200000.
EnvMap	

Env\<N\>	Parameter names (the part to the left of the equal sign) must contain sequential numbers, e.g. Env1, followed by Env2, Env3, and so on. These numbers are not the same as i_env in PortaBilling, which is specified inside the parameter's value.
	Parameter values are colon-separated strings in the following format:
	`Env<N>=<i_env>:<mail_domain>:`
	`<SIP_server_IP>:<DNIS_MailBOX>:`
	`<DNIS_MailBoxWithPass>:`
	`<DNIS_PlayBalance>:`
	`<DNIS_RechargeVoucher>`
	Example:
	`Env1=3:um.yourcompany.com:porta-sip:*98:1234567890:*11:*12`
	`Env2=4:um.othercompany.com:porta-sip2:*98:18001234567:*11:*12`