

Porta  **UM**
unified messaging



Administrator Guide

Maintenance Release 15

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**PortaUM Administrator Guide
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Table of Contents

Preface 3

Hardware and Software Requirements 4

Installation 5

What's New in Maintenance Release 15? 5

1. System Concepts 6

 PortaUM's Role in Your VoIP Network 7

 PortaUM Components 8

 Call Process 9

 Supported Services 11

2. Setting Up and Using UM Services 13

 Setting Up UM Services 14

 Setting Up and Using Prepaid IVR on PortaUM 27

 Auto Attendant 32

Preface

This document provides a general overview of PortaUM (Unified Messaging System), a platform for the delivery of enhanced business and residential communications services. PortaUM handles voice, fax, and regular email messages as objects in a single mailbox, which users 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 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



Caution indicates that the described action might result in program malfunction or data loss.

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



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



Tips provide information that might help you solve a problem.

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](#).

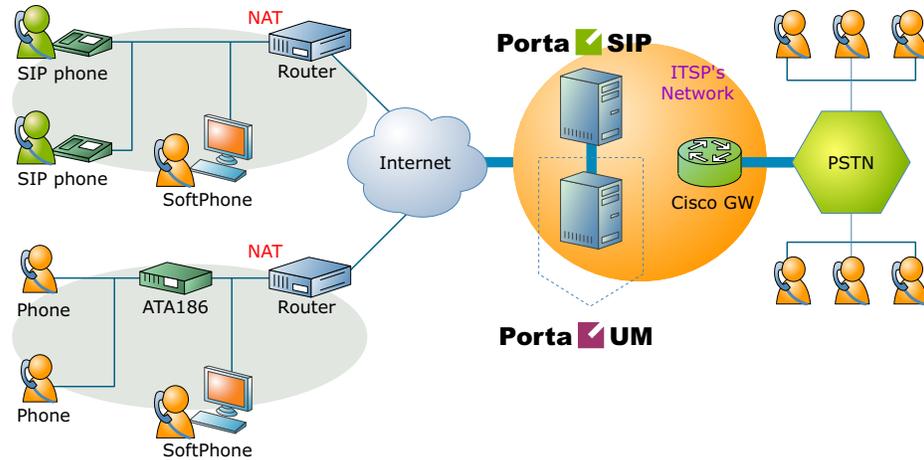
What's New in Maintenance Release 15?

This release includes several new features and improvements:

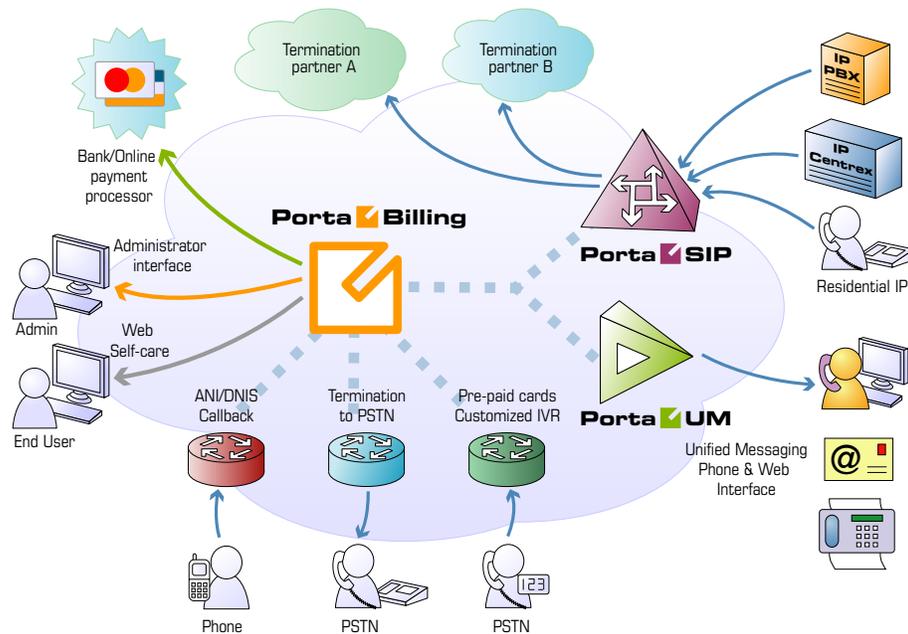
- Prepaid card IVR on PortaUM without using a Cisco gateway

1. System Concepts

PortaUM's Role in Your VoIP Network

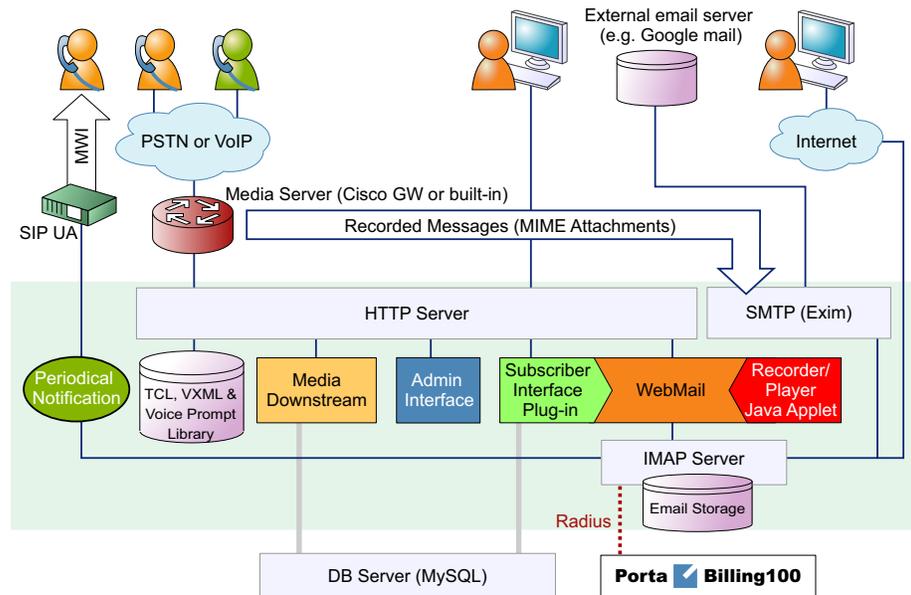


PortaUM (Unified Messaging system) is one of the key components of PortaSwitch, a software-based communication service and subscriber management platform that directly addresses the needs of modern communication network providers seeking new revenue streams from tools that unify voice, data, and fax traffic within a single network, and which enable diversification of their current offer of services.



PortaUM handles voice, fax, and regular email messages as objects in a single mailbox, which users can access via a web interface or by telephone.

PortaUM Components

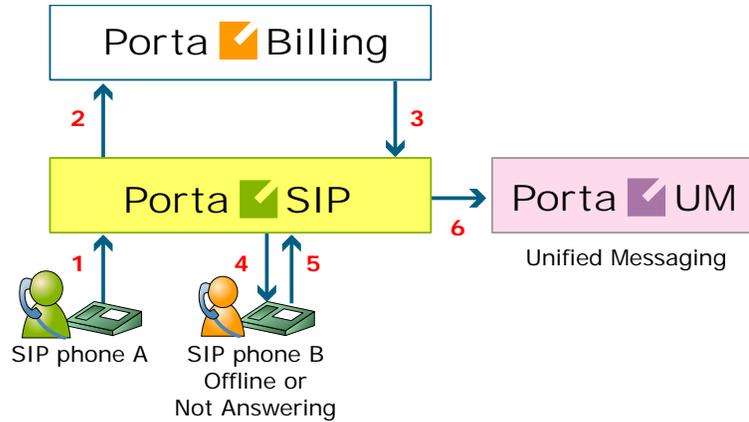


PortaUM consists of the following main components

- 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.

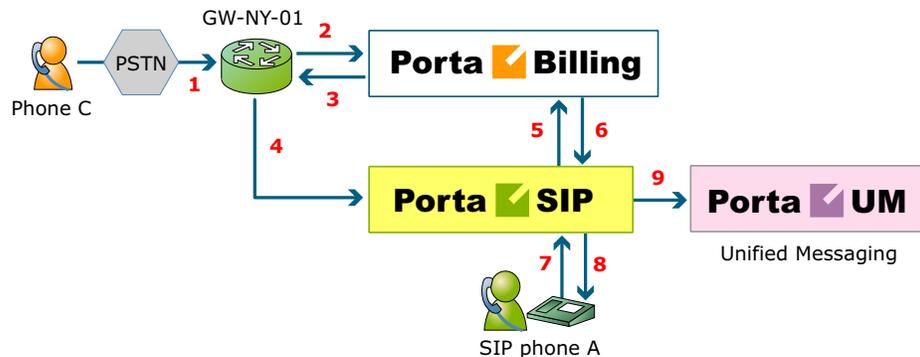
Call Process

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).

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

User of PortaUM can listen to the current state of his account by using PortaUM IVR menu. There is also IVR menu that enables vouchers to be recharged to add funds to user's account.

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.

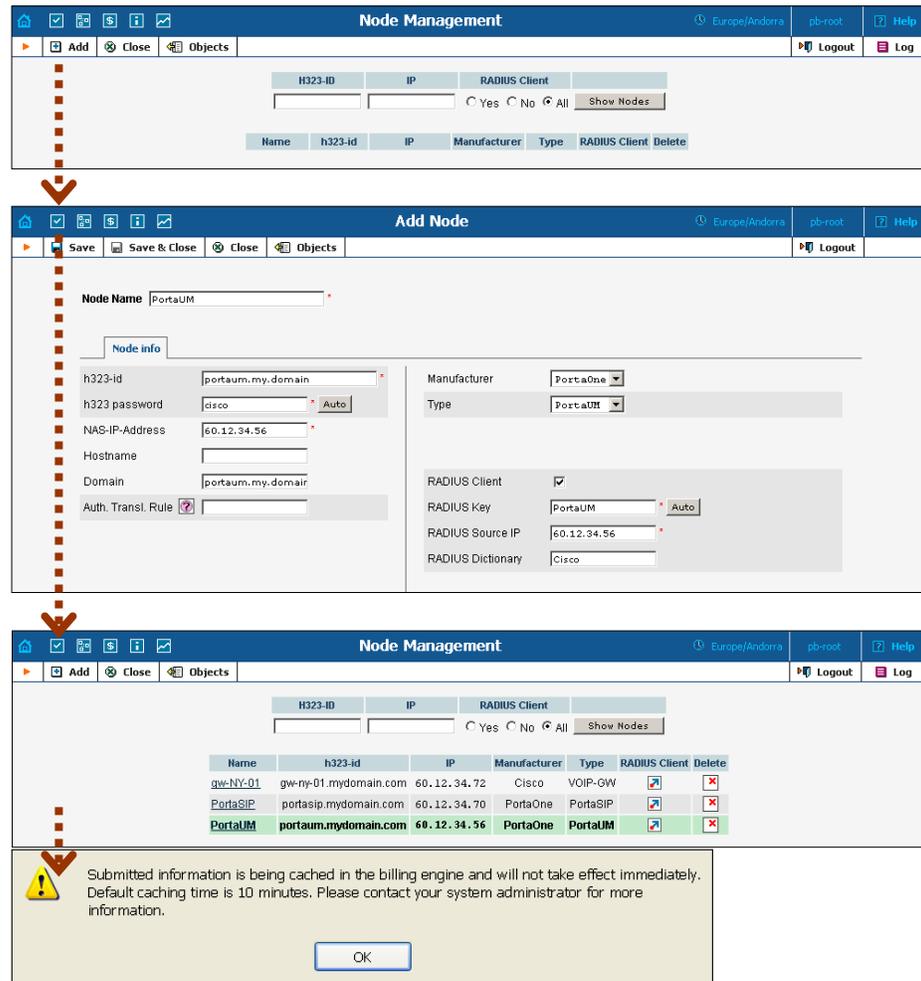


3. Enter your on-net UM number under **Prefix**, and **N/A** for the country and country subdivision. Put a comment in the **Description** column that clearly identifies this as a special prefix assigned to UM.
4. Click  **Save**.

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.



The screenshots illustrate the following steps:

1. Accessing the **Node Management** section and clicking the **Add** icon.
2. Filling out the **Add Node** form with the following details:
 - Node Name: PortaUM
 - h323-id: portaum.my.domain
 - h323 password: cisco
 - NAS-IP-Address: 60.12.34.56
 - Manufacturer: PortaOne
 - Type: PortaUM
 - RADIUS Client:
 - RADIUS Key: PortaUM
 - RADIUS Source IP: 60.12.34.56
 - RADIUS Dictionary: Cisco
3. Viewing the updated **Node Management** table, which now includes the new node:

Name	h323-id	IP	Manufacturer	Type	RADIUS Client	Delete
gw-NY-01	gw-ny-01.mydomain.com	60.12.34.72	Cisco	VOIP-GW	<input type="checkbox"/>	<input type="checkbox"/>
PortaSIP	portasip.mydomain.com	60.12.34.70	PortaOne	PortaSIP	<input type="checkbox"/>	<input type="checkbox"/>
PortaUM	portaum.mydomain.com	60.12.34.56	PortaOne	PortaUM	<input checked="" type="checkbox"/>	<input type="checkbox"/>

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)

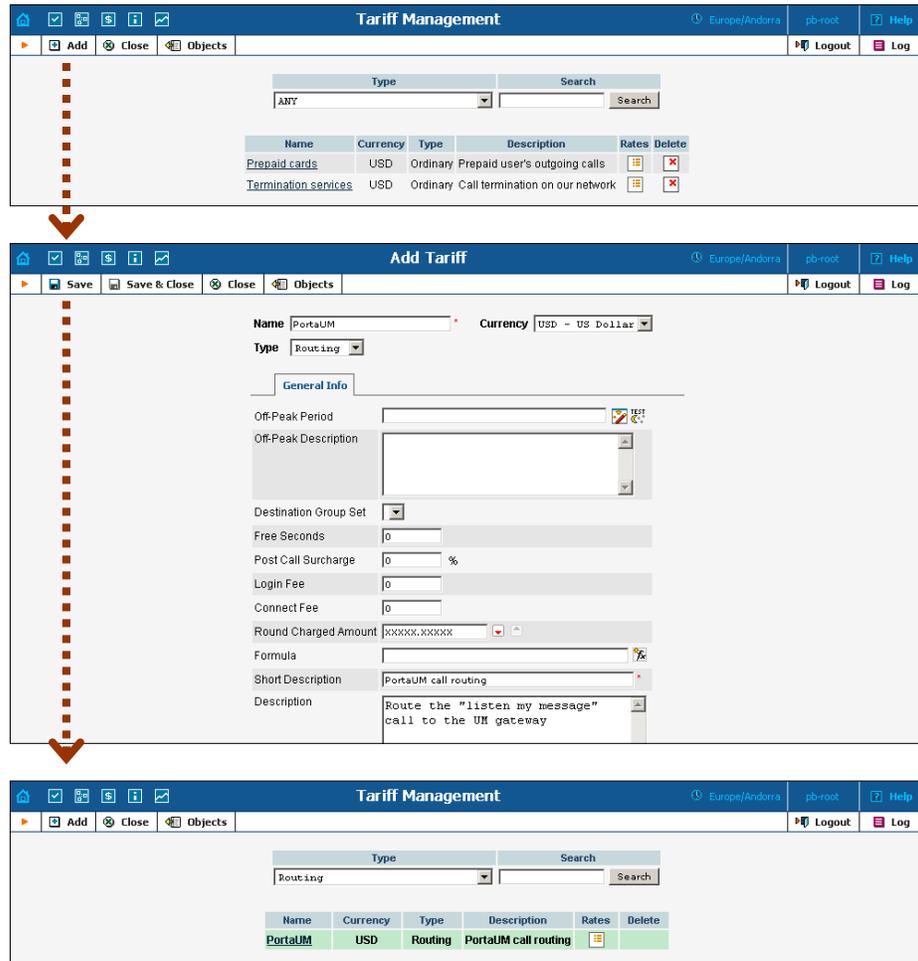


- **H323 Password** – If you plan to send calls from the UM gateway to your Cisco gateways where the default Cisco remote IP authentication script will be used, enter `cisco` here.
 - **NAS-IP-Address** – IP address of the UM gateway or PortaUM server.
 - **Auth. Translation rule** – You can just leave this empty.
 - **Manufacturer** – Select **PortaOne**.
 - **Type** – Node type; select **PortaUM**.
 - **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.
 - **Radius Client** – Check this, since both PortaUM and the UM gateway will need to communicate with the billing.
 - **Radius Key** – Enter the radius shared secret here; must be the same **key** as you entered during PortaUM installation.
 - **Radius Source IP** – See the *Node ID, NAS IP address, and Radius source IP* section in the [PortaBilling Administrator Guide](#) for more information. Unless your PortaUM server uses multiple network interfaces, the value here should be the same as the NAS-IP-Address.
4. Click  **Save&Close**.

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

Create a Tariff

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



The first screenshot shows the 'Tariff Management' page with a table of existing tariffs:

Name	Currency	Type	Description	Rates	Delete
Prepaid cards	USD	Ordinary	Prepaid user's outgoing calls		
Termination services	USD	Ordinary	Call termination on our network		

The second screenshot shows the 'Add Tariff' form with the following fields filled:

- Name: PortaUM
- Currency: USD - US Dollar
- Type: Routing
- Off-Peak Period: (empty)
- Off-Peak Description: (empty)
- Destination Group Set: (empty)
- Free Seconds: 0
- Post Call Surcharge: 0 %
- Login Fee: 0
- Connect Fee: 0
- Round Charged Amount: 0
- Formula: (empty)
- Short Description: PortaUM call routing
- Description: Route the "listen my message" call to the UM gateway

The third screenshot shows the 'Tariff Management' page with the new tariff added to the table:

Name	Currency	Type	Description	Rates	Delete
PortaUM	USD	Routing	PortaUM call routing		

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 **Type** – Since you need to create a tariff for routing calls to the UM gateway, make sure you choose **Routing** here.
 - o **Currency** – Since this is a “fake” tariff, just choose the same currency as your **Base Currency**.
 - o **Off-Peak Period** – Just leave this field is empty.

- **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.
 - **Destination Group Set** – Do not choose any value in the select menu.
 - **Free Seconds, Post Call Surcharge, Login Fee, Connect Fee, Formula** – Leave all these parameters empty.
 - **Short Description** – A short tariff description; enter “PortaUM routing” or something similar. This field is mandatory.
 - **Description** – An extended tariff description.
4. Click  **Save&Close**.

Enter Rates

Modifying tariffs for SIP users

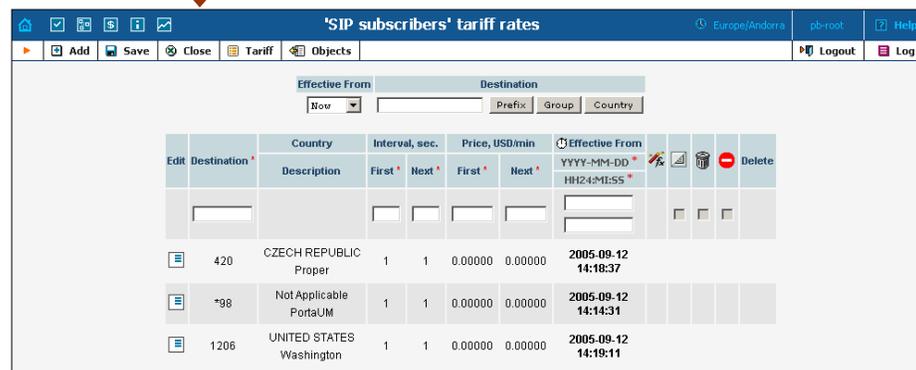
First of all, you should make sure that your customers are allowed to call voicemail. In order to do so, you should add a rate for calling the special voicemail number into the tariffs used for your SIP subscribers.



Name	Currency	Type	Description	Rates	Delete
PortaUM	USD	Routing	PortaUM call routing		
Prepaid cards	USD	Ordinary	Prepaid user's outgoing calls		
SIP_subscribers	USD	Ordinary	Outgoing SIP calls		
Termination services	USD	Ordinary	Call termination on our network		



Country	Interval, sec.	Price, USD/min	Effective From
<input type="text" value="*98"/>	1 1	0 0	Immediately



Country	Interval, sec.	Price, USD/min	Effective From
420 CZECH REPUBLIC Proper	1 1	0.00000 0.00000	2005-09-12 14:18:37
*98 Not Applicable PortaUM	1 1	0.00000 0.00000	2005-09-12 14:14:31
1206 UNITED STATES Washington	1 1	0.00000 0.00000	2005-09-12 14:19:11

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

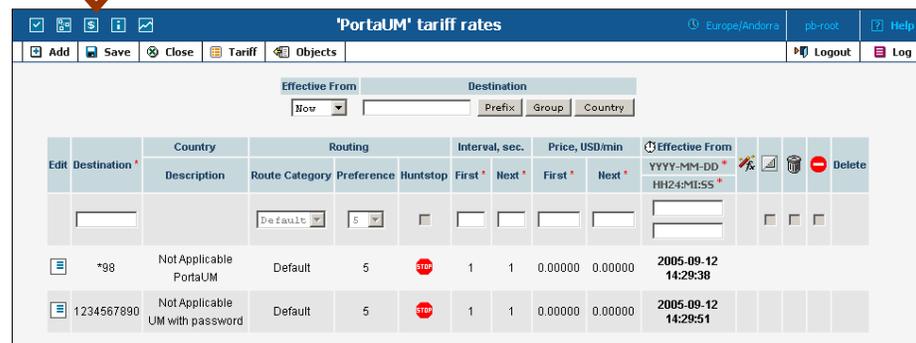
- **Effective from** – If you want the rate to take effect sometime in the future, you can either type in a date manually, or use the calendar (click on the DD-MM-YYYY link).

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.

- **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.
 - 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 the 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.

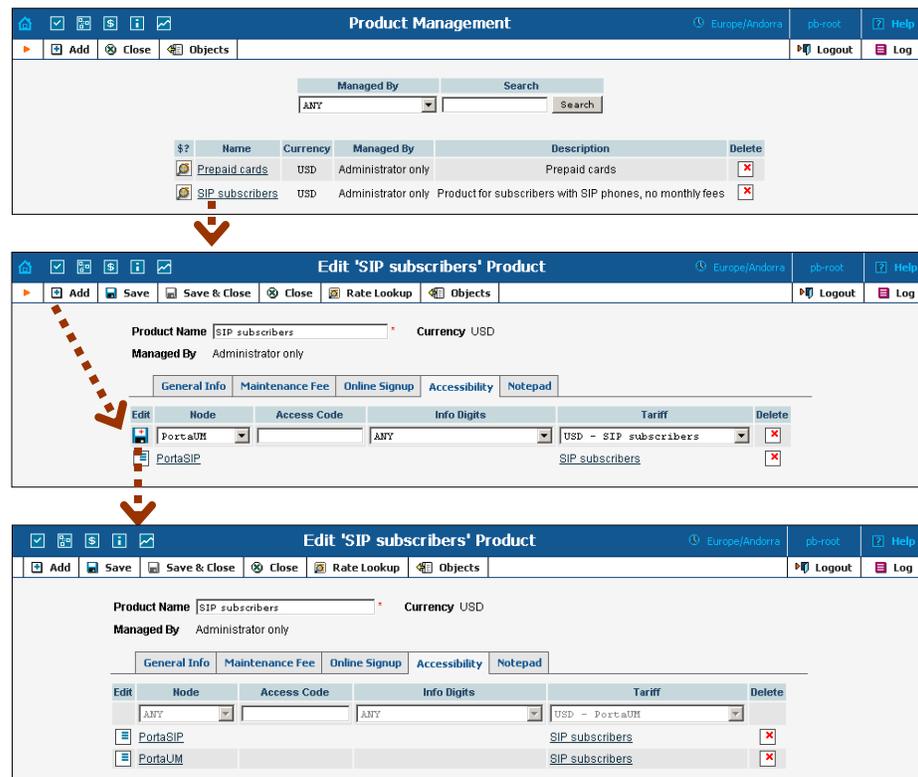


2. On the **Edit Rates** screen, click  **Add**.
3. Fill in the required information:
 - o **Destination** – The destination prefix for voicemail which you created earlier.
 - o **Route Category** – Leave “Default” here.
 - o **Preference** – Enter 10 in this field. This is the highest routing priority, since calls to voicemail should always be routed to the UM gateway.
 - o **Huntstop** – Check this box, since no other route for UM is used except a direct call to the UM gateway.
 - o **Interval 1, Interval N** – Enter 1 in both fields.
 - o **Price 1, Price N** – Enter 0 in both fields (UM is our own gateway, so termination costs are zero).
 - o **Effective from** – Leave **immediately** in this field.
 - 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.

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).



The screenshots illustrate the process of modifying product accessibility:

1. **Product Management**: A table listing products. The 'SIP subscribers' product is selected.
2. **Edit 'SIP subscribers' Product**: The 'Accessibility' tab is selected. A table shows existing accessibility entries for 'PortaSIP' and 'SIP subscribers'. A new entry for 'PortaUM' is added.
3. **Edit 'SIP subscribers' Product**: The final state after saving, showing 'PortaUM' added to the accessibility list.

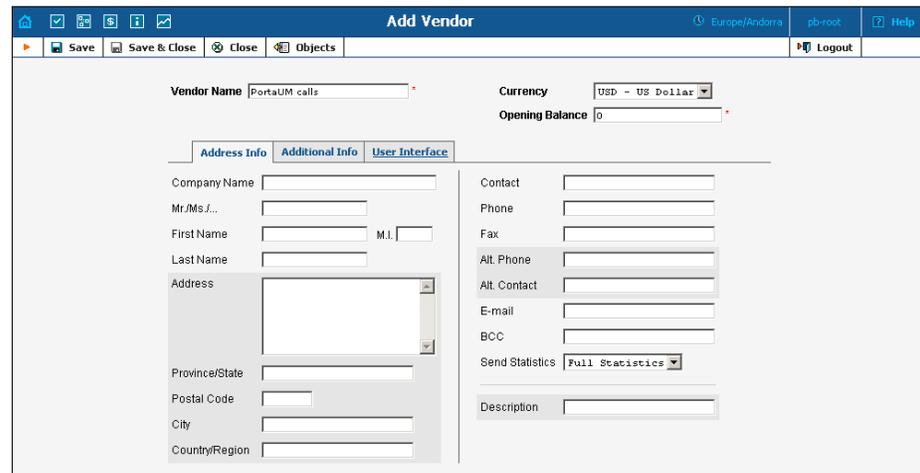
1. In the Management section of the Admin-Index page, choose **Products**.
2. On the Product Management page, click on the name of the corresponding product.
3. Click on the **Accessibility** tab to edit this product's accessibility.
4. After the **Accessibility** tab has been selected, click on the  **Add** icon.
5. 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.)
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 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**.

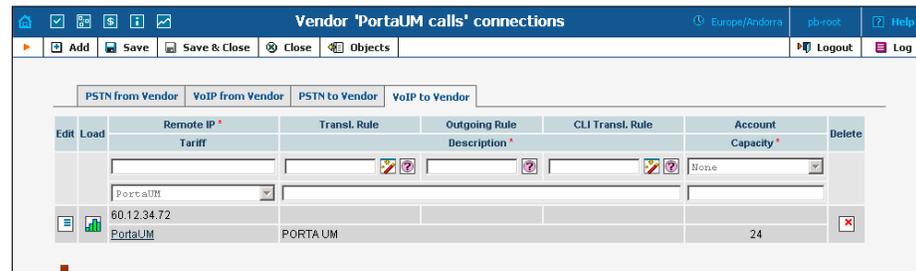
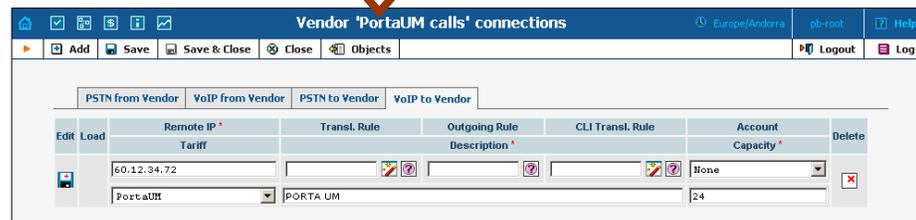


3. Fill in the New Vendor form. Please note that there are two tabs available on the screen. The most important fields are:
 - Main form (top):
 - **Vendor Name** – A short name for the vendor object which will be used on the web interface.
 - **Currency** – Enter the currency in which this vendor charges you.
 - **Opening balance** – Starting balance for the vendor; the default is zero.
 - Additional info:
 - **Billing period** – Split period for vendor statistics.
4. Click  **Save & Close**.

Define Connections

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.



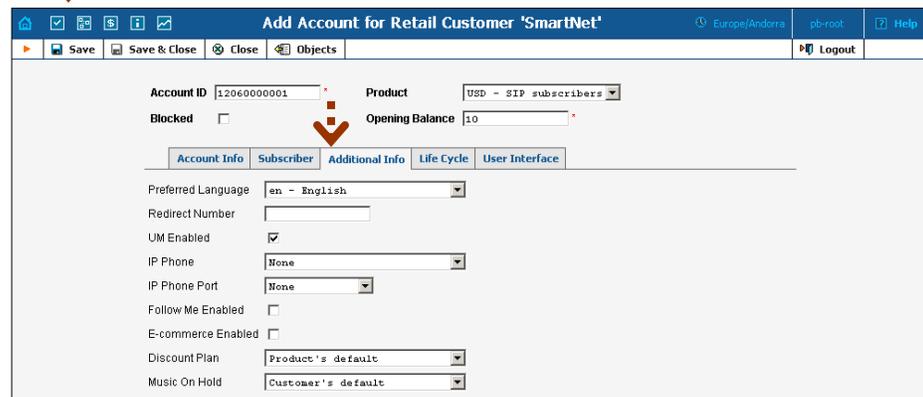
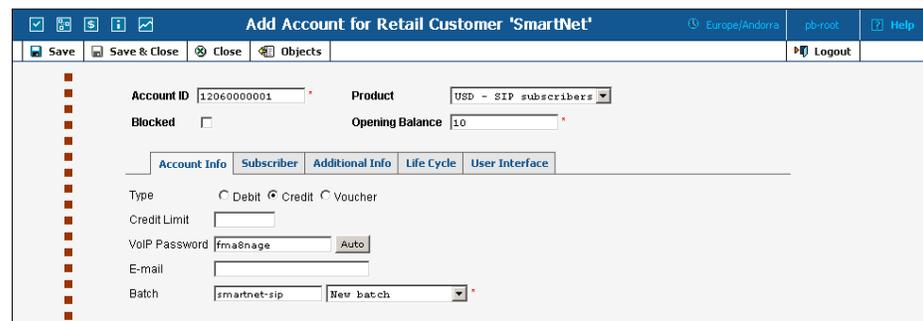
3. Choose the **VoIP to Vendor** connection type by clicking on the corresponding tab.
4. Press  **Add** to add a new connection.
5. Fill in the connection information. Type the address of the UM gateway in the **Remote IP** field. Choose the Porta UM tariff you created earlier from the select menu in the **Tariff** column. Enter a comment in the **Description**, and in **Capacity** give the number of simultaneous calls your UM gateway can handle.
6. Fill in the **Description** field with "PORTA UM". This description is used by PortaBilling to determine routing to PortaUM.
7. Click  **Save**.
8. 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:

- **Account ID** – The 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 tab

- **Account 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).
- **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.
- **Email** – Enter the account owner’s email address here. If he ever forgets his password for the web self-care, he will be able to reset it, and a new password will be sent to this email address. You can also leave this field empty.
- **Batch** – A management unit for accounts. The batch name is alphanumeric. You can type a new name here, or use an existing name in order to generate more accounts for the same batch.

Additional Info tab

- **Preferred 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).
- **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: the account will expire on “first usage date” + “life time” days. If you do not want to use this feature, leave the field blank.

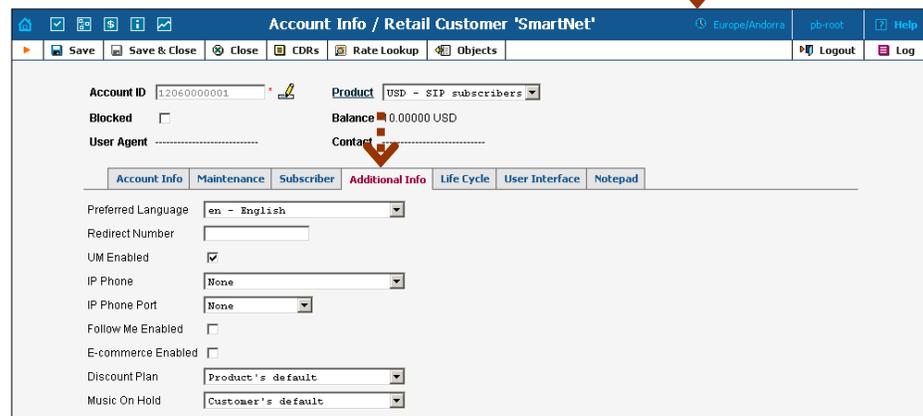
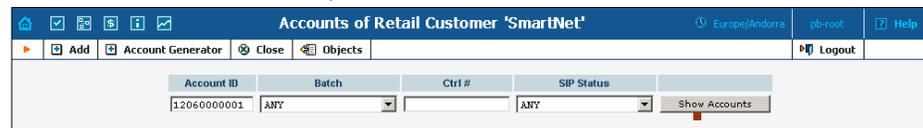
User Interface tab

- **Login** – Account login to the web self-care pages. This can be the same as the Account ID.
- **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 **Additional Info** 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 a 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'.

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

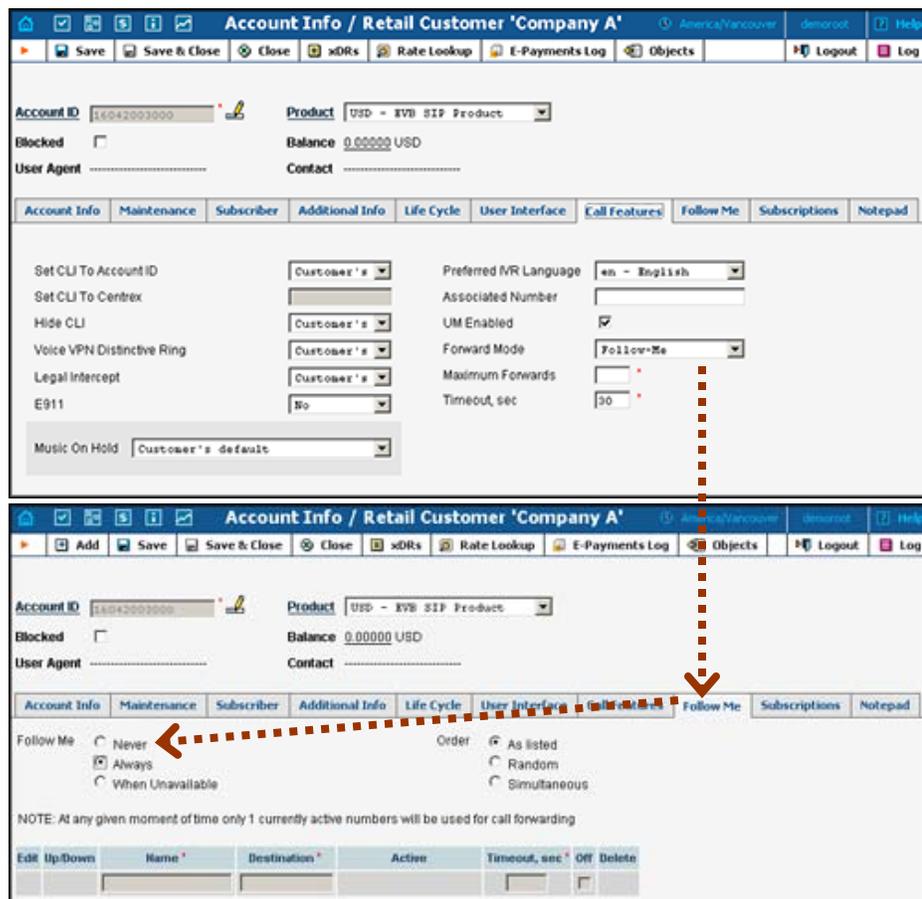
Optionally, you can configure any of the following additional parameters.

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

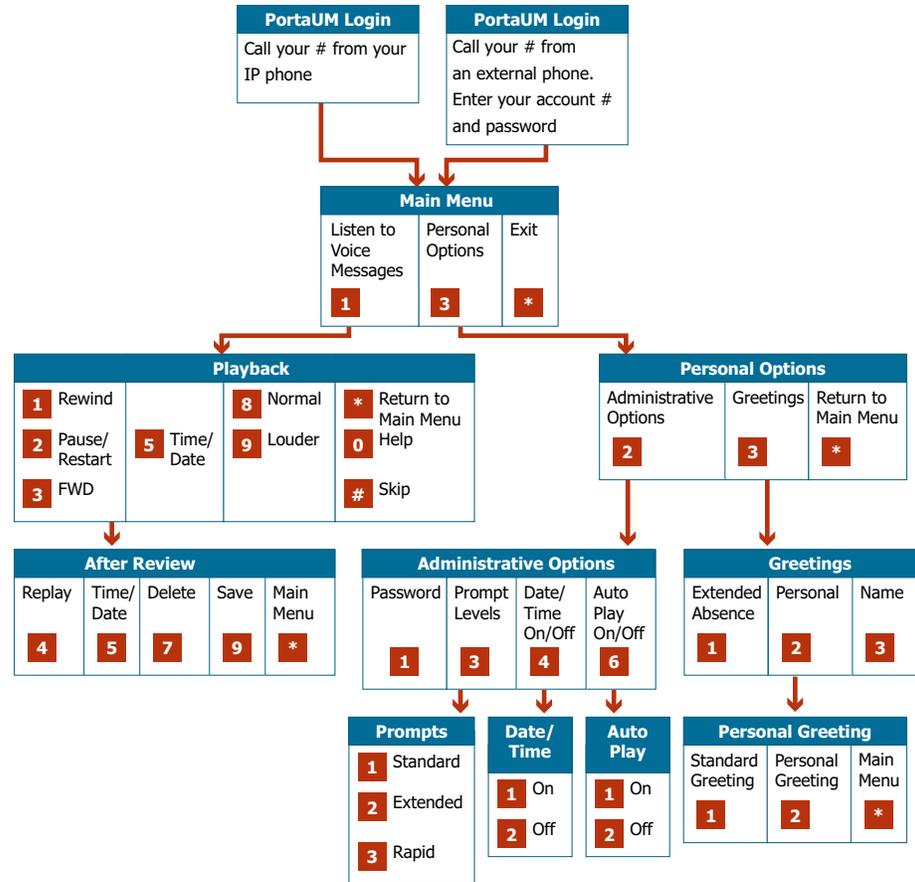
PortaUM allows Internet Telephony Service Providers (ITSPs) to offer their subscribers the ability to process emails and manage voicemail from within a web browser or using a favorite email client.

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.



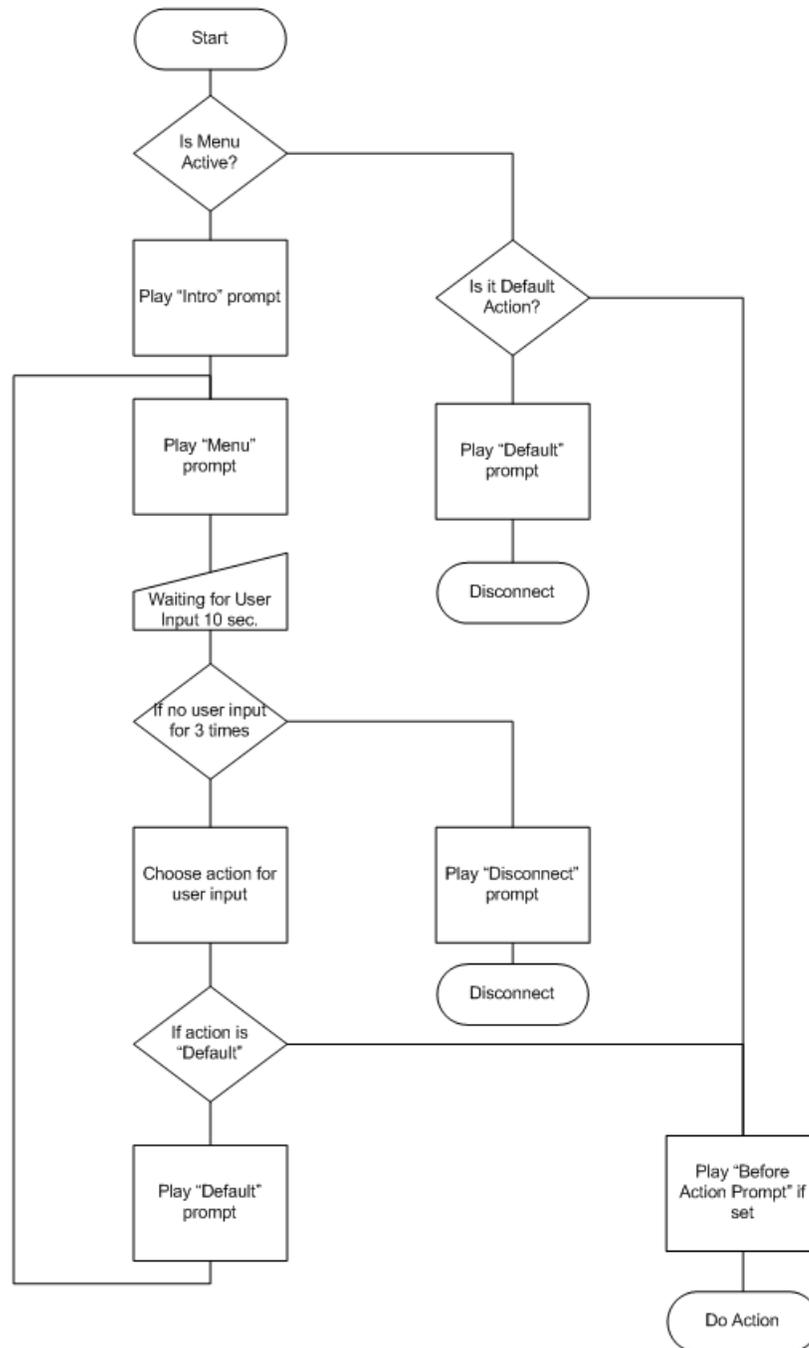
Auto Attendant

PortaUM 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 Auto Attendant (AA) 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, 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's input.
- The user's input will trigger execution of the following available actions:
 - o **Default** – Plays the default prompt from the current menu.

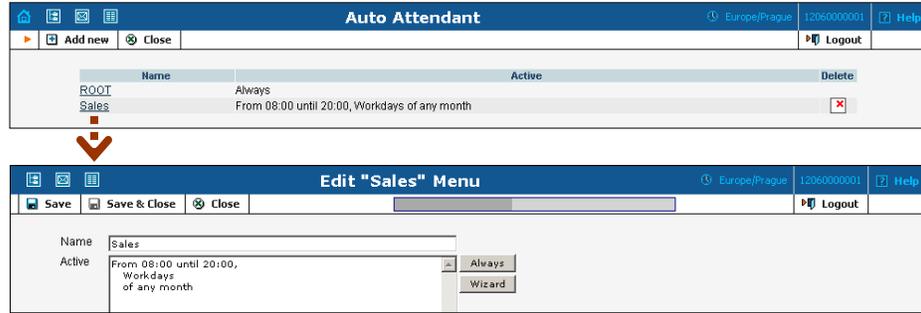
- **Transfer** – Transfers the call to a given telephone number or extension.
 - **Transfer to E164 number** – Transfers the call to a given number applying no translation rules.
 - **Transfer to extension** – Transfers the call to a number entered by the caller from his phone.
 - **Transfer to voicemail** – Switches to voicemail mode (this is the only way to receive fax in AutoAttendant mode).
 - **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 call menu flow chart is displayed in the diagram below:



Menu List Mode

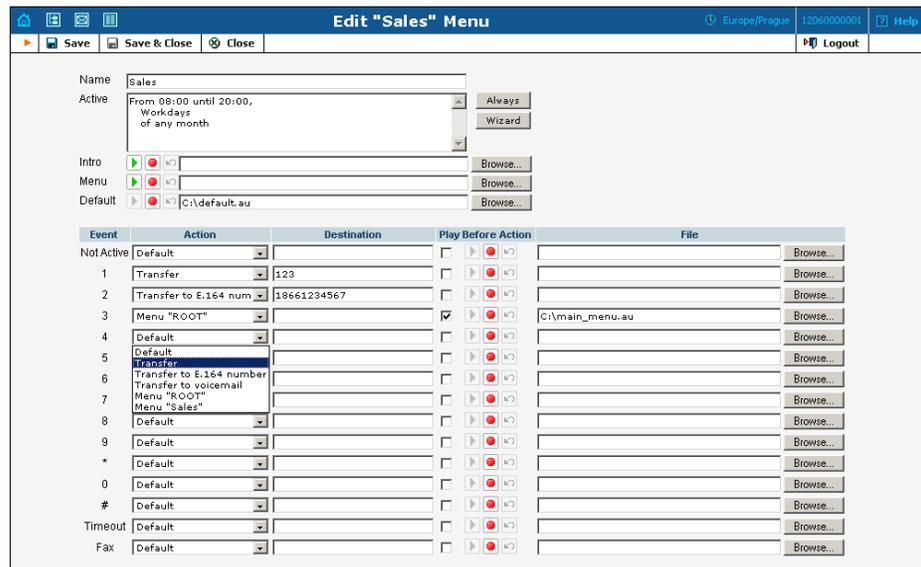
Auto Attendant can be selected from the Options menu. The main screen lists all the menus available in the system.

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 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

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’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

	<p>the active period definition above).</p> <p>0-9, #, * – User selection on telephone keypad.</p> <p>Timeout – No selection received from user.</p> <p>Fax – Fax CNG tone detected (works in Cisco-less PortaUM setup only).</p>
Action	<ul style="list-style-type: none"> ○ Default – Play the default prompt from the current menu. ○ Transfer – Transfer the call to a given telephone number or extension. ○ Transfer to E.164 number – Transfer the call to a given telephone number exactly (no translation rules will be applied). ○ Transfer to extension – Transfer the call to a number entered by the caller from his phone. ○ Transfer to voicemail – Switch to voicemail mode. ○ 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 an action is performed.
File	File name and path.