# Porta UM®



# Administrator Guide
Maintenance Release 24

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**Maintenance Release 24**
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# Preface

This document provides a general overview of PortaUM (Unified Messaging System), a media server platform for the delivery of enhanced business and residential communications services. PortaUM provides various IVR applications and 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`

The **exclamation mark** draws your attention to important information or actions.

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

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

**Tips** provide information that might help you solve a problem.

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# Hardware and Software Requirements

## Server System Recommendations

- An i386 processor (Xeon, Opteron) with 64bit support. Additional processor speed is needed for networks with a high call volume.
- A minimum of 500 GB of available disk space. On the average, 5,000 voice mailboxes take up about 1 GB of disk space, plus you need to reserve an amount of free space roughly equal to the projected database size for performing operations such as backup. RAID is recommended in order to improve performance and reliability.
- At least 8 GB of RAM, 12 GB recommended.

For additional details and configuration advice, see the *Hardware Recommendations* topic on our website:
**http://www.portaone.com/support/faq/hardware-requirements/hardware-requirements/**

For information about whether particular hardware is supported by Oracle Enterprise Linux from the JumpStart Installation CDs, consult the related document on the Oracle or RedHat website:
**https://hardware.redhat.com/**

## Client System Recommendations

- OS: Windows XP, Vista or 7, UNIX or Mac OS X
- Web browser: Internet Explorer 7.0 (or higher), Mozilla Firefox 3.6 (or higher)
- JavaScript and cookies enabled in web browser
- Spreadsheet processor (MS Excel or OpenOffice Calc)
- Display settings:
  - Minimum screen resolution: 1024 x 768

# Installation

Jumpstart installation CDs contain installation media for Oracle Enterprise Linux (64-bit version), supplementary packages necessary for convenient system administration and maintenance, and all required software packages. After the installation is complete you will assign roles (e.g. RADIUS, web interface, PortaUM, etc.) to individual servers using the configuration server tool – this will automatically enable the required components of PortaUM® software on each server.

For detailed installation instructions, please refer to the **PortaSwitch Installation Guide.**

# What's New in Maintenance Release 24?

This release includes the following new features and improvements:

- **Premium numbers IVR application** – Providing prepaid calling card service implicates the management, printing and distribution of cards or top-up vouchers. The Premium Numbers IVR application enables ITSP to offer cheap international calls to local telecom customers without administrative overhead. It mainly provides a special access number that any customer can reach from PSTN (via landline or mobile phone). Local telecom then delivers calls made to this access number to ITSP's network and delivers the payment (i.e. the telecom collects payments from customers and delivers payment to ITSP).

  Incoming calls are forwarded to PortaUM where the Premium Numbers IVR application is launched. IVR prompts for the destination and if allowed by the configuration, PortaBilling forwards the call to CLD according to the routing plan.

  The Premium Numbers IVR should be configured to allow calls to profitable destinations only (where the price that the telecom pays to ITSP is higher than call termination costs). Several Premium Numbers applications can be configured and each will have its own access number with a different rate and allowable destinations, respectively.

  If a telecom customer wants to call destinations that are not allowed by any Premium Numbers application he still can do this by registering an account (for ANI authentication) in the ITSP's system and transferring a part of his telecom balance to this account. This is done by sending an SMS with a certain command

to a premium number. Once registration is completed, the client can use the transferred amount for international calls via a standard ANI-based calling card IVR. If the account's balance falls below a specified threshold, the ITSP system initiates an account recharge by transferring a preconfigured amount from the customer's telecom balance.

- **Ability to make an outgoing call from one's mailbox (DISA functionality)** – The PortaUM *One's own voice mailbox access* application now allows customers to make outgoing calls from their mailboxes as if they were using a calling card. Without having a SIP phone nearby, a customer can access his voicemail from the PSTN (by dialing *your* access number from his landline or mobile phone). Apart from being able to listen to voice messages, the customer can make an outgoing call without taking extra actions, just by choosing a particular menu item.

- **Prepaid calling card IVR improvements** – The prepaid calling card application now allows the user to add the phone number he is calling from to his main account (identified by a PIN) for future PINless dialing. By keeping the original PIN, the user can associate additional phone numbers (e.g. cell phones and/or home phones) with his account. The next time he calls from any of those – the system will automatically recognize his number and allow him to make outgoing phone calls without having to manually enter his PIN.

  Another important improvement is that a customer can top-up his main account using another pre-paid card (i.e., from a debit account) in PortaBilling® instead of a top-up voucher. This allows the service provider to simplify the distribution process, since only one type of card would need to be generated, printed and delivered to distributors. The end-user would always buy the same brand of card and then simply use that one card for the initial registration and use their other cards to increase their balance.

  In order for the balance transfer between two accounts to function properly, the "donor" account must have the same currency and product as the "recipient" account and must never be used for making calls prior to the recharge (this is to prevent potential fraudulent attempts at "stealing" someone else's balance).

- **ANI authentication of calls originating from the PSTN network** – You may want to offer services such as voicemail and self-care for outside users calling your network. Usually you would configure a special access number that customers can call and then

identify themselves to using their PIN. But there are several drawbacks to this method.

Since MR24, the system is now able to authenticate calls coming into your system from vendors by using an ANI number appended with a certain prefix, by default: "<ANI>@pstn" (e.g. 44203123123@pstn). To allow your customers to authenticate themselves by using an ANI while calling to UM access numbers, assign accounts (or aliases) to them by using the format described above.

- **Support for new languages** – PortaUM now includes support for IVRs in German and Italian languages.

# Important Upgrade Notes

We try to make the process of upgrading as easy as possible, and to keep our releases backward compatible. Here are just a few things you should remember when upgrading:

- The use of the Cisco AS53xx gateway as a media server is no longer supported. Customers using the Cisco gateway as a media server for PortaUM should migrate to PortaUM's internal media server (which provides improved functionality and high call capacity).

# 1. System Concepts

# PortaUM's Role
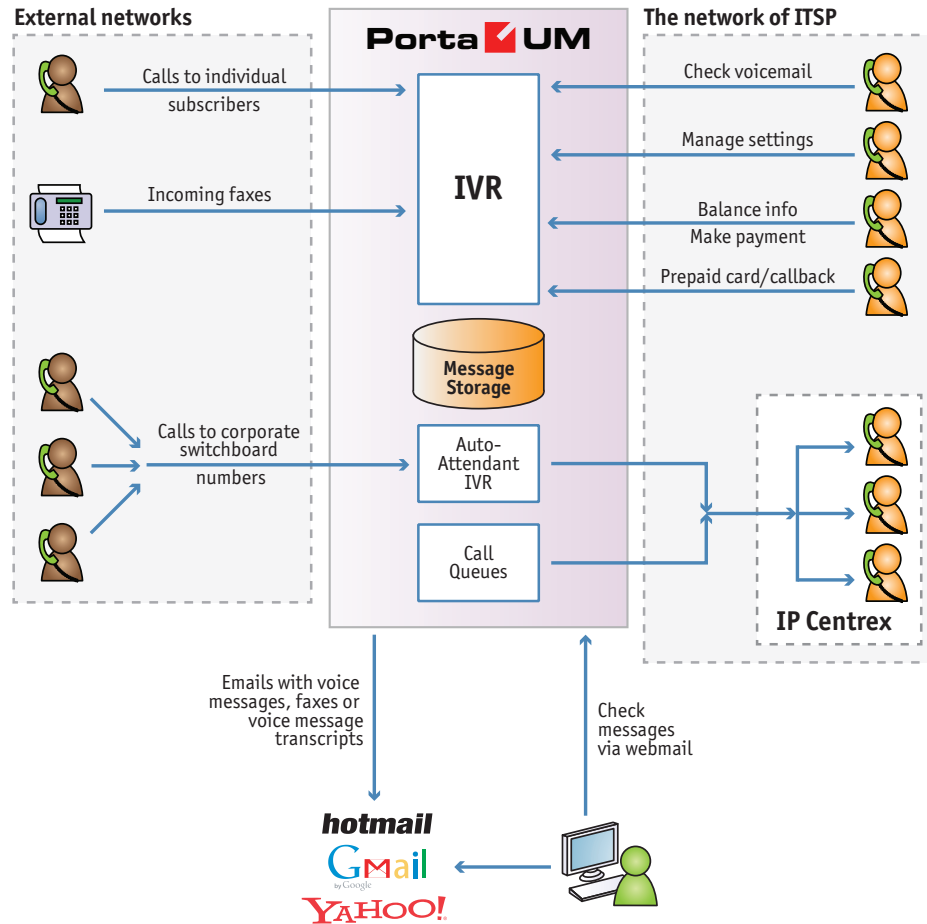


**PortaUM®** (Unified Messaging system) is one of the key components of PortaSwitch – it is a media server platform which provides users with various IVR applications: balance info, auto-attendant, prepaid card calling, conferencing, voicemail, fax-to-email, and so on.

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.

**PortaⓇUM**

# PortaUM Components
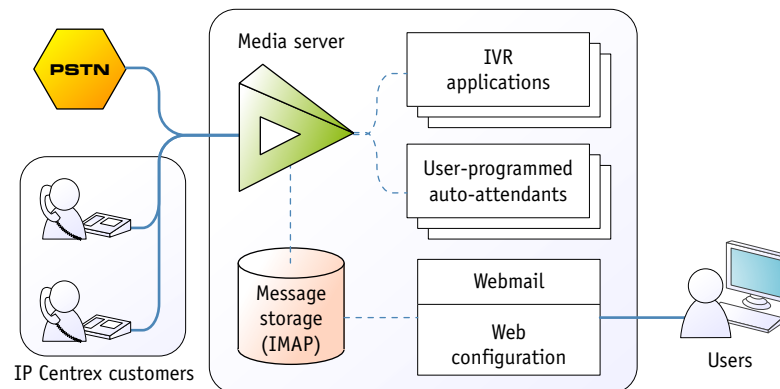


## PortaUM consists of the following main components

- A media server platform that provides IVR (interactive voice response), playing various voice prompts to users and then gathering users' input on the phone dial pad (DTMF) for menu navigation and entry of data such as credit card numbers. This component is also used to record voicemail messages and receive incoming faxes.

- A database server providing storage for all email, voice and fax messages, as well as the user's custom settings and voice prompts.

- An IMAP server which provides a unified interface for message retrieval and management, whether this is done from within PortaUM or by an external application (e.g. the user's Microsoft Outlook client).

- A web server to provide the user with web-based access to his mailbox.

- An SMTP server, used to receive incoming messages from other mail servers, as well as to send outgoing messages created by the user on the web interface, or when a message forwarding service is requested.

- A converter for converting audio and graphic attachments in messages received from the media server into formats compatible with the software available to end-users (e.g. Microsoft Windows operating system).

## PortaUM Architecture
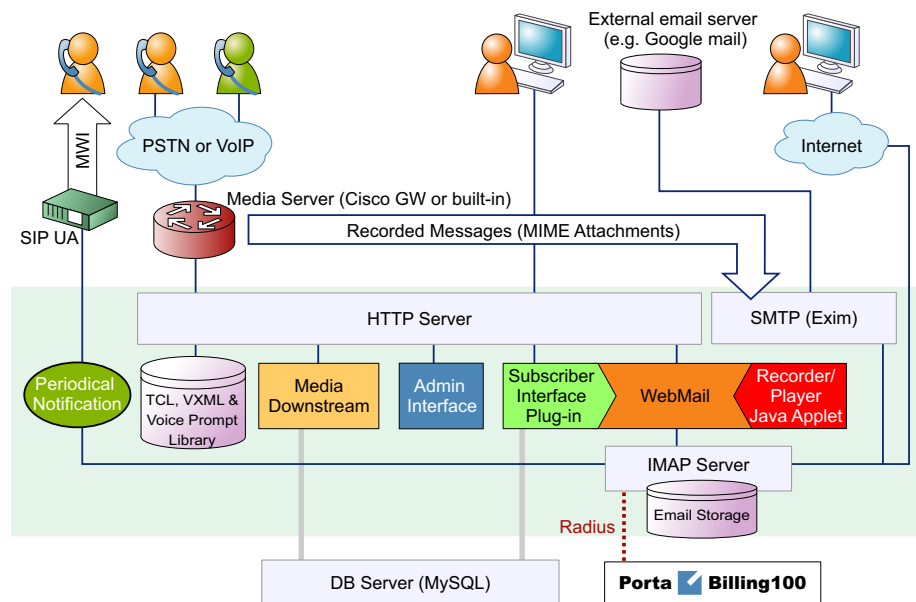


# PortaUM Performance

While for PortaBilling or PortaSIP the majority of resources are used during call initiation or call disconnect, the situation is different with PortaUM. PortaUM is a media server providing IVR services, and so all voice traffic must go through it. This means that while a call is in progress, hundreds of RTP packets must be processed each second with minimal delay, in order to maintain acceptable sound quality. This limits the number of concurrent calls a single PortaUM server can process. (Of course, you can scale up multiple PortaUM servers to achieve the required performance.)

On a server with the recommended configuration, PortaUM can handle about 1000 incoming IVR sessions (for applications such as auto attendant or balance info), or about 700 concurrent two-way IVR sessions, such as the prepaid card or callback applications, e.g. when there is an incoming call from a user to PortaUM and an outgoing call from PortaUM to an external destination.

Another limitation on PortaUM is the number of mailboxes the server can store. Each mailbox consumes a certain amount of disk space for data storage, and so the total amount of disk space used on the server quickly adds up. For instance, 20,000 users with an average mailbox size of 15

megabytes will require about 300 gigabytes of disk space. You can use quotas to set customers' maximum allowed mailbox size.

# PortaUM Clustering

When multiple PortaUM servers are installed as part of your PortaSwitch system, they all connect to the same database which contains IVR application settings, voice mailboxes, auto-attendant menus and all other data required to operate a unified communication service. Thus the function of a media server (playing voice prompts to the user, collecting DTMF input, establishing an outgoing call, etc.) can be performed by any server in the cluster. Incoming calls will be distributed among all of the available PortaUM servers – this provides the ability to perform load sharing and increase your total IVR capacity. If one of the servers is down because of a hardware failure, the remaining servers will continue processing the call.

To enable quick service recovery on the IVR service in case the database is down – a stand-by database server can be used (similar to the PortaBilling stand-by configuration).
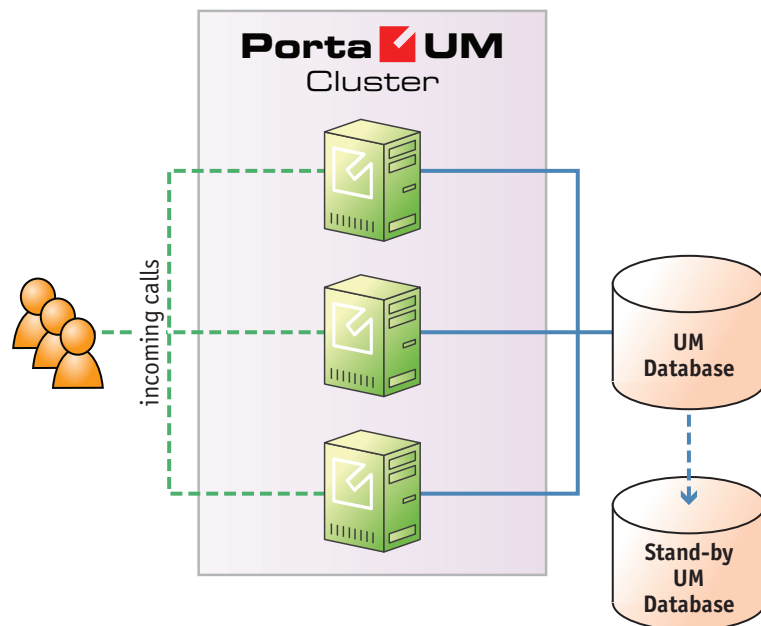


Since the PortaUM database contains unified communications data (including voice messages and faxes) for all customers – it may require a significant amount of disk storage space. At the same time, since most PortaUM operations include only simple data retrieval, the disk access speed is not crucial. Because of this, if you provide unified messaging services to a large number of customers it is recommended that you avoid

running the PortaUM database on the same physical server that is running the PortaBilling master database – it is better to set up a separate server for this, using larger storage disks. You can potentially even use slower disks (7200 RPM) to reduce the total server cost.

# 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.

## Access Number

When a pre-defined number is dialed from any phone connected to PortaSwitch, or calls to this number are received from an external network – they are forwarded to PortaUM IVR where an IVR application that is associated with this number is launched.

## IVR Applications

When PortaUM answers an incoming call, an IVR (Interactive Voice Response) application is deployed. This application defines how the call is to be further processed, i.e. what the user on the other end will hear. PortaUM IVR applications include, among others:

- Voicemail, allowing callers to leave a message which will be delivered to the user's mailbox;
- Fax-to-email;
- Auto attendant – a custom-designed set of voice menus (see below for more details);
- Check balance / top-up account;
- Prepaid card;
- Conferencing.

Each application normally has a set of parameters which customize its behavior (e.g. whether the prepaid calling card application should announce the available balance or not). The PortaBilling web interface is used to associate particular applications with a particular access number and then customize the parameters for further applications.

**PortaUM®**

## Auto Attendant

PortaUM's auto attendant is a flexible utility designed to provide IVR for callers and allow them to navigate among different options by pressing keys on their phones. Auto attendant capabilities include simple features such as playing a certain voice prompt to the end-user or collecting his input, as well as more advanced features such as incoming fax detection or call queues.

# Managing IVR Applications via Web GUI

Using PortaBilling web interface administrators can quickly and conveniently define how users access IVR applications by dialing phone numbers. Your customers would dial a phone number from the PSTN network or their IP phone to access a specific IVR application; for instance, *98 for voicemail, 12125551234 to access a prepaid calling card IVR, or 18005559876 to access the audio conferencing facility. So when PortaSwitch receives a call and matches the destination number with one of the access numbers - the call will be forwarded to PortaUM. PortaUM then will launch the specific IVR application associated with that access number.
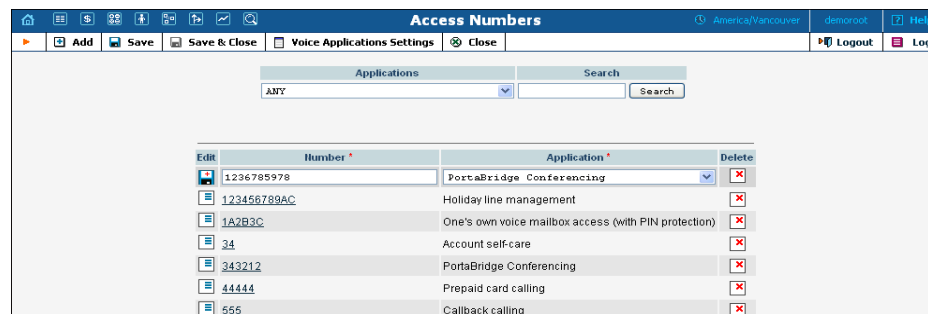


## Application options

Administrator can also customize properties of the application that is running on a specific access number.

**15**
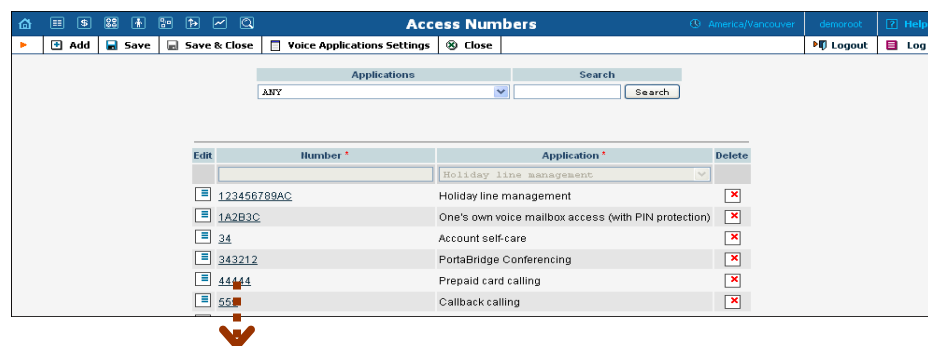
# Call Scenarios

## Call between Subscribers on VoIP Network

In this scenario a subscriber makes a call from his/her IP phone to a phone number that belongs to another subscriber on your VoIP network.
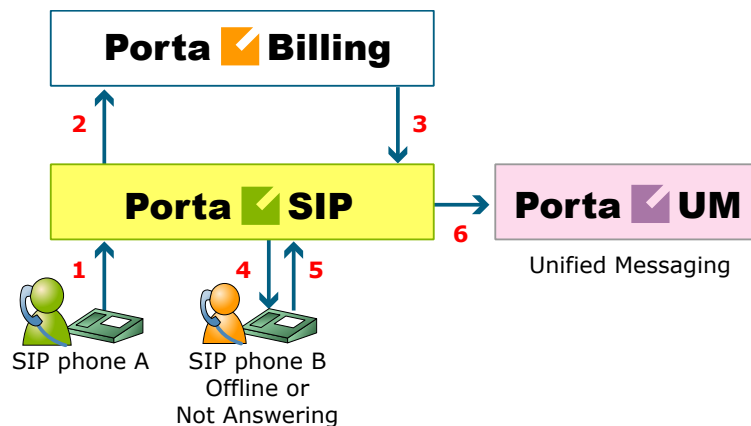


- User A dials 121 in an attempt to reach user B. A's SIP user agent sends an INVITE request to the SIP server (**1**).
- The SIP server performs authorization in the billing (**2**). For example, if A dialed 121, the billing will inform the SIP server that the actual number is 12027810009, and that 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**).

## A Call from the Outside Network (Via a VoIP DID Provider)
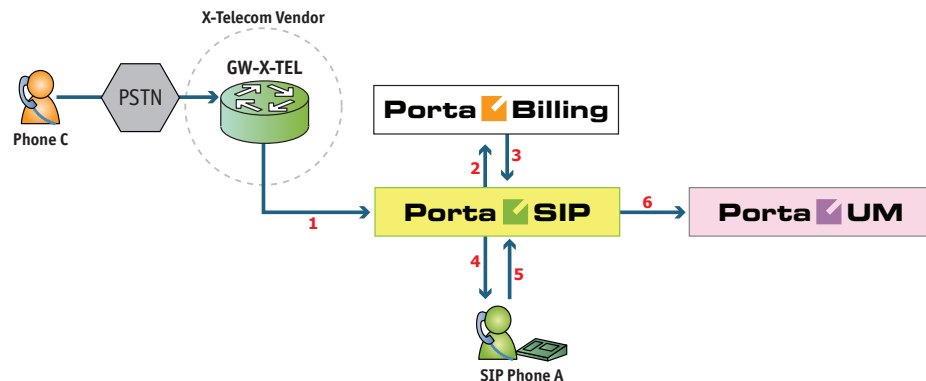


- User C (connected to the phone network of some other telco) wishes to call user A. He dials A's phone number 12025550003.
- The call is routed through the telecom network of the carrier providing the services to C, and then possibly via some transit operators. Eventually the call is delivered to the DID consolidator X-Telecom, which supplies the incoming DID calls to the ITSP. From the switch of the carrier X-Telecom, the call is sent to PortaSIP.
- When an incoming call arrives to PortaSIP (**1**), after gathering the required information, PortaSIP sends an authorization request to billing (**2**).
- On the PortaBilling side, several operations are performed:
  o First of all, PortaBilling detects that this is a call coming from a "VoIP from Vendor" connection which belongs to X-Telecom.
  o Then PortaBilling checks whether the number 12025550003 is actually serviced on the ITSP network; this is done to prevent a situation when, either by mistake or intentionally, X-Telecom starts sending transit voice traffic to the ITSP network. Only if there exists an account (or account alias) with ID 12025550003, meaning that this number indeed belongs to one of the customers, will the call be allowed to proceed; otherwise the authorization fails and the call is dropped.
  o Then PortaBilling checks if account 12025550003 is currently listed in the shared database of registered IP phones and what the contact IP address is.

- o Also, since account 12025550003 has the "Unified Messaging" service activated, PortaUM is added to the list of call delivery routes.
- o Finally, the authorization confirmation, which includes A's current location (IP address and port), call forwarding or voicemail information and the maximum allowed talk time, is sent back to PortaSIP (**3**).
- PortaSIP starts the outgoing call to 12025550003 (**4**).
  - o If A answers the call, the call is connected as usual. PortaSIP starts the timer, and will disconnect the call if the maximum call duration is exceeded.
  - o If A does not answer his phone (**5**), or his phone is not registered at this time, then the call is routed to PortaUM.
- PortaUM connects the incoming call (**6**) and, based on the number called (12025550003), retrieves the configuration settings (language, customized prompts, personal greeting, etc.) for this mailbox.
- The "Voicemail" IVR application is started, which announces that the called party is not available and allows the caller to leave a voice message or send a fax.
- The voice message or fax is processed, converted and delivered to the user's inbox.

## A Call from the Outside Network (Using a Gateway)
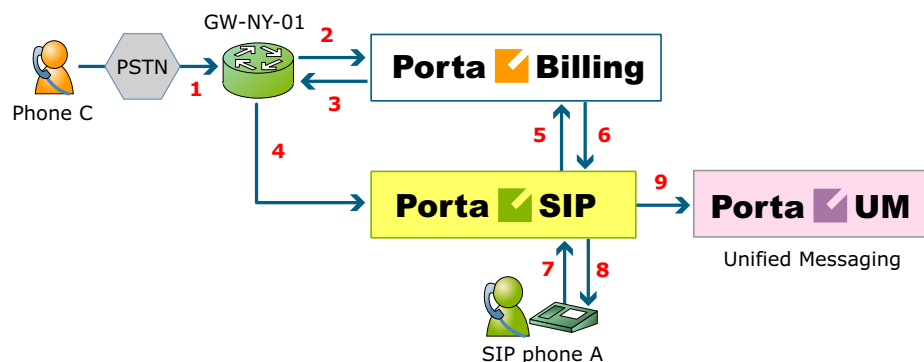


This is an alternative, when calls made on the PSTN network to your numbers are routed to your gateway via the telephony interface.

- User C (connected to the phone network of some other telco) 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:

- o Converts the phone number to E.164 format, so that 2027810003 becomes 12027810003.
- o 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.
- o Starts the outgoing call to 12027810003.
- o When the call is established, it starts the timer, and will disconnect the call if the maximum call duration is exceeded.
- o 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**).

## IVR Application
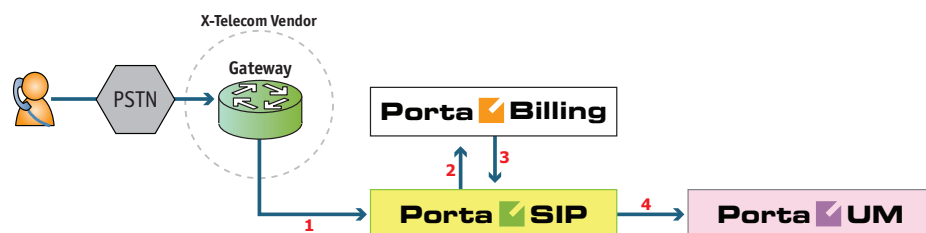


- The service provider wants to allow customers to access an IVR application (e.g. to check their voicemail from an external phone line). The number to be dialed by users (e.g. 18005555865 – 1800-555-5VML) is purchased from the DID provider.
- The administrator creates a new entry in the **Access Numbers** section in PortaBilling, assigning the "Voicemail Access (with PIN protection)" application to 18005555865, and configures the parameters of the application, if necessary.

- Customer C wishes to check his voice messages while out of the office; he dials 18005555865 from his cell phone.
- The call is routed through the telecom network of the cellular carrier providing the services to C, and then possibly via some transit operators. Eventually the call is delivered to the DID consolidator X-Telecom, which supplies incoming DID calls to the ITSP. From X-Telecom's switch the call is sent to PortaSIP.
- When an incoming call arrives to PortaSIP (**1**), PortaSIP checks the call handling rules to determine how this call should be authorized, i.e. based on the remote IP address or using the username and password. After gathering the required information, PortaSIP sends an authorization request to billing (**2**).
- On the PortaBilling side, several operations are performed:
  - First of all, PortaBilling detects that this is a call coming from a "VoIP from Vendor" connection which belongs to X-Telecom.
  - Then PortaBilling detects that there is an entry in the Access Numbers section which designates 18005555865 as a special IVR application number.
  - This call will be forwarded to PortaUM, and none of the other call routing options (vendor connections, LCR, etc.) applies in this case.
  - Finally, the authorization confirmation, which includes the routing list (with just PortaUM in it), is sent back to PortaSIP (**3**).
- PortaSIP routes the call to PortaUM.
- PortaUM connects the incoming call (**4**) and, based on the number called (18005555865), launches the "Voicemail Access" application.
- The application prompts the user to enter a mailbox ID (his phone number on the VoIP network) and PIN. Upon successful authentication, he can listen to his messages in the same way as he would from his IP phone

# Auto Attendant
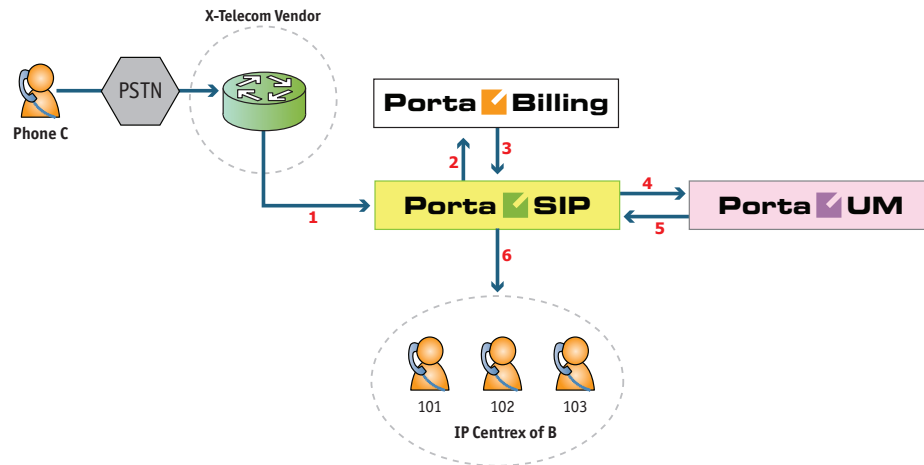


- Customer B, using IP Centrex services, purchases an extra DID number (18005551234) to serve as his main office number. An account with ID 18005551234 is created and the "Unified Messaging" service is enabled for it. This account will not be provisioned on any IP phone, since the goal is to let PortaUM handle the call. The customer logs in to the PortaUM self-care interface and configures the desired menu structure – which announcements should be made, which extensions/huntgroups calls should be forwarded to, etc.

- User C wishes to call company B. He dials B's phone number 18005551234.

- The call is routed through the telecom network of the carrier providing the services to C, and then possibly via some transit operators. Eventually the call is delivered to the DID consolidator X-Telecom, which supplies the incoming DID calls to the ITSP. From the switch of the carrier X-Telecom, the call is sent to PortaSIP.

- When an incoming call arrives to PortaSIP (**1**), PortaSIP checks the call handling rules to determine how this call should be authorized, based on the remote IP address or using the username and password. After gathering the required information, PortaSIP sends an authorization request to billing (**2**).

- On the PortaBilling side, several operations are performed:
  - First of all, PortaBilling detects that this is a call coming from a "VoIP from Vendor" connection which belongs to X-Telecom.
  - Then PortaBilling checks that an account (or account alias) with ID 18005551234 exists, meaning that this number indeed

belongs to one of the customers; otherwise the authorization fails and the call is dropped.

- o Since the account 18005551234 has the "Unified Messaging" service enabled, and this is not provisioned on any IP phone, PortaUM will be the only entry in the list of call delivery routes.
- o Finally, the authorization confirmation, which includes the routing list (with just PortaUM in it), is sent back to PortaSIP (**3**).
- PortaSIP routes the call to PortaUM.

- PortaUM connects the incoming call (**4**) and, based on the number called (18005551234), retrieves its configuration settings (e.g. auto-attendant activated for this number, voice prompts for menus, etc.).

- The auto-attendant IVR application starts up, plays the menu prompts (e.g. "Welcome to SmartDesign! Please press 1 for sales and 2 for technical support") and collects the user's input.

- If, after navigating the menu structure, user C chooses the option of being transferred to one of the extensions in the IP Centrex environment, PortaUM establishes a new outgoing call to PortaSIP (**5**).

- When an employee answers the call on that extension (**6**), PortaUM connects this call portion with the incoming call from user C directly, and is not involved in any further processing of this call (thus resources are free to process other calls).

# Auto-Attendant

## 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.

# Call Menu Flow Chart
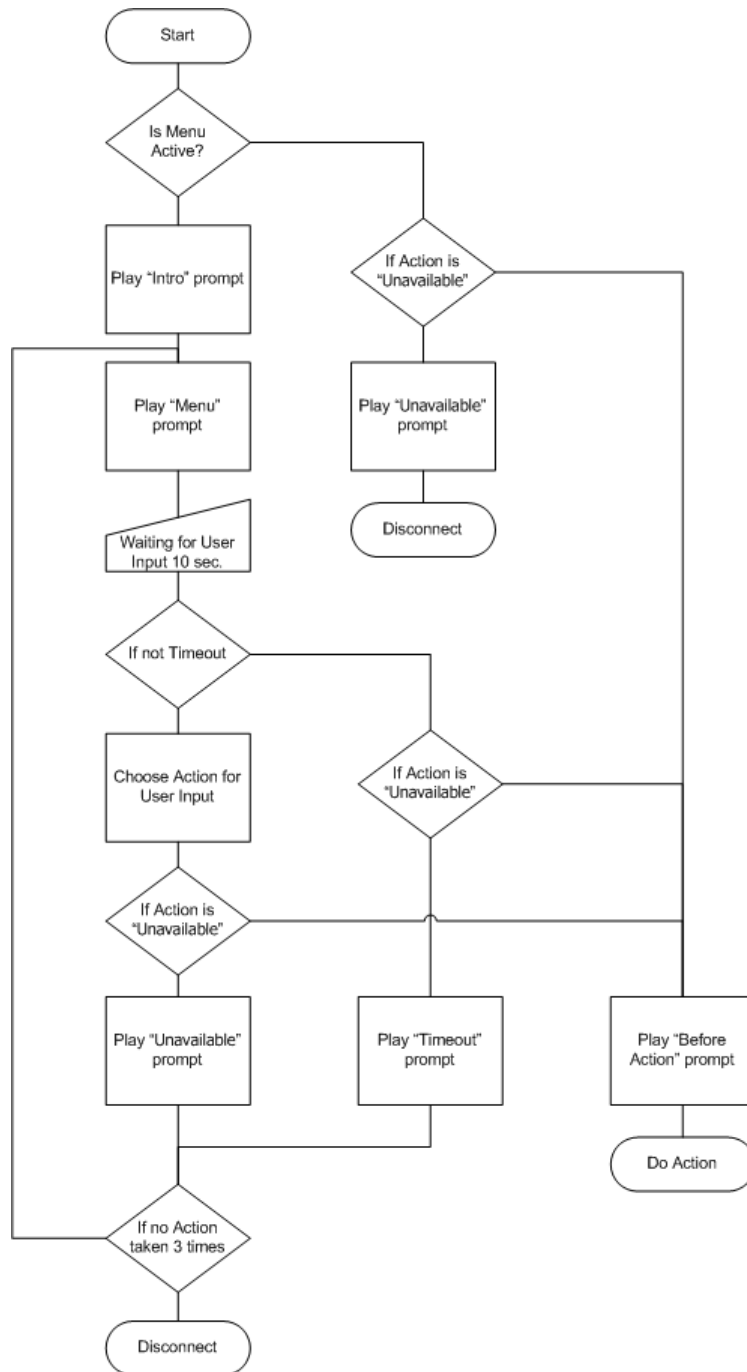


# Menu List Mode

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.
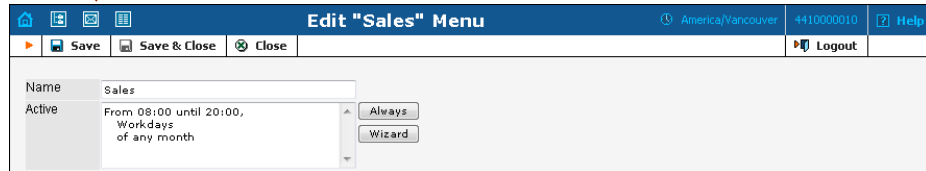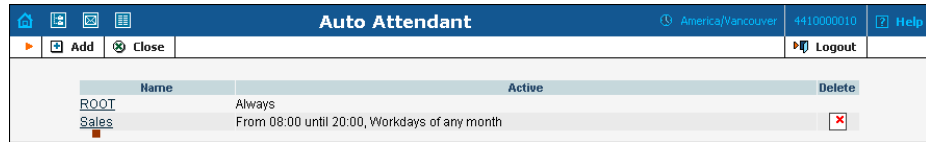


## Menu Edit Mode

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.
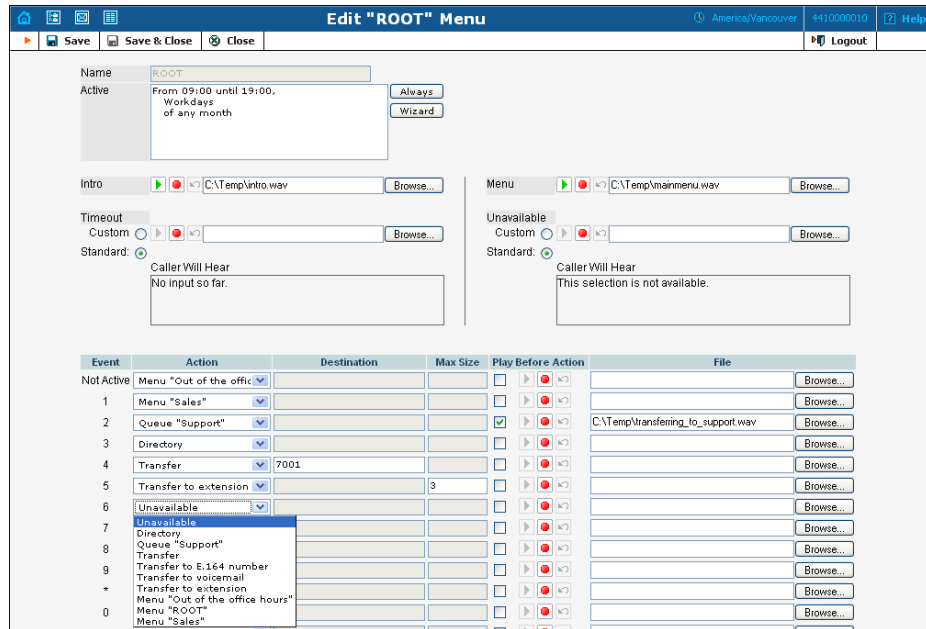


The fields of the Menu Edit screen are explained below:

**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, Timeout, Unavailable**
You can define four separate prompts (see previous section for an explanation of when each particular type of prompt is played). While you will need to provide content for the **Intro** and **Menu** prompts, you can use the default content for the **Timeout** and **Unavailable** prompts.

🔴 - **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. |
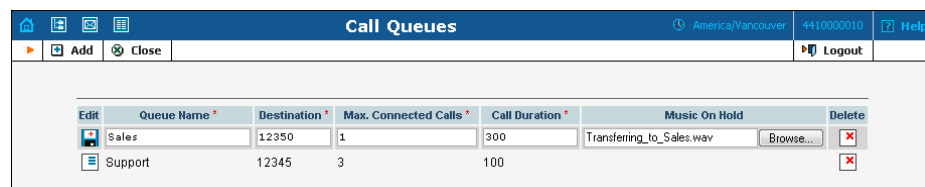
## Event table

| Column | Description |
|---|---|
| **Event** | **Not Active** – When the current menu is not active (see the active period definition above).<br>**0-9, #, \*** – User selection on telephone keypad.<br>**Timeout** – No selection received from user.<br>**Fax** – Fax CNG tone detected. |
| **Action** | See the description of available actions in the 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 arrive 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.
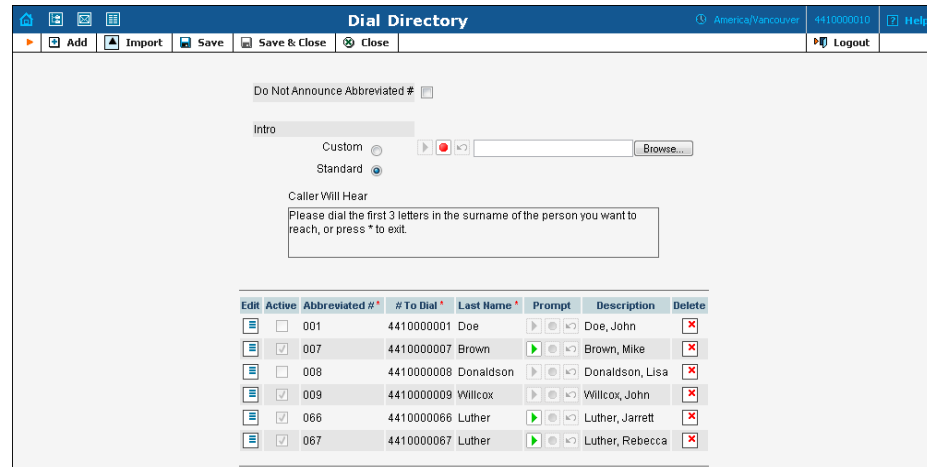
Every UM account has its own dial-by-name directory; however, an initial list of extensions may be imported from the main list in PortaBilling to save time and effort. You can upload a voice prompt with the actual person's name for each extension. You may also exclude certain extensions from being accessible 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.

Two modes of selecting the call transfer destination are available:
- The end-user hears the person's full extension (e.g. "Press 116 to reach John Brown, press 145 to reach Mary Broslavsky") and then enters it, thus learning the actual extension number for future use.
- If, for some reason, the actual extension numbers are to be hidden from end-users, then another mode is available. In this case, users can choose a person simply by dialing a sequential number from a list (i.e. the announcement in the previous example will now be: "Press 1 to reach John Brown, press 2 to reach Mary Broslavsky").

# Conferencing

## Conferencing Server Architecture

As part of the PortaUM solution, the conferencing server enables your customers to use voice-conferencing services. Multiple customers can use this service simultaneously on a single server, and each of them will have access to his own set of virtual conference rooms. The customer can manage his conferences (virtual conference rooms) via the PortaUM self-care interface.

A conference can be scheduled for a specific time, or a continually functioning conference (meeting room) can be created. Each conference is identified by a pair of unique access codes (one for the conference host, the other for conference guests). Although multiple conferences can be created on a PortaUM server, its resources are only used when a conference is in progress (i.e. at least one participant is in).
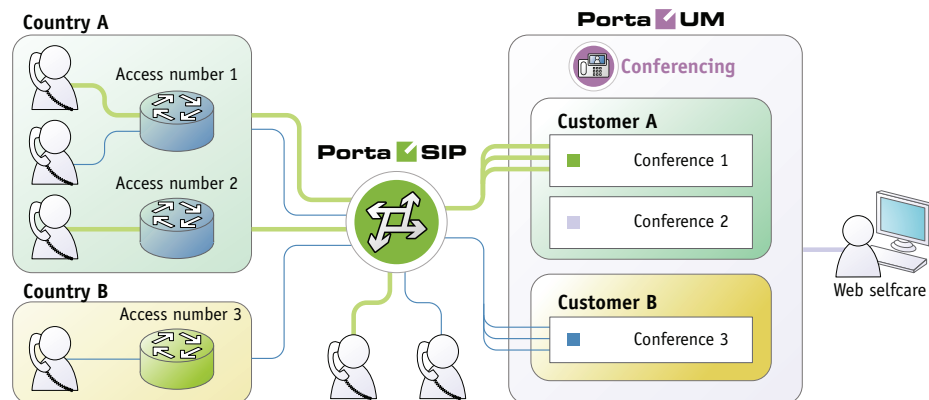


When a customer makes use of the conferencing facility, the PortaBilling administrator assigns a separate tariff for the conferencing service in the customer's product configuration, and the owner of the meeting room is then billed for each incoming call in a conference session. For example, if your conference rate is $0.03/min, and a customer organizes a conference for two participants where he (the host) stays connected for 30 minutes and his two partners stay connected for 25 and 20 minutes, respectively, then there will be three charge transactions in total: $0.90, $0.75 and $0.60. To prevent potential service abuse by guests, customers may create a moderated conference. In this case, guests can only participate in a conference when the host has already joined it.

Conferencing server carries out intensive calculations when mixing the audio from several meeting participants, and so requires significant amounts of CPU power. This is why, if you plan to use the conferencing

service extensively, you should consider allocating an additional PortaUM server just for running the conferencing service, so that your other IVR services (voicemail, auto-attendant, self-care, etc.) can run properly.

## Call Flow
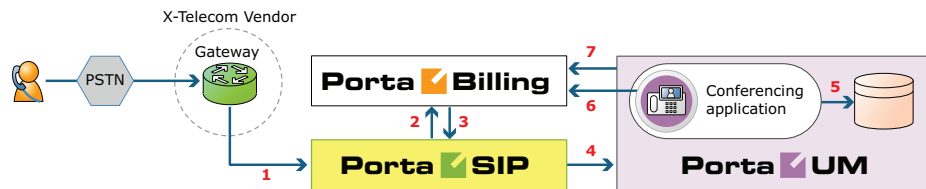


- The service provider wants to allow customers to access the audio conferencing service. The number to be dialed by users (e.g. 18665552637 – 1866-555-CNFR) is purchased from the DID provider.
- The administrator creates a new entry in the Access Numbers section in PortaBilling, assigning the "Conferencing" application to 18665552637, and configures the parameters of the application, if necessary.
- When a customer signs up for the conference service, an account conf1234 for managing the service is created in PortaBilling and provided to the customer. The customer then logins into their PortaUM self-care interface and creates specific conferences (virtual meeting rooms).
- The customer distributes the conference access information (access phone number and access code) to intended participants of the conference.
- A participant (connected to a network of some other telco) wishes to join the conference; he dials 18665552637 from his cell phone.
- The call is routed through the telecom network of the cellular carrier providing the services to the participant, and then possibly via some transit operators. Eventually the call is delivered to the DID consolidator X-Telecom, which supplies incoming DID calls to the ITSP. From X-Telecom's switch the call is sent to PortaSIP.
- When an incoming call arrives to PortaSIP (**1**), PortaSIP checks the call handling rules to determine how this call should be authorized, i.e. based on the remote IP address or using the username and password. After gathering the required information, PortaSIP sends an authorization request to billing (**2**).
- PortaBilling detects that there is an entry in the **Access Numbers** section which designates 18665552637 as a special IVR application number. The authorization confirmation, which includes the routing list (with just PortaUM in it), is sent back to PortaSIP (**3**). PortaSIP routes the call to PortaUM.

- PortaUM connects the incoming call (**4**) and, based on the number called (18665552637), launches the "Conferencing" application.
- The application prompts the user to enter the conference access code and validates it in the internal database (**5**).
- If a valid conference code is provided, the ID of the account that owns this conference room (conf1234) is retrieved. An authorization request is sent to PortaBilling (**6**) to check that usage of this particular conference number (18665552637) is allowed for this account and that the account has sufficient balance to cover the costs.
- If authorization is successful, the participant is allowed to join the conference.
- When the participant hangs up, an accounting request is sent to PortaBilling (**7**), so the account conf1234 (and the customer who owns it) are charged based on price per minute associated with this access number, and the call duration.

## Codec Licenses

Normally, a media server (such as the one used in PortaUM) will send pre-converted voice prompts to the user as a byte stream, so that no codec licenses are involved. In the case of conferencing server, however, the audio-stream with the voice of each participant has to be decoded in real time, and then sent back to each participant as an encoded audio-stream with the mixed sound. Thus, if commercial codecs (such as g729) are used, a codec license is required for each voice channel being used simultaneously. The conferencing server license does not include any codec licenses, and customers should procure such licenses independently.

# 2nd number service

Customers whose businesses receive many incoming calls may want to have one or more additional contact numbers for those purposes that are separate from their existing home or personal mobile numbers. At the same time, they may still want to be able to answer incoming calls to those numbers using their home or mobile phones. PortaBilling® now allows you to provide a so-called 2nd number service to satisfy those requirements.

Let's take a look at the following example:
- A customer with the mobile phone number 42073723123 buys a 2nd number service from you.

- You allocate another number for him, e.g. 42027777100, from the DID range and create an account in PortaBilling that represents that phone number.
- Add the alias 42073723123@pstn (the "Allow authentication/registration" flag should be disabled for this) to the account 42027777100 to associate the customer's external phone number with the DID.
- Set the default answering mode to "Forward then voicemail" for this account.

## Listening to voice mail and making an outgoing call
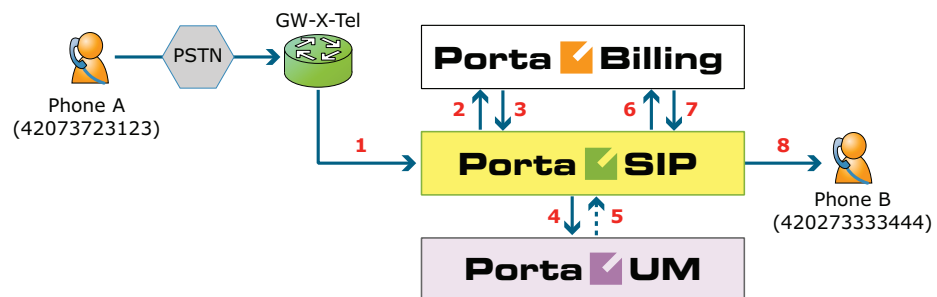


- From his 42073723123 mobile (**1**), the customer dials the phone number 42027777100 that was provided to him.
- PortaSIP sends an authorization request to billing (**2**).
- He is authorized by ANI 42073723123, and since 42073723123@pstn is an alias of the number dialed, the call is treated as an incoming call to the user's own mailbox.
- The authorization confirmation, including the routing list (with just PortaUM in it), is sent back to PortaSIP (**3**). PortaSIP routes the call to PortaUM.
- PortaUM connects the incoming call (**4**) and launches the "Voicemail Access" application.
- The customer can listen to messages the same way he would from his IP phone.
- If, after navigating the menu structure, the customer chooses the option of making an outgoing call, he will be prompted to input the destination.
- Customer inputs 420273333444.
- PortaUM® sends an INVITE request to the SIP server (**5**).
- The SIP server sends an authorization request to billing (**6**).
- Billing sends an authorization response to the SIP server (**7**).
- The SIP server sends the call (**8**) to 420273333444 (the party being called will see 42027777100 as the CLI).

### Receiving an incoming call
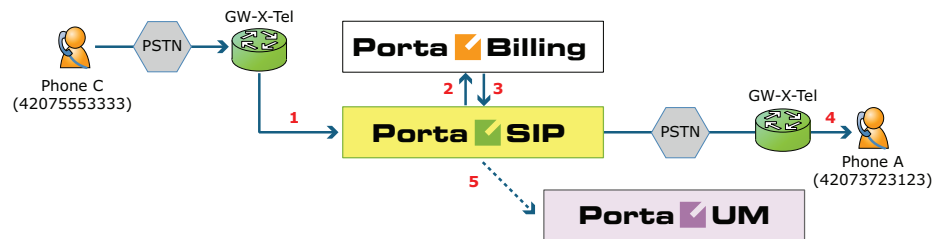


- From PSTN (42075553333), the user dials (**1**) customer's number: 42027777100.
- PortaSIP sends an authorization request to billing (**2**).
- The account for ANI authentication is not found and the call is treated as an incoming call to the 42027777100 account. PortaBilling recognizes that this account has Follow-me services enabled, and produces a list of routes, accordingly.
- The authorization confirmation including the routing list (with the Follow-me number and PortaUM in it), is sent back to PortaSIP (**3**).

The SIP server redirects the call (**4**) to a Follow-me number (customer will see 42027777100 as the CLI). If the call is not answered it will be redirected to voicemail (**5**), and the user can leave a message.

# Calling via Premium Number

Beginning with Maintenance Release 24, service providers can significantly increase their customer base by allowing subscribers of any other telco operator in the country to make cheap international calls via the service provider's network without any initial registration. Providers do not need to print and distribute calling cards or top-up vouchers; customers do not have to make a contract, perform online sign-up or purchase a card beforehand. This drastically simplifies the customer acquisition process and reduces associated costs.

A person can call a premium number that is registered on your network from his existing mobile phone or landline. For this call he will be charged by his operator according to the rate you assign to that particular premium number. In return, you can enable outgoing international calls to any of the destinations where your cost is lower than the revenue from the premium number. For instance, if you set the rate for premium number 9001234567 to $0.10/min, you can allow calls made to any destination where you pay your vendors less than $0.10/min to terminate the call. There is no need for the end-user to be provisioned in your system, you do not need to generate an invoice or request a payment: the telco, which

"owns" the end-user, will collect the funds from the end-user and then forward that amount (minus commission, if applicable) to you.

Usually there are several "tiers" of pricing – e.g. the price of calling landlines in most European countries is about the same, but the price of calling a mobile number in those countries is significantly higher, and some exotic destinations may even be more expensive. Multiple premium access numbers are utilized to address this, so for each group of countries / destination that fall within the same pricing tier, the end-user can be offered a premium number with a competitive price (while this price will still be higher than the average termination price in the group). For instance, the end-user can utilize a premium number with a $0.20/minute price to call landlines in Europe, and another premium number with a $0.50/minute price to call mobile networks in Europe. In the tariff for each premium number the administrator will put:

- Ordinary rates for destinations that customers can call (the price that the end-user pays to call a premium number is sufficient to cover the outgoing call there).
- Destinations that are too expensive to call using this premium number are listed in the tariff as "forbidden." Nonetheless, price per minute information is still included in those rates and plays an important role. It suggests a per-minute rate for a premium number to call those destinations. This price will be used to search for the number that can be used to call those destinations from among all of the available premium numbers.

Let's assume the average termination cost for destination 442 (UK-London) is $0.12 and for 447 (UK Mobile) it's $0.22. There are two premium numbers, 12320 and 12350, with per minute rates of $0.20 and $0.50, respectively. Each of these numbers is configured in the Access Numbers section and the price per minute is set in the "Access Number Price" attribute. There are two separate tariffs, A and B – each linked to one of the numbers (see the examples in the **PortaSwitch Converged Services handbook** for detailed configuration examples). In tariff A, associated with the 12320 number, the service provider will enter the rate for 442 as normal, so users are allowed to make calls to UK-London via this number. The rate for 447 is marked as "forbidden" since otherwise, the service provider would lose money (the $0.20 price per minute that the customer pays for the call to the 12320 number is less than the provider's cost of $0.22). So customers are "forbidden" from calling the 447 (UK-Mobile) destination with that premium number. In the "Price" attributes of the rate for 447 the administrator enters $0.50, which means that the customer will be advised to use the premium number at the $0.50 price per minute rate.
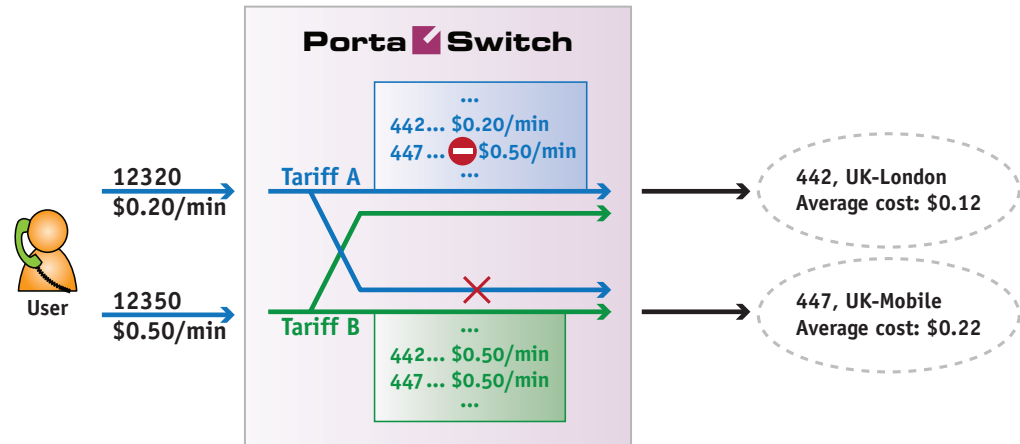
According to our example, that would be the 12350 number. The 12350 number has tariff B associated with it, and tariff B's rates are "normal" for both destinations, so either destination can be called by the customer using that 12350 premium number. This offers additional flexibility to the service provider since different destinations can be mapped using either the more or the less expensive premium numbers.

## Call Flow

- A user dials a premium number from his mobile phone or landline, the call arrives to PortaUM. The end-user is prompted to enter the destination number.
- PortaUM attempts to authorize a call to this number using PortaBilling.
- If PortaBilling confirms that calling that destination using this premium number is allowed, PortaUM will connect the outgoing call.
- If PortaBilling rejects the request (the destination is marked as forbidden in the tariff), PortaUM will retrieve the price per minute entered in the forbidden rate – this is the "suggested" rate that we should collect for a call. Then PortaUM will search for a premium number IVR among the list of configured access numbers that has a "price per minute" attribute with the same value as the suggested price.
    - If an access number is found, the user will be informed that in order to complete the call he should use that premium number – the number and the rate for it are announced.
    - Otherwise the user is informed that a call to that destination is not possible and is advised to contact customer support for further assistance.

# 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.

### DISA functionality for One's own voice mailbox access application

When a customer accesses an IVR application to check his voicemail he may also want to make an outgoing call right away (e.g. to call back the person who left the voicemail). Now the customer can make an outgoing call from the voice mailbox by simply choosing that option from the application menu.

### Fax-to-email services

In addition to voice messages, UM-enabled SIP customers will be able to receive faxes on their "answering machine". When a call is connected to PortaUM and the remote party initiates a fax transmission, PortaUM automatically detects this event and switches into fax receiving mode. The fax will be processed and then delivered to the user's mailbox as a .PDF or .TIFF document attached to an email message.

### 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. Also, they can hear their current balance by dialing the access number for their mailbox and pressing option "0." 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).
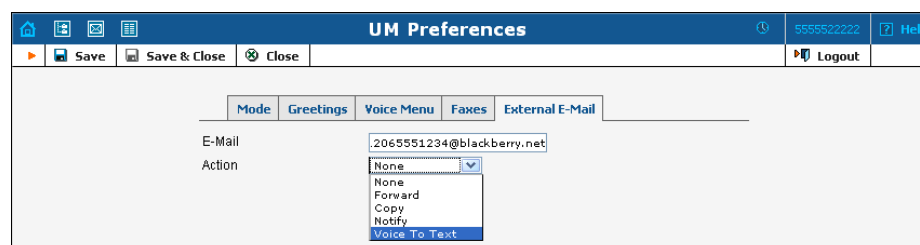
### Auto attendant

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.

## Voice-to-text Service

PortaUM is now integrated with the **GotVoice** service to provide "visual voicemails" to customers. Once a recorded voice message is delivered to PortaUM, it can be transferred to one of the GotVoice servers for processing. GotVoice uses state-of-the-art technology to recognize message contents, combining computerized speech recognition with live human assistance for those parts which cannot be processed automatically. In a matter of minutes the customer receives an email containing a transcript of the message text. This makes it possible to access the voicemail information via an email client or an SMS sent to a mobile phone.



VoIP providers using PortaSwitch can now reliably offer subscribers the ability to read their voicemail and process the information instantly. Using this service requires a contract with GotVoice (in order to be assigned an account giving access to the conversion API) and the basic PortaSwitch configuration (to create rates for this service, etc.).

An ITSP must sign a contract with GotVoice for voice-to-text service and obtain the username/password information for accessing the conversion API. This information should be provisioned in porta-um.conf configuration file on PortaUM.

Since voice-to-text is a paid service, where ITSP is charged for every conversion, charges should be applied to the customer, who is using the service. PortaUM performs authorization in PortaBilling prior to conversion (to verify that the customer has sufficient funds). Once conversion has been completed, it sends a "charge" event to PortaBilling.

## Registration for PINless Dialing Service via SMS

This is an extension to the "classic" PINless dialing service, in which the user is identified by his caller ID (ANI/CLI), maintains a balance in PortaSwitch and is able to call any destination with charges applied to his balance. Traditionally, the registration for PINless dialing service is performed by the end-user via the purchase of a top-up card and then

calling the special IVR. When the user's balance is low, he needs to buy more top-up cards or pay his bill online. The registration via SMS (simply send an SMS message to a certain number) is an easier alternative. A simplified registration process and further management of the account make this service especially convenient for users of mobile networks. The service provider has two options for collecting funds from the end-user:

- Send a request to the mobile operator that the end-user is connected to for a balance transfer. This option must be supported by the mobile carrier – its advantage is that any "ordinary" number can be used by the customer for sending an SMS.

- Establish a premium number where the end-user can send an SMS. Just like calls to premium numbers are charged differently than normal calls, messages to premium numbers are charged according to the higher rate (set by the service provider who owns the number). So for each message sent by the end-user, the mobile carrier will collect the money from the end-user and then pass it to you as the owner of the premium number.

To start using the service, the end-user simply sends an SMS with the message SUBSCRIBE to the registration number. Upon receiving the message, the SMS processing module in PortaSwitch automatically provisions his account with the CLI/ANI of the mobile phone as the authentication ID; requests a balance transfer from the mobile operator and then adds these funds to the customer's balance in PortaSwitch so they can be used again in the future. End-user will receive an SMS that confirms the service activation and provides important service information, such as the number to call for PINless dialing or a username/password for accessing the web self-care. Once that occurs, the end-user can start using the service by dialing the access number and then entering the final destination number. The entire registration process is very easy to perform and can be completed in just a few seconds. PortaSwitch can also perform an automated balance top-up: when the balance of the created account goes below a specified threshold, the system initiates another balance transfer from the user's balance on the mobile network. The telecom client can disable this automatic recharge by sending an SMS to the SMS access number with the message STOP.

# 2. Appendices

# APPENDIX A. PortaUM IVR Applications Preferences

## Prepaid Card Calling

| Field | Description |
|---|---|
| ANI Authentication | If **ANI Authentication** is turned on, the IVR application will attempt to authenticate the customer's phone number (ANI). If authentication is successful, the IVR does not ask for a PIN. |
| ANI Translation Rule | Caller identification (ANI or CLI number) is usually delivered in the "local" format, without the country code, etc., e.g. **021234567**. In this case, you need to translate the number into a unified format so it will match the ID of the billing account. The recommended format is a non-numeric prefix followed by an E164 number. In our earlier example, assuming that the country code is **44**, we applied ANI Translation Rule **s/^0/ani44/** to obtain "**ani4421234567**" as the authorization ID. This allows you to manage numbers from different countries (so they will not overlap) and the "ANI" prefix will ensure that these accounts are not confused with the actual phone numbers provisioned for on IP phones. |
| Manual Authentication | If **ANI authentication** is disabled or fails for some reason, users will still be able to authenticate their identities by using their PIN numbers when this option is enabled. |
| Auto Registration | If the user's ANI is not found and PIN authentication is successful, the user will be prompted to register their ANI for future PINless dialing. If the user confirms, the ANI number he is calling from is added as an alias to his account. The next time a call is made from that phone number, it will be authenticated by the ANI and he will not be asked to enter a PIN number. |
| Minimum Card Length | Minimum length of a card number. By default: 11. |
| Maximum Card Length | Maximum length of a card number. By default: 11. |
| Maximum Login Attempts | Specify maximum number of attempts for user to enter a card number. By default: 3. |

| Authorize with IVR Session ID | PortaBilling provides the same Routing Plan within a session. If you are using the Routing Plan function, you may need to disable this option in order to allow the Billing Engine to recover its separate routing plan for each authorization. Note: When disabled, PortaUM will generate a new h323-conf-id for every authorization request. |
|---|---|
| Check Service Password | Use this option to increase the security of PIN-less dialing service or simply to prevent unauthorized ANI usage when ANI Authentication is enabled. Upon collecting an ANI number, IVR prompts customer for their password and uses it for account validation (the password provided must match the account's Service Password). Note: If the account has an empty service password field, any user input matches it. |
| Minimum Password Length | Minimum number of characters the valid service password consists of. By default: 4. |
| Maximum Password Length | Maximum number of characters the valid service password consists of. By default: 4. |
| Languages | This enables you to select language(s) to be used as voice prompts. Click the wizard icon to open the language selection window. |
| Conf Language List is Preferred | If this box is checked, the language list defined on this page has a higher preference than the account's preferred language. |
| Prompts Brand | IVR prompts can be customized per Access Number. Specify the directory where the custom prompt files are located. PortaUM will look for the prompts in that folder first and if found – use them. Otherwise, it will use the default files. (For example, put customized English prompts under the directory /porta_var/porta-um/prompts/brands/ my_company/en/ and specify "my_company" in this field). |
| Play Welcome Message | If enabled, IVR will play the welcome message to a user once he reaches an access number. |
| Toll Free | If unchecked, the prepaid application will exit immediately if the account has insufficient funds. |
| Offer to Recharge | If **never** is defined, IVR never prompts for a recharge. When **on low balance** is defined and the account's balance is less than product breakage, IVR announces that the balance is low and offers to top-up the balance. If **always** is defined, IVR always prompts for a top-up. |

| Minimum Voucher Length | Minimum length of a voucher number. By default: 11 characters. |
|---|---|
| Maximum Voucher Length | Maximum length of a voucher number. By default: 11 characters. |
| Announce Balance | If enabled, the IVR will announce the current balance for the user before prompting for the phone number he wishes to reach. |
| Check Associated Number | When enabled, the prepaid card application checks if the account has specified an associated number.. If it has, the call is redirected to this number. For example, the user buys a PSTN number (e.g. 1800123), a corresponding account is created in PortaBilling and the user specifies his own mobile number as the associated number. In this case if anyone calls his PSTN number (e.g. 1800123), the call is automatically redirected to his mobile phone. |
| Announce Time | When a user inputs the destination, IVR announces the maximum call duration allowed for this destination and then connects the call. |
| Use Announcement Tariff | It is possible to use different tariffs for the announcement of the maximum call duration allowed and the rate for this call. The first (transparent) tariff should be associated with the "ANNOUNCE+DNIS" access code in the product's accessibility list. The second tariff should be associated with an actual access code. Thus, the user hears the time announced according to the first tariff, while the call is actually authorized and rated according to the second tariff. |
| Seconds Rounding Mode | This option defines the way the seconds portion of "available time" is announced to a customer:<br>• **none** – there is no rounding, so the time is announced including seconds, e.g. "Five minutes and 49 seconds."<br>• **up** – the time is rounded up to the nearest minute, so 5:49 will be announced as "Six minutes." |
| Play Pre-ring MOH | By default, the calling party hears a ringtone while waiting until the remote party answers. You may want to replace the ringtone with music on hold (MOH). Check this option to allow IVR to play a special MOH ring when the call has already been placed but the ring signal has not yet been received. |

| | |
|---|---|
| **Music on Hold Class** | Specify a Pre-ring MOH class here. Note: Ringtone files for custom MOH should be placed under /porta_var/porta-um/prompts/moh/OPTION_VALUE/ (e.g. for the MOH class "my-ringtone" the path will be /porta_var/porta-um/prompts/moh/my-ringtone/). |
| **End Announce Time, sec** | This feature announces the remaining time in seconds or simply plays a "beeping" sound. This "time left" warning is played when there are a specified number of seconds left before the call is disconnected. |
| **End Announce Type** | Choose one of the following types of "time left" warnings:<br>• beep – plays "beeping" sound;<br>• voice – announces the remaining time in seconds. |
| **Disconnect Call Before, sec** | The call can be disconnected before an account's balance is actually depleted. Specify in seconds (till the calculated end of the call) when to disconnect the call. |
| **Maximum Dial Attempts** | Maximum number of dial attempts within one session. By default: 3. |

## Callback

| Field | Description |
|---|---|
| **Authenticate by** | This option defines a user-name for an authentication request. Choose one of the following:<br>• **ani** – Calling-Station-ID (caller's number) is used for an authentication request.<br>• **dnis** – Called-Station-ID (access number which was dialed) is used for an authentication request. This application establishes a callback to the associated number parameter of the account equal to the access number.<br>• **none** – Do not immediately authenticate. This application prompts the user to enter a card number for authentication after calling the user back. |

| | |
|---|---|
| **ANI Translation Rule** | If ANI authentication is used, the voice application will attempt to authenticate the customer's phone number (so he does not have to enter a PIN). Unfortunately, caller identification (the ANI or CLI number) is usually delivered in a specific format (e.g. "local" - without the country code and area code: **021234567)**. In this case, you need to translate the number into a unified format so it matches the ID of the billing account. The recommended format is a non-numeric prefix followed by an E164 number - in our earlier example (assuming that the country code is **44**) we applied ANI Translation Rule **s/^0/cb44/** to obtain "**cb4421234567**" as the authorization ID. This allows you to manage numbers from different countries (so they do not overlap) and the "cb" prefix ensures that these accounts are not confused with actual phone numbers provisioned for on IP phones. |
| **Send Authorization** | When enabled, an authorization request will be sent along with authentication. It allows you to: <br>• check if the caller has sufficient balance to initiate a call leg (i.e., user will not be called back if the balance is insufficient). <br>• block some destinations for callback calls in the account's tariff (mark destinations as forbidden in the appropriate tariff rates). <br>• use a **Minimum Sufficient Time** option. |
| **Minimum Sufficient Time** | Specify the minimum amount of seconds a user needs for leg A in order to use the callback service (applicable when **Send Authorization** is set to **Yes** and **Minimum Sufficient Time** > 0). |
| **Dial Translation Rule** | When the caller identification (ANI or CLI number) is delivered in a specific format (e.g. "local") and should be modified for a callback, this translation rule is used. <br>For example, when caller identification is delivered in a domestic European dialing format (0 + area code + phone number, e.g. **0 5 888 7766**) and we need to call the user back using an E164 format. In this case (assuming that the country code is **44**) we would apply Dial Translation Rule **s/^0/44/** to obtain **44 5 888 7766** as the number to call back. |

| | |
|---|---|
| **Wait Time** | Delay (in seconds) before the callback application calls the user back. |
| **Auto Registration** | When enabled, it allows a caller to use callback with the auto registration feature. When the customer dials an access number, the system drops the call to initiate the callback. Customer is prompted to enter a voucher number and after validation, the system creates a new account with the exact same options/product as an etalon account (an etalon account is defined as a parameter for each separate access number). |
| **Etalon Account** | Specify the account ID here to use its configuration properties as an etalon for all auto registered accounts. |
| **Minimum Card Length** | Minimum length of a card number. By default: 11. |
| **Maximum Card Length** | Maximum length of a card number. By default: 11. |
| **Maximum Login Attempts** | Maximum number of attempts for a user to enter a card number. By default: 3. |
| **Use Fake Account** | During callback, the user is asked to enter his card number. In case of unsuccessful authorization (e.g. the user has not entered a valid number) a fake account will be used for charging leg A. Enable the **Use Fake Account** option to use a fake account for a PIN-based callback scenario. (Default: no) |
| **Fake Account** | Specifies a fake account. Also, you can allow/deny certain destinations in this account's tariff as the initial callback will be authorized with the 'fake account'. Note: If it is not defined, PortaUM uses the application access number by default. |
| **Languages** | This enables you to select language(s) to be used for voice prompts. Click the wizard icon  to open the language selection window. |
| **Conf Language List is Preferred** | If this box is checked, the language list defined on this page has a higher preference than the account's preferred language. |

| | |
|---|---|
| **Prompts Brand** | IVR prompts can be customized per Access Number. Specify the directory where custom prompt files are located. PortaUM will look for the prompts in that folder first and if found – use them. Otherwise, it will use default files. (For example, put customized English prompts under the directory /porta_var/porta-um/prompts/brands/ my_company/en/ and specify "my_company" in this field). |
| **Play Welcome Message** | If enabled, IVR will play the welcome message to the user once an access number is reached. |
| **Check Low Balance** | When account balance is less than product breakage and **disconnect** is defined, IVR plays the "you have insufficient funds" voice prompt and disconnects the call. If **no calls** is defined, IVR prompts for the destination number but gives a busy tone and drops the call immediately after dialing it. Option **none** disables this feature. |
| **Announce Balance** | If enabled, the IVR will announce the current balance for the user before prompting for the phone number he wishes to reach. |
| **LegA CLD Prefix** | Leg A (callback to a user's number) access code prefix. Allows billing leg A with different tariff. A separate accessibility entry with the appropriate access code and tariff should exist in the product's accessibility list. Note: Access code must contain DNIS after the Prefix. For example, if the **LegA CLD Prefix** is LEG_A and the Access Number is 12125551234, then the Access Code should be LEG_A12125551234. |
| **LegB CLD Prefix** | Leg B (user's call to destination) access code prefix. Allows billing leg B with different tariff. A separate accessibility entry with the appropriate access code and tariff should exist in the product's accessibility list. Note: Access code must contain DNIS after the Prefix. For example, if the **LegB CLD Prefix** is LEG_B and the Access Number is 12125551234, then the Access Code should be LEG_A12125551234. |
| **Expected Connect Time** | This option verifies that the account's balance is sufficient for the leg B setup time. If the **Expected Connect Time** value is more than the maximum leg A duration, leg B will not be allowed. Default: 20 seconds. |

| Special Charge if LegB Fails | By enabling this, you can charge users differently for calls with failed leg B (e.g. incomplete call is not charged). Just add a new accessibility entry to the product's accessibility list, set its access code to "CALLBACK_FAIL" and associate it with a special tariff. |
|---|---|
| Strict Authorization | When enabled, call leg A will be disconnected if caller doesn't have sufficient funds to place call leg B. |
| Announce Time | When user inputs the destination, IVR announces the maximum call duration allowed to this destination and then connects the call. |
| Use Announcement Tariff | It is possible to use different tariffs for the announcement of the maximum call duration allowed and for rating this call.<br>The first (transparent) tariff should be associated with the "ANNOUNCE+DNIS" access code in the product's accessibility list.<br>The second tariff should be associated with an actual access code.<br>Thus, the user hears the time announced according to the first tariff, while the call is actually authorized and rated according to the second tariff. |
| Seconds Rounding Mode | This option defines the way the seconds portion of "available time" is announced to a customer:<br>• **none** – there is no rounding, so the time is announced including seconds, e.g. "Five minutes and 49 seconds."<br>• **up** – the time is rounded up to the nearest minute, so 5:49 will be announced as "Six minutes." |
| Play Pre-ring MOH | By default, the calling party hears a ringtone while waiting until the remote party answers. You may want to replace this ringtone with music on hold (MOH).<br>Check this option to allow IVR to play a special ringing MOH when the call has been placed but the ring signal has not yet been received. |
| Music on Hold Class | Specify a Pre-ring MOH class here.<br>Note: Ringtone files for custom MOH should be placed under /porta_var/porta-um/prompts/moh/OPTION_VALUE/ (e.g. for the MOH class "my-ringtone" the path will be /porta_var/porta-um/prompts/moh/my-ringtone/). |

| | |
|---|---|
| **End Announce Time** | This feature announces the remaining time in seconds or simply plays a "beeping" sound. This "time left" warning is played when a specified number of seconds is left before the call is disconnected. |
| **End Announce Type** | Choose one of the following types of "time left" warnings: <br> • beep – play "beeping" sound; <br> • voice – announce the remaining time in seconds. |
| **Disconnect Call Before** | You can configure the system to disconnect a call before an account's balance is actually depleted. Specify in seconds (till the calculated end of the call) when to disconnect the call. |
| **Maximum Dial Attempts** | Maximum number of dial attempts within one session. By default: 3. |

## Account Top-up via Credit Card

| Field | Description |
|---|---|
| **Languages** | This enables you to select language(s) to be used for voice prompts. Click the wizard icon ![icon] to open the language selection window. |
| **Conf Language List is Preferred** | If the box is checked, the language list defined on this page has a higher preference than the account's preferred language. |
| **Minimum Sum** | Specify minimum amount of funds to be transferred from a customer's credit card. |
| **Minimum Card Length** | Minimum length of a card number. By default: 11. |
| **Maximum Card Length** | Maximum length of a card number. By default: 11. |
| **Maximum Login Attempts** | Specify maximum number of attempts for a user to enter a card number. By default: 3. |
| **Minimum Voucher Length** | Minimum length of a voucher number. By default: 11. |
| **Maximum Voucher Length** | Maximum length of a voucher number. By default: 11. |
| **Customer Center Number** | Specify the number where a user may reach an operator. |

## Account Top-up via Voucher

| Field | Description |
|---|---|
| **Maximum Login Attempts** | Specify maximum number of attempts for a user to enter a card number. By default: 3. |

| Minimum Voucher Length | Minimum length of a voucher number. By default: 11. |
|---|---|
| Maximum Voucher Length | Maximum length of a voucher number. By default: 11. |

## One-stage Calling

Note that the number for this application must be in the format <special-prefix>%, e.g. 5061999%.

The process of One-stage calling to an Access Number is the following:

- The customer dials any number (e.g. 12065551234) from his home phone;
- The local telco sends this call to your system with a special 5061999 prefix (so, for example, the CLD is 506199912065551234); alternatively this special prefix can be added in the **Translation Rule** field of the particular connection or even dialed by the customer (this depends upon the agreement between the local telco and your company);
- The call is routed to PortaUM;
- PortaUM starts the One-stage calling application;
- The application strips the 5061999 prefix from the CLD and sends an authorization request to PortaBilling;
- PortaBilling performs the authentication by ANI, checks whether the customer is allowed to call this particular destination and the length of the maximum allowed call duration, and sends back a reply;
- IVR announces the maximum allowed call duration (if enabled);
- The application sends the call to PortaSIP (the CLD is 12065551234); and
- PortaSIP routes the call according to the system routing plan.

| Field | Description |
|---|---|
| **ANI Translation Rule** | If ANI authentication is turned on, the voice application will attempt to authenticate the customer's phone number (so he does not have to enter a PIN). Unfortunately, caller identification (ANI or CLI number) is usually given in the "local" format, without a country code, etc.: **021234567**. In this case, you need to translate the number into a unified format so it matches the ID of an account for billing. The recommended format is a non-numeric prefix followed by an E164 number in our earlier example (assuming that the country code is **44**) we applied ANI Translation Rule **s/^0/ani44/** to obtain "**ani4421234567**" as the authorization ID. This allows you to manage numbers from different countries (so they do not overlap) and the "ANI" prefix ensures that these accounts are not confused with actual phone numbers provisioned for on IP phones. |
| **Languages** | This enables you to select language(s) to be used for voice prompts. Click the wizard icon ![icon] to open the language selection window. |
| **Conf Language List is Preferred** | If the box is checked, the language list defined on this page has a higher preference than the account's preferred language. |
| **Prompts Brand** | IVR prompts can be customized per Access Number. Specify the directory where the custom prompt files are located. PortaUM will look for the prompts in that folder first and if found – use them. Otherwise, it will use default files. (For example, put customized English prompts under the directory /porta_var/porta-um/prompts/brands/ my_company/en/ and specify "my_company" in this field). |
| **Play Welcome Message** | If enabled, IVR will play the welcome message to a user once he reaches an access number. |
| **Announce Time** | When user inputs the destination, IVR announces the maximum call duration allowed to this destination and then connects the call. |

| | |
|---|---|
| **Use Announcement Tariff** | It is possible to use different tariffs for the announcements of maximum call duration allowed and the call's rating. The first (transparent) tariff should be associated with the "ANNOUNCE+DNIS" access code in the product's accessibility list.. The second tariff should be associated with an actual access code. Thus, the user hears the time announced according to the first tariff while the call is actually authorized and rated according to the second tariff. |
| **Seconds Rounding Mode** | This option defines the way the seconds portion of "available time" is announced to a customer: <br> • **none** – there is no rounding, so the time is announced including seconds, e.g. "Five minutes and 49 seconds." <br> • **up** – the time is rounded up to the nearest minute, so 5:49 will be announced as "Six minutes." |

## One's Own Voice Mailbox Access

| Field | Description |
|---|---|
| **Languages** | This enables you to select language(s) to be used for voice prompts. Click the wizard icon  to open the language selection window. |

You can allow customers to make outgoing calls from their mailboxes. This can be enabled on the **Voice Applications Settings** tab.

## Premium Numbers

| Field | Description |
|---|---|
| **Special Username** | Specifies the ID of a specially configured premium number account. The account must have no credit limit (to allow unlimited chargeable calls); account's product must list all access numbers in its services and rating list and each one must be associated with a corresponding tariff. |
| **Customer Care Number** | Specifies the number a user can call if he experiences any difficulties using the service. This number will be announced to the user by the IVR. |

| | |
|---|---|
| **Access Number Price** | Specifies the price for this particular access number. If the user tries to dial a destination that is not covered by the access number he has called, PortaBilling will try to find the proper access number by matching the price for the desired destination (in the tariff assigned to the current access number) with the value specified here. The user is then informed of it. |

# APPENDIX B. Supported DTMF Methods

PortaUM® supports the following DTMF methods:

- Inband
- RFC 2833
- SIP INFO

# APPENDIX C. Supported Languages

The following tables show which languages are supported by specific PortaUM IVR applications.

## Self-care IVR Applications

| LANGUAGE APPLICATION | English (en) | Spanish (es) | French (fr) | Swiss French (fr-CH) | Hebrew (he) | Brazilian Portuguese (pt-BR) | Russian (ru) | Cantonese Chinese (yu) | Mandarin Chinese (cm) | Greek (el) | Arabic (ar) | Swedish (sv) | German (de) | Italian (it) |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| **Account top-up via credit card** | Yes | Yes | No | No | No | No | No | No | Yes | No | Yes | Yes | Yes | Yes |
| **Simplified account self-care** | Yes | Yes | No | No | No | No | No | No | Yes | No | Yes | Yes | Yes | Yes |
| **Call Forwarding management** | Yes | Yes | Yes | No | Yes | No | Yes | No | Yes | No | Yes | Yes | Yes | Yes |
| **Account Self-care** | Yes | Yes | Yes | No | Yes | No | No | No | Yes | No | Yes | Yes | Yes | Yes |
| **Account top-up via voucher** | Yes | Yes | Yes | Yes | Yes | Yes | Yes | No | Yes | No | Yes | Yes | Yes | Yes |
| **Balance information** | Yes | Yes | Yes | No | No | No | Yes | No | Yes | No | Yes | Yes | Yes | Yes |
| **One's own voice mailbox access (with PIN protection)** | Yes | Yes | P | No | P | No | P | No | Yes | No | Yes | Yes | Yes | Yes |
| **One's own voice mailbox access** | Yes | Yes | P | No | P | No | P | No | Yes | No | Yes | Yes | Yes | Yes |
| **Payment Remittance – TransferTo** | Yes | Yes | No | No | No | No | No | No | No | No | No | No | Yes | Yes |

## Outgoing Calling Applications

| LANGUAGE / APPLICATION | English (en) | Spanish (es) | French (fr) | Swiss French (fr-CH) | Hebrew (he) | Brazilian Portuguese (pt-BR) | Russian (ru) | Cantonese Chinese (yu) | Mandarin Chinese (cm) | Greek (el) | Arabic (ar) | Swedish (sv) | German (de) | Italian (it) |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| One-stage calling | Yes | Yes | Yes | Yes | Yes | Yes | Yes | P | Yes | Yes | Yes | Yes | Yes | Yes |
| Callback calling | Yes | Yes | Yes | Yes | Yes | Yes | Yes | P | Yes | P | Yes | Yes | Yes | Yes |
| Prepaid Card Calling | Yes | Yes | Yes | P | P | Yes | Yes | P | Yes | P | Yes | Yes | Yes | Yes |
| Premium Numbers | Yes | Yes | No | No | No | No | No | No | No | No | No | No | Yes | No |

## Other IVR Applications

| LANGUAGE / APPLICATION | English (en) | Spanish (es) | French (fr) | Swiss French (fr-CH) | Hebrew (he) | Brazilian Portuguese (pt-BR) | Russian (ru) | Cantonese Chinese (yu) | Mandarin Chinese (cm) | Greek (el) | Arabic (ar) | Swedish (sv) | German (de) | Italian (it) |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| Holiday line management | Yes | Yes | Yes | No | No | No | No | No | Yes | No | Yes | Yes | Yes | Yes |
| Conferencing | Yes | No | No | No | No | No | No | No | No | No | No | No | No | No |

o  No – not supported
o  Yes – fully supported
o  P – partially supported (incomplete prompts list or prompts are out of date)

# APPENDIX D. IVR Flow Diagrams

## Accessing your Own Mailbox IVR

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.
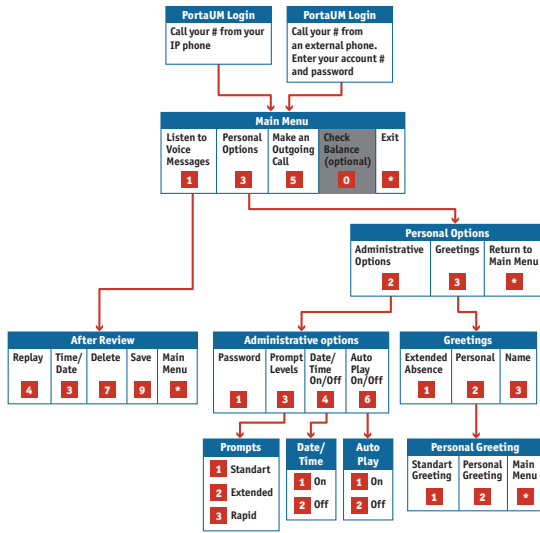


## Personalization IVR (Short Schema)

Below is a brief IVR path diagram to help users to navigate and configure the system. A full IVR diagram can be found in the **Personalization IVR (Full Schema)** section.

**Porta UM®**
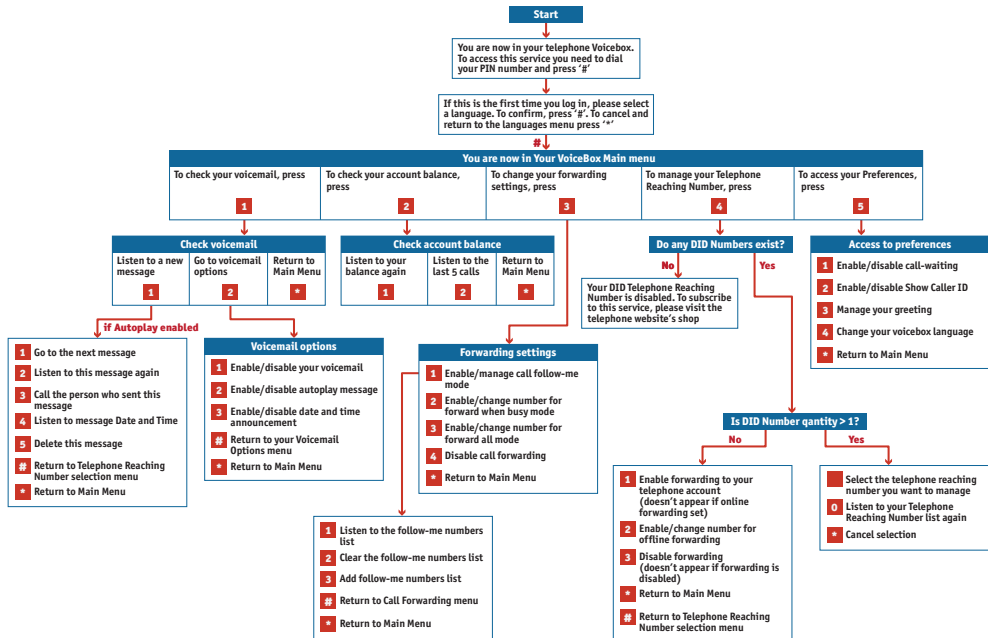
## Voucher Recharge IVR
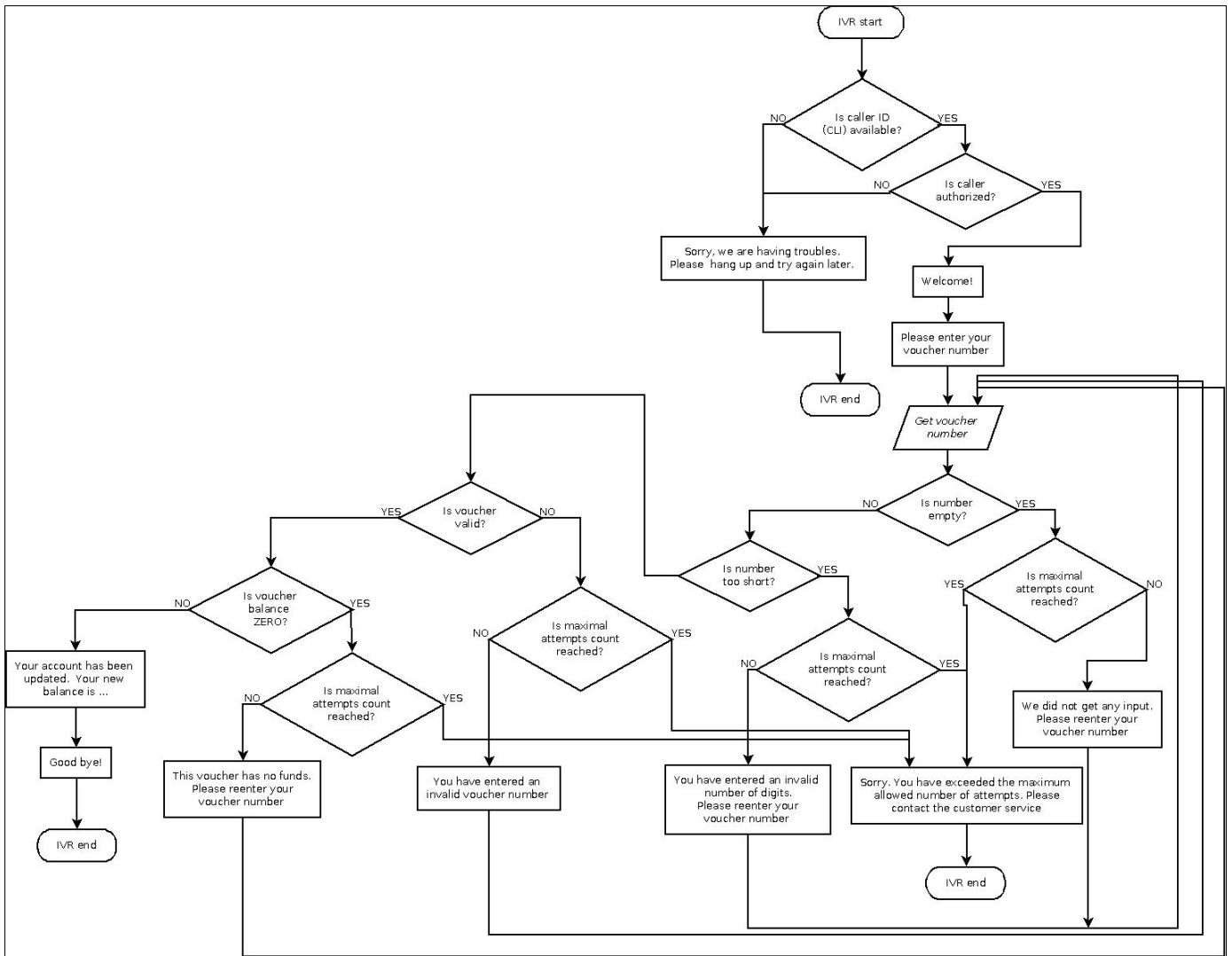
## Pay by Credit Card IVR

## Account Management IVR

Application note.
The DNIS format should be:
*77*n*123456789,
where *77 - Account Management IVR DNIS;
n – operation (1 - recharge account, 2 – add alias, 3 - delete alias);
123456789 – passed Alias ID or Voucher ID

Start

Check ANI, DNIS

ANI available — no

yes

DNIS format valid — no

yes

Check Passed parameters

Required operation is recharge — yes / no

Check Voucher

Voucher valid — no / yes

Top-up account with a voucher

Success — no / yes

Play: "The PIN number is invalid"
File: man_auth_voucher_failed

Play: "Your balance has been updated for ..."
File: man_balance_upd

Required operation is add_alias — yes / no

Check Alias ID

Alias ID already exists — yes / no

Play: "The number 12345 already exists in the database"
File: number + ...+ already_exists

Add a new alias

Success — no / yes

Play: "The number 12345 is not valid, please try again"
File: number + ... + alias_add_failed

Play: "The number ... has been added to your account"
File: number + ... + alias_added

Required operation is delete_alias — yes / no

Check Alias ID

Alias ID exists — no / yes

Play: "The number 12345 is not linked to your account"
File: number + ...+ no_alias

Delete alias

Success — yes / no

Play: "The number 12345 has been successfully removed from your account"
File: number + ... + alias_deleted

Play: "The number 12345 is not valid, please try again"
File: number + ... + alias_delete_failed

End

## Prepaid Card IVR

**PortaUM®**

## ANI/DNIS/PIN Callback IVR

Porta**UM**®
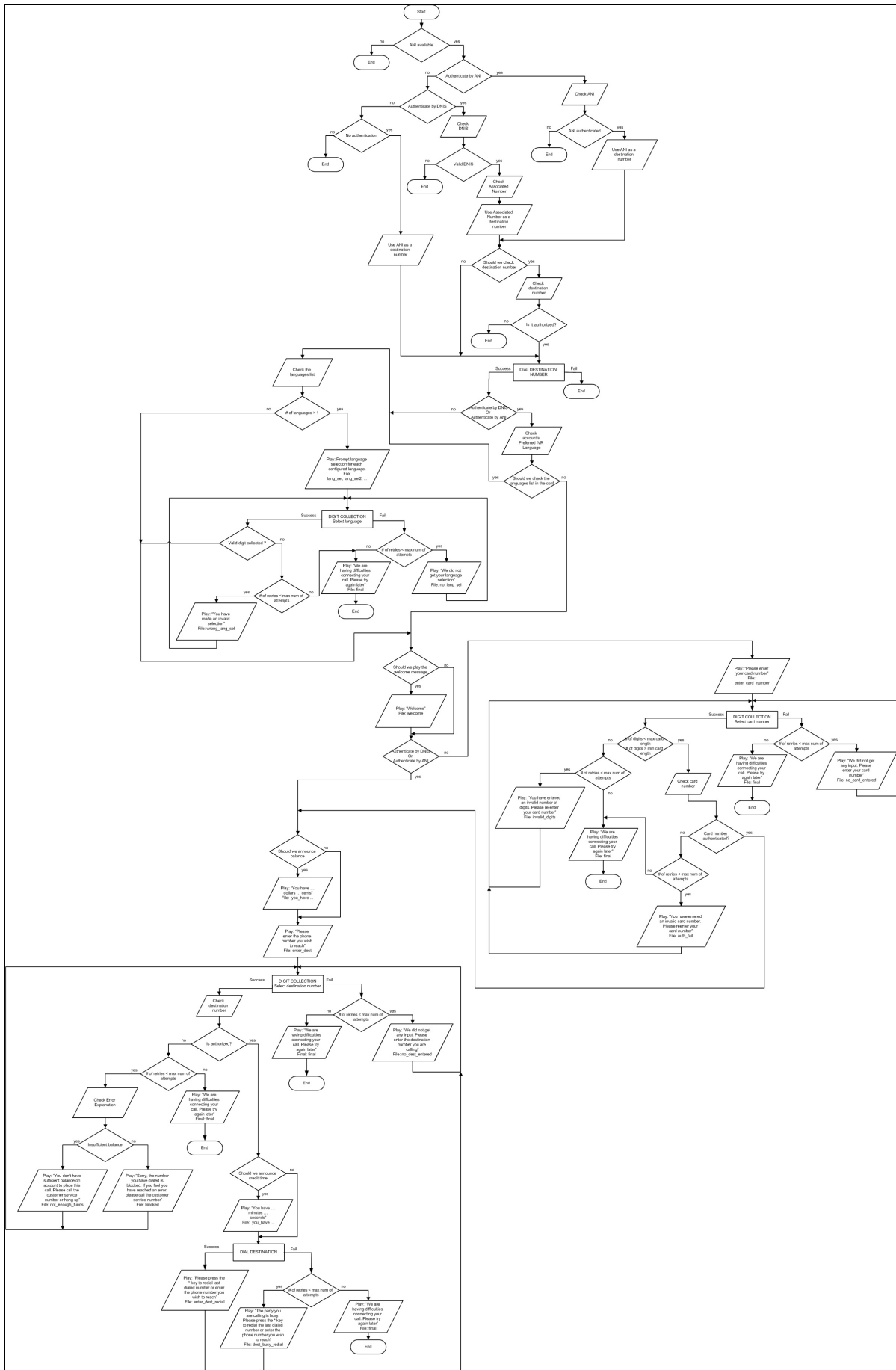
**Callback with Registration IVR**

**Premium Numbers IVR**

PortaUM®

**One-stage Dialing IVR**

```
                              ┌──────────┐
                              │  Start   │
                              └──────────┘
                                   │
                              ╱──────────╲
                             │  Check CLI, │
                             │     CLD     │
                              ╲──────────╱
                                   │
                              ╱──────────╲
                             │   Check     │
                             │ destination │
                             │   number    │
                              ╲──────────╱
                                   │
            no              ◇───────────◇              yes
      ┌──────────────────── │ Is authorized? │ ────────────────────┐
      │                     ◇───────────◇                          │
      │                                                            │
```

**Is authorized?** — no / yes

**Check Error Explanation**

**Insufficient balance** — yes / no

Play: "You don't have sufficient balance on account to place this call. Please call the customer service number or hang up"
File: not_enough_funds

Play: "Sorry, the number you have dialed is blocked. If you feel you have reached an error, please call the customer service number"
File: blocked

**Should we play the welcome message** — no / yes

Play: "Welcome"
File: welcome

**Should we announce credit time** — no / yes

Play: "You have … minutes … seconds"
File:  you_have ...

**DIAL DESTINATION**

**End**

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## Personalization IVR (Full Schema)

Is Number Empty ?

YES
You have entered an empty number.

To reenter the number press 1 to exit number entrance and go back to the Telephones DID service menu press #

"1"

"#"

NO

Please dial the number you want to add after the tone and press the '#' key

"2"

You have entered <NUMBER> To save this number press '#' To cancel and redial press '*'

"#"

>>Play message date and time<<

Is Mailbox Full?

"NO"

"4"

Your message will be deleted ------ To continue press # To return to your messages press '*'

#

The message has been deleted

"YES"

Your voicemail box is full. To receive new messages please delete some existing ones

Message N from ##### on DATE_TIME >>Message content<< + 'End of message'.

To go to the next message press '1' To listen to this message again press '2' To call the person who sent this message press '3' To listen to message date and time press '4' To delete this message press '5' To return to your Voicemail Menu press '#' To return to the Main Menu press '*'

"3"

Is there senders number?

"YES"

Is Dest. Allowed?

"YES"

-->Place a call<--

Stop IVR

"NO"

Sorry. There is no number to dial

"NO"

Sorry, the number you have dialed is blocked.

"1"

Select the Telephone Reachin number you want to manage.
------
To manage Telephone Reachin number '###' press '1'
To manage Telephone Reachin number '###' press '2'
------
To listen to your Telephone Reachin number list again press '0'
To cancel selection press '*'

"1"

You are now in Your Telephone Reachin number menu
Your Telephone Reachin number <number> is currently - not forwarded to any number - forwarded to your Telephones account - forwarded to <NUMBER>
------
To enable forwarding to your Telephone account press '1' (doesn't appear if online forwarding set)
To enable/change number for offline forwarding press '2'
To disable forwarding press '2' (doesn't appear if forwarding disabled).
------
To return to the Main Menu press *
To return to the Telephone Reachin. number selection menu press '#' (appears if DID numbers quantity >1)

"1"

Your Telephone Reachin number has been forwarded to your Telephone account

"2"

Forwarding of your Telephone Reachin number has been disabled

"1"
or "Auto-Play enabled"

You are now in your Voicemail Menu. You have <number> (new) messages.
>> Go to "Listen to new messages" if auto-play enabled <<
To listen to your (new) messages press '1' >>Prev. entry played only if some messages/emails exist <<
To go to your voicemail options press '2'
To return to the Main Menu press *

Last message?

YES

"There are no more (unread) messages"

NO

Is DID Number quantity > 1

YES

NO

Is DID selected?

"YES"

"NO"

Do any DID numbers exist?

"YES"

"NO"

Your DID Telephone Reachin number service is disabled. To subscribe to a Telephone Reachin number please visit the telephone website's shop.

START

Welcome to your telephone Voicebox. To access this service you need to provide us with your Pin Number. Please dial the PIN number and press '#'

Welcome to your telephone Voicebox.

"4"

You are now in Your VoiceBox. Main Menu
------
To check your voicemail press '1'
To check your account balance press '2'
To change your forwarding settings press '3'
To manage your Telephone Reachin numbers press '4'
To access your preferences press '5'

"5"

Call forwarding has been disabled

"2"

Your balance is <amount>

"1"

Your balance is <amount>

"1"

You are now in your account menu
------
To listen to your balance again press '1'
To listen to the last 5 calls you have made press '2'
To return to the Main Menu press '*'

"2"

Repeat for 5 calls.

Call N made on <DATE_TIME>. You called <NUMBER>, call duration is <duration>, call cost is <amount>

You are now in Your Voicemail Options menu
------
Voicemail is disabled/enabled. To enable/disable your voicemail press '1'
Autoplay is disabled/enabled. To enable/disable autoplay message press '2'
Date and time announcement is disabled/enabled. To enable/disable date and time announcement press '3'
------
To return to your Voicemail Menu press '#'
To return to the Main Menu press '*'

"1", "2", "3"

Voicemail / Autoplay / Date And Time announcement has been enabled / disabled

You are now in Your Call Forwarding Menu
Your call forwarding is:
- disabled
- set to forward when busy to <NUMBER>
- set to forward all to <NUMBER>
- set to follow-me to <N> numbers
------
To enable/manage call follow-me mode press '1'
To enable/change number for forward when busy mode press '2'
To enable/change number for forward all mode press '3'
To disable call forwarding press '4' (doesn't appear if disabled)
To return to the Main Menu press '*'

"2"

Is a single number set?

NO

YES

Please dial the number you want to add after the tone and press the '#' key

"#"

Forwarding When Busy <NUMBER> has been enabled
------
To change your forwarding number press '1'
To return to the Call Forwarding Menu press '#'
To return to the Main Menu press '*'

"1"

Is Number Empty?

NO

YES

You have entered an empty number.

"*"

To reenter the number press 1 to exit the number entrance and go back to the previous menu press #

"1"

"#"

You have entered <NUMBER> To save this number press '#' To cancel and redial press '*'

"#"

Is Previous Number set?

"NO"

"YES"

"3"

Is Any Number Set?

YES

NO

The same as "Forward When Busy"

"4"

"1"

"3"

You are now in Your Call Follow-Me menu
a) Call Follow-Me is enabled to # numbers (in random order/simultaneously)as provided.
OR
b) You have not entered any valid follow-me numbers, follow me mode will not be enabled.
------
To listen to your follow-me numbers list press '1'
To clear your follow-me numbers list press '2'
To add follow-me numbers press '3'
To change the order of your follow me numbers press '4'
To return to the Call Forwarding Menu press '#'
To return to the Main Menu press '*'

Is any number set?

"NO"

"NO"

"YES"

No numbers set

The follow-me numbers entered are number one: <NUMBER1> number two: <NUMBER2>

Enter follow-me numbers one by one

"3"

Please dial the number you want to add after the tone and press '#' key.

Is Number Empty?

NO

You have entered <NUMBER> To save this number press '#' To cancel and redial press '*'

"#"

Your number has been saved

To add another number press '1' To continue press '#'

"1"

"#"

YES

You have entered an empty number

"*"

To reenter the number press 1 to exit number entrance and go back to previous menu press #

"1"

"#"

Is numbers quantity >0?

You have entered # numbers for follow me mode
If you want these numbers to be used randomly press '1'
If you want these numbers to be used simultaneously press '2'
If you want these numbers to be used in the order you have saved them press '3'

"4"

"2"

To confirm deleting of your follow-me numbers press '#' To cancel and return to the Call Follow-Me menu press '*'

"#"

You have deleted all your follow-me numbers, call forwarding is now disabled. You can enable it again from the call forwarding menu.

Current order of your follow-me numbers is set to randomly/ simultaneously/ as provided.
------
If you want follow-me numbers to be used randomly press '1'
If you want follow-me numbers to be used simultaneously press '2'
If you want follow-me numbers to be used in the order you have provided press '3'
To return to the Call Follow-Me menu press '*'

"1"

Call waiting has been enabled/disabled

You are now in Your Preferences Menu
------
Call-waiting is disabled/enabled. To enable/disable call-waiting press '1'
Show Caller ID is disabled/enabled. To enable/disable Show Caller ID press '2'
To manage your greetings press '3'
To change your Voicebox language press '4'
To return to the Main Menu press '*'

"2"

Show caller ID has been enabled/disabled

"3"

To record a custom greeting press '#' and wait for the tone, say your greeting and press '#'. To cancel press '*'

"1"

Your new greeting is: <GREETING AUDIO> To save this greeting press '#' To record another one press '*'

"#"

Your greeting has been saved

Your greeting is set to the standard telephone/custom greeting: <GREETING AUDIO>
------
To record a new custom greeting press '1'
To set a standard greeting press '2' (doesn't appear if standard greeting set)
To return to Your Preferences Menu press '#'
To return to the Main Menu press '*'

"2"

The standard telephone greeting has been set

"4"

The <<English>> language has been set ( on selected language )

"NO"

Is PIN valid?

"YES"

"NO"

Is maximal attempts count reached?

"YES"

Sorry the PIN number you provided is not valid. Please re-enter your PIN number followed by #

You have exceeded the number of attempts available. Please try calling again later.

Stop IVR

First Login?

"NO"

"YES"

This is the first time you log in to your voicebox. Please select a language

( List of languages - each menu choice on corresponding language):
For English press '1'

You have selected the "English" language
To confirm press '#'
To cancel and return to the Languages menu press '*'
------
( The same as above but on new selected language )

"#"

The English language has been set ( on selected language)

The <English> language has been set ( on selected language)

You have selected the "English" language
To confirm press '#'
To cancel and return to the Languages menu press '*'
------
( The same as above but on new selected language )

"#"

"*"

( List of languages - each menu choice on corresponding language) For English press '1'
------
To return to Your Preferences Menu press '#'
To return to the Main Menu press '*'

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## Payment Remittance IVR