

**Porta  UM®**



# **Administrator Guide**

Maintenance Release 21

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**PortaUM Administrator Guide, September 2010  
Maintenance Release 21  
V.1.21.2**

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## Preface

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

### Where to Get the Latest Version of This Guide

The hard copy of this guide is updated at major releases only, and does not always contain the latest material on enhancements occurring between minor releases. The online copy of this guide is always up-to-date, and integrates the latest changes to the product. You can access the latest copy of this guide at [www.portaone.com/support/documentation/](http://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 performing 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 4 GB of RAM, 8 GB recommended.
- At least one USB port.

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 DVD, 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
- Browser: Internet Explorer 7.0, FireFox 3.x with JavaScript enabled.
- Spreadsheet processor (MS Excel)
- Display settings:
  - Minimum screen resolution: 1024 x 768
  - Color palette: 16 bit color (minimum)

## Installation

A jumpstart installation DVD is provided for all PortaOne products. This DVD contains 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](#).

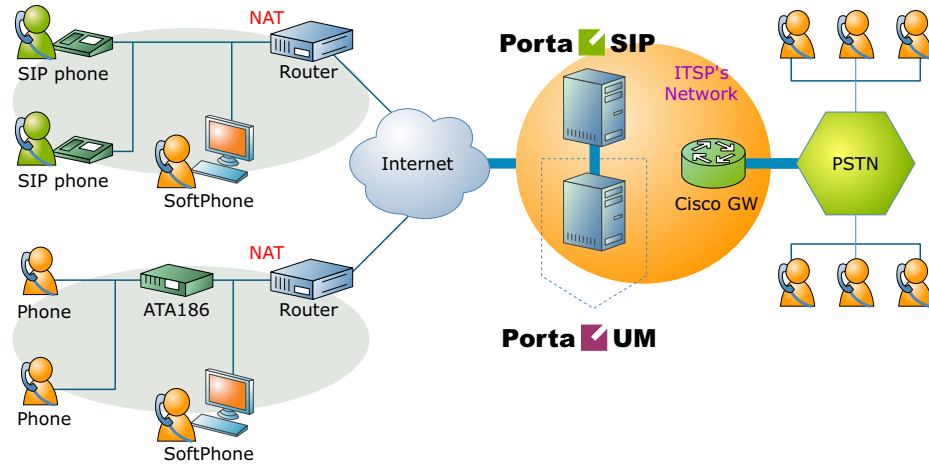
## What's New in Maintenance Release 21?

This release includes the following new features and improvements:

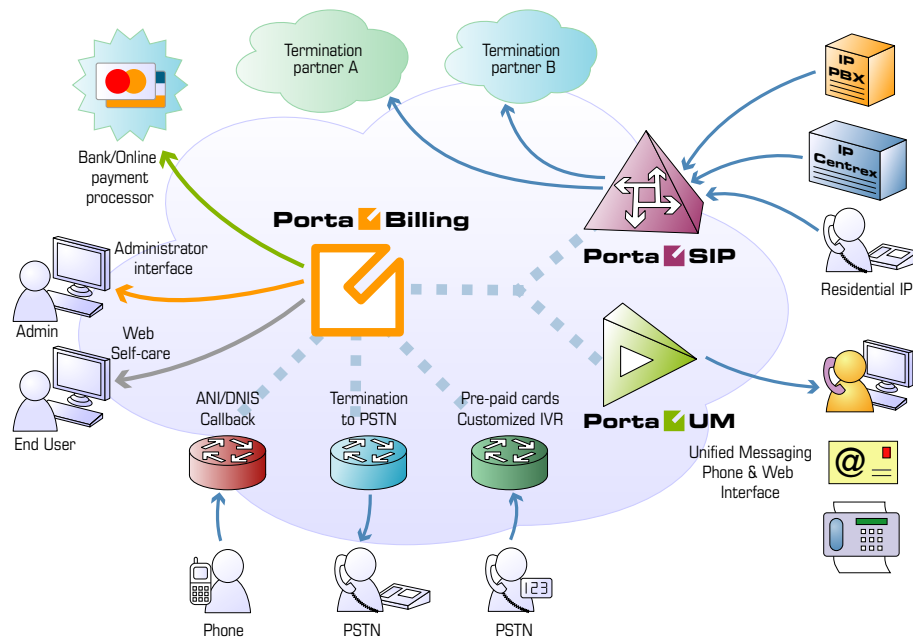
- Management of the IVR access numbers, associated applications and parameters of an individual application is now done entirely from the PortaBilling® web interface.
- PortaUM cluster is redesigned to allow easy failover and dynamic load sharing between the available servers.
- DBMAIL (<http://www.dbmail.org/>) is now used as the storage engine for all voicemail data, and the actual data is stored in MySQL database. This provides improved performance and scalability, as well as the ability to use a different database storage solution (e.g. Oracle).

# 1 . System Concepts

## PortaUM's Role in Your VoIP Network



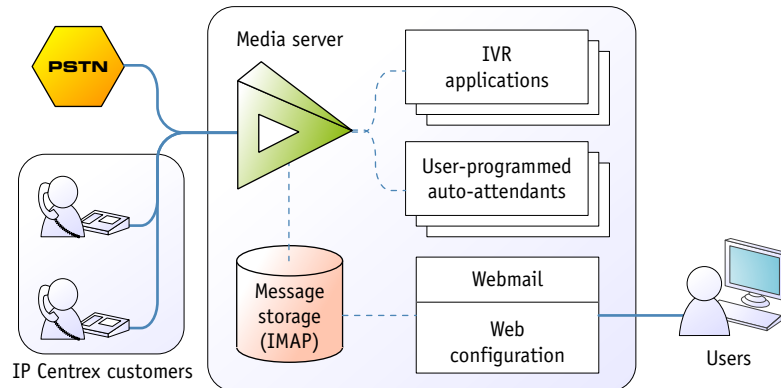
**PortaUM** (Unified Messaging system) is one of the key components of PortaSwitch, a software-based communication service and subscriber management platform that directly addresses the needs of modern communication network providers seeking new revenue streams from tools that unify voice, data and fax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



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



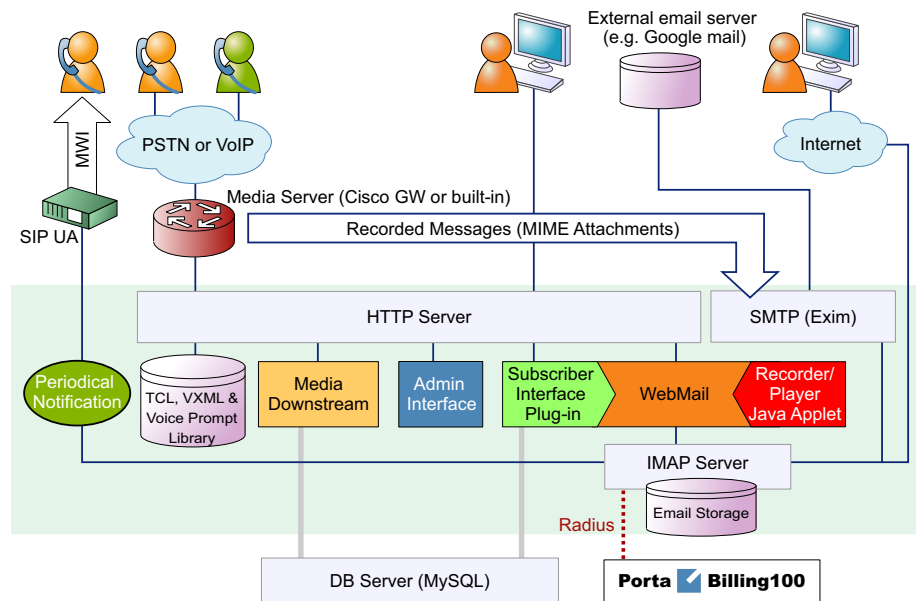
## PortaUM Components



### PortaUM consists of the following main components

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

## PortaUM Architecture



## PortaUM Performance

While for PortaBilling100 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.)

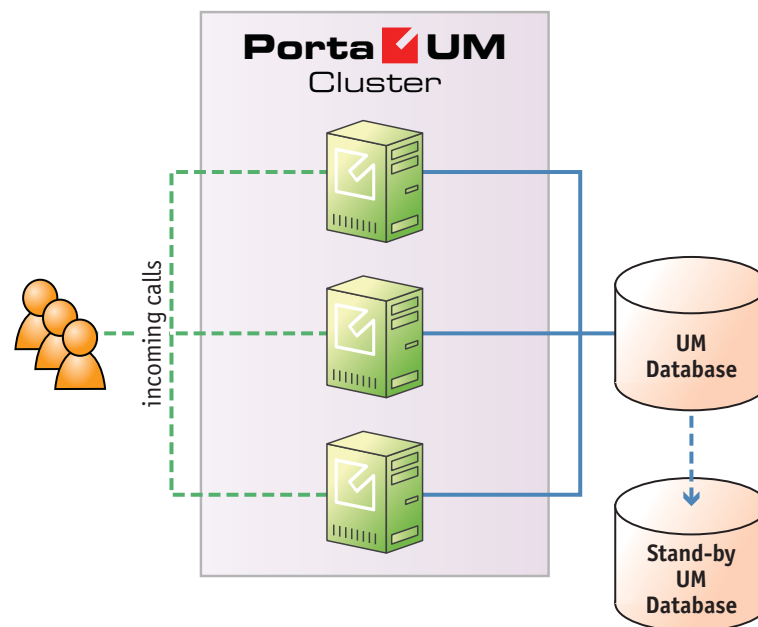
On a standard server, PortaUM can handle about 1000 incoming IVR sessions (for applications such as auto attendant or balance info), or about 600 concurrent two-way IVR sessions, such as the prepaid card or callback applications, i.e. when there is an incoming call from a user to PortaUM and an outgoing 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 and 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.

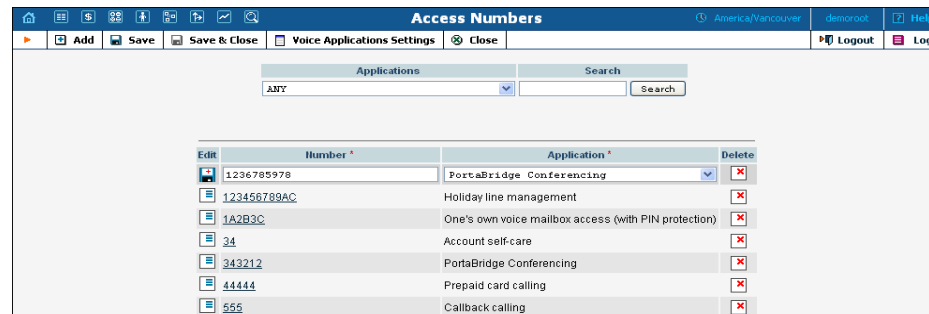
Each application normally has a set of parameters which customize its behavior (e.g., whether the prepaid calling card application should announce the available balance or not). The PortaBilling web interface is used to associate particular applications with a particular access number and then customize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

## 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.

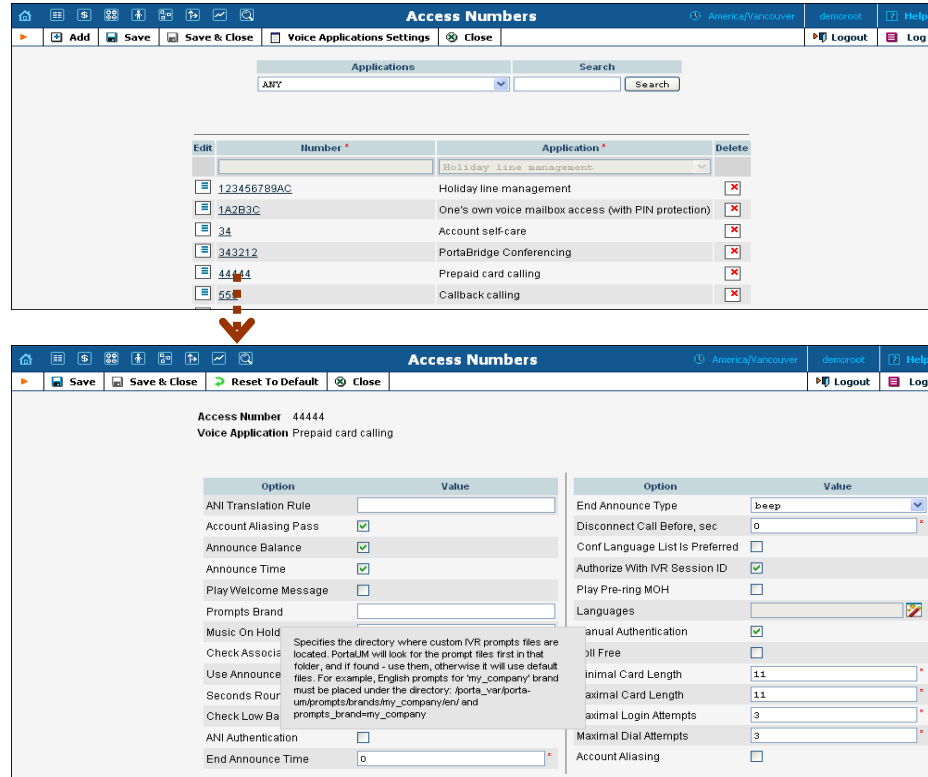
## Managing IVR applications via Web GUI

Using PortaBilling100 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.



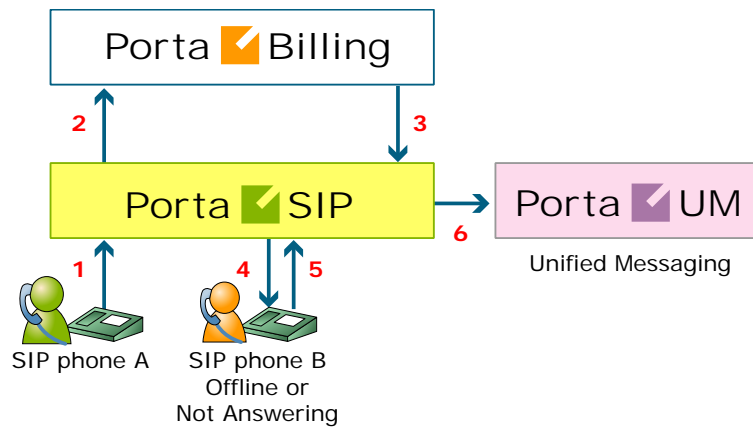
### Application options

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



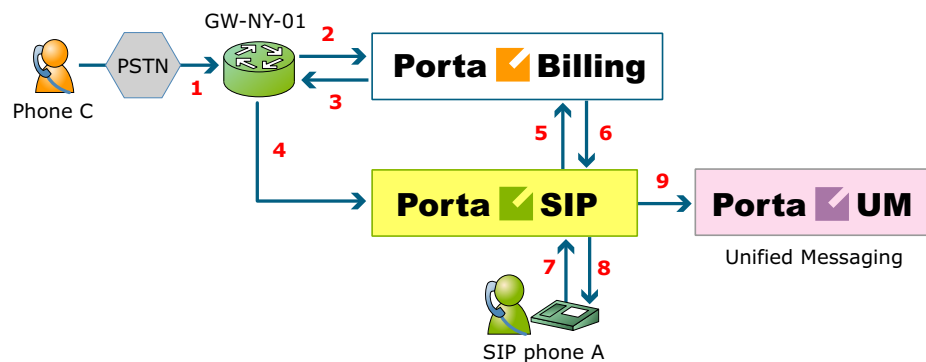
### Call Process

SIP UA <-> SIP UA



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

## SIP UA -> PSTN



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

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

- User C wishes to call user A. He dials A's phone number; since C is in the USA, he dials it in North American format, i.e. 2027810003.
- The call is routed through the telecom network to gateway GW-NY-01. When the incoming call arrives on the gateway (1), it starts a special TCL application to handle this call. This application does several things:
  - Converts the phone number to E.164 format, so that 2027810003 becomes 12027810003.

- Performs authorization in the billing (2), i.e. checks whether A is allowed to receive incoming telephony calls from GW-NY-01 and, if you charge for incoming calls, what is the maximum allowed call time based on A's current balance (3). One important point is that authorization must happen without a password check, since the application does not know the valid password for the SIP account.
- Starts the outgoing call to 12027810003.
- When the call is established, it starts the timer, and will disconnect the call if the maximum call duration is exceeded.
- The gateway is configured so that it knows that calls to 1202781.... numbers should be sent to the PortaSIP server. Thus it sends an INVITE to the PortaSIP (4).
- PortaSIP receives the INVITE, but without authorization information. Thus the PortaSIP server performs authentication based on the IP address (5), (6). Since this call comes from a trusted node, i.e. gateway GW-NY-01, it is authorized.
- The SIP server checks its registration database, but it appears that this account is not online at the moment. If A has Unified Messaging services enabled, the call will be redirected to the voicemail system, and C can leave a message for him (9). The same thing would happen if A were online, but did not answer his phone (7), (8).



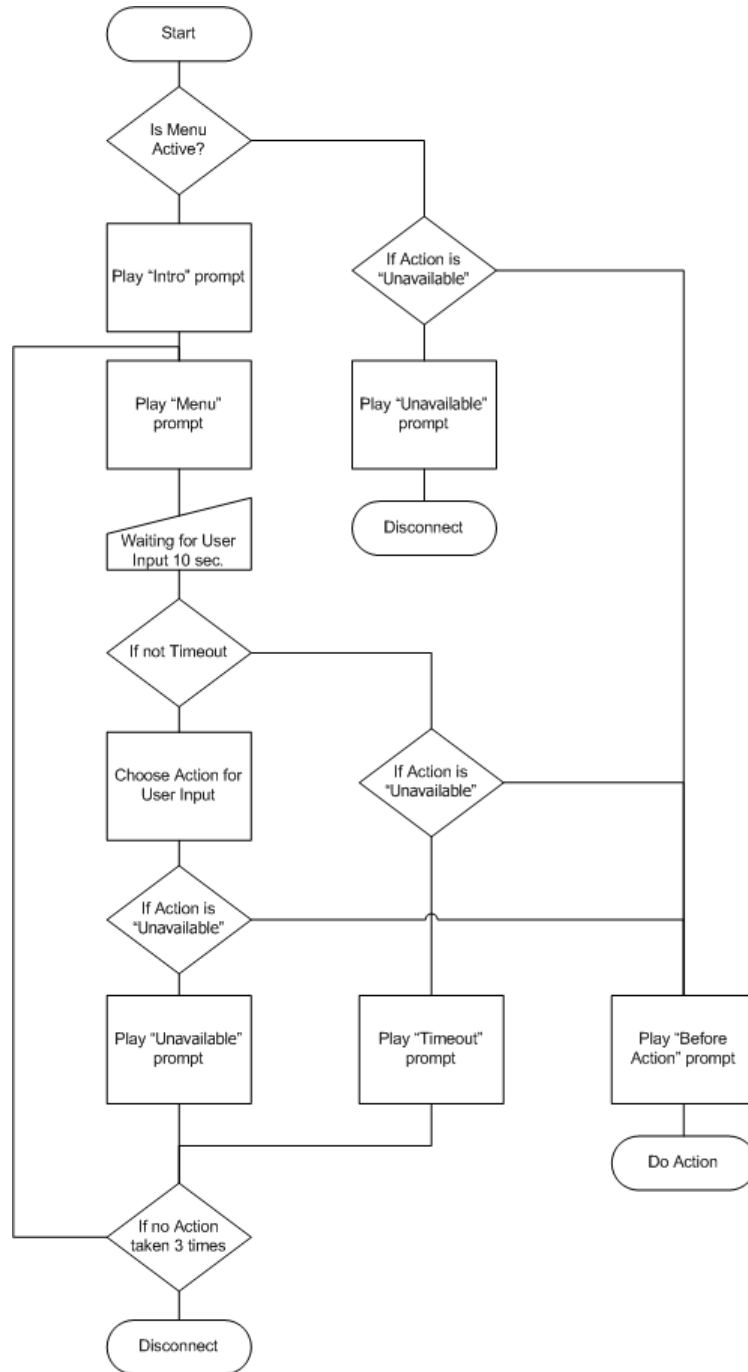
# Auto Attendant

## 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**).


- **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.
- **Menu** – Transfers the user to the selected menu.
- **Directory** – Launches this company’s dial-by-name directory.
- **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.

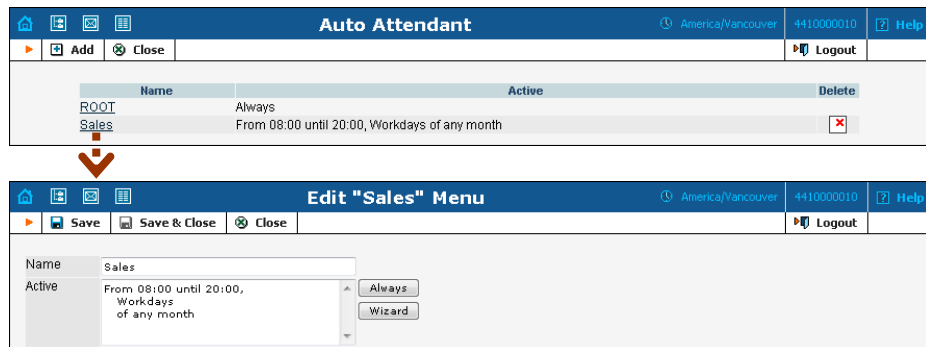
### Call Menu Flow Chart



### Menu List Mode

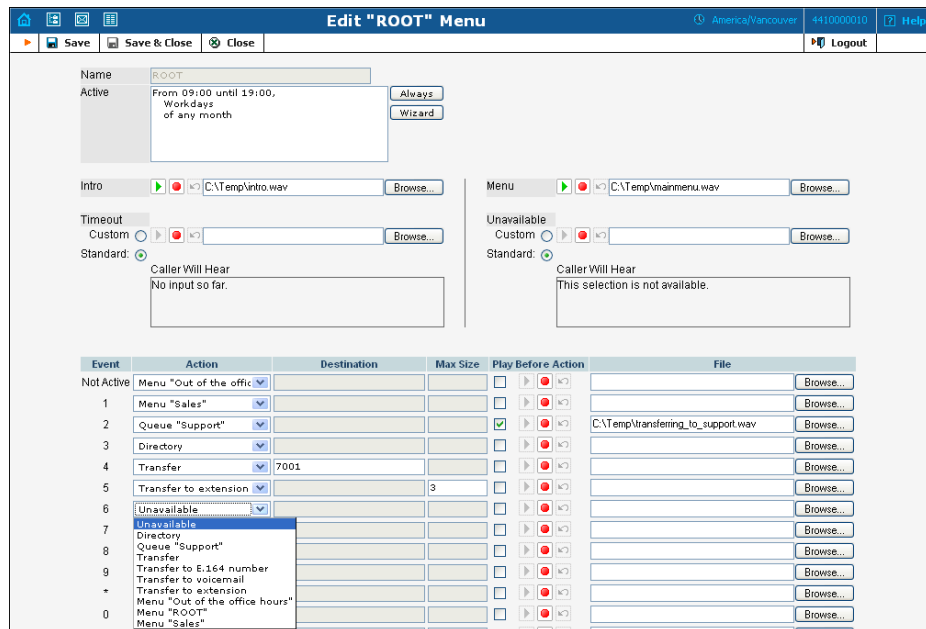
Auto attendant can be selected from the Options menu. The main screen lists all the menus available in the system.

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.

<b>.raw</b>	Raw files (no header).
<b>.wav</b>	Microsoft .WAV RIFF files.

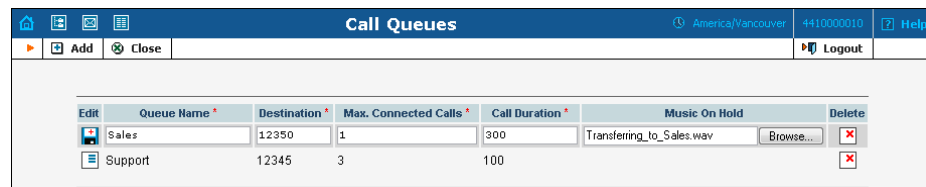
Event Table

Column	Description
<b>Event</b>	<b>Not Active</b> – When the current menu is not active (see the active period definition above). <b>0-9, #, *</b> – User selection on telephone keypad. <b>Timeout</b> – No selection received from user. <b>Fax</b> – Fax CNG tone detected.
<b>Action</b>	See the description of available actions in the <i>Basic Concept</i> section above.
<b>Play before action</b>	Check this box if the corresponding prompt is to be played before an action is performed.
<b>File</b>	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.

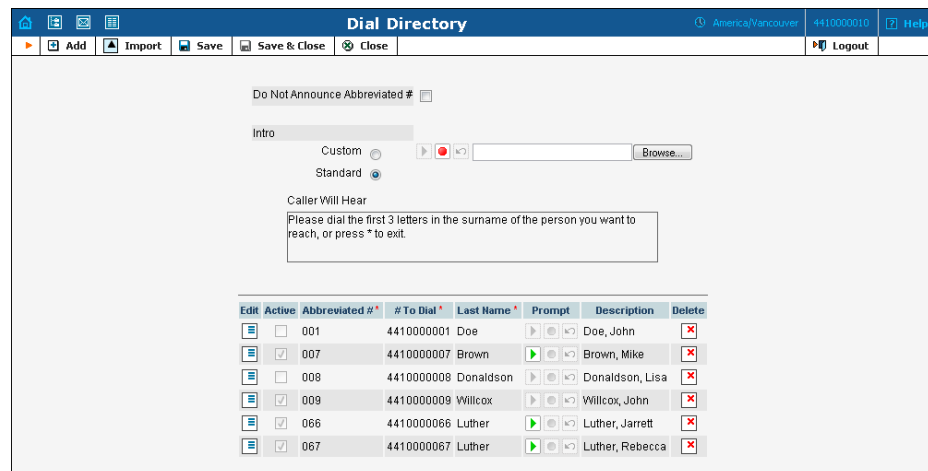
- **Call Duration** – The average expected processing time for each call (used to calculate the estimated waiting time).
- **Music on Hold** – A melody (or announcement) which is played to users waiting to be connected.

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.



The screenshot shows the 'Dial Directory' web interface. At the top, there is a navigation bar with 'Add', 'Import', 'Save', 'Save & Close', and 'Close' buttons. Below this, there are options for 'Do Not Announce Abbreviated #' and 'Intro' (Custom and Standard). A text box for 'Caller Will Hear' contains the message: 'Please dial the first 3 letters in the surname of the person you want to reach, or press \* to exit.' Below the text box is a table of contacts.

Edit	Active	Abbreviated #*	# To Dial*	Last Name *	Prompt	Description	Delete
<input type="checkbox"/>	<input type="checkbox"/>	001	4410000001	Doe		Doe, John	
<input type="checkbox"/>	<input checked="" type="checkbox"/>	007	4410000007	Brown		Brown, Mike	
<input type="checkbox"/>	<input type="checkbox"/>	008	4410000008	Donaldson		Donaldson, Lisa	
<input type="checkbox"/>	<input checked="" type="checkbox"/>	009	4410000009	Willcox		Willcox, John	
<input type="checkbox"/>	<input checked="" type="checkbox"/>	066	4410000066	Luther		Luther, Jarrett	
<input type="checkbox"/>	<input checked="" type="checkbox"/>	067	4410000067	Luther		Luther, Rebecca	



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").

## 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.

### 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. There is also an IVR menu enabling the use of vouchers to add funds to an account.

### Prepaid card IVR

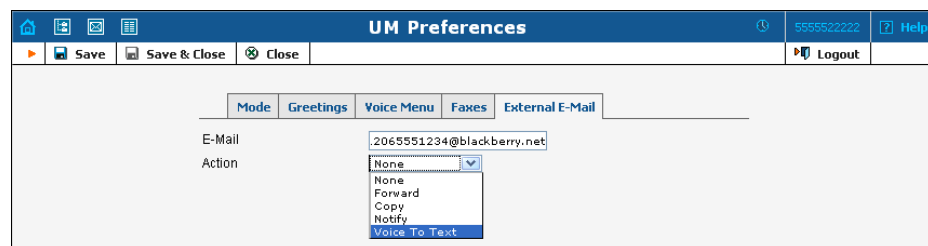
PortaUM includes an IVR system that allows you to offer a prepaid card service for incoming VoIP calls. This IVR supports all the PortaBilling extensions for the prepaid card application (access codes, real and announced call duration, and so on).

### 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.

## Supported Languages

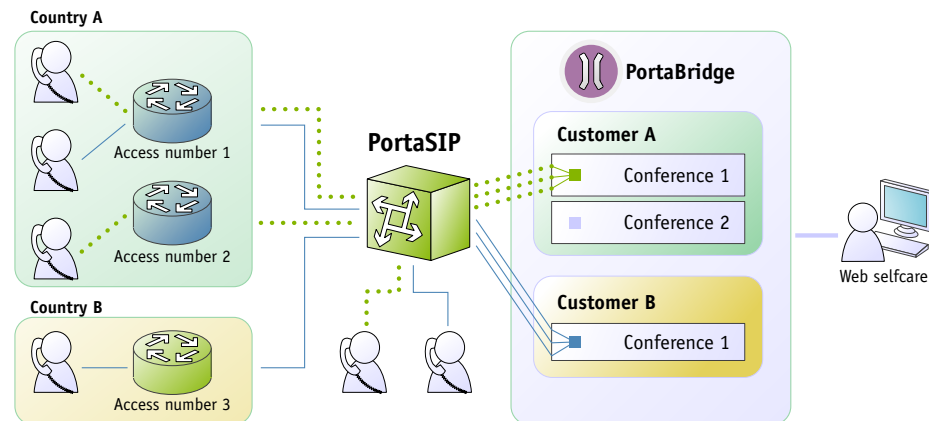
The following table shows which languages are supported in PortaUM applications and web interface.

<b>LANGUAGE</b>	<b>English (en)</b>	<b>Spanish (es)</b>	<b>French (fr)</b>	<b>Swiss French (fr-CH)</b>	<b>Hebrew (he)</b>	<b>Brazilian Portuguese (pt-BR)</b>	<b>Russian (ru)</b>	<b>Cantonese Chinese (yu)</b>
<b>APPLICATION</b>								
<b>Web Interface</b>	Yes	No	Yes	No	Yes	No	Yes	No
<b>Voicemail</b>	Yes	Yes	Yes	No	Yes	No	No	No
<b>Calling Card</b>	Yes	Yes	Yes	Yes	Yes	Yes	No	Yes
<b>Attendant Features</b>	Yes	Yes	Yes	No	Yes	No	Yes	No
<b>Account Self-care</b>	Yes	No	Yes	No	Yes	No	No	No
<b>Payment by Credit Card</b>	Yes	No	No	No	No	No	No	No

# 2. PortaBridge

## PortaBridge Architecture

As part of the PortaUM solution, PortaBridge enables your customers to use voice-conferencing services. Multiple customers can use this service simultaneously on a single PortaBridge server. A customer account with this service enabled has access to the PortaBridge self-care interface, where multiple conferences can be managed. 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 PortaBridge 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.

PortaBridge carries out intensive calculations when mixing the audio from several meeting participants, and so requires significant amounts of CPU power. This is why it should be installed on a dedicated server.

## PortaBridge and Codec Licenses

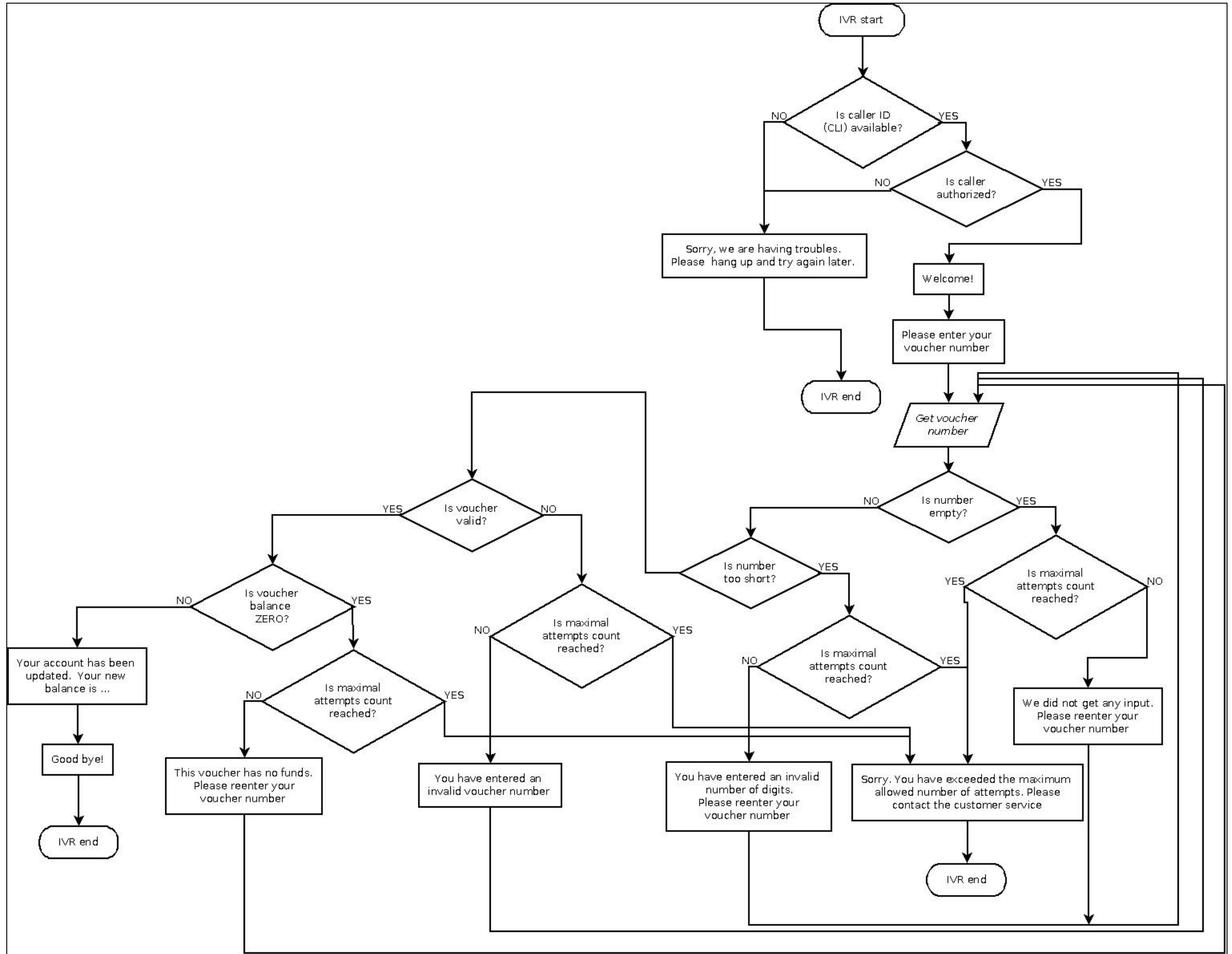
Normally, a media server (such as the one used in PortaUM) will send pre-converted voice prompts to the user as a byte stream, so that no codec licenses are involved. In the case of PortaBridge, however, the audio-stream with the voice of each participant has to be decoded in real time, and then sent 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 PortaBridge license **does not include any codec licenses**, and customers should procure such licenses independently.

# 3. IVR Flow Diagrams

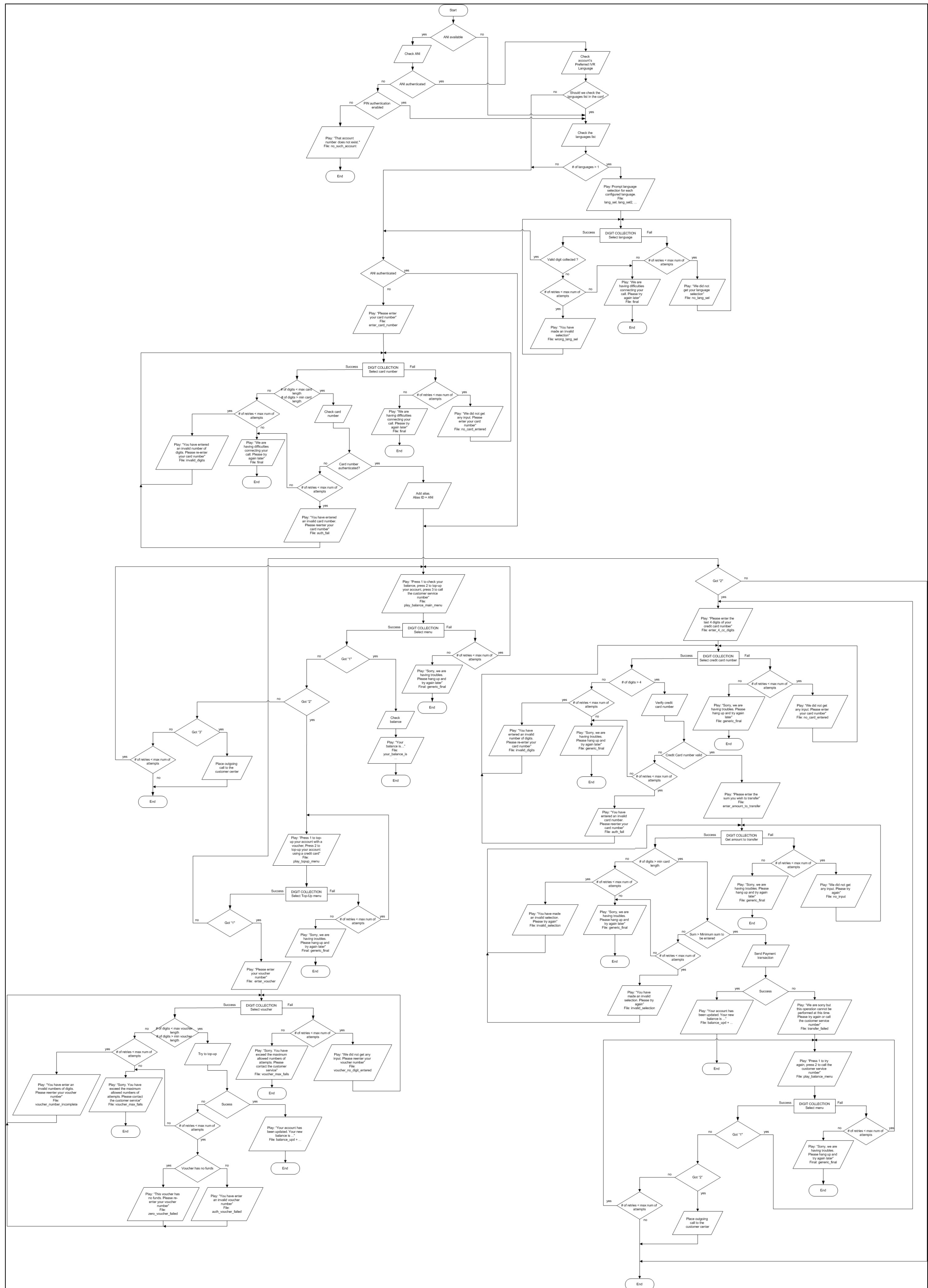




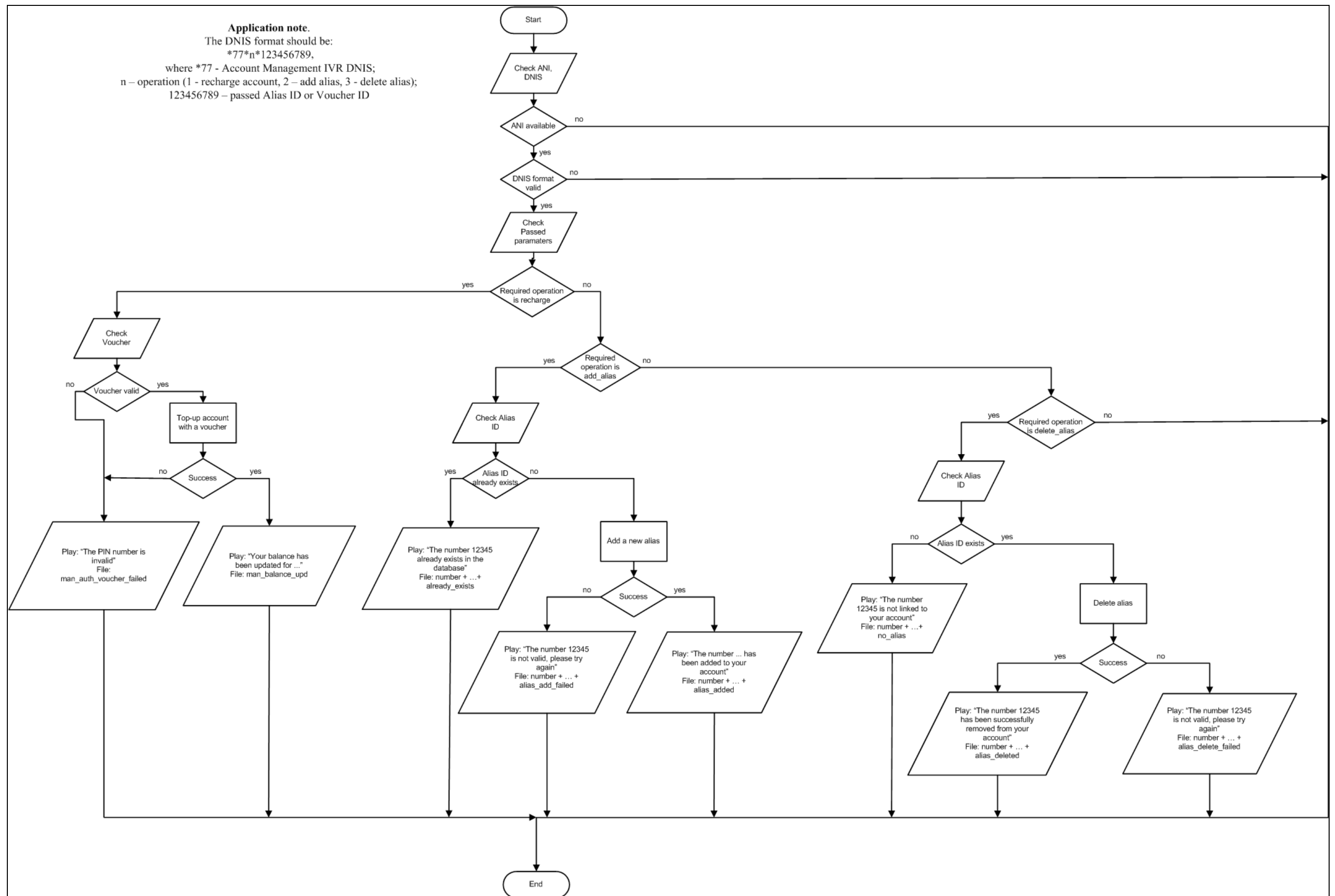
### Voucher Recharge IVR



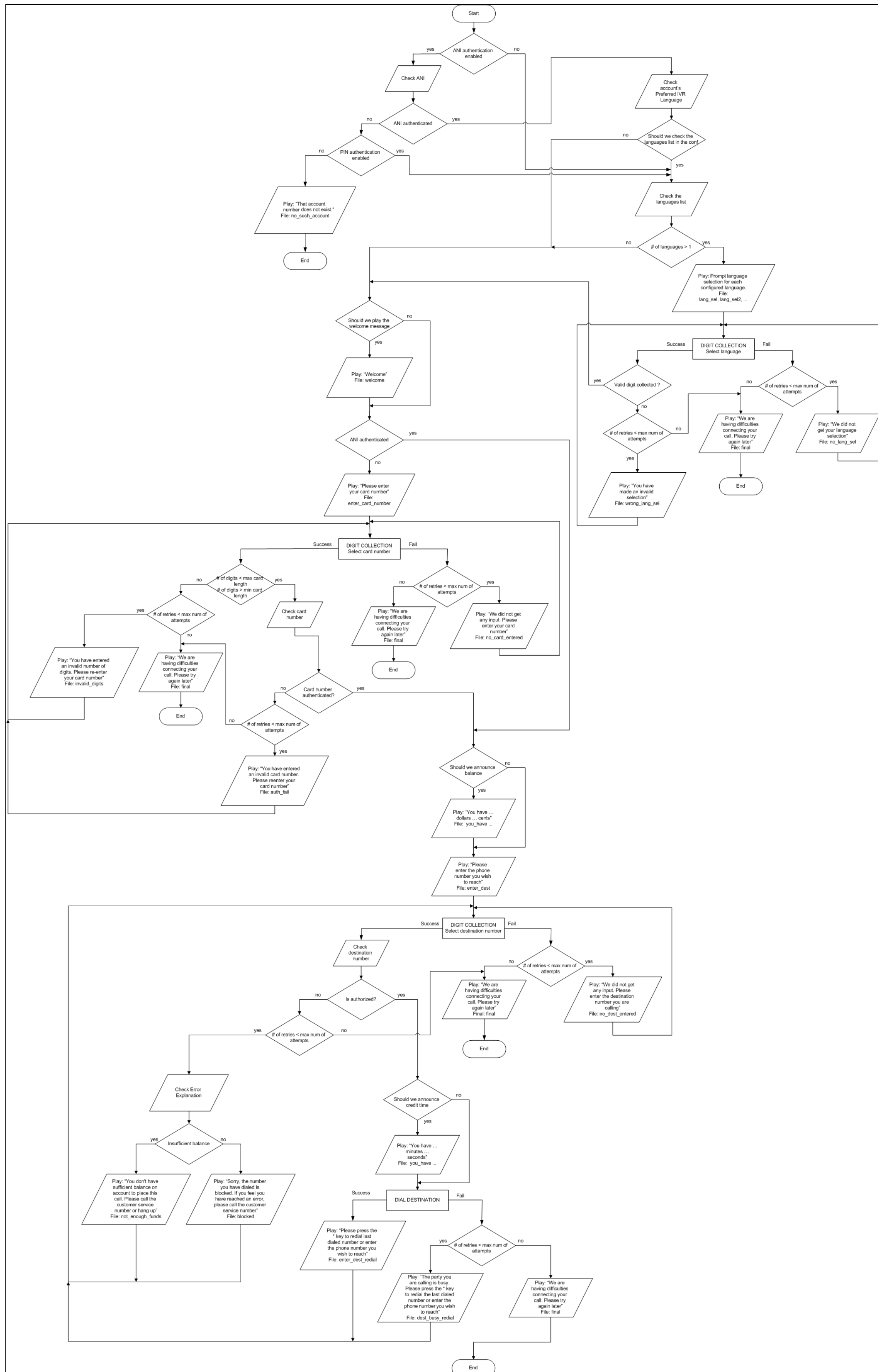
# Pay by Credit Card IVR



# Account Management IVR



# Prepaid Card IVR





### Callback with Registration IVR

