

Porta  **UM**
unified messaging



Administrator Guide
Maintenance Release 20

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**PortaUM Administrator Guide
V.1.20.3, September 2009**

Please address your comments and suggestions to: Sales Department,
PortaOne, Inc. Suite #408, 2963 Glen Drive, Coquitlam BC V3B 2P7
Canada.

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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/

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.

Hardware and Software Requirements



PortaUM can use its own built-in media server, or it can make use of an external Cisco gateway as a media server. Should you decide to use a Cisco gateway for this purpose, please note that this requires a *dedicated* Cisco AS 5300/5350, i.e. you cannot run other applications on it (such as wholesale termination or calling cards). No Cisco gateway is required if you wish to run the IVR embedded into PortaUM.

Cisco Requirements

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

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

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

Client System Recommendations

- OS: Windows 2000/XP/Vista, UNIX or Mac OS X
- Java Virtual Machine Support (JRE 1.4.2_01 or later)
- Browser: Internet Explorer 7.x/8.x or Mozilla Firefox 3.x with JavaScript and cookies enabled.
- Display settings:
 - Min Screen Resolution: 1024 x 768
 - Color Palette: 16 bit color (minimum)

NOTE: To view downloaded CDR files in Windows, proceed as follows in order to match the PortaBilling default list separator: My Computer -> Control Panel -> Regional Settings -> Number -> List Separator type “,”.

Installation

PortaUM is provided via a jump-start installation CD. This CD contains installation media for FreeBSD (6.3-stable branch, with the latest security bug fixes), supplementary packages necessary for convenient system administration and maintenance, and PortaUM software packages.

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

For detailed installation instructions, please refer to the [PortaUM Installation Guide](#).

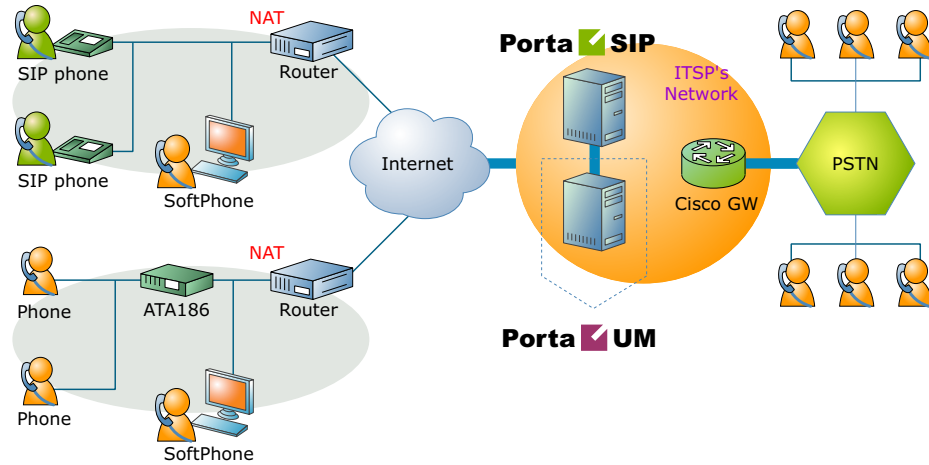
What's New in Maintenance Release 20?

This release includes the following new features and improvements:

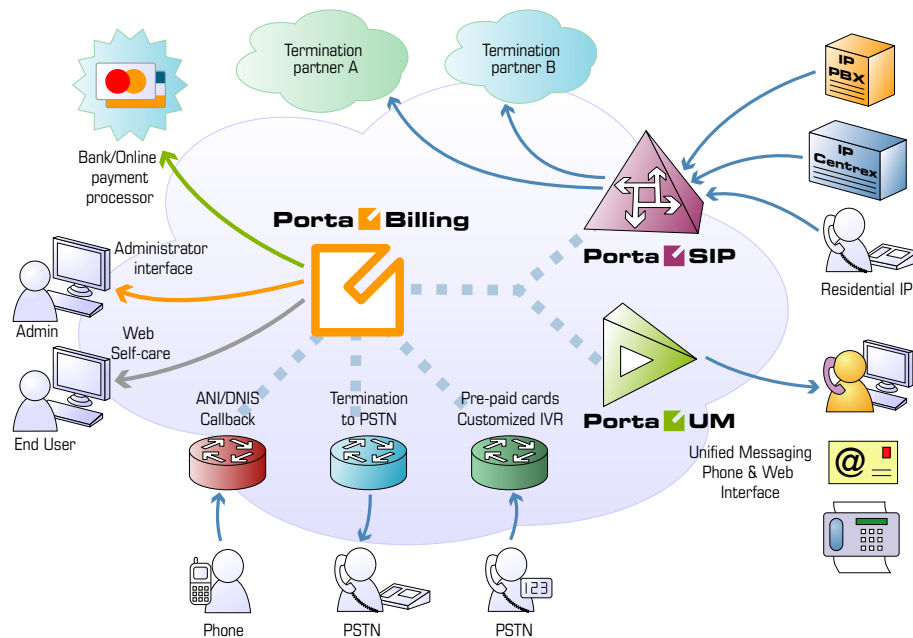
- In order to allow ITSPs using PortaSwitch to expand their customer base even further, PortaUM now offers language support for French (for all IVR applications and the web interface) and Brazilian Portuguese (for applications such as calling card and balance recharge).
- There is a new IVR application, which allows to submit a payment to an account using a credit card.

1 . System Concepts

PortaUM's Role in Your VoIP Network

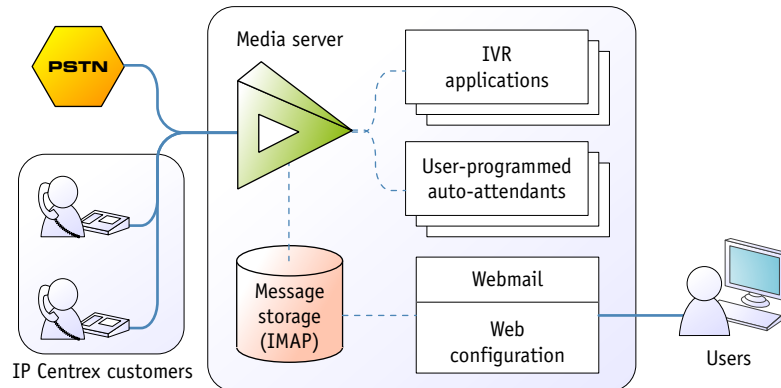


PortaUM (Unified Messaging system) is one of the key components of PortaSwitch, a software-based communication service and subscriber management platform that directly addresses the needs of modern communication network providers seeking new revenue streams from tools that unify voice, data and fax traffic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

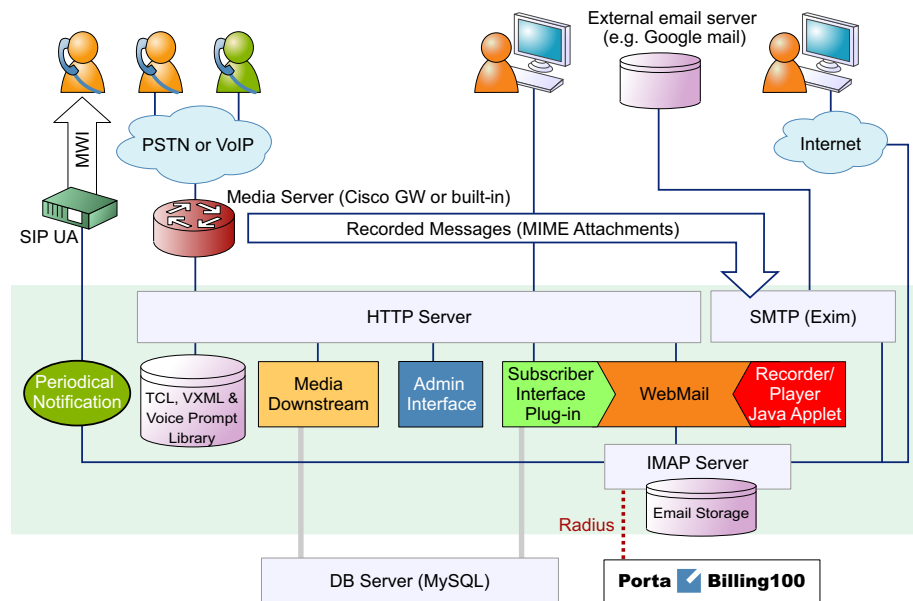
PortaUM Components



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.)

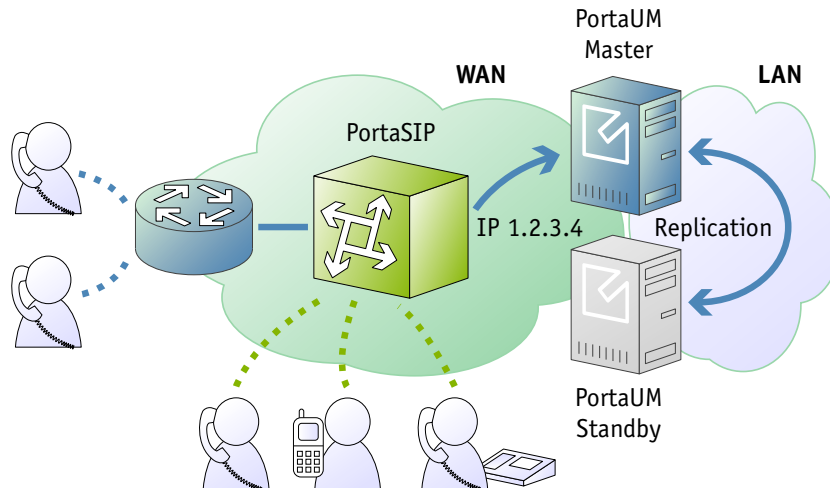
On a standard server, PortaUM can handle about 500 incoming IVR sessions (for applications such as auto attendant or balance info), or about 250 concurrent two-way IVR sessions, such as the prepaid card or callback applications, i.e. when there is an incoming call from a user to PortaUM and an outgoing 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.

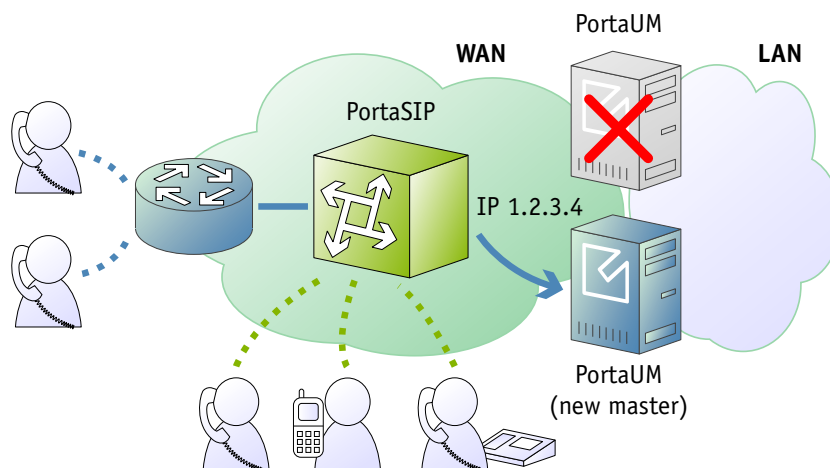
PortaUM Clustering

PortaUM Stand-by

To enable quick service recovery on the PortaUM server in case of hardware problems, you can set up a second physical server to act as a stand-by.



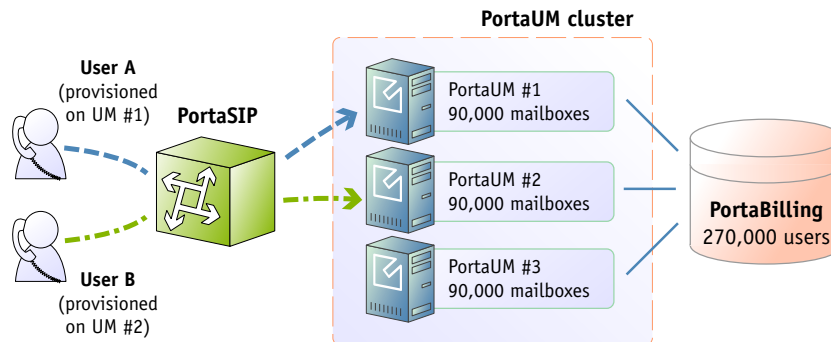
Connected via a private LAN to the primary PortaUM server, the stand-by server constantly replicates all changes made (new messages, user preferences, auto attendant configuration, etc.). If the main server goes down due to a hardware problem, the stand-by server can simply take over the primary server's IP address and begin operating as a new primary.



This ensures primary/stand-by switchover without the need for any network, IP phone or PortaSwitch reconfiguration.

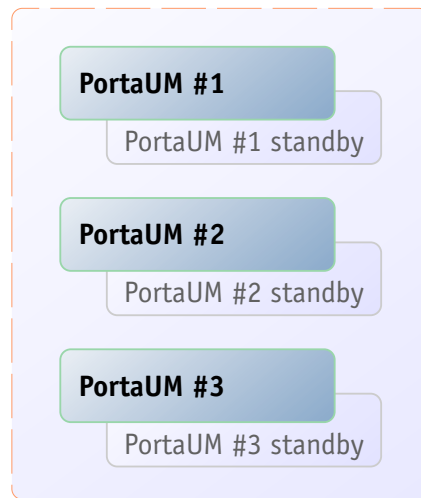
Distributed PortaUM

Although the PortaUM license does not limit the number of mailboxes or IVR sessions, there are some limitations based on CPU processing power, disk space, or other hardware parameters. Should your business require more UM mailboxes or concurrent IVR sessions than a single PortaUM server can handle, you may use several PortaUM servers to distribute your customers or IVR calls between them.



When a cluster of PortaUM servers is deployed, and the UM service has been enabled for a certain account, PortaBilling100 will automatically provision the user's mailbox on one of the available PortaUM servers. (The administrator can override these settings, if necessary, and manually assign a mailbox on a specific server to a given account.) When a call to this account is to be forwarded to the mailbox, or when the account owner dials a PortaUM self-care IVR, the call is automatically forwarded to the specific PortaUM server handling this account. For services which do not require association with a specific mailbox (e.g. prepaid card IVR), and so can be handled by any available server, incoming calls are forwarded to PortaUM servers in the cluster in a "round-robin" fashion. This ensures both load-sharing between servers (each server receives only a certain portion of all calls) and call failover (if one server is down, calls will go to the others).

PortaUM cluster



Of course, you can combine the distributed and standby approaches by creating a PortaUM cluster where the total load is spread among several PortaUM servers, each of which has its own standby server for failover purposes.

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.

IVR Scenarios

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;

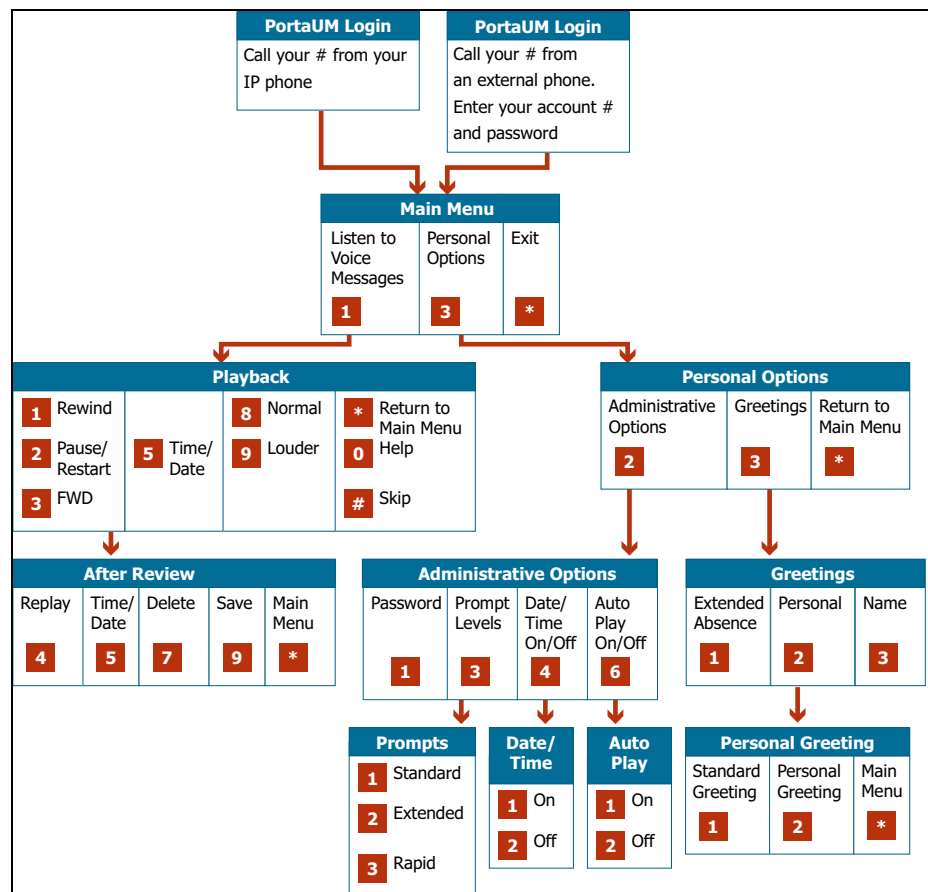
- Prepaid card.

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.

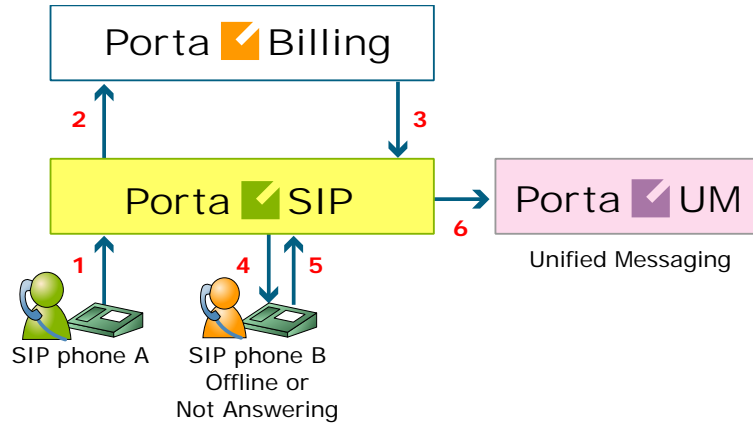
IVR Path Diagram

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



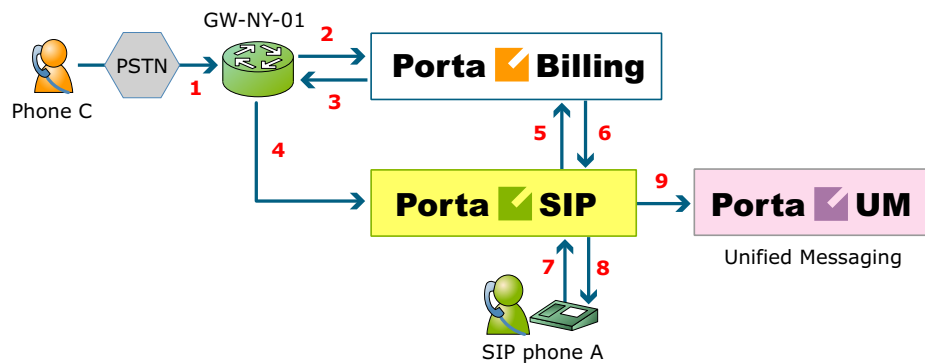
Call Process

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).

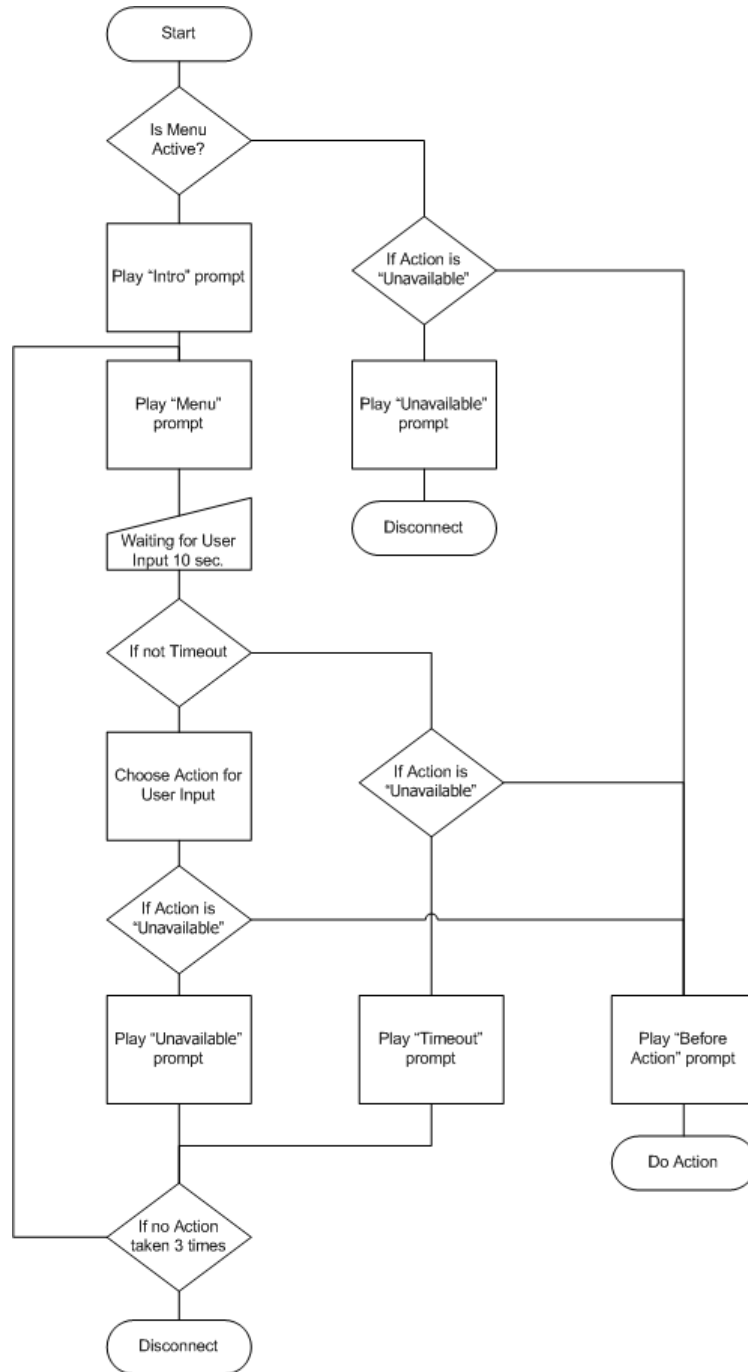
Auto Attendant

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.

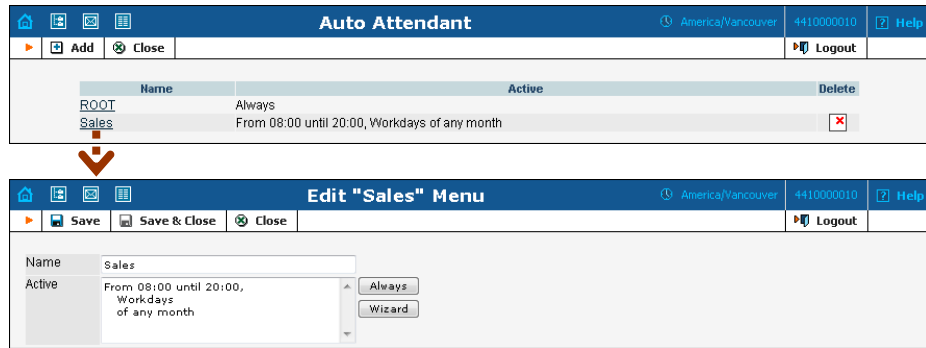
Call Menu Flow Chart



Menu List Mode

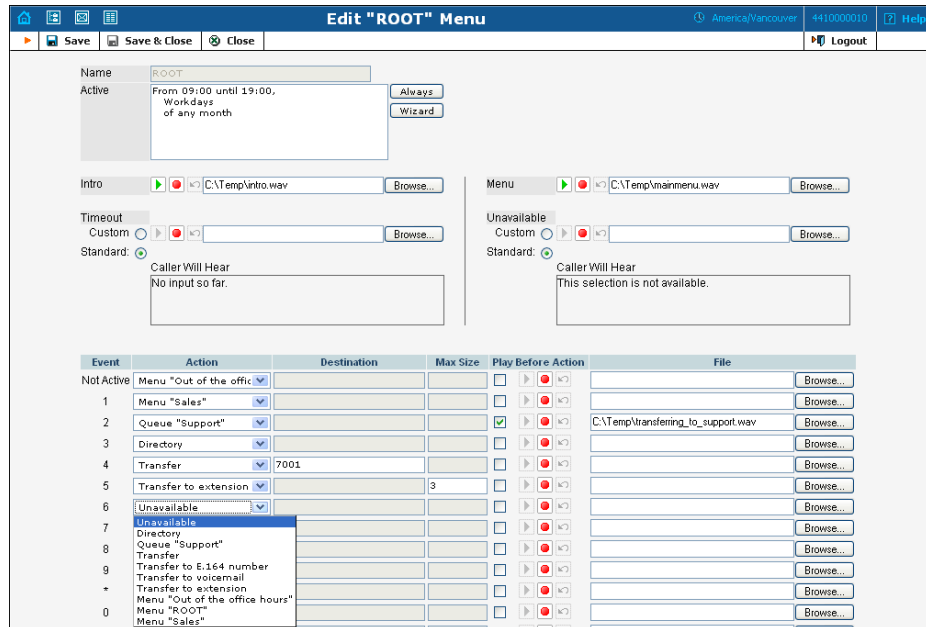
Auto attendant can be selected from the Options menu. The main screen lists all the menus available in the system.

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.

.raw	Raw files (no header).
.wav	Microsoft .WAV RIFF files.

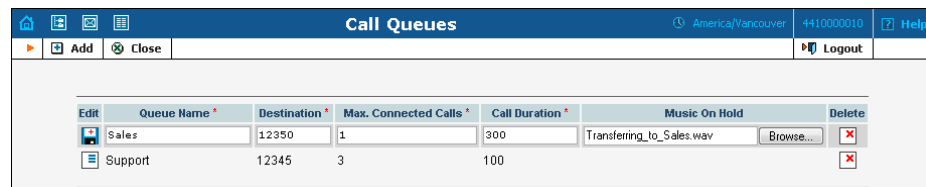
Event Table

Column	Description
Event	Not Active – When the current menu is not active (see the active period definition above). 0-9, #, * – User selection on telephone keypad. Timeout – No selection received from user. Fax – Fax CNG tone detected.
Action	See the description of available actions in the <i>Basic Concept</i> 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.

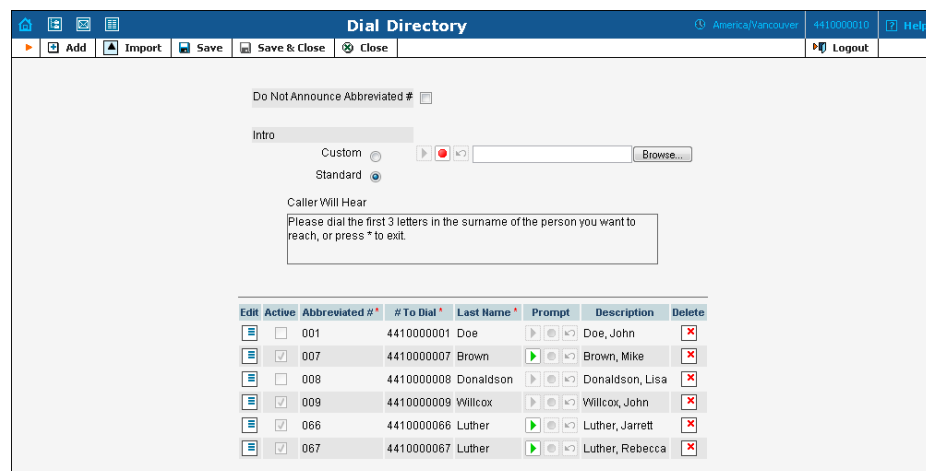
- **Call Duration** – The average expected processing time for each call (used to calculate the estimated waiting time).
- **Music on Hold** – A melody (or announcement) which is played to users waiting to be connected.

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.



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

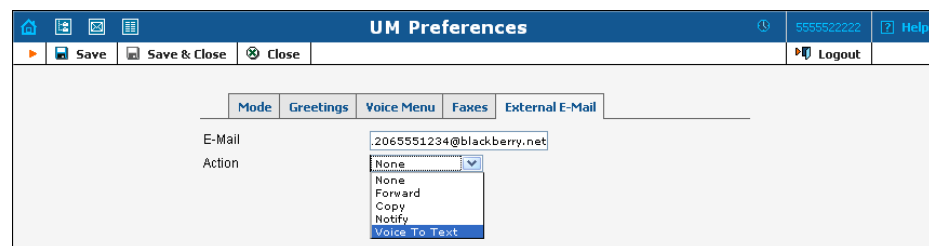
PortaUM includes an IVR system that allows you to offer a prepaid card service for incoming VoIP calls. This IVR supports all the PortaBilling extensions for the prepaid card application (access codes, real and announced call duration, and so on).

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

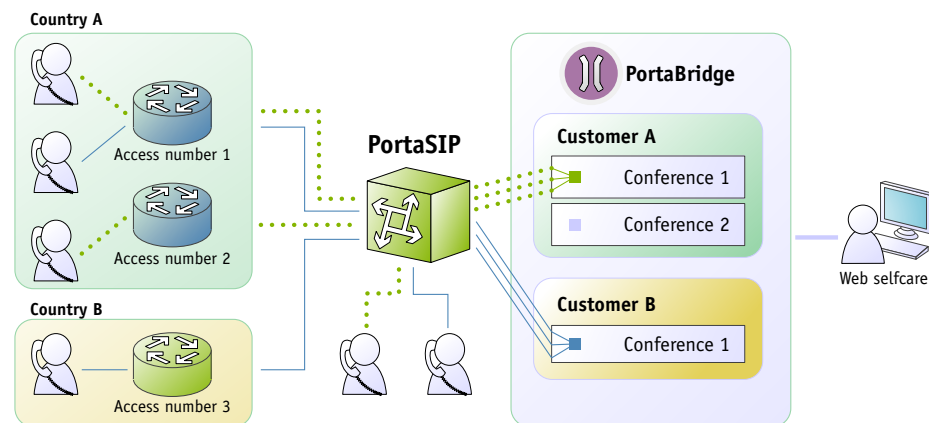
The following table shows which languages are supported in PortaUM applications and web interface.

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

PortaUM Configuration

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

porta-um.conf

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

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

Variable	Description
Global	
DefaultPassword	The default voicemail password for new accounts. Initially 777, but can later be changed by customers on their self-care interface.
DefaultOffset	15
TmpDir	PortaUM system temp directory; default: /var/tmp
AttachDir	PortaUM system upload directory; default: /var/tmp/upload
FilePath	/home/porta-um/apache
PasswdRecoveryURL	https://193.28.87.35:8445/forgot.html
DefaultAlwaysPassword	Default: no.
IMAP	
NewuserQuota	Maximum allowed size of mailbox; assigned to all mailboxes when they are created.
Web	
BrandPane	The BrandPane option places the contents of a given file at the top of the PortaUM web interface. If, for example,

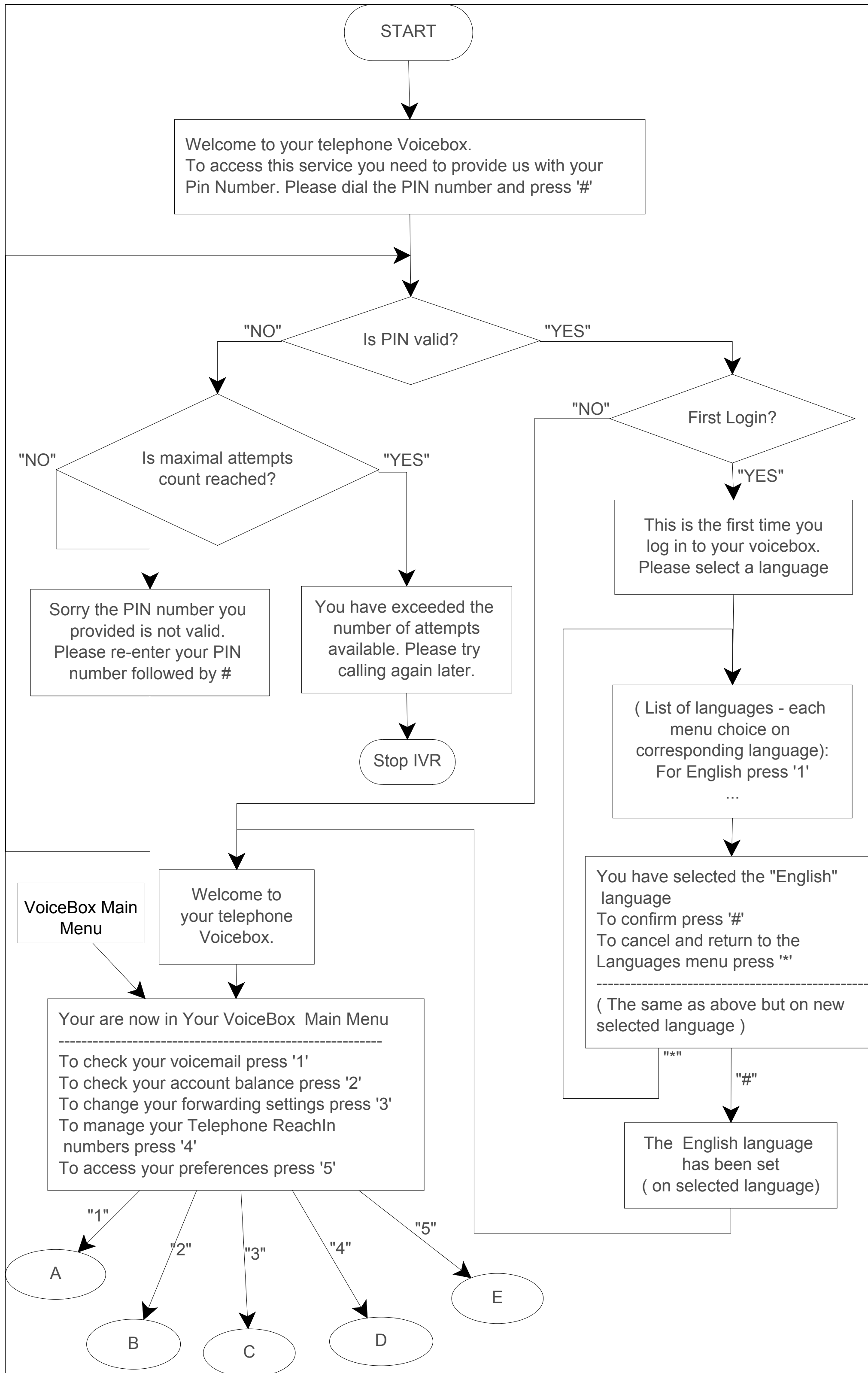
	you choose “BrandPane=/brandpane.html”, then this file must be placed under the web server document root (e.g. /home/porta-um/SquirrelMail/brandpane.html).
JavaPolicyInstallerURL	http://www.portaone.com/resources/PortaUM_Client_1.0.exe
RechargeVoucher	
MaxVoucherLen	Maximum length of voucher number; default: 11.
MinVoucherLen	Minimum length of voucher number; default: 11.
DebitCard	
MaxCardLen	Maximum length of card number; default: 11.
MinCardLen	Minimum length of card number; default: 11.
AniAuth	Is ANI authentication enabled? Default: no.
ManualAuth	Is manual authentication enabled? Default: yes.
MaxLoginAttempts	Maximum number of attempts to enter a card number; default: 3.
MaxDialAttempts	Maximum number of dial attempts within one session; default: 3.
TollFreeAllowed	If this parameter is set to ‘false’, then the Debit Card application will exit immediately if the account has insufficient funds. Default: false.
Ext<N>	Description of extensions used for the prepaid card application. These can contain several lines in the following format: Ext<N>=<i_env>:<extension>:<ANI translation rule> Where: N - arbitrary unique number i_env - ID of the environment extension - extension for the prepaid card application ANI translation rule – a regular expression (used only if AniAuth=yes) Example: Ext1=1:1234567891:s/^8.*(.)\$/8888888888\$1/;
Voicemail	
MinServers	200 Daemon

MaxServers	300	options for Cisco-less setup
MinSpareServers	20	
MaxSpareServers	30	
MinDuration	Minimum length of voice message, in seconds. If the voice message is shorter or equal to the MinDuration value, the message will be canceled. The default is 3 seconds. A value equal to '0' will disable this check.	
InitialPause	Initial delay in seconds before first playout; default: no pause.	
SilenceDetection	Enable silence detection during voicemail recording; default: on.	
SilenceSeconds	How many seconds of silence will result in canceling of voicemail recording; default: 2.	
FaxDetection	Enable fax detection during voicemail recording; default: on.	
MailTemplate	Mail message template. Specifies the file that will be used as the body template for voicemail and fax messages. Messages will have no body if MailTemplate is not set. Default: no message body. Example: MailTemplate=/home/porta-um/etc/mail-template.txt	
CleanupPeriodINBOX	Cleanup of user's Inbox folder. All messages older than the selected value (in days) will be moved to the Trash folder. A zero value disables cleanup. Default: 30.	
CleanupPeriodTrash	Cleanup of user's Trash folder. All messages older than the selected value (in days) will be permanently deleted. A zero value disables cleanup. Default: 30.	
DefaultPromptTimeout	Default timeout for all voice prompts, in milliseconds; default: 9000.	
RecordTimeout	Default timeout for recording voicemails and greetings, in milliseconds; default: 200000.	
EnvMap		
Env<N>	Parameter names (the part to the left of the equal sign) must contain sequential numbers, e.g. Env1, followed by Env2, Env3, and so on. These numbers are not the same as i_env in PortaBilling, which is specified inside the parameter's value.	

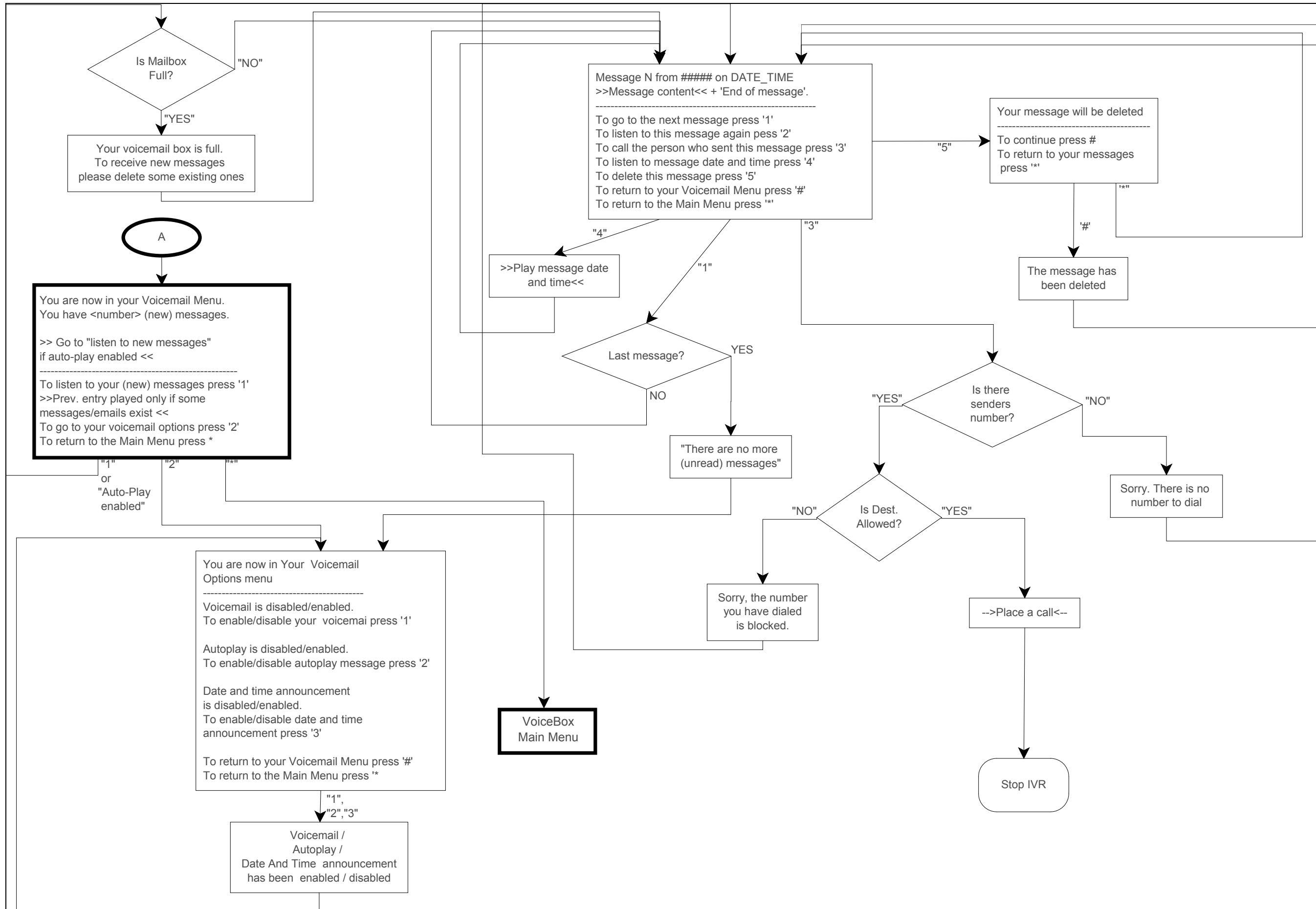
	<p>Parameter values are colon-separated strings in the following format:</p> <pre>Env<N>=<i_env>:<mail_domain>: <SIP_server_IP>:<DNIS_MailBOX>: <DNIS_MailBoxWithPass>: <DNIS_PlayBalance>: <DNIS_RechargeVoucher></pre> <p>Example:</p> <pre>Env1=3:um.yourcompany.com:porta- sip:*98:1234567890:*11:*12 Env2=4:um.othercompany.com:porta- sip2:*98:18001234567:*11:*12</pre>
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4. IVR Flow Diagrams

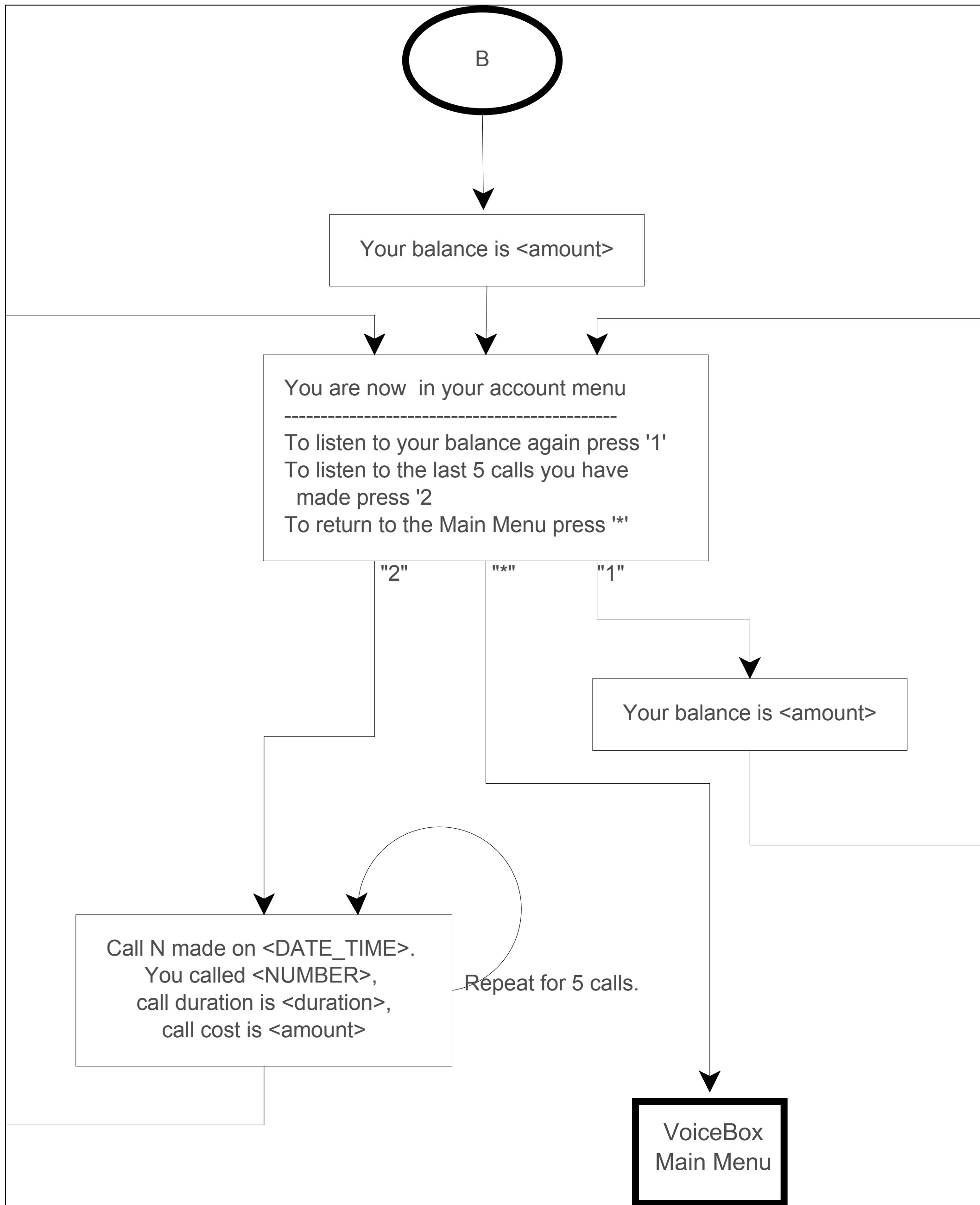
Personalization IVR



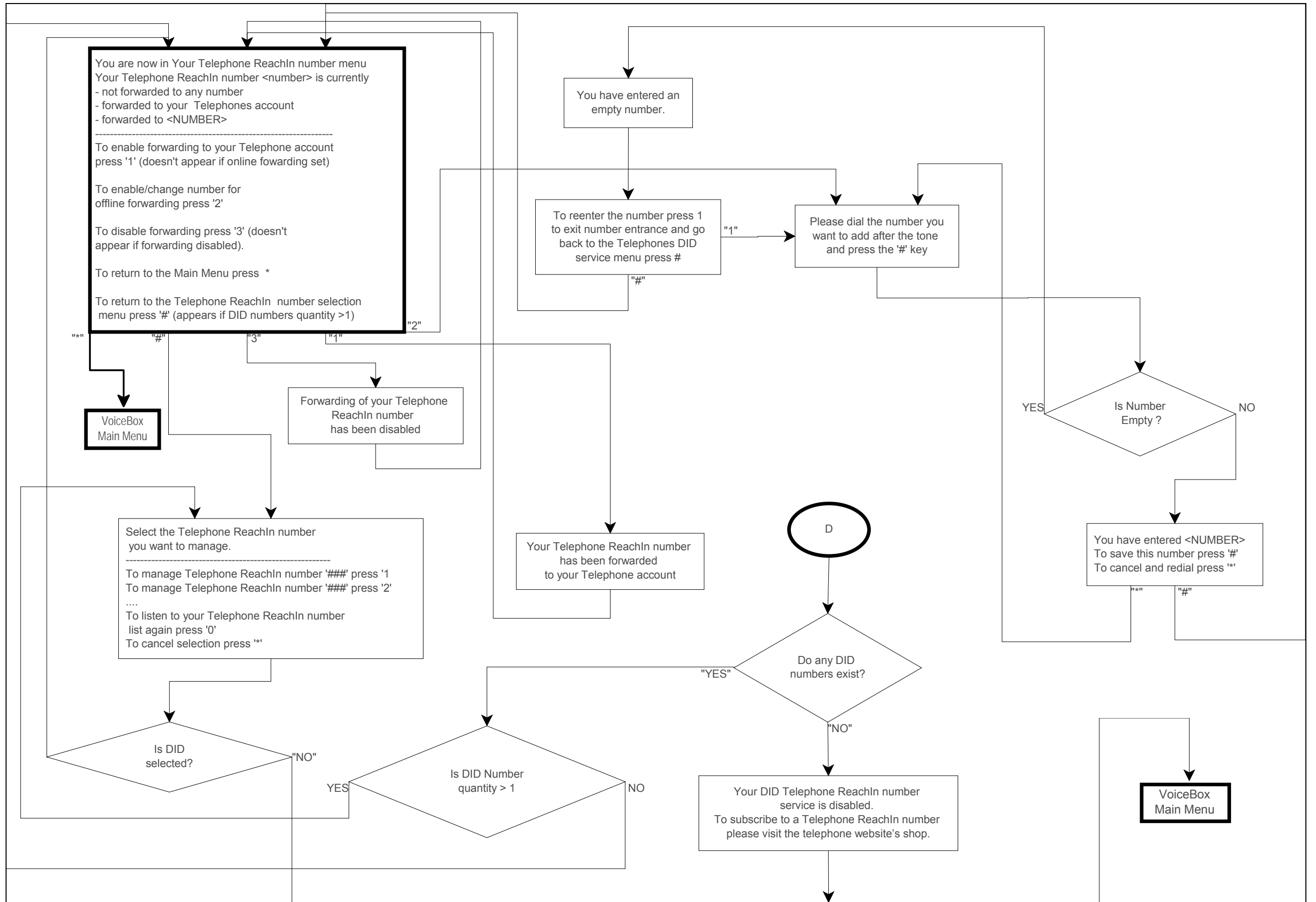
A Sub-diagram – Checking Voicemail



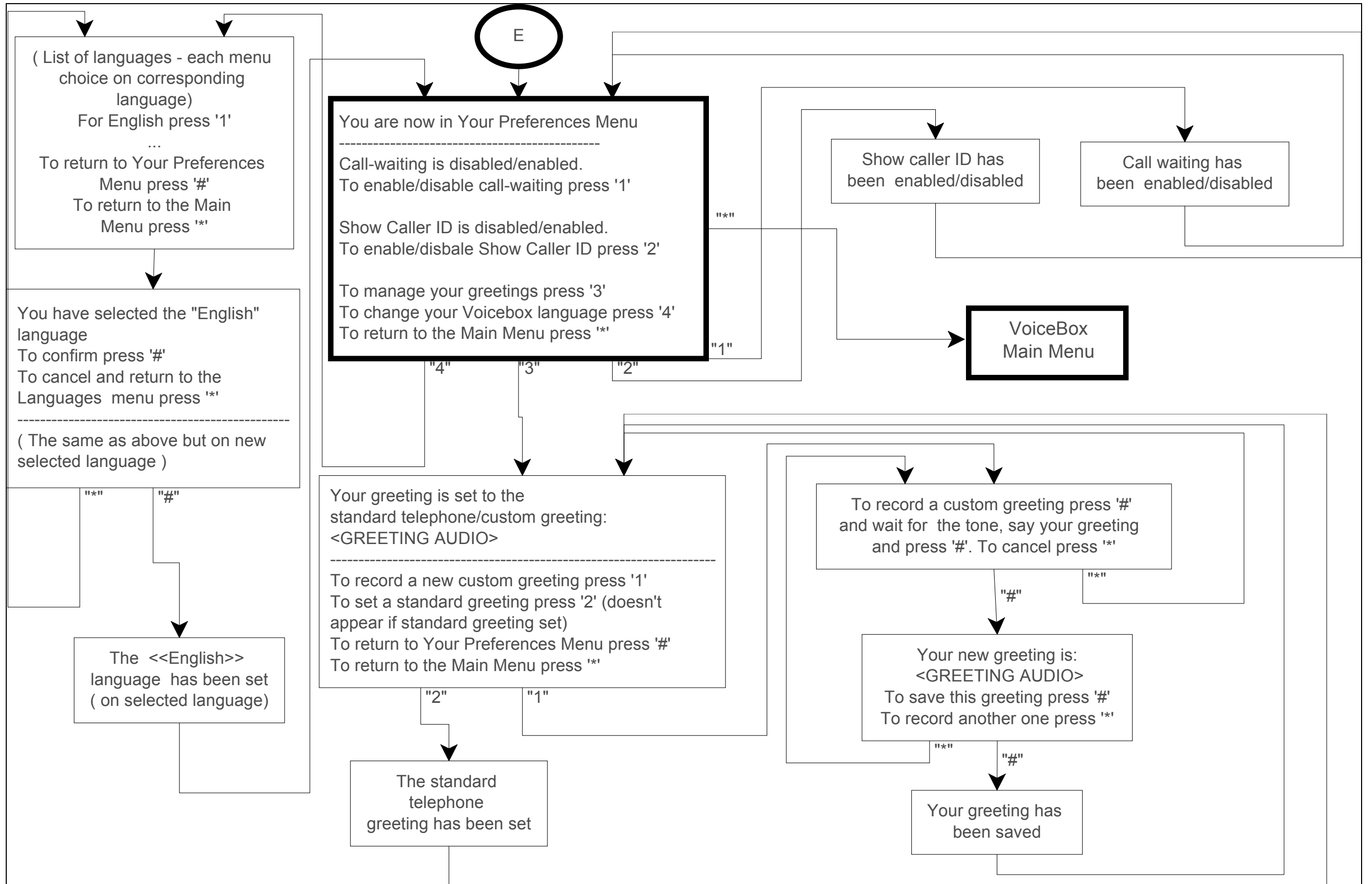
B Sub-diagram - Checking Account Balance



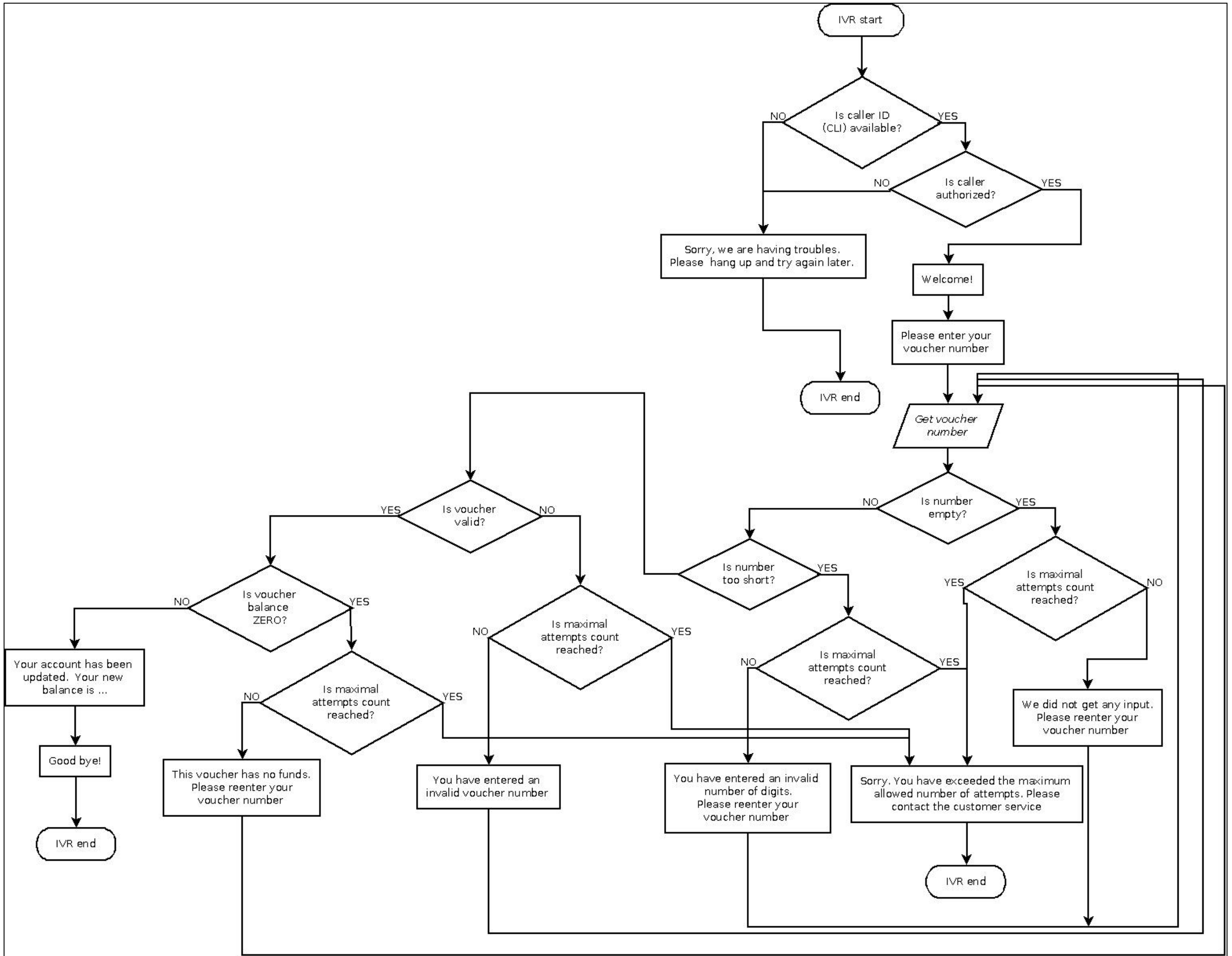
D Sub-diagram – Managing Telephone ReachIn Numbers



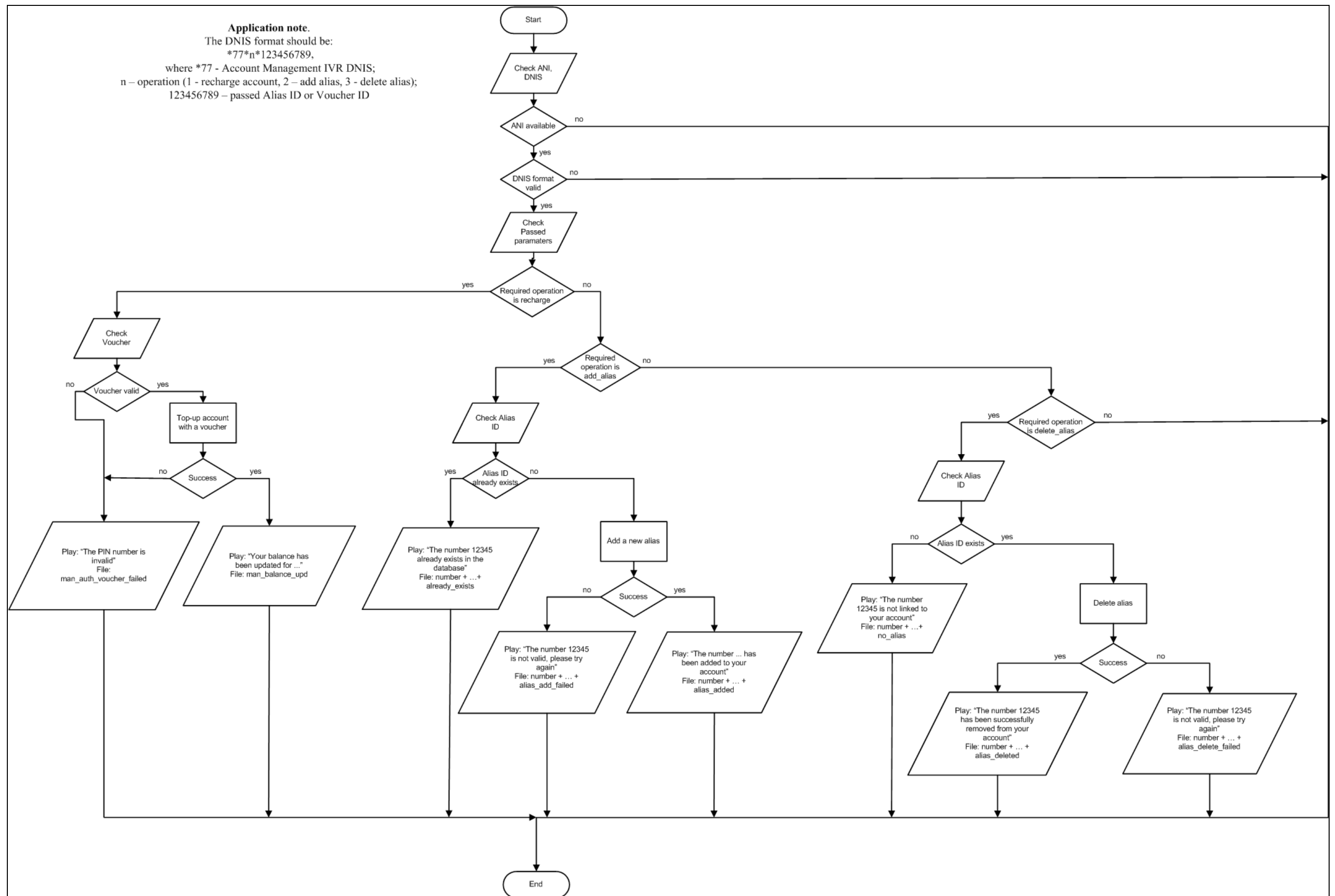
E Sub-diagram - Accessing Preferences



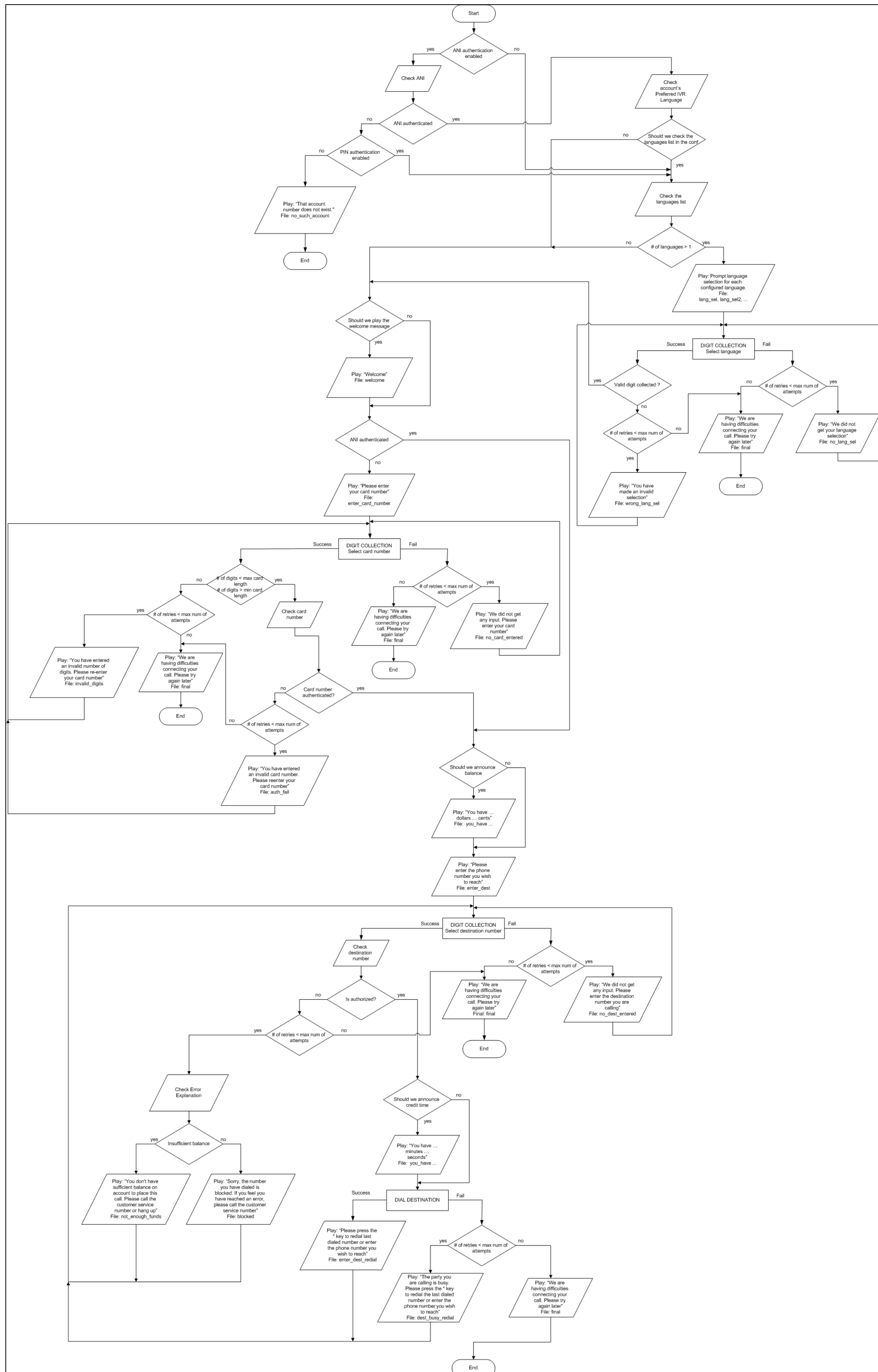
Voucher Recharge IVR



Account Management IVR



Prepaid Card IVR



ANI/DNIS/PIN Callback IVR

