PORTA ONE

PortaSIP

Media Server Administrator Guide

Maintenance Release

Documentation

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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

PortaSIP® consists of the PortaSIP® Switching Server and the PortaSIP® Media Server. This guide mainly covers the PortaSIP® Media Server; consult the **PortaSIP® Switching Server Administrator Guide** for information about the PortaSIP® Switching Server.

This document provides a general overview of the PortaSIP® Media Server (hereinafter referred to as the Media Server), a platform for the delivery of enhanced business and residential communications services. The Media Server provides various IVR applications and handles voice, fax, and regular email messages, 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 upon major releases only, and does not always contain the latest material on enhancements that occur in-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`.

Exclamation mark draws your attention to important actions that must be taken for proper configuration.

NOTE: Notes contain additional information to supplement or accentuate important points in the text.

Timesaver means that you can save time by taking the action described here.

Tips provide information that might help you solve a problem.

Archivist explains how the feature worked in previous releases.

Gear points out that this feature must be enabled on the Configuration server.

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Hardware and Software Requirements

Server System Recommendations

- A 64-bit processor (Xeon, Opteron). Additional processors are recommended 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/hw-requirements/**.

For information about whether particular hardware is supported by Oracle Enterprise Linux used as the operating system in PortaSwitch®, consult the related document on the Oracle or RedHat website: **https://hardware.redhat.com/**.

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Installation

PortaSwitch® installation ISO files contain everything required for installing Oracle Enterprise Linux (64-bit version), PortaSwitch® and the supplementary packages that are necessary for convenient system administration and maintenance.

After the installation is complete you will assign roles (e.g. RADIUS, web interface, PortaSIP®, etc.) to individual servers using the configuration server tool – this will automatically enable the required components of the PortaSIP® Media Server software on each server.

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

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1. System Concepts

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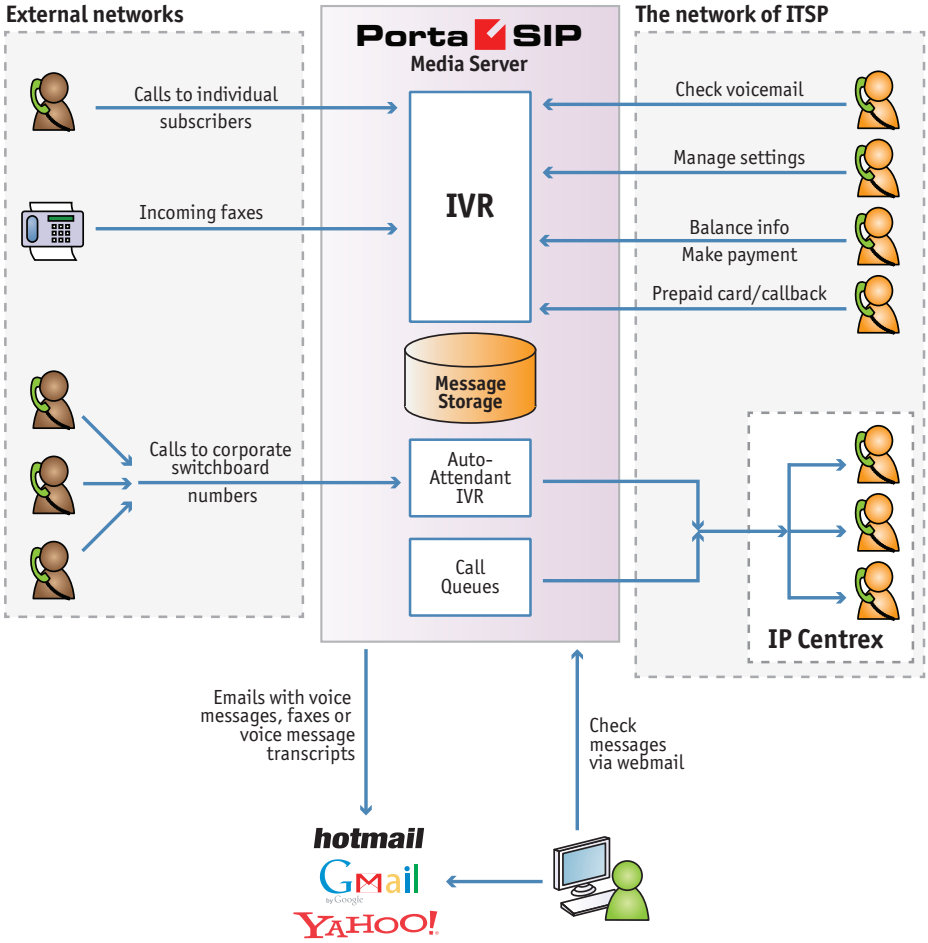
The Media Server's Role



The **Media Server** is one of the key components of PortaSwitch® – it is a platform which provides users with various IVR applications: balance info, auto attendant, prepaid card calling, conferencing, voicemail, fax-to-email, and so on.

The Media Server handles voice, fax and regular email messages, which users can access via a web interface or by telephone.

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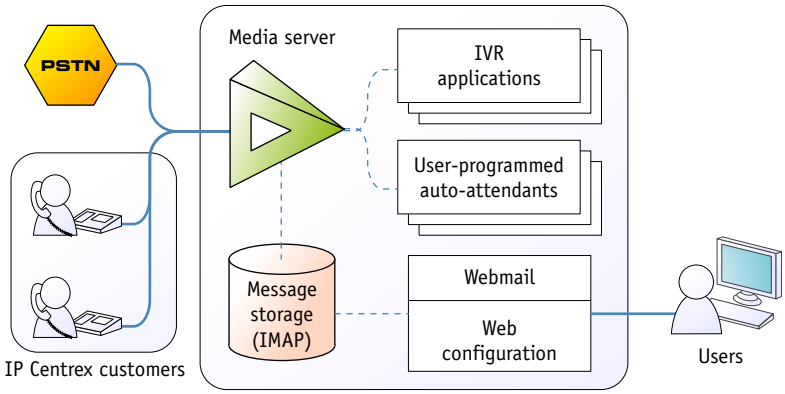
The Media Server Components



The Media Server 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 the Media Server 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).

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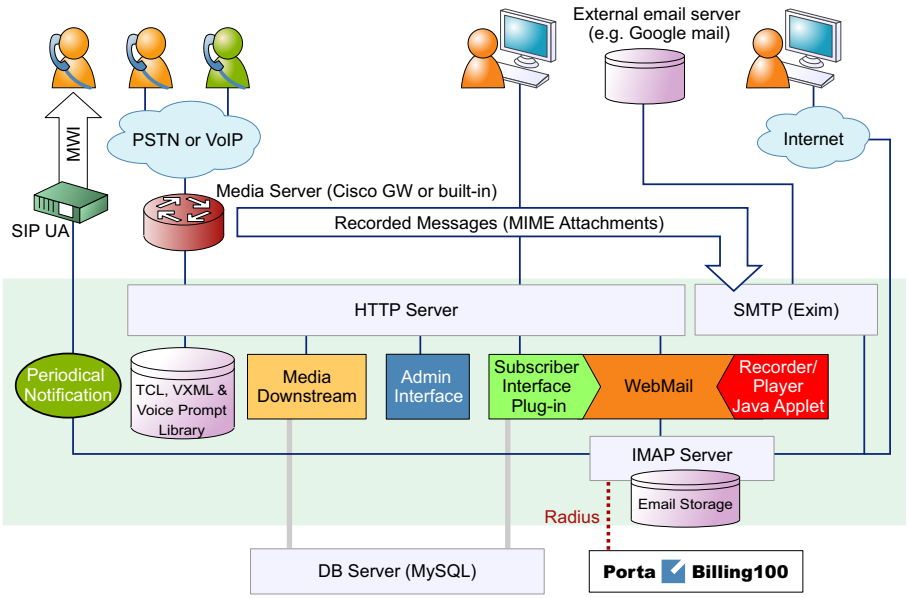
The Media Server Architecture



The Media Server Performance

While for PortaBilling® or the Switching Server the majority of resources are used during call initiation or call disconnect, the situation is different with the Media Server. The Media Server provides 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 Media Server can process. (Of course, you can scale up by using multiple Media Servers to achieve the better performance.)

On a server with the recommended configuration, a Media Server 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 the Media Server and an outgoing call from the Media Server to an external destination.

Another limitation on the Media Server 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.

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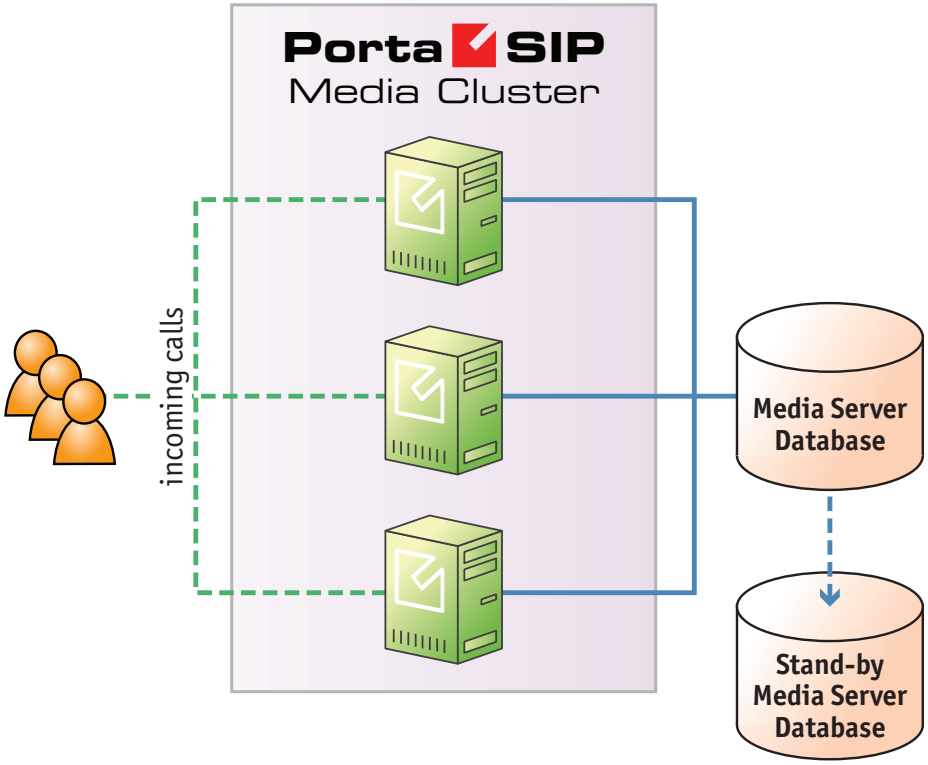
The Media Server Clustering

When multiple Media 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 Media 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 Media Server 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 Media Server 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 Media Server database on the same physical server that is

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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 Media 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 the Media Server where an IVR application that is associated with this number is launched.

IVR Applications

When the Media Server answers an incoming call, an IVR (Interactive Voice Response) application is launched. This application defines how the call is to be further processed, i.e. what the user on the other end will hear. The Media Server 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 the **Auto Attendant** section 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 card calling application should announce the available balance or not). Using the PortaBilling® web interface, administrators can quickly and conveniently define how users access IVR applications.

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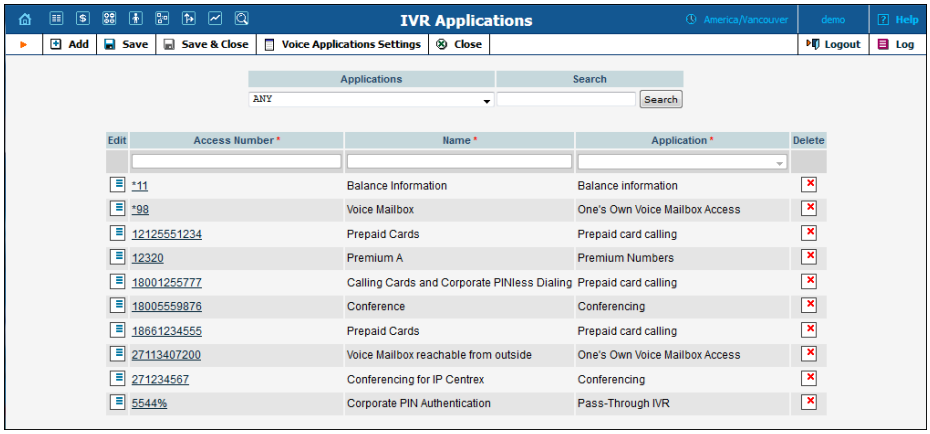
Auto Attendant

The Media Server'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 card calling 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 the Media Server. The Media Server then will launch the specific IVR application associated with that access number.



Application options

The administrator can customize properties of the application that is running on a specific access number.

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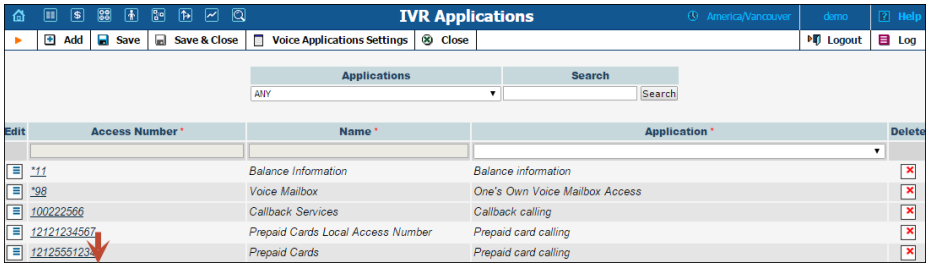

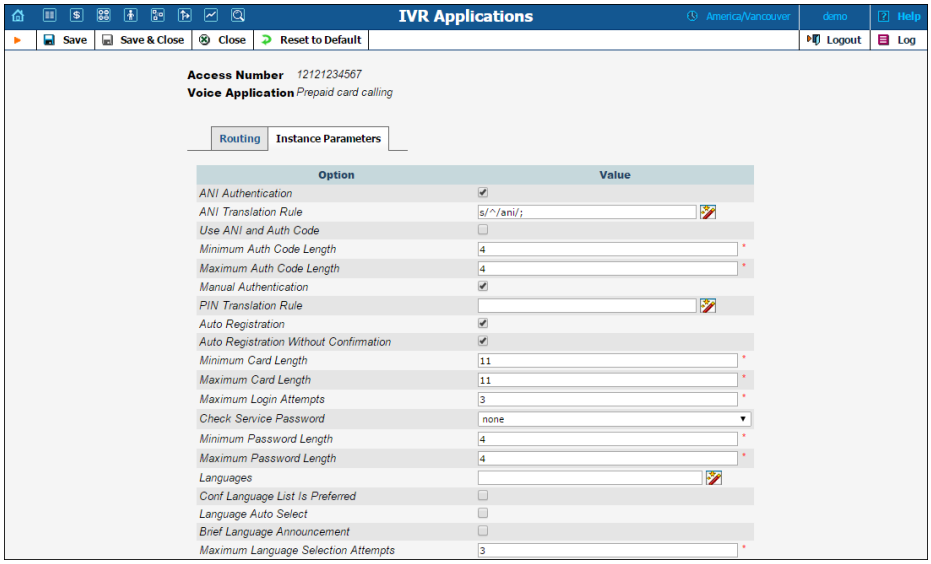

Distributing incoming IVR calls to specific Media Servers

The administrator can also assign incoming calls on a particular access number to a chosen subset of available Media Servers. The list of available servers – or groups of servers – can be prioritized so calls will first be handled by the server at the top of the list and then in descending order based on availability. A typical case scenario is when Media Servers are installed in multiple geographic locations and it is desirable to handle calls arriving from local telco partners on the Media Servers in a specific region.

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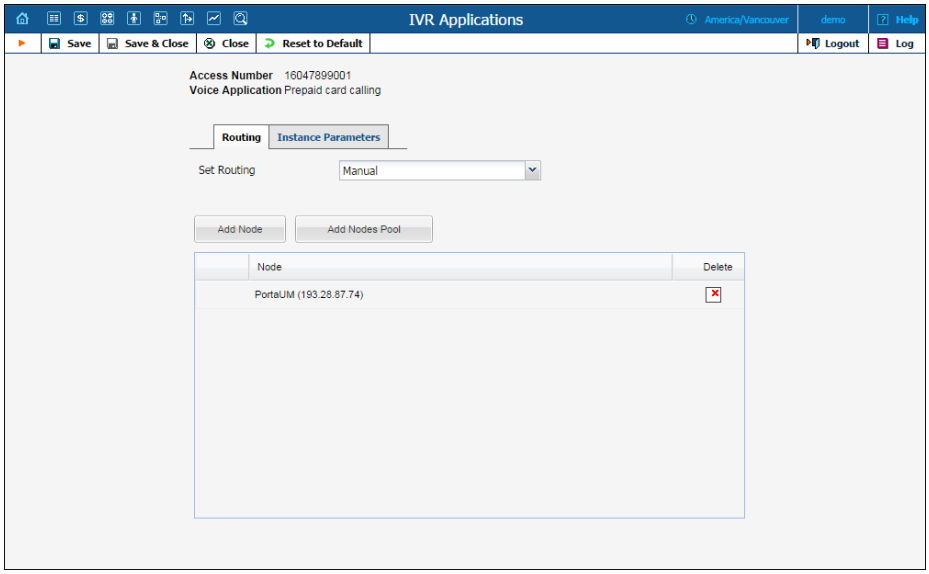
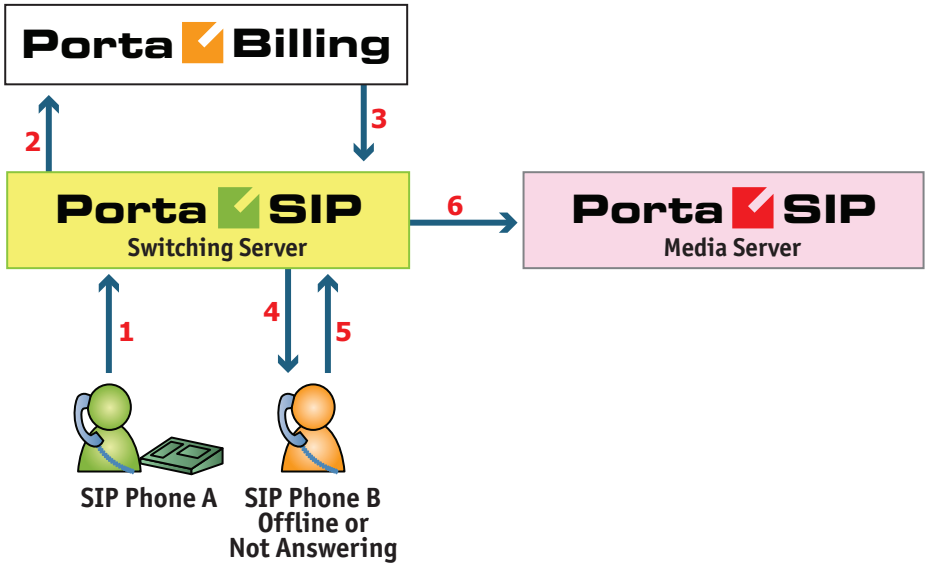


This will improve call quality by reducing network delay between the telco's originating gateway and the Media Server.

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 Switching Server (1).

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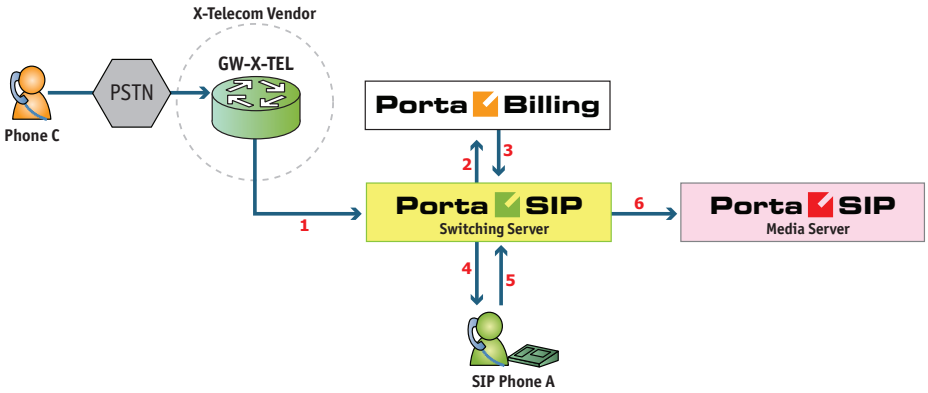
- The Switching Server performs authorization in the billing (2). For example, if A dialed 121, the billing will inform the Switching Server that the actual number is 12027810009, and that this number belongs to B (3).
- The Switching Server checks the registration database, but it appears that this 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 the Switching Server.
- When an incoming call arrives to the Switching Server (1), after gathering the required information, the Switching Server 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 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

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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.
    - Then PortaBilling checks if account 12025550003 is currently listed in the shared database of registered IP phones and what the contact IP address is.
    - Also, since account 12025550003 has the "Unified Messaging" service activated, the Media Server is added to the list of call delivery routes.
    - 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 the Switching Server (**3**).

- The Switching Server starts the outgoing call to 12025550003 (**4**).
    - If A answers the call, the call is connected as usual. The Switching Server starts the timer, and will disconnect the call if the maximum call duration is exceeded.
    - If A does not answer his phone (**5**), or his phone is not registered at this time, then the call is routed to the Media Server.

- The Media Server 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.

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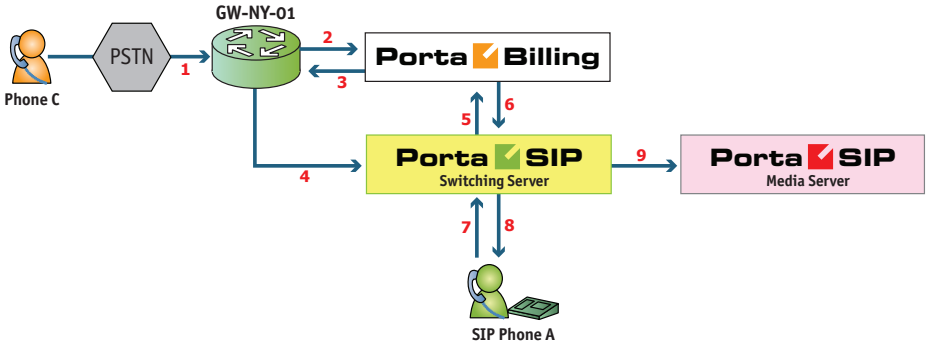
A Call from the Outside Network (Using a Gateway)



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:
    - 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 Switching Server. Thus it sends an INVITE to the Switching Server (4).

- PortaSIP receives the INVITE, but without authorization information. Thus the Switching Server performs authentication

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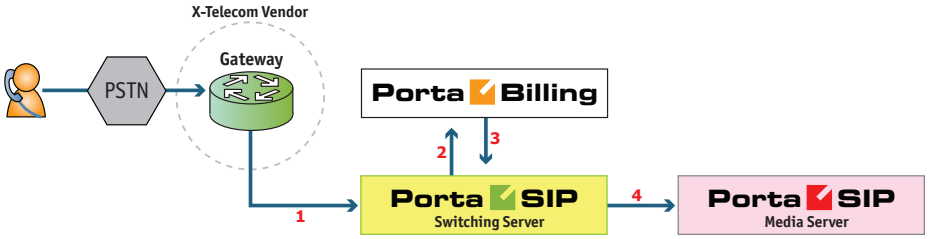
based on the IP address (5), (6). Since this call comes from a trusted node, i.e. gateway GW-NY-01, it is authorized.

- The Switching 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 mobile 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 the Switching Server.
- When an incoming call arrives to the Switching Server (1), the Switching Server 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, the Switching Server 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.

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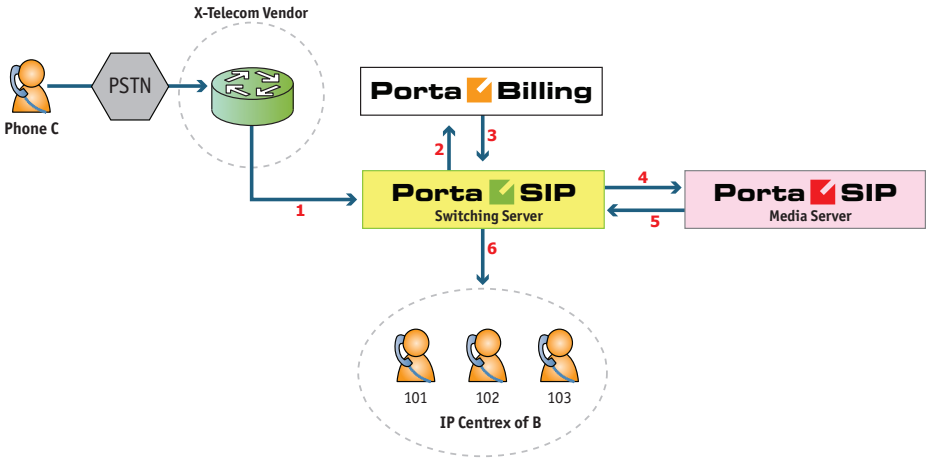
- Then PortaBilling detects that there is an entry in the Access Numbers section which designates 18005555865 as a special IVR application number.
- This call will be forwarded to the Media Server, and none of the other call routing options (vendor connections, LCR, etc.) applies in this case.
- Finally, the authorization confirmation, which includes the routing list (with just the Media Server in it), is sent back to the Switching Server (3).

- The Switching Server routes the call to the Media Server.
- The Media Server 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 auto attendant service is enabled for it. This account will not be provisioned on any IP phone, since the goal is to let the Media Server handle the call. The customer logs in to the 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.

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- 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 the Switching Server.
- When an incoming call arrives to the Switching Server (1), the Switching Server 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, the Switching Server 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 "Calls from Vendor via SIP" 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.
    - Since the account 18005551234 has the auto attendant service enabled, and this is not provisioned on any IP phone, the Media Server will be the only entry in the list of call delivery routes.
    - The authorization confirmation including the routing list (containing only the Media Server), is sent back to the Switching Server (3).

- The Switching Server routes the call to the Media Server.
- The Media Server 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, the Media Server establishes a new outgoing call to the Switching Server (5).
- When an employee answers the call on that extension (6), the Media Server 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).

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Auto Attendant

Basic Concept

- The Media Server'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 **Timeout** action is performed. If **Timeout** is not configured beforehand by the administrator, then the system plays the prompt about missed input 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:
    - **Disabled** – This informs a user that the selection is unavailable and returns to the "Play Menu prompt" (this is the action used for all menu items when the initial value has not been modified).
    - **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.
    - **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).
    - **Prompt for extension #** – This transfers the call to an extension number entered by the caller from his dialpad. For example, the menu item is 5 and the extension number is 123. When the voice prompt requests that the user input a menu

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item and then the extension number, the caller inputs 5123 and is connected with the callee.

- **Dial extension directly** – This transfers the call to an extension number entered by the caller from his phone. Note that this option is only applicable if the first number of an extension coincides with the current action digit. For example, the menu item is 5 and the extension number is 5123. The voice prompt requests only the extension number to be input, so the caller inputs 5123 and is connected with the callee.
- **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.
- **Directory** – Launches this company's dial-by-name directory.
- **Menu** – This transfers the user to the selected auto attendant menu.
- **DISA** (Direct Inward System Access) – This asks caller for their DISA password. If the DISA password entered is valid, the system facilitates the outgoing call.
- **Call Queue** – This transfers the caller to the selected 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.

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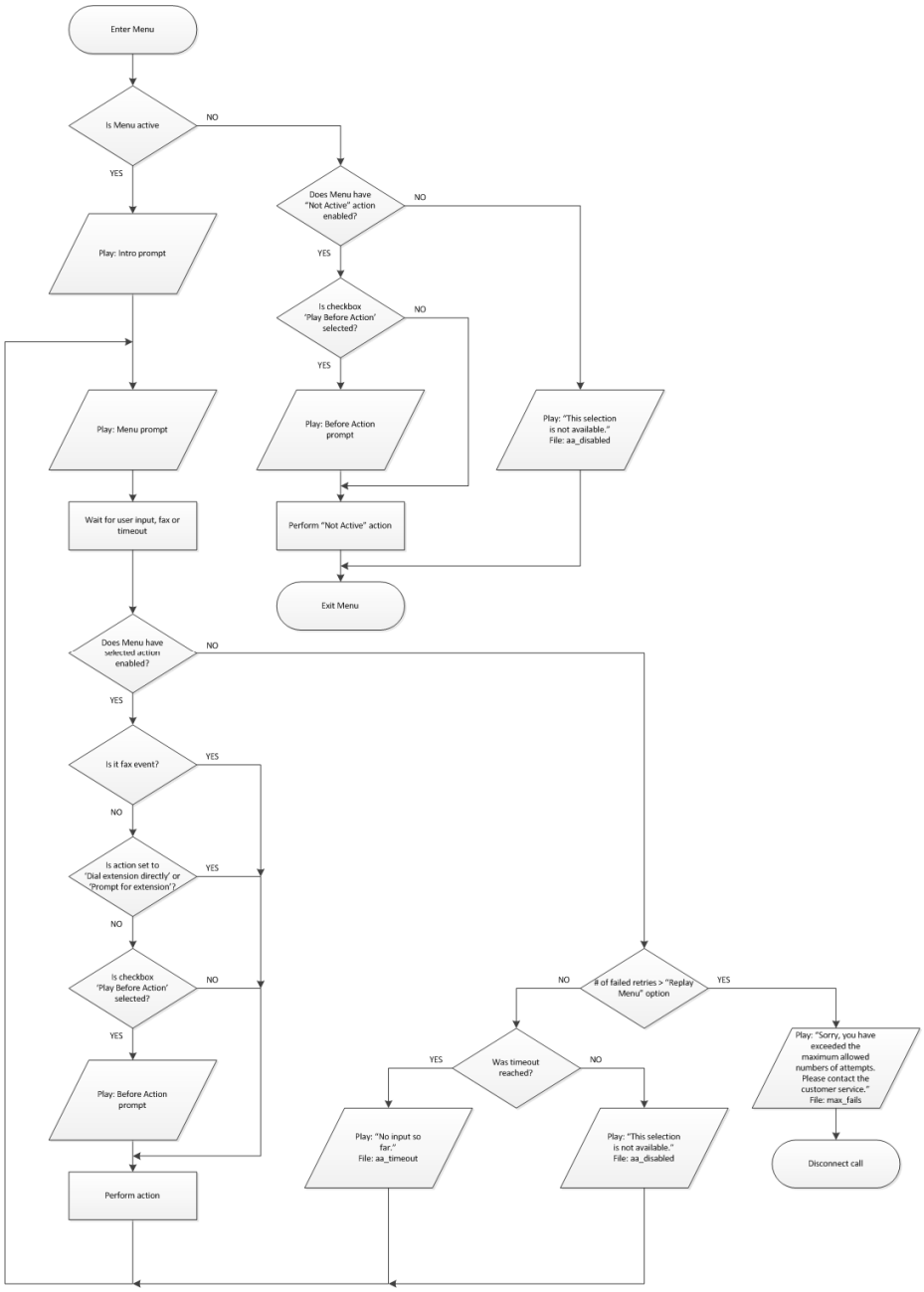
Call Menu Flow Chart



Detailed information on how to configure the auto attendant is provided in the **Configuring Auto Attendant Functionality** handbook.

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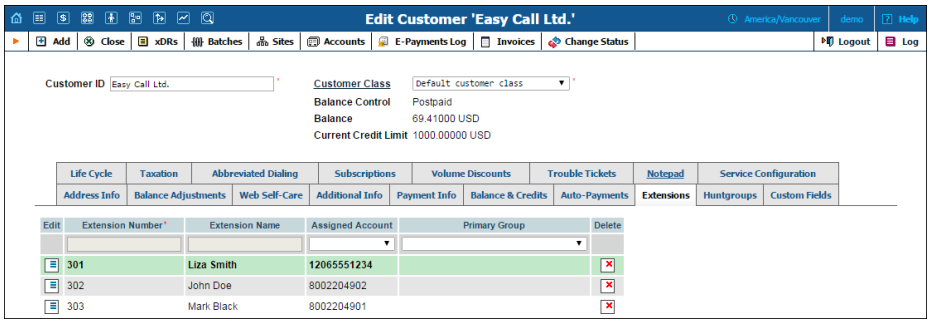
Direct Access to Extension from Auto Attendant

In addition to the existing mode that requires an end user to select a "Dial extension" action first, you may set up direct dialing for the extension from the auto attendant. For example, when a customer wants to contact a personal manager, he dials the company's corporate number (e.g. 18005559876) and the manager's extension (e.g. 301, found on the manager's business card). The customer dials 18005559876 from his mobile and reaches the corporate menu, which offers to initiate a call with, e.g. sales by pressing "1," or the support dept. by pressing "2" or by directly dialing the extension. The customer may now dial "301" without having to listen to all of the options and the call to his manager will be immediately initiated. If the manager is unavailable, the call may be redirected to voicemail (if such an option is activated).

To configure the Direct Access to Extension from Auto Attendant, follow the steps below:



1. Create the required extension (for example, 302 for account 8002204902).
2. Set up the Auto Attendant option for a number (for example, 8002204900).

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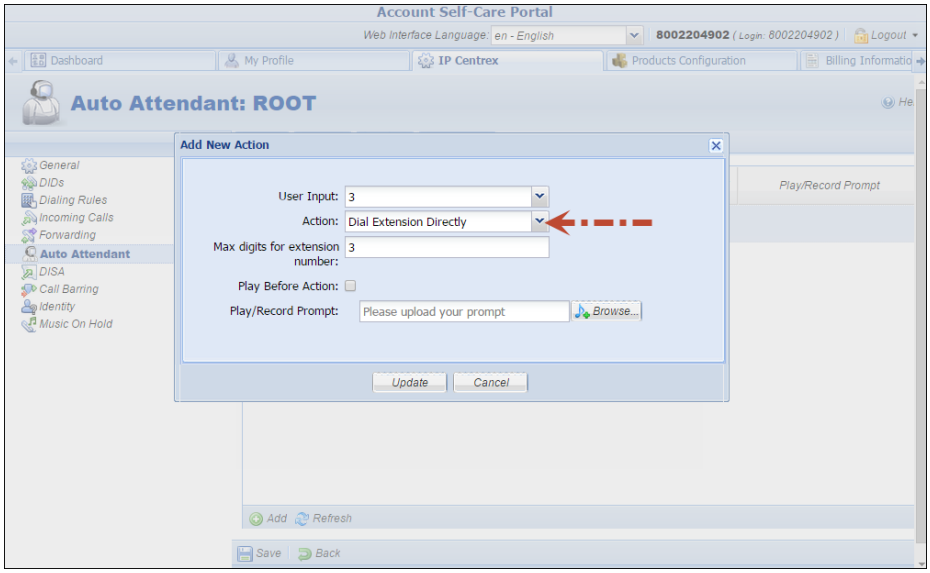
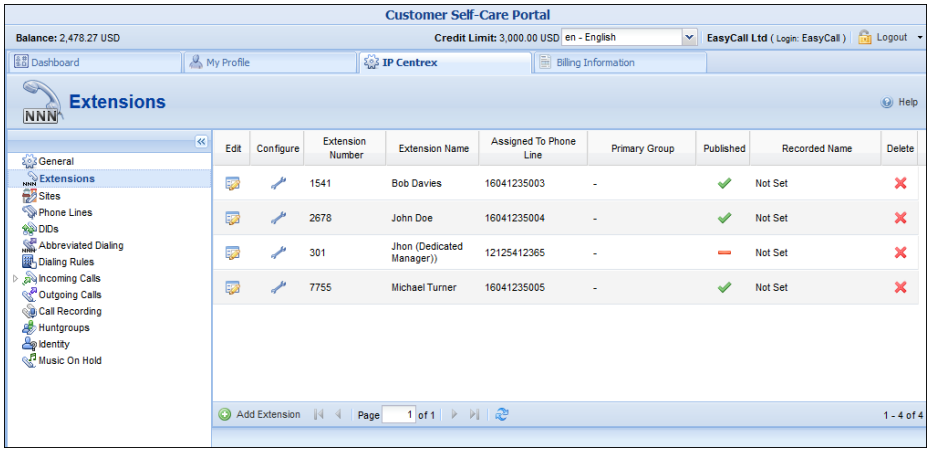


3. Add the action **Dial extension directly** for event 3.
4. When a caller dials 8002204900 and then 302, the call will be immediately redirected to John Doe (8002204902).

NOTE: The feature is feasible when an extension number starts with the same digit as the "Dial extension directly" option (e.g. 3). Dial 301 but not 3-302, otherwise the extension won't be found.

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.



There is a single, unified dial-by-name directory for each IP Centrex environment. Basically it is linked to the list of extensions, and so when creating an extension you can mark it as "published" so that it is included

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in the dial-by-name directory, and upload voice prompts with the person's name. 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.

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, PortaSIP® can forward these calls to the actual agents (customer service representatives) in a regulated fashion.

Consider the following example:

A broker company receives a lot of calls from its clients. To retain all incoming calls *and* give full attention to all their clients, the company's administrator creates the call queue "Sales" and assigns it to the Sales department.

So, when clients reach the company and are transferred to the "Sales" department, they are placed on hold and wait for an agent to become available and accept the call.

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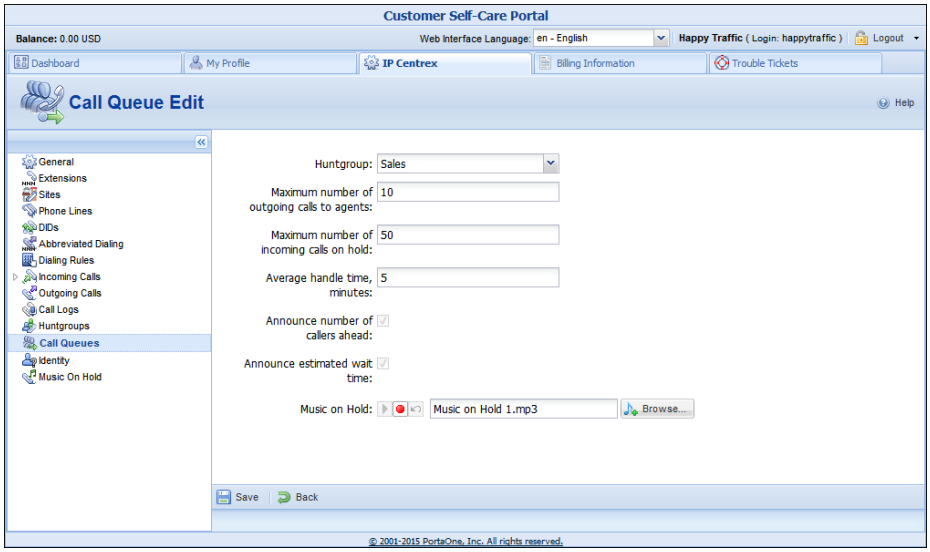
The whole Call Queue configuration is performed at the customer level (on the **IP Centrex** tab of the customer self-care interface):



Every call queue contains several configuration parameters:

- **Huntgroup** – When creating a new call queue, a customer must select a huntgroup so that when a call arrives to the call queue, it is transferred to the corresponding huntgroup.
- **Maximum number of outgoing calls to agents** – The maximum number of calls that agents can process simultaneously. You can think of it as the number of agents who answer calls made to a call center (e.g. if you have 5 agents who answer calls made to Sales, they can process a maximum of 5 calls, simultaneously). This option is used only by the previous-generation Media Server.
- **Maximum number of incoming calls on hold** – The maximum number of calls that can be placed on hold within this queue. When this number is reached, the next call is disconnected. This option is used only by the new Media Server.
- **Average handle time, minutes** – The average expected processing time for each call (used to calculate the estimated wait time).
- **Announce number of callers ahead** – When this check box is selected, callers will hear an announcement about number of callers ahead of them in the queue. Note that this check box is currently always selected and dimmed.
- **Announce estimated wait time** – When this check box is selected, callers will hear an announcement about the estimated wait time. Note that this check box is currently always selected and dimmed.

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- **Music on Hold** – A melody (or announcement) which is played to users waiting to be connected. The maximum file size allowed is 3 MB.

Call Flow

The new Media Server

The Media Server interacts with B2BUA to obtain information about available agents.

The number of registered extensions to the SIP Server in the hunt group is adopted as the maximum number of available agents. If there are 5 registered extensions in the hunt group, then B2BUA tries to immediately connect the first 5 calls to the agents, queues the 6th call and returns its position in the queue to the Media Server. The Media Server plays announcements to the 6th caller about their position in the queue and the estimated wait time. The number of calls that can be placed in the queue is limited by the number specified in the **Maximum number of incoming calls on hold** option**.**

As soon as one of the agents finishes the phone conversations and becomes available, B2BUA attempts to connect the first queued call and then sends updated information to the Media Server regarding the positions of the remaining queued calls. B2BUA also notifies the Media Server when one of the callers who is on hold in the queue drops the call. The Media Server updates callers about their positions in the queue and the estimated wait time every 5 minutes (by default).

If an available agent doesn't answer a call, the call won't be placed in the queue. When there are available agents, the Media Server attempts to connect the first call in the queue until it is answered by an agent or dropped by the caller.

Estimated wait time is calculated for the new Media Server as (average handle time)*(number of callers ahead).

The previous-generation Media Server

The Media Server uses the **Maximum number of outgoing calls to agents** option to determine how many calls can be simultaneously connected to agents. If its value is 6, then the Media Server attempts to immediately connect the first 6 calls and queues the 7th call. It plays an announcement to the 7th caller about their position in the queue and the estimated wait time.

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The Media Server has no information about currently available agents. It uses the general number of calls that were transferred to the queue and are now either queued or connected. When this number decreases (i.e. one of the calls ends or is dropped), the Media Server attempts to connect the first call from the queue.

If agents fail to pick up a call, the Media Server rechecks whether the **Maximum number of outgoing calls to agents** has been reached, and if so, queues the call, and if not – attempts to connect the call again.

Every 5 minutes (by default) the Media Server announces up-to-date information to callers about their positions in the queue and estimated wait time.

Estimated wait time is calculated as (Average handle time)* [(Number of calls that arrived to the call queue and are currently on hold or connected to agents) - (Maximum number of outgoing calls to agents)].

Conferencing

Conferencing Server Architecture

As part of the PortaSIP® 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 account 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 Media Server, its resources are only used when a conference is in progress (i.e. at least one participant is in).

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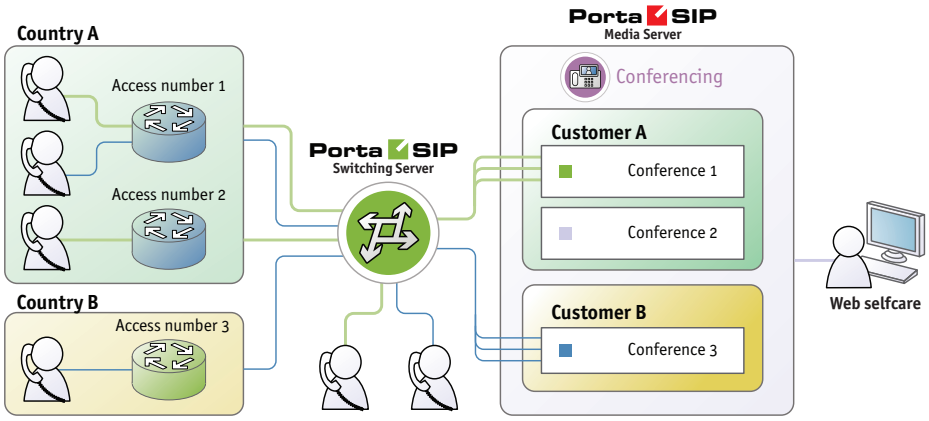
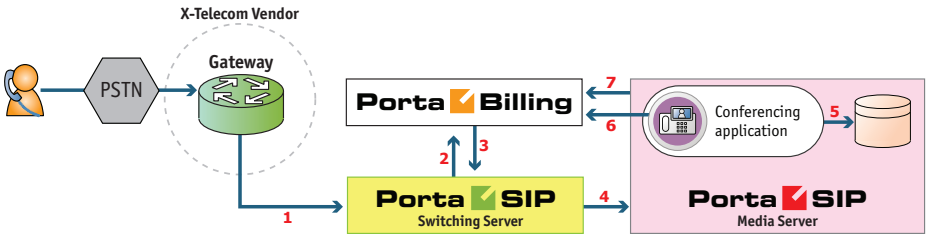


When a customer makes use of the conferencing facility, the PortaBilling® administrator assigns a separate tariff for the conferencing service in the customer's product configuration, and the owner of the meeting room is then billed for each incoming call in a conference session. For example, if your 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 Media 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.

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- 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 account 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 mobile 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 the Switching Server.
- When an incoming call arrives to the Switching Server (**1**), the Switching Server 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, the Switching Server 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, including the routing list (containing only the Media Server), is sent back to the Switching Server (**3**). The Switching Server routes the call to the Media Server.
- The Media Server 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.

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- 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 PortaSIP®) 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. Prompts used should be available in both G.711 and G.729 formats (the media server chooses the most suitable format for each specific call). Thus, if commercial codecs (such as G.729) 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.

Required License Calculation

- If none of the conference participants use a G.729 codec, then no G.729 transcoders are required.
- If at least one of the conference participants uses a G.729 codec, the number of required G.729 licenses must be equal to the number of conference participants and then multiplied by 2 plus 1. For example, if you have 10 conference participants, but only 5 of them use a G.729 codec, the number of required G.729 licenses will be equal to 21.

Video Conferencing

PortaSwitch® supports video conferencing so that during the video conference all participants can see the other participants in the conference.

This technology is useful for business customers because it offers them the opportunity to hold meetings, conferences, presentations and trainings remotely, without the need for all parties to be in one single location, thus saving time and money. Apart from the business sphere, this service can be useful in other life situations: for education, medicine, mass media, and of course, for personal use (e.g. communicating with distant relatives).

Video conferencing services provided by PortaSwitch® are manageable by the end user via the account self-care interface. Aside from the options available for audio conferences, the user can record a welcome video to be

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displayed during the authentication or Music on Hold. Video conferencing users can open the OSD (on-screen display) menu by pressing the # key on the phone keypad. The OSD menu provides a list of conference participants and instructions for how to switch among them.

Video conferencing service supports video in CIF (352*288) resolution and one of the most commonly used video codecs – H.264. PortaSIP serves as a multiplexer, so each participant can choose a specific video stream to watch and only that stream is sent to them - this optimizes bandwidth usage. Since entire video streams are switched / duplicated and no actual video content conversion is done - that translates to high performance (a single PortaSIP server can support up to 50 simultaneous video streams) with no additional costs for codec licenses.

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" check box 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.

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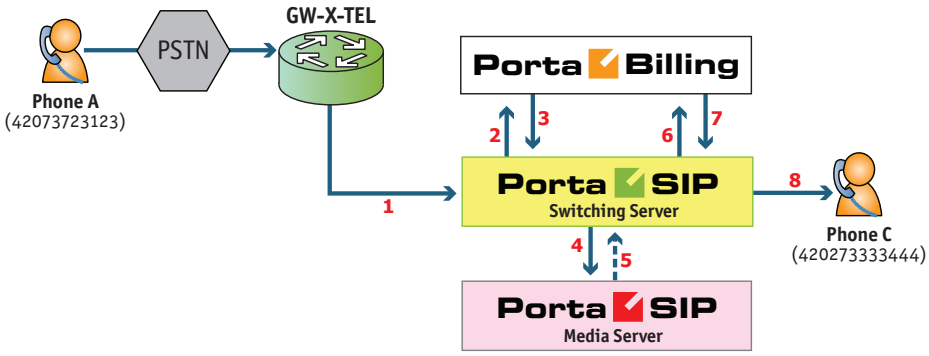
Listening to voice mail and making an outgoing call



- From his 42073723123 mobile (1), the customer dials the phone number 42027777100 that was provided to him.
- The Switching Server 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 the Media Server in it), is sent back to the Switching Server (3). The Switching Server routes the call to the Media Server.
- The Media Server 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.
- The Media Server sends an INVITE request to the Switching Server (5).
- The Switching Server sends an authorization request to billing (6).
- Billing sends an authorization response to the Switching Server (7).
- The Switching Server sends the call (8) to 420273333444 (the party being called will see 42027777100 as the CLI).

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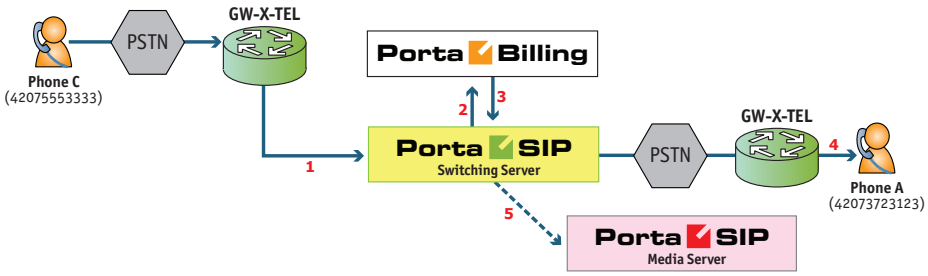
Receiving an incoming call



- From PSTN (42075553333), the user dials (1) customer's number: 42027777100.
- The Switching Server 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 the Media Server in it), is sent back to the Switching Server (3).

The Switching 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.

Prepaid Card Application

Prepaid services enable Internet telephony service providers (ITSPs) to offer calling card services that customers can pay for in advance.

The market for prepaid services includes tourists, immigrant communities, mobile populations such as military personnel, and people with limited credit histories who cannot otherwise get a private telephone line in their homes. These users can all gain immediate access to long-distance or international calling services from wherever they are located by using prepaid calling cards purchased at supermarkets and other types of retail outlets.

Separate Charge for Incoming Leg in the Prepaid Card Application

Incorporating incoming leg costs into the price of an outgoing call works well when there are just a few different cost levels for an incoming call leg (e.g. zero cost for local access numbers and $0.01/min for toll-free). If there are variations in incoming call costs (e.g. price differences

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between toll-free numbers and access numbers from different parts of the USA, or costs that depend on origination locations), then the number of required outgoing tariffs grows quickly and becomes hard to manage. That's why PortaBilling® provides separate pricing for incoming (calls to an access number) and outgoing (calls to a final destination) call legs in the Prepaid Card Calling Application. Two xDRs will be generated for such a call: one with a charge for the incoming call leg (leg A) and the other with a charge for the outgoing call leg (leg B.)

You may find out how to configure the prepaid card application to apply a separate charge for incoming call legs in **APPENDIX F. Separate Charge for Incoming Call Leg in the Prepaid Card Application** of the *Prepaid Services* handbook.

This feature allows you to manage prices for calling card services without administrative overwhelm, improve product price lining and thus increase your revenue.

Prepaid Card ANI Number Reassignment Provisioned via the Self-Care Menu

Depending on their business models, ITSPs can configure PINless dialing service so that customers who purchase a new prepaid card must reassign the ANI number associated with the previous card to the new card.

ANI number reassignment takes place through the self-care menu of the Prepaid Card Calling IVR application, allowing end users to perform more personalized account management.

It works as follows:

- The administrator configures the Prepaid Card Calling IVR application.
- The end user dials the access number and enters the self-care menu.
- By following the IVR prompts, the user associates his or her ANI number with a new prepaid account.

NOTE: ANI number reassignment is implemented only for the Prepaid Card Calling IVR application that runs on the new Media Server.

To illustrate:
End user John has signed up for PINless dialing service. His mobile phone number 16045558521 (the ANI number) is associated with prepaid card number 72313070131 worth $10.

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His account balance drops to $0.02 and is not enough to make further calls. He therefore buys another prepaid card, number 77854126029 worth $20. He dials the access number, enters the self-care menu, and associates his ANI number with 77854126029. Now John's balance is $20 and he can continue to use the PINless dialing service.

Alternatively, ITSPs can enable end users to delete their existing ANI number and associate this deleted number (or a new number) with a prepaid card. For example, if Jane Doe has associated the ANI number of a temporary location (e.g., a hotel phone number) with her prepaid card, once the number is of no further use to her, she simply deregisters it from her card.

Thus, with this functionality ITSPs can provide their customers with a flexible tool for ANI number management. An additional benefit is that, by making customers abandon depleted accounts, ITSPs receive additional funding from residual account balances.

The description of the Prepaid Card Calling IVR application configuration can be found in the **Prepaid Calling Cards (Using a PortaSIP® Media Server)** section of the **Unified Handbooks Collection**.

Supported Services

Leaving voice messages for a PortaSIP® user

The Media Server 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 Scenarios** section for detailed call flow descriptions.

Direct Inward System Access (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). 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 the Media Server and the remote party initiates a fax transmission, the

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Media Server 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 PortaSIP® user

The Media Server 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 Media Server IVR menu. This menu allows them to listen to recorded messages. Additionally, for SIP phones that support a message waiting indicator (MWI), the Media Server can automatically manage the SIP phone's MWI status, so that the user is notified when he has new messages.

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 Media Server 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 UM settings using a SIP phone

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

Managing personal UM settings using a web browser

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

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Managing user's current balance

PortaSIP® users can listen to the current state of their accounts by using the 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. In *APPENDIX B. Balance Announcements*, there are balance announcement examples for different IVR configurations.

Balance top-up

When your customers' balance has gotten too low, remind them that it is necessary to refill it.

This can be done in the following way:

- Select the **Play an announcement about self-care menu** and **Enable self-care menu** check boxes.
- Select **None** in the **Restriction for Calling with Low Balance;**
- The acceptable balance level is specified in the **Breakage** field of the product.
- Your users will hear the "Your balance is low, please refill your account soon" prompt and they can refill their balance within the self-care IVR menu by pressing the **#** key.

Prepaid card IVR

The Media Server 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.

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

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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 collects funds from the end user's mobile phone (by sending the balance transfer request to the mobile operator that the end user is connected to.) However, this must be supported by the mobile carrier. One advantage of this feature is that a customer can use any "ordinary" number to send an SMS.

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.

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2. Callback

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Callback Principle

The main idea behind callback is that when user A wishes to have a phone conversation with user B, he does not make an outgoing call to user B (as he would do with normal telephony service); instead:

- He initiates a callback call (we will discuss the available methods for triggering a callback later).
    - The system establishes a call to A, and instead of originating the call, A actually answers it (in most telephony networks a subscriber does not pay for incoming calls, so this call is free for A).
    - The system establishes a call to B and then bridges the two calls together so that A and B can communicate.

Main Components

The previous section described the callback service from an end-user perspective. Now let's look at the internal architecture of this solution. There are three main components of callback:

- Callback trigger.
- Callback engine.
- Authorization and billing.

Callback Trigger

The user informs the system of his desire to make a callback call in one of several available ways. The part of the system capable of interacting with the end user in some way and, upon receiving certain information from him, initiating the callback process, is called the callback trigger. Several types of callback triggers are available, and of course new ones can be invented and developed.

ANI (missed call) Callback Trigger

This is perhaps the most common trigger type; it is available on the PortaSIP® Media Server as a built-in IVR application.

The task of this module is to:

- Intercept an incoming call and collect information about the ANI number.
- Authenticate this number with PortaBilling®.
- Disconnect the incoming call.

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- Establish an outgoing call to the destination (the caller's number).
- Once this call is connected, hand over call processing to the callback engine which will do the rest (give the balance, ask for destination, etc.).

The user dials a certain access number, the call is forwarded by his telco operator and the incoming call is delivered to the PortaSIP® Media Server where a callback trigger application is configured. The application does not answer the call; instead it records the originating phone number (ANI or CLI number) and then disconnects it. Since the call is never connected, the end user is not charged by the telco operator. Now the application has the user's phone number and can therefore initiate a call back to him. Then an IVR is initiated which gives the user his current balance and prompts him for the destination number, etc.

Needless to say, this service only works when the correct caller info is delivered to PortaSwitch®. If information about a user's phone number is not available during an incoming call or if it is garbled (for instance, only the four last digits are discernible) then there is no way to call back the user. Usually you can only rely on ANI information for calls made within the same country. Thus, if you have an access number in the Czech Republic and your users make calls from within the Czech Republic (fixed or mobile phones), you receive correct ANI information. But if someone calls your access number from Slovakia or France, there is a high chance that either no ANI will be provided at all, or else it will be provided in a local format such as 02345464 (no country code), in which case it is impossible to determine whether this number is actually in the Czech Republic, Slovakia or France unless a correct translation rule is applied to the ANI number.

DNIS (DID) Callback Trigger

This is a slightly different approach to the callback function described above. In a situation where a correct ANI number is not likely to be available, one of the remaining options is to allocate an access number to each of your callback users.

Each access number is associated with a **Callback calling** IVR application and is provisioned as the account ID in PortaBilling®. Therefore, every callback user has an account in PortaBilling® with an account ID that is equal to the callback access number. The user then enters the user's actual phone number into the account's properties (in the **Associated Number** field on the **Service Configuration** tab).

When the user wishes to make a callback call, he simply makes a call to his dedicated access number. Every incoming call to this number will be dropped, but the system is triggered to initiate a call to the phone number

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associated with this access number. After that, the call is handled similarly to an ANI callback.

Thus, the DNIS callback functions as follows:

- It intercepts the incoming call and gets information about the DNIS number (access number dialed by the customer).
- Authenticates this number with PortaBilling®.
- Disconnects the incoming call.
- Establishes an outgoing call to the number specified in the **Associated Number** property for the account in PortaBilling.
- After the call is connected, the trigger relays call processing to the callback engine which does the rest (gives the balance, asks for destination, etc.).

One definite advantage of this method is that it can be used by callers located in one country who have been provisioned access numbers from a different country.

A disadvantage is that you will need a relatively large quantity of access numbers.

PIN callback trigger

This is yet another type of callback. Here a caller is not authenticated by the ANI number. The ANI number is noted by the trigger and the call is dropped. The so-called internal account is authenticated and authorized for the call leg to the party initiating the callback. Therefore, the internal account must be provisioned in the system and defined for the trigger.

After the callback is established to the party that initiated it, the IVR is brought up, prompting the caller to enter the PIN. After the PIN number is collected, the system searches for an account in PortaBilling® with the same account ID as the PIN entered by the caller. Once found, the account is authorized for the outgoing call leg. Upon successful authorization, the call is handled the same way as in the ANI callback.

The PIN callback trigger functions as follows:

- It intercepts the incoming call and records information about the ANI number.
- Checks for the internal account.
- Authenticates the internal account with PortaBilling®.
- Disconnects the incoming call.
- Establishes an outgoing call to the destination (the caller's number).
- And after the call is connected, relays call processing to the callback engine.

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- The callback engine collects the PIN number and authenticates the PIN with PortaBilling®.
- Upon successful PIN authentication, the callback engine performs further call processing (gives the balance, asks for destination, etc.).

The advantage of this method is the ability of the service provider to prevent service abuse, allowing the system to initiate callbacks to only a certain number of destinations. This is achieved by assigning a special tariff to the internal account's product.

Configuration

The ANI / DNIS / PIN callback trigger configuration consists of associating an access number with a callback IVR application and customizing the general parameters for the application. These include languages and authentication by ANI, DNIS or PIN.

The step-by-step configuration of the ANI Callback trigger can be found in the **Setting up an ANI Callback** handbook. For the complete overview of the available IVR application configuration parameters, please also consult the *APPENDIX A. Configuration Parameters for IVR Applications* section.

SMS Callback Trigger

Most mobile phones on the market have the ability to send short text (SMS) messages. These messages can also be used to initiate callback. When a user needs to make a callback call, he simply sends a message with the relevant information to your access number. This message is wirelessly delivered to the PortaSIP® Media Server where the information is processed instantaneously and the callback is initiated.

An SMS message sent from a user's mobile phone reaches the PortaSIP® Media Server via a triggered component that therefore allows SMS messages to be relayed via a hosted SMS provider connection.

The hosted SMS provider interconnects with wireless carriers around the world, providing you with consolidated SMS flow (using a well-documented API) and a centralized management and billing interface. One good example of such a company is Connection Software, **http://www.csoft.co.uk/**.

Another supported hosted SMS provider is Mega-M, **http://www.mega-m.si**. It is supported as an outgoing provider, i.e. for the dispatch of SMS flow from the PortaSIP® Media Server to end users.

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NOTE: The incoming SMS flow from end users to PortaSwitch® is provided via Connection Software, **http://www.csoft.co.uk/**.

There are several advantages to this method:

- It is possible to include the destination (B) number in the text message. Thus the call can be connected immediately and the user does not have to manually enter this information in the IVR.
- It is possible to include extra information in the text message. For instance, the user can include a password in the message and then initiate a call from a friend's phone. Or he can initiate a call from his mobile phone, but tell the system to connect his hotel phone in Singapore with a destination number in the USA.
- SMS messages have a low fixed cost so there is no risk of being charged for an outgoing call. Also, while traveling in foreign countries, outgoing calls may be disabled unless a special roaming service has been activated, and SMS messages are normally allowed.

Other features of SMS Callback:

SMS messages can not only be used to initiate calls, but to perform other actions, too, such as:

- New user registration.
- Balance top-up.
- Balance check.
- Password change.

Configuration

The configuration of the SMS trigger is performed via the admin web interface. A detailed description of the SMS trigger instance configuration is described in the **Setting up SMS Callback Services** handbook.

Usage

After the SMS trigger is configured, your customers can send SMS messages to your access number that contain any of the commands described in *Appendix B – SMS callback commands*.

Web Callback Trigger

The purpose of this module is to allow end users to initiate callback calls by filling in a form on the web interface.

When a user wants to initiate a callback call, he just goes to your website, enters all the required information (username, password, phone number, etc.) and clicks the Submit button to initiate the call. This request is

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delivered to the callback trigger via HTTP or HTTPS, and the callback call is initiated.

The workflow of the web callback trigger is as follows:

- The customer retrieves the web form and fills in values such as account ID, password, first number, second number, etc.
- By pressing the submit button the customer's request goes to the CGI script (hosted on the PortaSIP® Media Server).
- The script processes the input parameters and then relays the callback request.

A sample HTML document that contains a form with callback parameters is available as part of the PortaSIP® Media Server, i.e. built in, and available upon the configuration of web callback. This document is only an example of the required parameters; you can customize it (change text and colors, insert pictures, etc.) or even design the form to be a part of your existing customer portal. However, it is important that all of the form's parameters and their names be left unchanged.

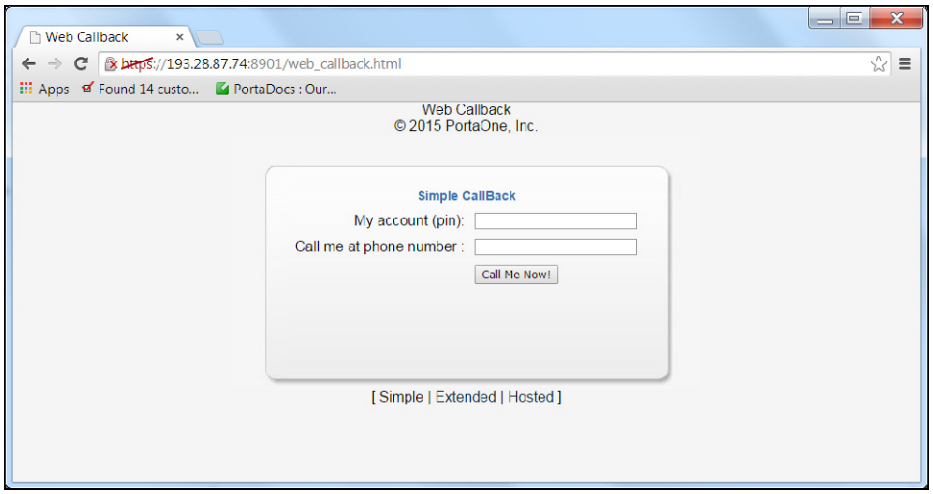
You may also entirely omit this web form where users fill in data and press the submit button. In this case, the HTTP request for callback would be sent from another application.

An advantage of this method is that (similar to SMS callback) the user can enter all the required information in one place. A disadvantage is that the user needs access to the Internet in order to use the service.

The following are the types of web callback:

Simple

This type of callback provides a convenient way for a user to get connected to the helpdesk or central office.



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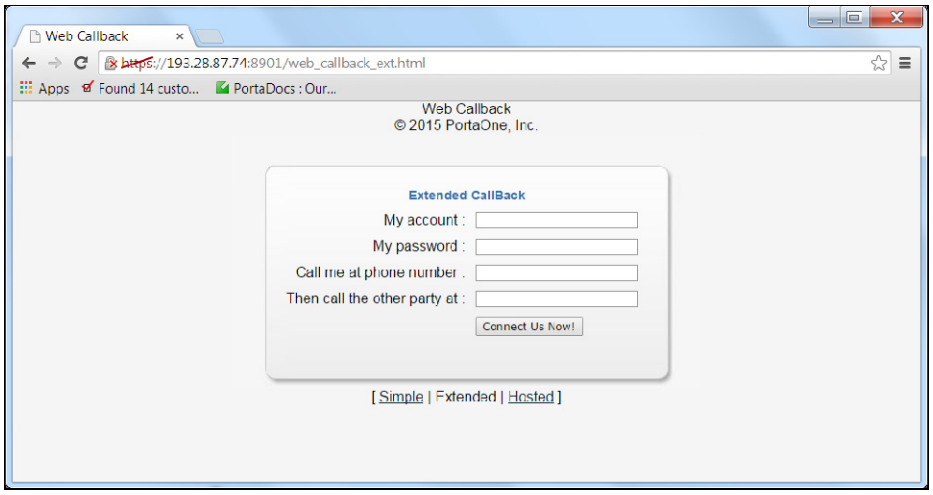
- The user provides his account ID (PIN) and the phone number.
- The second number is pre-defined in the application configuration.
- The system checks with billing that the account provided is valid and allowed to establish a call to the given number (the user's phone number).
- When the user answers, the second call (to the pre-defined number) is established.
- After the second call is established, the two calls are bridged together.

Extended

This is the typical callback scenario, in which all of the information is provided by filling in the web form.



- The user provides his account ID (PIN), service password and two phone numbers (the first is the number the user wants to be called back to and the second is the number of the remote party the user wants to reach).
- The system checks with PortaBilling® that the account provided is valid and allowed to initiate a call to the first number.
- When the user answers, he hears the current balance and maximum allowed call duration while the second number is being dialed.
- When the second call is answered, the two calls are bridged together.

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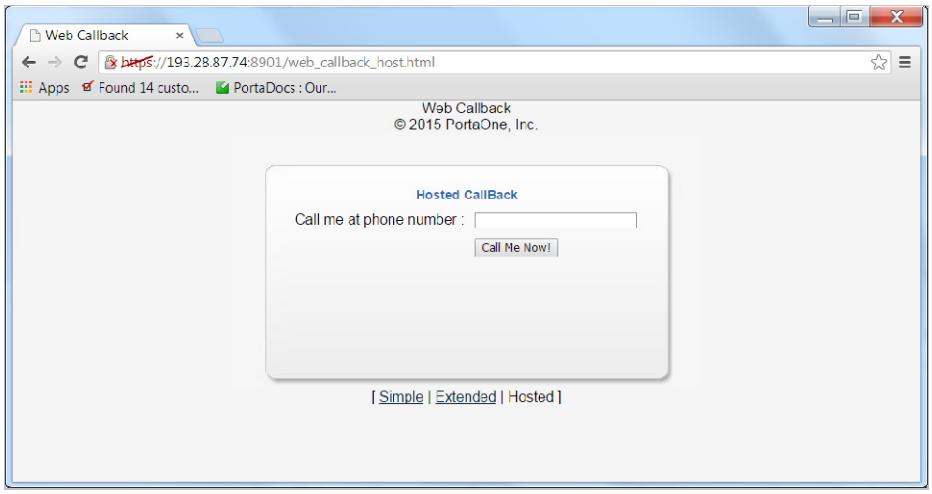
Hosted

This type of callback is a convenient way for a user to request a call to the user's phone from the service provider's helpdesk (at the expense of the service provider).



- The user provides the phone number where he wants to be called.
- The system authenticates the internal account for the call to the user's phone number.
- After the call is established, the system establishes an outgoing call to the pre-defined number in the application configuration.

NOTE: Since the first outgoing call is established without any authentication, to prevent service abuse, make sure that unauthorized users may not access the web callback trigger for hosted mode.

Configuration

The configuration of a web callback trigger is performed via the admin web interface. A detailed description of the web callback trigger instance configuration is described in the **Setting up WEB Callback Services** handbook.

Email Callback Trigger

The purpose of this module is to allow users to request callback calls via sending an email.

The workflow of the email callback is as follows:

- The user composes an email according to the pre-defined format in which the user provides his account number, password and phone numbers inside a message and sends it to a specific email address (e.g. callback@yourdomain.com). The format of the

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message must correspond to the format of the "Initiate callback – type C" SMS command. Please refer to *Appendix B – SMS callback commands* for the detailed command description.

- When the email arrives to the PortaSIP® Media Server, the mail filter parses the information contained in it and passes it to the callback trigger.
- The trigger authenticates the account provided in the email and relays the call processing to the callback engine.
- The callback engine then initiates the call legs.

One small advantage of this method over web callback is that while in some office environments users are not permitted to browse the web, they can still send and receive emails.

Configuration

The configuration of the email callback trigger is performed via the admin web interface. A detailed description of the email callback trigger instance configuration is described in the **Setting up Email Callback Services** handbook.

Callback Engine

The component of the VoIP network that actually establishes calls to A and B and bridges them together is known as the callback engine. The functions that the callback engine must perform are as follows:

- It includes an interface for configuring a callback trigger which transfers the required call information to the callback engine.
- Performs call authorization in PortaBilling® to ensure that the user is allowed to make the call and has sufficient funds to cover it.
- Correctly establishes and connects outgoing calls.
- Disconnect calls when caller has exceeded his credit limit.
- Reports call duration and other relevant call information to PortaBilling® so the call will be correctly charged.

The Media Server callback engine does just that. It works in close conjunction with the callback trigger. After the callback trigger gathers the initial information (ANI number, account and password) and performs the initial account authentication / authorization, the callback engine performs further call processing.

A detailed description of the callback call flow will better illustrate its functions:

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- When a new request is received from the callback trigger (e.g. for phone numbers 12065554123 and 16047770456) including the username and password for the account trying to establish the call, a request to establish call leg A (to number 12065554123) is sent to the PortaSIP® Switching server.
- The PortaSIP® Switching Server authorizes call leg A in PortaBilling®.
- Upon successful call leg A authorization, PortaSIP® asks PortaBilling® for the optimal routing for destination number 12065554123 and establishes the outgoing call to the first destination number.
- After call leg A is connected (i.e. the user picks up the phone), it sends a request to PortaSIP® to establish call leg B (to number 16047770456) if no additional actions are required from the user's side (i.e. to enter the PIN, the destination number, etc.).
- PortaSIP® authorizes call leg B as well, but differently: PortaBilling® is informed that there are two simultaneous calls, i.e. to 12065554123 and 16047770456. Thus PortaBilling® calculates the maximum allowed call duration so that the sum of the charges for both calls according to the appropriate rates will not be greater than the current balance.
- If authorization is successful, PortaSIP® obtains the routing for the second destination number and establishes the outgoing call.
- When the user of the second phone number (16047770456) answers, PortaSIP® starts a timer. Notification that call leg B is connected is sent to the Media Server callback engine.
- Call legs A and B are bridged together and users of phone numbers 12065554123 and 16047770456 can now start their conversation.
- When one of the users hangs up, the Media Server callback engine sends a request to PortaSIP® to disconnect the other call leg as well. The same happens if call leg B is disconnected because the time has run out (the time for call leg B is calculated based on the time for call leg A); the Media Server callback engine will request that call leg A be disconnected as well.
- PortaSIP® sends accounting information to PortaBilling® so that the account is charged for two call legs (to 12065554123 and 16047770456). Two xDRs appear in the database, and its balance is modified accordingly.

Advantages

Compared to similar software solutions, the Media Server callback engine has the following principal advantages:

- No specialized hardware. The Media Server callback engine is used on the same server on which PortaSIP® is installed.

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- This is the "out of the box" solution. To enable callback, simply perform the application configuration on the web interface.

Billing

Call authorization

The billing system must support a type of call authorization which is specific to callback services. For normal service (e.g. using prepaid cards) the question for authorization is, "If account 12065554123 has $5 available, and tries to call 861234567, is he allowed to talk at all and, if yes, for how long?" In this case, the billing system finds the applicable rate for 861234567 (let us assume it is $0.10/min, rounded to 60-second increments) and then calculates the allowed maximum duration as 5/0.10=50 minutes. This task becomes more complicated as the rating formula is more complex, including multiple intervals and special surcharges (a.k.a. billing tricks).

With callback, it becomes more complicated still. Now the question is, "If account 12065554123 has $5 available, and tries to call 861234567 and 42029876543 at the same time, for how long should he be allowed to talk?"

Thus the maximum allowed call duration must be calculated in such a way that the sum of the charges for the first and second calls does not exceed $5. Since rates for China and the Czech Republic will most certainly be different, and the rates will include different intervals and surcharges, this task becomes quite complicated and is therefore not supported by most billing systems. However, PortaBilling® supports special callback authorization, and is available to provide authorization for advanced callback services.

Charging a call

After the call has been completed, billing should take appropriate action to calculate the call charges, write an xDR to the database and modify the account's balance as well as any other applicable tasks. PortaBilling® supports all of the callback features described and allows you to implement flexible rating for callback services.

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FAQ

Can I use ANI callback when my users from country A call my access number in country B?

Yes, you can set up this service, however, there is a high chance that the customer's ANI information will not be relayed correctly to PortaSwitch® – so the system may be unable to determine the customer's callback number. In this type of case you can offer your customers DNIS callback.

Can I use your SMS callback on CDMA-based wireless networks?

Yes. The wireless protocol used for sending SMS messages is irrelevant to callback as long as the message is delivered to the PortaSIP® Media Server. If you use one of the hosted SMS providers they will deal with the protocol details on their side, and all messages will be delivered to the PortaSIP® Media Server in a unified format.

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3. Appendices

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APPENDIX A. Configuration Parameters for IVR Applications

Prepaid Card Calling

Field	Description
ANI Authentication	If this option 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	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 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. For example, assuming that the country code is 44, we can apply 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.
Use ANI and Auth Code	This feature enables service for enterprise customers when multiple individuals place outgoing calls using the same caller number (typically from the company's main phone line). The IVR application finds the customer's phone number (ANI) and his company ID (in this account's **associated number** field) and collects the user's Authentication Code. The call is then authorized and billed to the account with an ID consisting of a company ID followed by the # sign and the authentication code, e.g. 534564#555.
Minimum Auth	The minimum number of characters the

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Code Length	authentication code consists of. By default: 4.
Maximum Auth Code Length	The maximum number of characters the authentication code consists of. By default: 4.
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.
PIN Translation Rule	When the **Check Service Password** option is enabled and the **PIN Translation Rule** is set, the IVR application modifies the collected PIN according to the PIN translation rule that allows the use of an ANI-based account (for example, ani442031234567) for PIN authentication. This provides your end users, normally identified by caller IDs (ANI/CLI), with the option of making calls from other phone numbers without having to input hard-to-remember PIN numbers.
Auto Registration	If the user's ANI is not found and PIN authentication is successful, the user will be prompted to register his 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.
Auto Registration Without Confirmation	When this option is enabled, the IVR will not ask for confirmation before registering a user's ANI for future PINless dialing. This option is available only when the **Auto Registration** option is selected.
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.
Check Service Password	Use this option to increase the security of a PIN-less dialing service or to allow end users to input their "ANI numbers" confirmed by service passwords for PIN authentication: • **None** ○ **For ANI authentication** – The IVR application will never prompt the user for their service password after

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	successful authentication by ANI. ○ **For PIN authentication** – It will be successful *only* if the user's account doesn't have a service password specified. This is a common use case for prepaid card services. • **Always** ○ **For ANI authentication** – The IVR application will always prompt the user for their service password. ○ **For PIN authentication** – The IVR application will prompt the user for their service password as long as the password is defined. • **Only PIN** ○ **For ANI authentication** – The IVR application will never prompt the user for their service password. This is a common use case for ANI services. ○ **For PIN authentication** – The IVR application will prompt the user for their service password as long as the password is defined. • **Only ANI** ○ **For ANI authentication** – The IVR application will always prompt the user for their service password. ○ **For PIN authentication** – It will be successful *only* if the user's account doesn't have any service password defined. This is a common use case for PIN services. For example, in case of the **Always** or **Only ANI** options, the IVR will do the following: upon collecting an ANI number, the IVR prompts the user 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 will match it.
Minimum	Minimum number of characters the valid service

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Password Length	password consists of. By default: 4.
Maximum Password Length	Maximum number of characters the valid service password consists of. By default: 4.
Languages	This option allows the user to select the language(s) to be used for voice prompts. Click the **Wizard** icon to open the language selection dialog box and define languages in a specific order, and then upon reaching an access number, the end user can choose one of these languages.
Conf Language List is Preferred	Select this check box to give the language list defined on this page a higher preference than the account's preferred language has.
Language Auto Select	Select this check box to allow the IVR application to proceed in the first language on the list if a caller does not choose any of the available languages.
Brief Language Announcement	Select this check box to play a brief announcement, "Please select your language or press 1 to proceed in *the first language on the list,*" instead of listing all of the available languages.
Maximum Language Selection Attempts	Type the maximum number of times that the IVR application replays the language menu prompt. By default: 3. This means that if a caller doesn't choose any of the available languages after the first time the menu is played, the IVR application replays the menu two more times.
Prompts Brand	IVR prompts can be customized per access number. Specify the directory where the custom prompt files are located. The Media Server 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 /var/lib/porta-um/prompts/brands/my_company/en/ and specify "my_company" in this field.
Play Welcome Message	If enabled, the IVR will play the welcome message to the user once he reaches an access number.
Restriction for Calling with 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.

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	• If **no outgoing 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.
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.
Credit Accounts Balance Announcement	You can configure your IVR applications to announce the current balance *for credit accounts* in one of two modes: • **Funds / balance** – For prepaid customers (those who have a credit limit of 0 or less), announce "available funds" (calculated as credit limit minus balance) and announce the current balance for others (postpaid customers); • **Balance** – Always announce balance (this is the classic mode, ideal for typical postpaid customers). • Note that a third **Backward compatibility** mode is enabled by default. It preserves the announcements exactly the way they worked in releases prior to MR35. In *APPENDIX B. Balance Announcements*, there are balance announcement examples for different IVR configurations.
Announce Credit Limit	If this option is enabled, the balance announced to the end user will be followed by an announcement of the credit limit. This allows you to separate cases where credit limits are assigned only as a precaution (and so the end user should not really know about them) from cases where they are an integral part of the service (e.g. loyal customers are rewarded with an increased credit limit the longer they stay with you).
Enable Self-care menu	Enables the menu with additional options such as making a call using another prepaid card or the

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	balance top-up. User can enter this menu by pressing the # key.
Play an announcement about Self-care menu	If this option is enabled, the IVR will allow the user to enter the self-care menu by pressing the # key.
Enable Account Top-Up	Selecting this option allows end users to top up their account balances by using vouchers on the self-care menu. The feature is enabled by default.
Allow ANI Deregistration	Selecting this option allows end users to disassociate ANI numbers from their prepaid cards through the self-care menu. When users next make a call, they are authenticated by PIN.
Check Associated Number	When this option is enabled, the prepaid card application checks if the account has specified an associated number. If it has, the call is redirected to this number. This field can also be used in different scenarios, e.g. together with the **Use ANI and Auth Code feature**.
Announce Time	When a user inputs the destination, IVR announces the maximum call duration allowed for this destination and then connects the call.
Use special access code for first calls	If this option is enabled, a special tariff will be used for the first outgoing call made by users with prepaid cards. This tariff should be associated with the appropriate access code in the product's **Services and Rating** list. This access code depends on the **Use fixed access code for first calls** option.
Use fixed access code for first calls	This option specifies the special access code format that must be applied for the *first* outgoing call with prepaid cards. • If this option is enabled, then the "FIRSTCALL" access code will be used. • If this option is disabled, then the "FIRST#12345" access code will be used, where 12345 is the access number dialed.
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

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	product's **Services and Rating** 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.
Separate Incoming Leg Cost	This option allows for the setting of separate prices for outgoing and incoming call legs.
Rate By	This specifies which number should be used for charging an incoming call leg. Available options are: CLI, CLD.
Incoming Access Code Prefix	This prefix must be added to an access number to permit the use of a different tariff for an incoming call leg cost.
Charge an Incoming Leg for Idle Time	If this option is enabled, the IVR application will include idle time in its charge for an incoming call leg. If this is disabled, then the length of the incoming call leg will be equal to the sum of the outgoing call length and be charged accordingly.
No Charge if Call Is Not Connected	If this option is set, the caller will not be charged for an incoming call leg if the outgoing call is not connected.
Seconds Rounding Mode	This option defines the way the seconds portion of "available time" is announced to a customer: • **none** – there is no rounding, so the time is announced including seconds, e.g. "Five minutes and 49 seconds." • **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). Select this check box 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.
Dynamic Reauthorization	Enable this feature to perform dynamic reauthorization for calls made though this access number. Dynamic reauthorization is required in cases where an account is concurrently using multiple services and making a payment while the session is still in progress.

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	Please consult the *Overdraft Protection* section in the **PortaBilling® Administrator Guide** for more information.
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/).
Disconnection Warning Interval, sec	This feature allows you to provide a warning when a call is about to be disconnected because of insufficient funds. This ensures that the customer has a sufficient amount of time to react and either finish the call or top up their balance.
Repeat Every, sec	This feature allows you to provide multiple warnings before a call is disconnected because of insufficient funds. In this case, IVR repeats the warning every specified number of seconds until the end of the call. These warnings help your customers to estimate whether there is enough time to finish their conversation or not, or if necessary, to top up their balance.
Type of Disconnection Warning	Choose one of the following warning types: • **beep** – plays "beeping" sound. • **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 Calling

Field	Description
Authenticate by	This option defines a user-name for an authentication request. Choose one of the following: • **ani** – Calling-Station-ID (caller's number) is used for an authentication request. • **dnis** – Called-Station-ID (access number which was dialed) is used for an authentication request. This application

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	establishes a callback to the associated number parameter of the account equal to the access number. • **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. For example, assuming that the country code is 44 we can apply 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: • 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). • Block some destinations for callback calls in the account's tariff (mark destinations as forbidden in the appropriate tariff rates). • 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. This option is applicable when **Send Authorization** is set to **Yes** and **Minimum Sufficient Time** > 0).
Destination 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.

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	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 *(for new media server)*	When enabled, this 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 a callback. Customer is prompted to enter a PIN number and after successful PIN validation, the system adds the caller's number as an alias to the customer's account.
Auto-Registration *(only for previous generation media server)*	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 the template account (a template account is defined as a parameter for each separate access number).
Template Account *(only for previous generation media server)*	Specify the account ID here to use its configuration properties as a template 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 Internal 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) an internal account will be used for charging leg A. **NOTE**: The PortaSIP Media Server regards the **Use Internal Account** check box as always selected.
Internal Account	Specifies an internal account.

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	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, the Media Server uses the application access number by default.
Languages	This option allows the user to select the language(s) to be used for voice prompts. Click the **Wizard** icon to open the language selection dialog box and define languages in a specific order, and then upon reaching an access number, the end user can choose one of these languages.
Conf Language List is Preferred	If this check box is selected, 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. The Media Server 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 /var/lib/porta-um/prompts/brands/my_company/en/ and specify "my_company" in this field.
Play Welcome Message	If enabled, the IVR will play the welcome message to the user once an access number is reached.
Restriction for Calling with 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 outgoing 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.
Credit Accounts Balance Announcement	You can configure your IVR applications to announce the current balance *for credit accounts* in one of two modes: • **Funds / balance** – For prepaid customers (those who have a credit limit of 0 or less), announce "available funds" (calculated as

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	credit limit minus balance) and announce the current balance for others (postpaid customers). • **Balance** – Always announce balance (this is the classic mode, ideal for typical postpaid customers. • Note that a third **Backward compatibility** mode is enabled by default. It preserves the announcements exactly the way they worked in releases prior to MR35. In *APPENDIX B. Balance Announcements*, there are balance announcement examples for different IVR configurations.
Announce Credit Limit	If enabled, the IVR will announce the credit limit as well as the account balance This option is applicable when **Credit Accounts Balance Announcement** is set to **balance** or **funds / balance**.
Minimum Voucher Length	Minimum length of a voucher number. By default: 11.
Maximum Voucher Length	Maximum length of a voucher number. By default: 11.
LegA CLD Prefix	Leg A (callback to a user's number) access code prefix. Allows billing leg A with different tariff. A separate rating entry with the appropriate access code and tariff should exist in the product's list of rating entries. **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_A.12125551234.
LegB CLD Prefix	Leg B (user's call to destination) access code prefix. Allows billing leg B with different tariff. A separate rating entry with the appropriate access code and tariff should exist in the product's list of rating entries. **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_B.12125551234.
Expected	This option verifies that the account's balance is

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Connect Time	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. By 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 rating entry to the product's rating 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. The first (transparent) tariff should be associated with the "ANNOUNCE+DNIS" access code in the product's rating 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: • **none** – there is no rounding, so the time is announced including seconds, e.g. "Five minutes and 49 seconds." • **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). Select this check box 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. **NOTE**: Ringtone files for custom MOH should be placed under

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	/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/).
Disconnection Warning Interval	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.
Type of Disconnection Warning	Choose one of the following types of "time left" warnings: • **beep** – play "beeping" sound; • **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
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 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. For example, assuming that the country code is 44, we can apply 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

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	provisioned for on IP phones.
Use Account Information Provided by PortaSIP	If this option is enabled, an IVR application securely identifies a user if his account was originally authenticated by the PortaSIP® Switching Server instead of attempting to authenticate the customer's phone number (ANI). Only enable this for access numbers dialed from SIP phones connected to PortaSwitch® (calls within your VoIP network). This option is useful in cases when normal ANI authentication by the IVR application can either be very difficult or unsafe (as the calling account may be mismatched): • The end user makes a call from his / her IP phone with a CLI that is either hidden or different from the authenticated account ID. • The account ID is an internal ANI number (e.g. 000999123) or even a non-E164 identifier like an email address that can't be used as a normal CLI for outgoing calls.
Languages	This option allows the user to select the language(s) to be used for voice prompts. Click the **Wizard** icon to open the language selection dialog box and define languages in a specific order, and then upon reaching an access number, the end user can choose one of these languages.
Conf Language List is Preferred	If this check box is selected, 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. The Media Server 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 /var/lib/porta-um/prompts/brands/my_company/en/ and specify "my_company" in this field.
Minimum Card	Minimum length of a card number.

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Length	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.
Minimum Sum	Specify minimum amount of funds to be transferred from a customer's credit card.
Credit Accounts Balance Announcement	You can configure your IVR applications to announce the current balance *for credit accounts* in one of two modes: • **Funds / balance** – For prepaid customers (those who have a credit limit of 0 or less), announce "available funds" (calculated as credit limit minus balance) and announce the current balance for others (postpaid customers); • **Balance** – Always announce balance (this is the classic mode, ideal for typical postpaid customers). • Note that a third **Backward compatibility** mode is enabled by default. It preserves the announcements exactly the way they worked in releases prior to MR35. In *APPENDIX B. Balance Announcements*, there are balance announcement examples for different IVR configurations.
Announce Credit Limit	If this option is enabled, the IVR will announce the credit limit as well as the account balance (applicable when **Credit Accounts Balance Announcement** is set to "balance" or "funds / balance").
Customer Center Number	Specify the number where a user may reach an operator.

Account Top-up via Voucher

Field	Description
ANI Translation Rule	Caller identification (ANI or CLI number) is usually delivered in the "local" format, without the

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	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. For example, assuming that the country code is 44, we can apply 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.
Use Account Information Provided by PortaSIP	If this option is enabled, an IVR application securely identifies a user if his account was originally authenticated by the PortaSIP® Switching Server instead of attempting to authenticate the customer's phone number (ANI). Only enable this for access numbers dialed from SIP phones connected to PortaSwitch® (calls within your VoIP network). This option is useful in cases when normal ANI authentication by the IVR application can either be very difficult or unsafe (as the calling account may be mismatched): • The end user makes a call from his / her IP phone with a CLI that is either hidden or different from the authenticated account ID. • The account ID is an internal ANI number (e.g. 000999123) or even a non-E164 identifier like an email address that can't be used as a normal CLI for outgoing calls.
Prompts Brand	IVR prompts can be customized per access number. Specify the directory where the custom prompt files are located. The Media Server 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

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	under /var/lib/porta-um/prompts/brands/my_company/en/ and specify "my_company" in this field.
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.
Allow to transfer the balance from debit accounts	Select this check box to allow a customer to top-up his main account using another pre-paid card (i.e. from a debit account) in PortaBilling® instead of a top-up voucher. 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).
Credit Accounts Balance Announcement	You can configure your IVR applications to announce the current balance *for credit accounts* in one of two modes: • Funds / balance – For prepaid customers (those who have a credit limit of 0 or less), announce "available funds" (calculated as credit limit minus balance) and announce the current balance for others (postpaid customers); • Balance – Always announce balance (this is the classic mode, ideal for typical postpaid customers). Note that a third "Backward compatibility" mode is enabled by default. It preserves the announcements exactly the way they worked in releases prior to MR35. In *APPENDIX B. Balance Announcements*, there are balance announcement examples for different IVR configurations.
Announce Credit Limit	If enabled, the IVR will announce the credit limit as well as the account balance (applicable when **Credit Accounts Balance Announcement** is set to "balance" or "funds / balance").

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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 the Media Server.
- The Media Server 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 the Switching Server (the CLD is 12065551234).
- The Switching Server 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 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. For example, assuming that the country code is 44, we can apply 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

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	will ensure that these accounts are not confused with the actual phone numbers provisioned for on IP phones.
Languages	This option allows the user to select the language(s) to be used for voice prompts. Click the **Wizard** icon to open the language selection dialog box and define languages in a specific order, and then upon reaching an access number, the end user can choose one of these languages.
Conf Language List is Preferred	If this check box is selected, 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. The Media Server 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 /var/lib/porta-um/prompts/brands/my_company/en/ and specify "my_company" in this field.
Play Welcome Message	If this option is enabled, the 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 **Services and Rating** 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: • **none** – there is no rounding, so the time is announced including seconds, e.g. "Five minutes and 49 seconds." • **up** – the time is rounded up to the nearest

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	minute, so 5:49 will be announced as "Six minutes."

One's Own Voice Mailbox Access

Field	Description
Languages	This option allows the user to select the language(s) to be used for voice prompts. Click the **Wizard** icon to open the language selection dialog box and define languages in a specific order, and then upon reaching an access number, the end user can choose one of these languages.
Use Account Information Provided by PortaSIP	If this option is enabled, an IVR application securely identifies a user if his account was originally authenticated by the PortaSIP® Switching Server instead of attempting to authenticate the customer's phone number (ANI). Only enable this for access numbers dialed from SIP phones connected to PortaSwitch® (calls within your VoIP network). This option is useful in cases when normal ANI authentication by the IVR application can either be very difficult or unsafe (as the calling account may be mismatched): • The end user makes a call from his / her IP phone with a CLI that is either hidden or different from the authenticated account ID. • The account ID is an internal ANI number (e.g. 000999123) or even a non-E164 identifier like an email address that can't be used as a normal CLI for outgoing calls.

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

Account Self-Care

Field	Description
Languages	This option allows the user to select the language(s) to be used for voice prompts. Click the **Wizard** icon to open the language

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	selection dialog box and define languages in a specific order, and then upon reaching an access number, the end user can choose one of these languages.
Prompts Brand	IVR prompts can be customized per access number. Specify the directory where the custom prompt files are located. The Media Server 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 /var/lib/porta-um/prompts/brands/my_company/en/ and specify "my_company" in this field.
Follow Me Timeout	This option specifies how long the IP phone will ring before the call goes to follow-me numbers.
Play Welcome Message	If this option is enabled, the IVR will play the welcome message to the user once he reaches an access number.

Balance information

Field	Description
Use Early Media	If this option is enabled, the IVR will announce the account's balance without establishing a call so the end user is not charged by their telco.
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. For example, assuming that the country code is 44, we can apply 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.
Use Account Information Provided by	If this option is enabled, an IVR application securely identifies a user if his account was originally authenticated by the PortaSIP®

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PortaSIP	Switching Server instead of attempting to authenticate the customer's phone number (ANI). Only enable this for access numbers dialed from SIP phones connected to PortaSwitch® (calls within your VoIP network). This option is useful in cases when normal ANI authentication by the IVR application can either be very difficult or unsafe (as the calling account may be mismatched): • The end user makes a call from his / her IP phone with a CLI that is either hidden or different from the authenticated account ID. • The account ID is an internal ANI number (e.g. 000999123) or even a non-E164 identifier like an email address that can't be used as a normal CLI for outgoing calls.
Prompts Brand	IVR prompts can be customized per access number. Specify the directory where the custom prompt files are located. The Media Server 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 /var/lib/porta-um/prompts/brands/my_company/en/ and specify "my_company" in this field.

Call forwarding management

Field	Description
Prompts Brand	IVR prompts can be customized per access number. Specify the directory where the custom prompt files are located. The Media Server 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 /var/lib/porta-um/prompts/brands/my_company/en/ and specify "my_company" in this field.

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Conferencing

Field	Description
Languages	This option allows the user to select the language(s) to be used for voice prompts. Click the **Wizard** icon to open the language selection dialog box and define languages in a specific order, and then upon reaching an access number, the end user can choose one of these languages.
Maximum Login Attempts	Specify maximum number of attempts for a user to enter a card number. By default: 3.
Prompts Brand	IVR prompts can be customized per access number. Specify the directory where the custom prompt files are located. The Media Server 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 /var/lib/porta-um/prompts/brands/my_company/en/ and specify "my_company" in this field.
Record with Review	If this option is enabled, the IVR will play back the name that the end user recorded; if preferred, it can be re-recorded.
Play Welcome Message	If enabled, the IVR will play the welcome message to the user once he reaches an access number.

Pass-through IVR

This IVR application introduces prepaid card calling features to traditional residential VoIP service. For example, the application can play a "time left" warning when a specified number of seconds is left – before the call is disconnected. It can also announce the maximum call duration allowed to the dialed destination and then connect the call. In addition, the Pass-through IVR permits additional verification of the outgoing toll (e.g. on international) calls.

The administrator controls whether the outgoing calls are made in a normal fashion or if the calls are handled by the Pass-through IVR application. This is specified in the account's **Service Features** configuration section.

Please consult the *Voiceover Announcements* section in the **PortaSIP® Switching Server Administrator Guide** for more information.

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Field	Description
Maximum Length of the Auth Code	Maximum allowable length for an authorization code. By default: 4.
Maximum Login Attempts	Specify maximum number of attempts for a user to enter a card number. By default: 3.
Announce Time	When user inputs the destination, the IVR announces the maximum call duration allowed to this destination and then connects the call. By default this feature is disabled.
Type of Disconnection Warning	Choose one of the following types of "time left" warnings: • **beep** – play "beeping" sound. • **voice** – announce the remaining time in seconds.
Disconnection Warning Interval	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.
Authorize Non-Free Calls	Enable this option to perform an additional verification for toll calls. This will prevent unauthorized usage of phones available in public areas. Furthermore, if a single phone is shared among multiple users, then the outgoing calls can be billed to separate accounts; this will allow for accountability for each user based on the input authorization code. By default: disabled. Note that if this option is disabled and the **Treat Auth Code as Part of Account ID** option is enabled at the same time, the user will only be able to make toll-free calls.
Use Associated Number to Produce Auth ID	When this option is enabled, the application checks for an associated number specified for the account. If there is one, the authorization is done using the associated number and authorization code input by the user (i.e. account's associated number followed by # followed by authorization code, e.g. 74951234567#1234) will be used to authenticate the call). By default this is disabled.

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Treat Auth Code as Part of Account ID	Enable this feature if a single phone is shared among multiple users and there is a need to bill outgoing calls to separate accounts; the account with ID *AccountID#AuthCode* (account ID followed by # followed by authorization code, e.g. 12121234567#1234) will be used to authenticate the call. If this feature is disabled, the input authorization code will be compared with the account's **Service Unblock Code** (defined in the **Fraud Detection** tab) to confirm that a legitimate user is attempting to make a call. By default this feature is enabled.

Payment remittance – TransferTo

Field	Description
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.
Play Welcome Message	If this option is enabled, the IVR will play the welcome message to the users once they reach an access number.
Prompts Brand	IVR prompts can be customized per access number. Specify the directory where the custom prompt files are located. The Media Server 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 /var/lib/porta-um/prompts/brands/my_company/en/ and specify "my_company" in this field.

Simplified account self-care

Field	Description
Play Announcements	If enabled, the IVR will play the announcements without establishing a call so the end user will not be charged by their telco.
Use Account	If this option is enabled, an IVR application

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Information Provided by PortaSIP	securely identifies a user if his account was originally authenticated by the PortaSIP® Switching Server instead of attempting to authenticate the customer's phone number (ANI). Only enable this option for access numbers dialed from SIP phones connected to PortaSwitch® (calls within your VoIP network). This feature is useful in cases when normal ANI authentication by the IVR application can either be very difficult or unsafe (as the calling account may be mismatched): • The end user makes a call from his / her IP phone with a CLI that is either hidden or different from the authenticated account ID; • The account ID is an internal ANI number (e.g. 000999123) or even a non-E164 identifier like an email address that can't be used as a normal CLI for outgoing calls.
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. For example, assuming that the country code is 44, we can apply 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.
Etalon Account	Specify the account ID here to use its configuration properties as an etalon for all accounts.
Prompts Brand	IVR prompts can be customized per access

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	number. Specify the directory where the custom prompt files are located. The Media Server 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 /var/lib/porta-um/prompts/brands/ my_company/en/ and specify "my_company" in this field.

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APPENDIX B. SMS Callback Commands

Command syntax

- Most of the commands have the syntax **`<code> <value>`**, where <code> is an abbreviated parameter code such as **`R`** or **`DN`** and <value> is a required parameter. Note that <code> and <value> can be separated by a space; if more than one <value> is to be provided, the individual elements can be separated by a space as well.
- The callback trigger is not case-sensitive, so **`DN 420212345`** is equal to **`dn 420212345`**
- Phone numbers should be specified in the E.164 format (country code, followed by area code, followed by the phone number). You should not put 00 or + in front of the phone number; if you do so, the callback trigger will automatically strip them off.

In the description below, the following terminology is used:

- **Source phone number** means the phone number that was used to send the SMS message.
- **Registered phone number** means the phone number that is subscribed to the service (there is an account in billing with such an ID).

Registration of a new account

Format of the SMS message: **`R`** or **`R NNNNNNN`** (note the space after R!)

Action: The system will take the source number and attempt to create a new account with such an ID (parameters such as product, billing model, account prefix, etc. will be taken from the config file). The new account is created with a zero balance; but if the voucher code NNNNNNN is provided, the new account will immediately be recharged using this voucher.

Following registration, the user will receive an SMS message confirming his subscription and providing him with basic information on the use of the service.

If an account with such an ID already exists and no voucher ID is provided, this command will be ignored; otherwise it is executed as an account recharge.

Examples:
R

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```
R 234873640324
```

Initiate callback call – type A

Connect a registered phone number with the provided destination number.

Format of the SMS message: **NNNNNNN** or **DN NNNNNNN**

Action: Establish a call to a registered phone number. When the call is connected, establish a call to destination number NNNNNNN and bridge the two calls together. This is the most commonly used service, so it allows for easy use of an SMS message: the user just includes the phone number.

Examples:
```
6581433653
DN 420212345678
```

Initiate callback call – type B

Connect a provided source phone number with a provided destination number.

This service is typically used when SMS messages can be sent from the user's phone (the one registered in the system) though he does not wish to receive incoming calls to it. For example, he may be abroad, so an incoming GSM call would be very expensive for him. Instead, he can instruct the system to call him on a local access number.

Format of the SMS message: **SN XXXXXXX DN NNNNNNN**

Action: Establish a call to phone number XXXXXXX. When the call is connected, establish a call to destination number NNNNNNN and bridge the two calls together.

Examples:
```
SN 6581433653 DN 420212345678
```

Initiate callback call – type C

It may happen that the user is not able to send SMS messages from his mobile phone (phone not working, prepaid balance exceeded, etc.). Instead he may use another phone to send an SMS message. In this case, of course, in addition to providing his original phone number he must also provide a valid password, so that no misuse of this service can occur.

Format of SMS message: **P AAAA YYYY DN NNNNNNN SN XXXXXXX** or **P AAAA YYYY DN NNNNNNN**

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Action: Authenticate by account AAAA with password YYYY. If source number XXXXXXX is not provided, then assume that the first call must be established to the registered number. Establish the first call. When the call is connected, establish a call to destination number NNNNNNN and bridge the two calls together.

Examples:
```
P 6192345678 1ak45 SN 6581433653 DN 420212345678
P 6192345678 1ak45 DN 420212345678
```

Check current balance

Format of SMS message: **CB**

Action: If the phone number from which the SMS was sent is registered, send back the current balance information by SMS.

Examples:
```
CB
```

Change current password

Format of the SMS message: **CP XXXXX YYYYY**

Action: If the old password XXXXX provided is correct, change the password for the account with the registered phone number to YYYYY.

Examples:
```
CP 1ak45 andrew1
```

APPENDIX C. Voice Applications Settings

One's Own Voice Mailbox Access

Field	Description
Access Code	This allows you to apply a special rating to outgoing calls made by customers from their voice mailboxes. For example, if an administrator specifies the **VOICEMAIL** access code in the application settings, then PortaBilling® will use a rating entry with this very access code from the account's product to authorize outgoing calls from users' voice mailboxes.

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APPENDIX D. Balance Announcements

In the table below you can see available variants for balance announcements with the **Credit Accounts Balance Announcement** option set to **backward compatibility** or **funds / balance** value:

	Backward Compatibility	Funds/Balance
Balance = 10; Credit Limit = 300	"You have 290 dollars."	"You have 290 dollars available."
Balance = 10; Credit Limit = n/a	"You have unlimited available funds."	"Your balance is 10 dollars."
Balance = -10; Credit Limit = 300	"You have 310 dollars."	"Your balance is minus 10 dollars."
Balance = -10; Credit Limit = 0	"You have 10 dollars."	"You have 10 dollars available."

In the following table you can see available variants for balance announcements with the **Credit Account Balance Announcement** option set to **balance** value (with the **Announce Credit Limit** option enabled / disabled):

	Balance	
	Without Credit Limit Announcement	With Credit Limit Announcement
Balance = 10; Credit Limit = 300	"Your balance is 10 dollars."	"Your balance is 10 dollars, your credit limit is 300 dollars."
Balance = 10; Credit Limit = n/a	"Your balance is 10 dollars."	"Your balance is 10 dollars, there is no credit limit."
Balance = -10; Credit Limit = 300	"Your balance is minus 10 dollars."	"Your balance is minus 10 dollars, you credit limit is 300 dollars."
Balance = -10; Credit Limit = 0	"Your balance is minus 10 dollars."	"Your balance is minus 10 dollars, you credit limit is 0 dollars."

APPENDIX E. Supported DTMF Methods

PortaSIP® supports the following DTMF methods:

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- Inband (as normal audio tones in the RTP stream with no special coding or markers).
- RFC 2833 (as named telephone events in the RTP stream).
- SIP INFO method – This is the preferred DTMF method.

Please note that to use these DTMF methods they must be enabled on the configuration server web interface.

APPENDIX F. Fax Transmission-Supported Codecs

PortaSIP® supports high-quality fax transmission in the following codecs:

- T.38
- G.711 (G.711a and G.711u)

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APPENDIX G. Supported Languages

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

- No – Not supported.
- Yes – Fully supported.
- P – Partially supported (incomplete prompts list or prompts are out of date).

NOTE: With every release, we update the English prompts. For other languages, we usually do batch updates of prompts every few releases because of high studio recording costs. If you wish to expedite the prompts recording for a specific language, please contact our sales team to discuss our agenda.

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)	Persian (fa)
Account top-up via credit card	Yes	Yes	No	No	No	No	No	No	Yes
Simplified account self-care	Yes	Yes	No	No	No	No	No	No	Yes
Call Forwarding management	Yes	Yes	Yes	No	Yes	No	Yes	No	Yes
Account Self-care	Yes	Yes	Yes	No	Yes	No	No	No	Yes
Account top-up via voucher	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No	Yes
Balance information	Yes	Yes	Yes	No	No	No	Yes	No	Yes
One's own voice mailbox access (with PIN protection)	Yes	Yes	P	No	P	No	P	No	Yes
One's own voice mailbox access	Yes	Yes	P	No	P	No	P	No	Yes
Payment Remittance	Yes	Yes	No	No	No	No	No	No	Yes

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LANGUAGE / APPLICATION	Mandarin Chinese (cm)	Greek (el)	Arabic (ar)	Swedish (sv)	German (de)	Italian (it)	Myanmar (my)	Turkish (tr)
Account top-up via credit card	Yes	No	Yes	Yes	Yes	Yes	No	Yes
Simplified account self-care	Yes	No	Yes	Yes	Yes	Yes	No	Yes
Call Forwarding management	Yes	No	Yes	Yes	Yes	Yes	No	Yes
Account Self-care	Yes	No	Yes	Yes	Yes	Yes	No	Yes
Account top-up via voucher	Yes	No	Yes	Yes	Yes	Yes	Yes	Yes
Balance information	Yes	No	Yes	Yes	Yes	Yes	Yes	Yes
One's own voice mailbox access (with PIN protection)	Yes	No	Yes	Yes	Yes	Yes	No	Yes
One's own voice mailbox access	Yes	No	Yes	Yes	Yes	Yes	No	Yes
Payment Remittance	No	No	No	No	Yes	Yes	No	Yes

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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)	Persian (fa)
One-stage calling	Yes	Yes	Yes	Yes	Yes	Yes	Yes	P	Yes
Callback calling	Yes	Yes	Yes	Yes	Yes	Yes	Yes	P	Yes
Prepaid Card Calling	Yes	Yes	Yes	P	P	Yes	Yes	P	Yes

LANGUAGE / APPLICATION	Mandarin Chinese (cm)	Greek (el)	Arabic (ar)	Swedish (sv)	German (de)	Italian (it)	Myanmar (my)	Turkish (tr)
One-stage calling	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Callback calling	Yes	P	Yes	Yes	Yes	Yes	Yes	Yes
Prepaid Card Calling	Yes	P	Yes	Yes	Yes	Yes	P	Yes

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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)	Persian (fa)
Conferencing	Yes	No	Yes	No	No	No	No	No	No

LANGUAGE / APPLICATION	Mandarin Chinese (cm)	Greek (el)	Arabic (ar)	Swedish (sv)	German (de)	Italian (it)	Myanmar (my)	Turkish (tr)
Conferencing	No	No	No	No	No	No	No	Yes

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Supported currency

If account's currency is not supported by the IVR application, then IVR uses a default currency:

Language	Supported currency (ISO 4217 code)	Default currency (ISO 4217 code)
Arabic	BHD, USD, EUR	BHD
Cantonese Chinese	USD, CNY	USD
English	USD, RUB, EUR, GBP, THB, INR, MYR, BND, AED, IRR, ZAR, CHF, CNY, DKK, MXN, TRY, NGN, BHD, AUD, HKD, NZD, SGD, PKR, JOD, IQD, EGP, PHP, IDR, BDT, CRC	USD
French	USD, EUR, RUB, GBP, THB, INR, MYR, BND, AED	USD
German	USD, RUB, EUR, GBP, CHF	EUR
Greek	USD, EUR	EUR
Hebrew	ILS, USD, EUR	ILS
Italian	USD, RUB, EUR, GBP, CHF	EUR
Mandarin Chinese	USD, CNY	USD
Myanmar	USD, EUR, THB, MMK	MMK
Portuguese	USD, BRL, EUR	EUR
Russian	USD, RUB	RUB
Spanish	USD, EUR, CRC	EUR
Swedish	SEK, USD, RUB, EUR, GBP, THB, INR, MYR, BND, AED, IRR	USD
Swiss French	USD, EUR, FRF	USD

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ISO 4217 code

AED – United Arab Emirates dirham
AUD – Australian dollar
BDT – Bangladeshi taka
BHD – Bahraini dinar
BND – Brunei dollar
BRL – Brazilian real
CHF – Swiss franc
CNY – Renminbi
CRC – Costa Rican colón
DKK – Danish krone
EGP – Egyptian pound
EUR – Euro
FRF – French franc
GBP – Pound sterling
HKD – Hong Kong dollar
IDR – Indonesian rupiah
ILS – Israeli new shekel
INR – Indian rupee
IQD – Iraqi dinar
IRR – Iranian rial
JOD – Jordanian dinar
MMK – Burmese kyat
MXN – Mexican peso
MYR – Malaysian ringgit
NGN – Nigerian naira
NZD – New Zealand dollar
PHP – Philippine peso
PKR – Pakistani rupee
RUB – Russian ruble
SEK – Swedish krona
SGD – Singapore dollar
THB – Thai baht
TRY – Turkish lira
USD – United States dollar
ZAR – South African rand

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APPENDIX H. IVR Applications Migration to the New Media Server

The table below demonstrates which IVR applications have been migrated to the New Media Server.

IVR Application	Migrated (Yes /No/P)
Simplified Account Self-Care	Yes
Callback Calling	Yes
Conferencing	No
Prepaid Card Calling	Yes
Auto Attendant – Dial Directory	Yes
Auto Attendant – DISA	Yes
Auto Attendant – Call Queues	Yes
Call Forwarding Management	Yes
Intro IVR	Yes
One-stage Calling	Yes
Account Top-up via Credit Card	Yes
Account Top-up via Voucher	Yes
Account Self-Care	Yes
Balance Information	Yes
Record Voicemail	Yes
Payment Remittance – TransferTo	Yes
One's Own Voice Mailbox Access	Yes
Pass-through IVR	Yes

- **No** – Not migrated.
- **Yes** – Fully migrated.
- **P** – Partially migrated.

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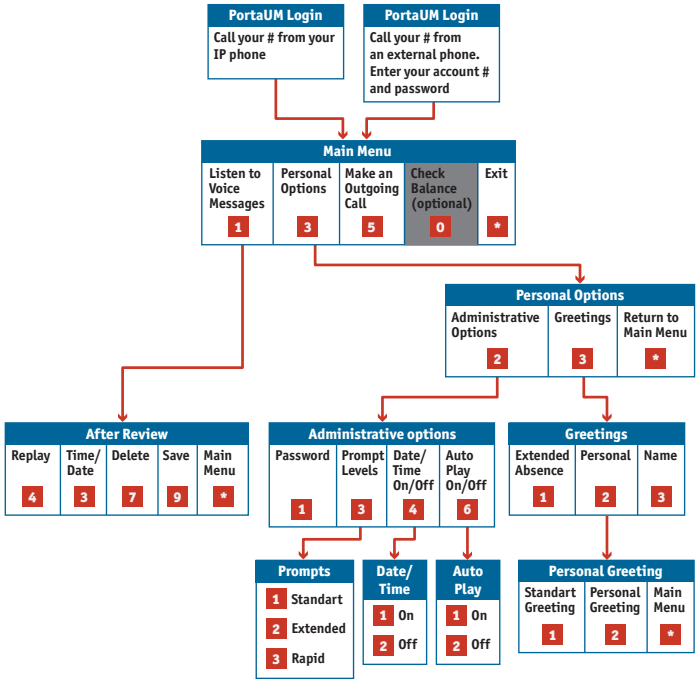
APPENDIX I. IVR Flow Diagrams

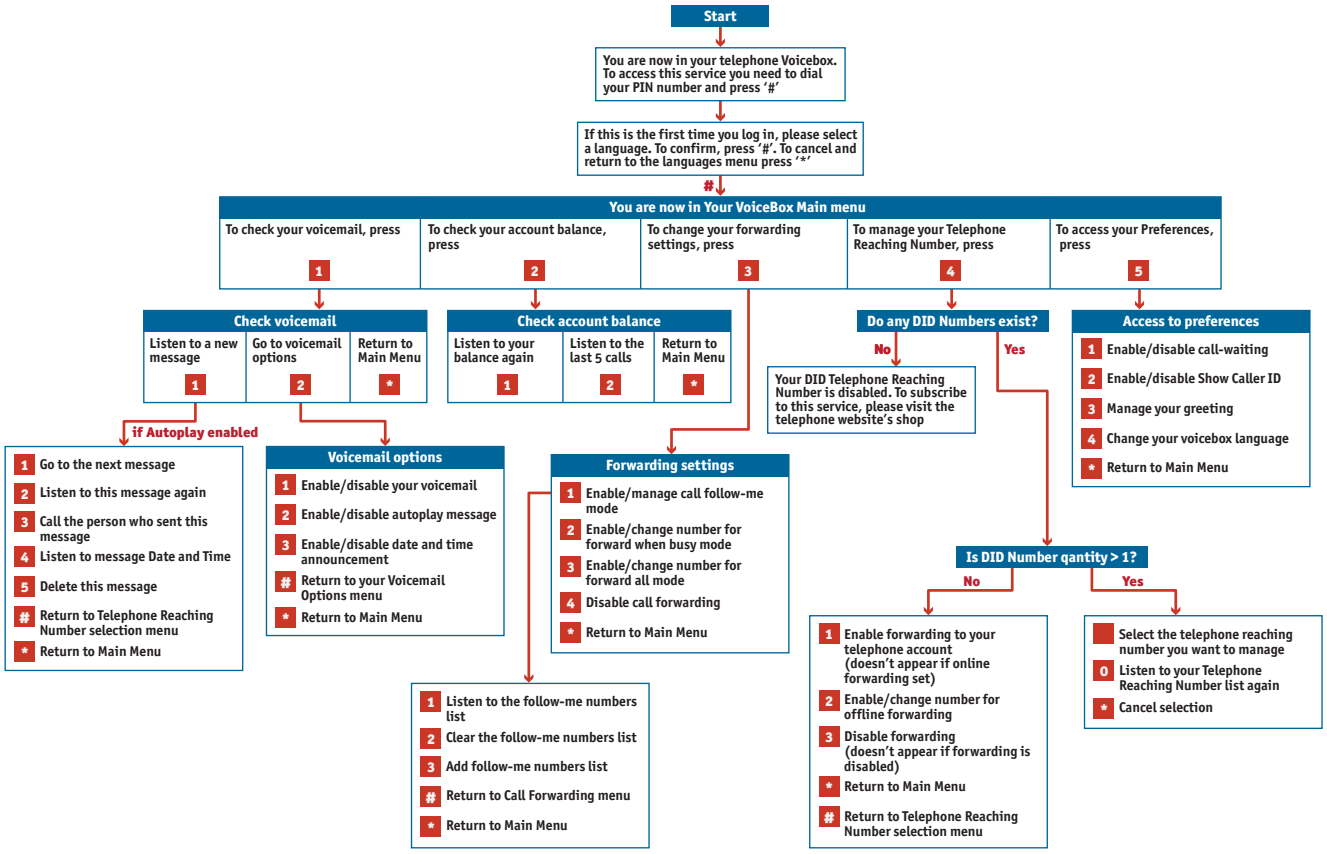
One's Own Voice Mailbox Access IVR (Short Schema)

The Media Server mailbox can be accessed by phone by dialing *98.
Below is an IVR path diagram to help users in navigating and configuring the system. A full IVR diagram can be found in the **One's Own Voice Mailbox Access IVR (Full Schema)** section.



Account Self-care 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 **Account Self-care IVR (Full Schema)** section.



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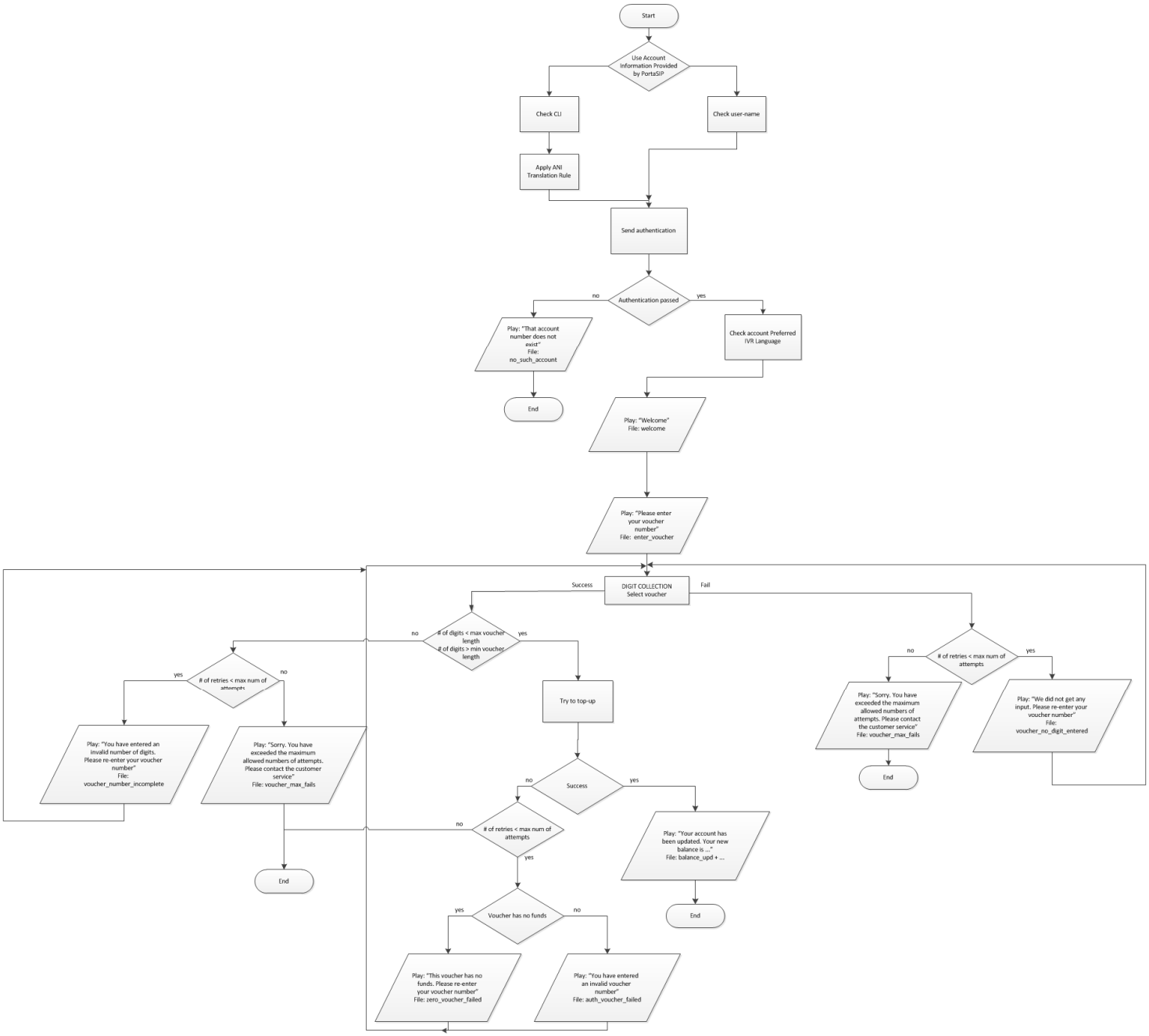
Account Top-up via Voucher IVR



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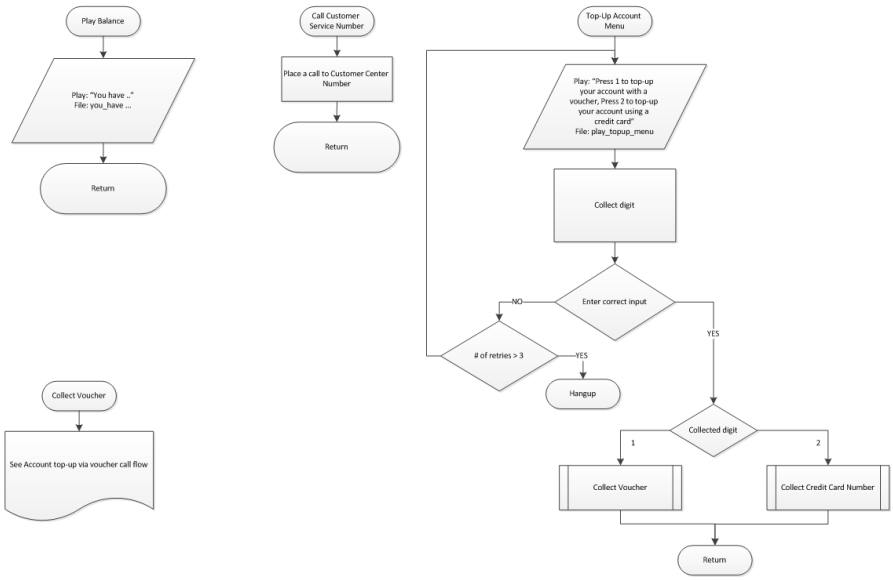
Account Top-up via Credit Card IVR



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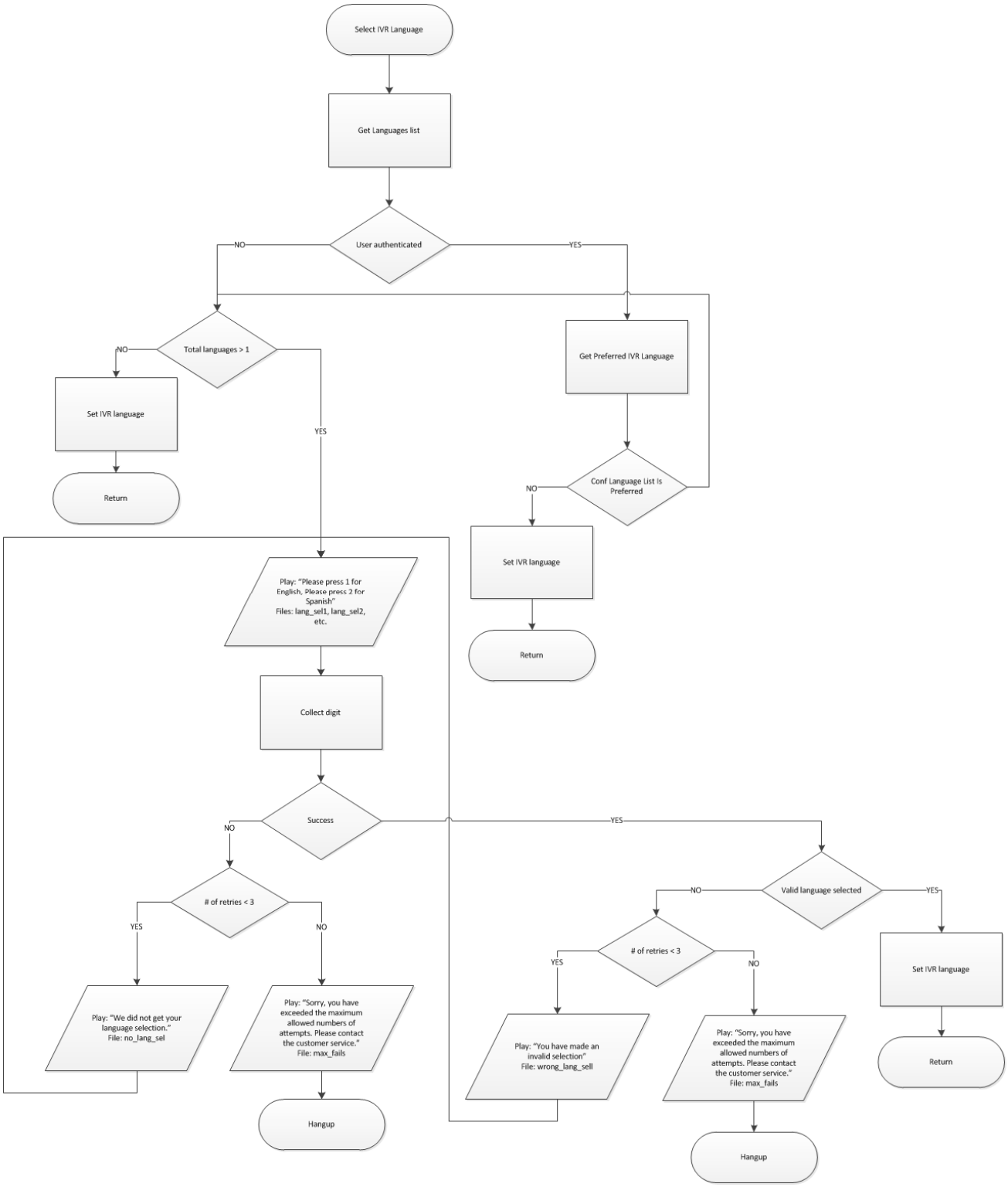
Account top-up via credit card IVR . Select IVR language



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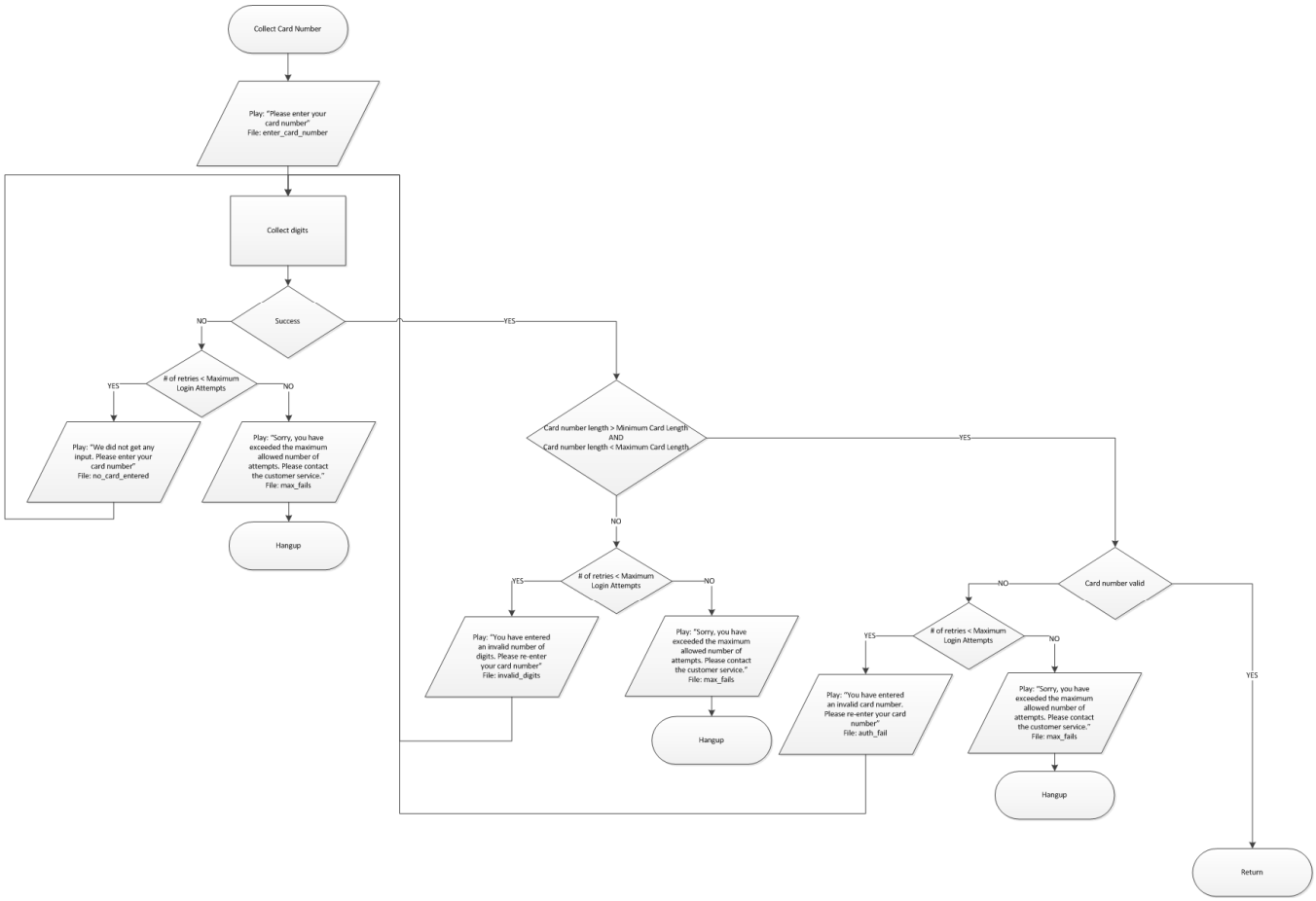
Account top-up via credit card IVR. Collect card number



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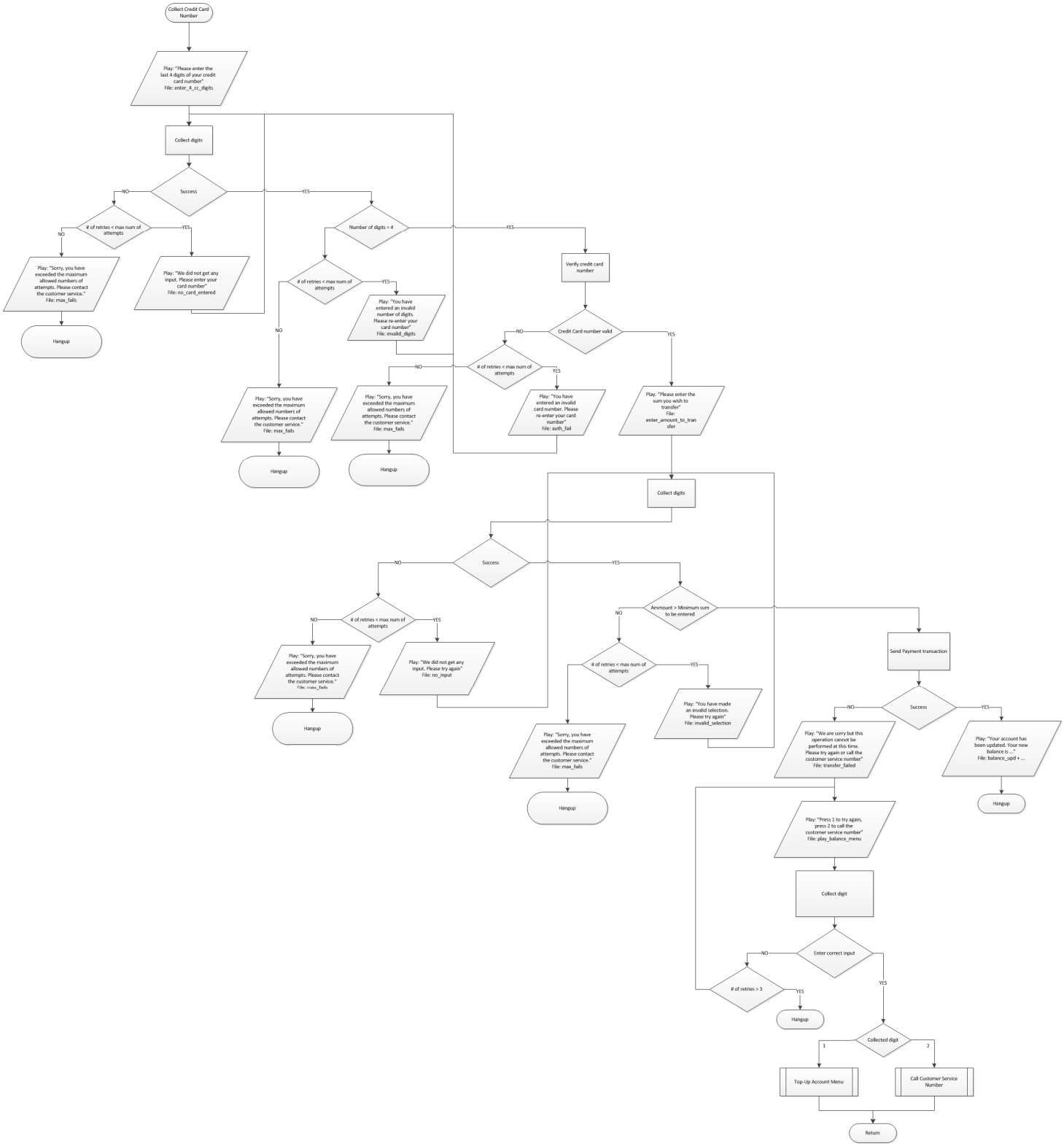
Account top-up via credit card IVR. Collect credit card number



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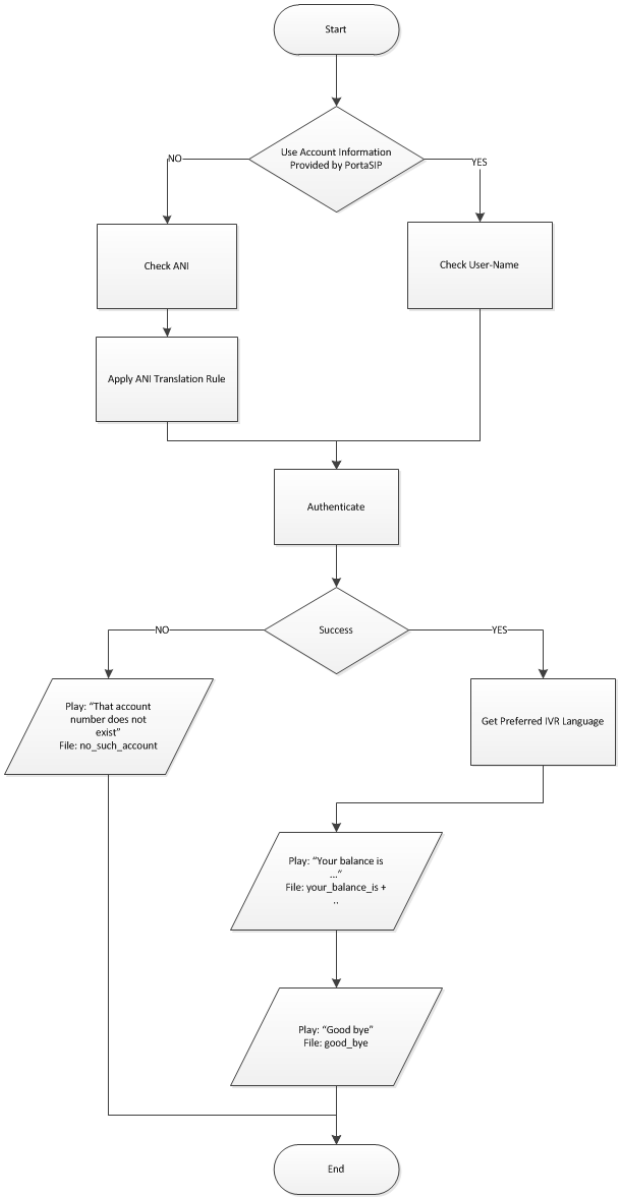
Balance Information



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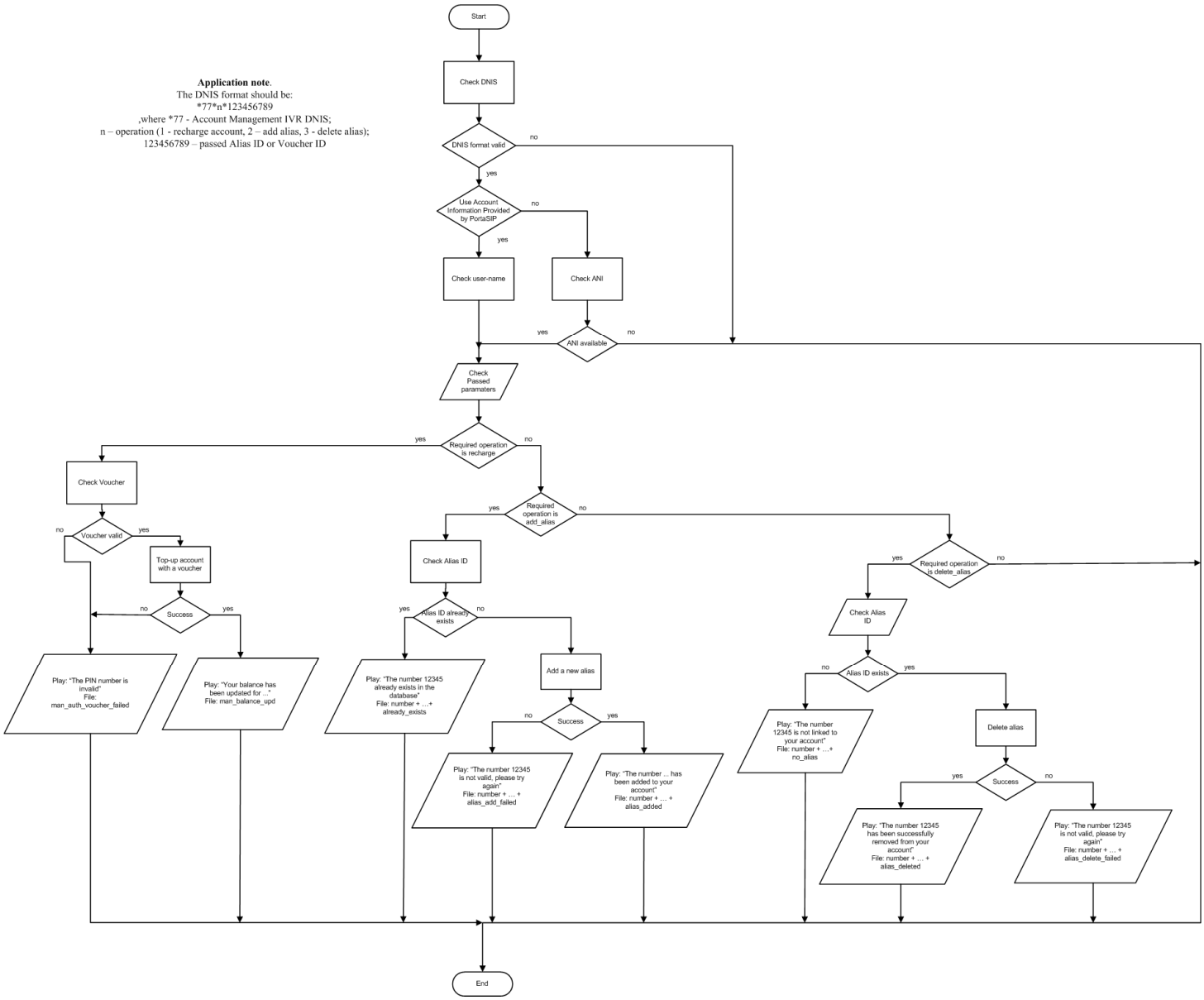
Simplified Account Self-care IVR



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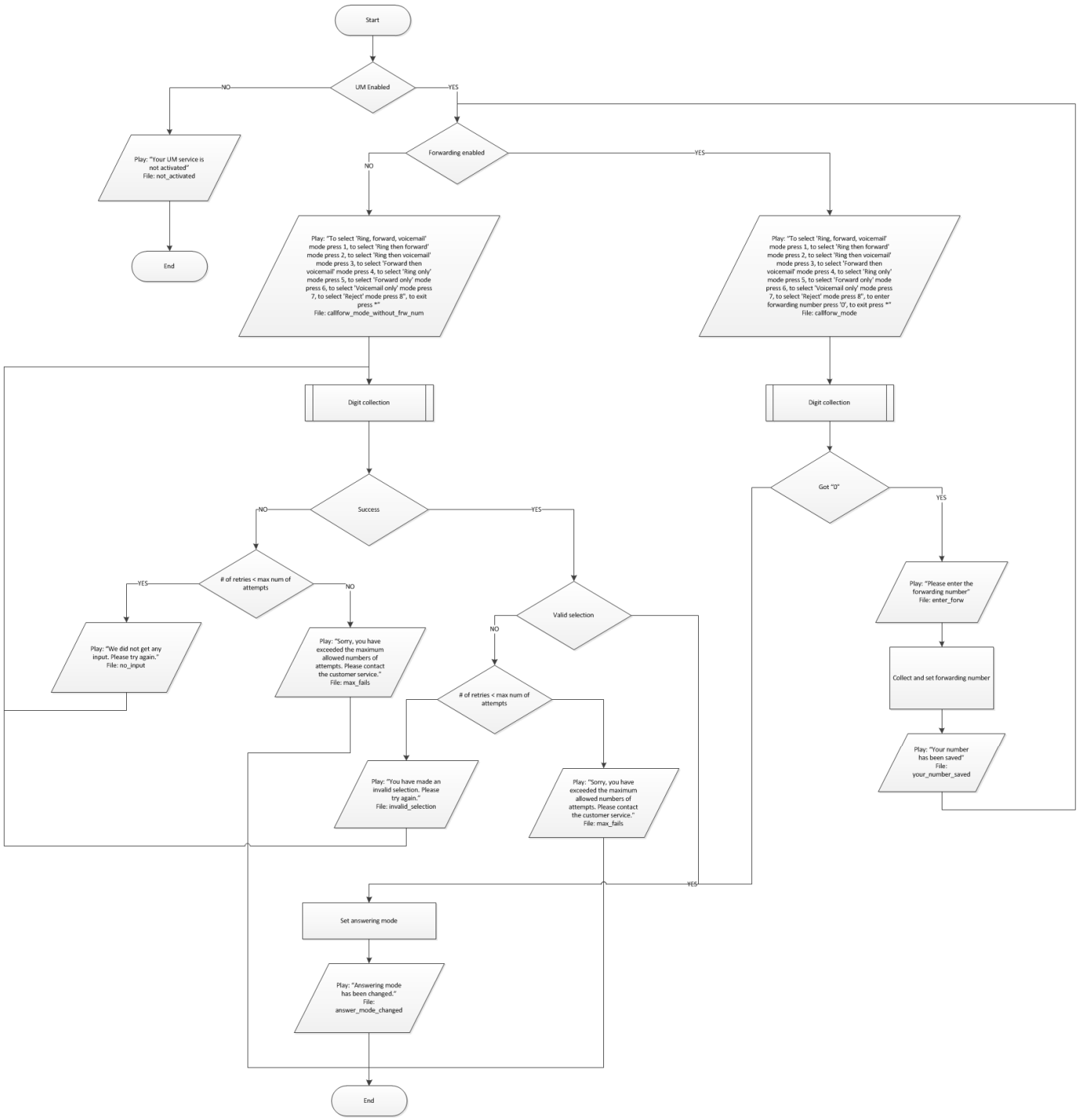
Call Forwarding Management IVR



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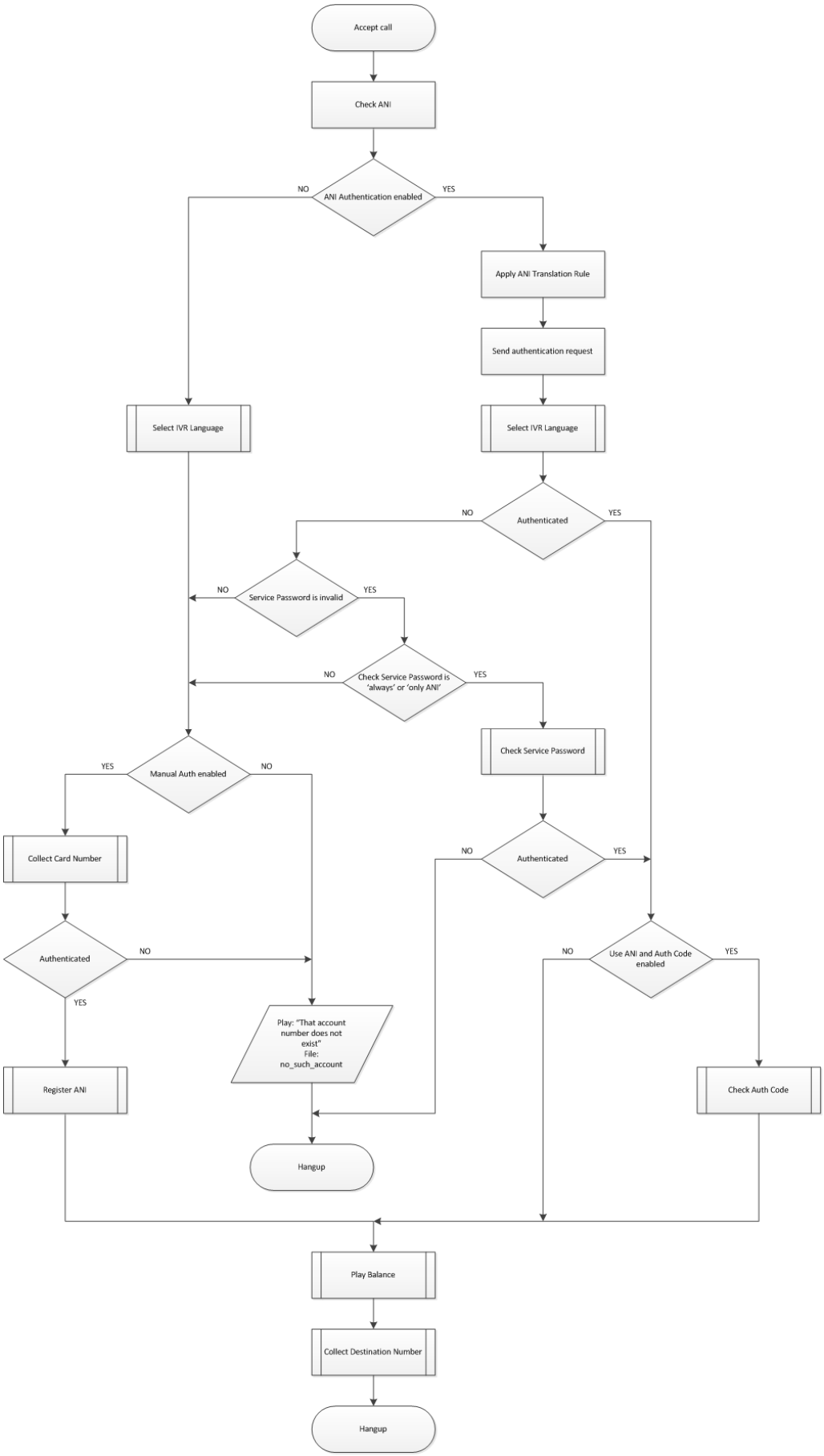
Prepaid card calling IVR



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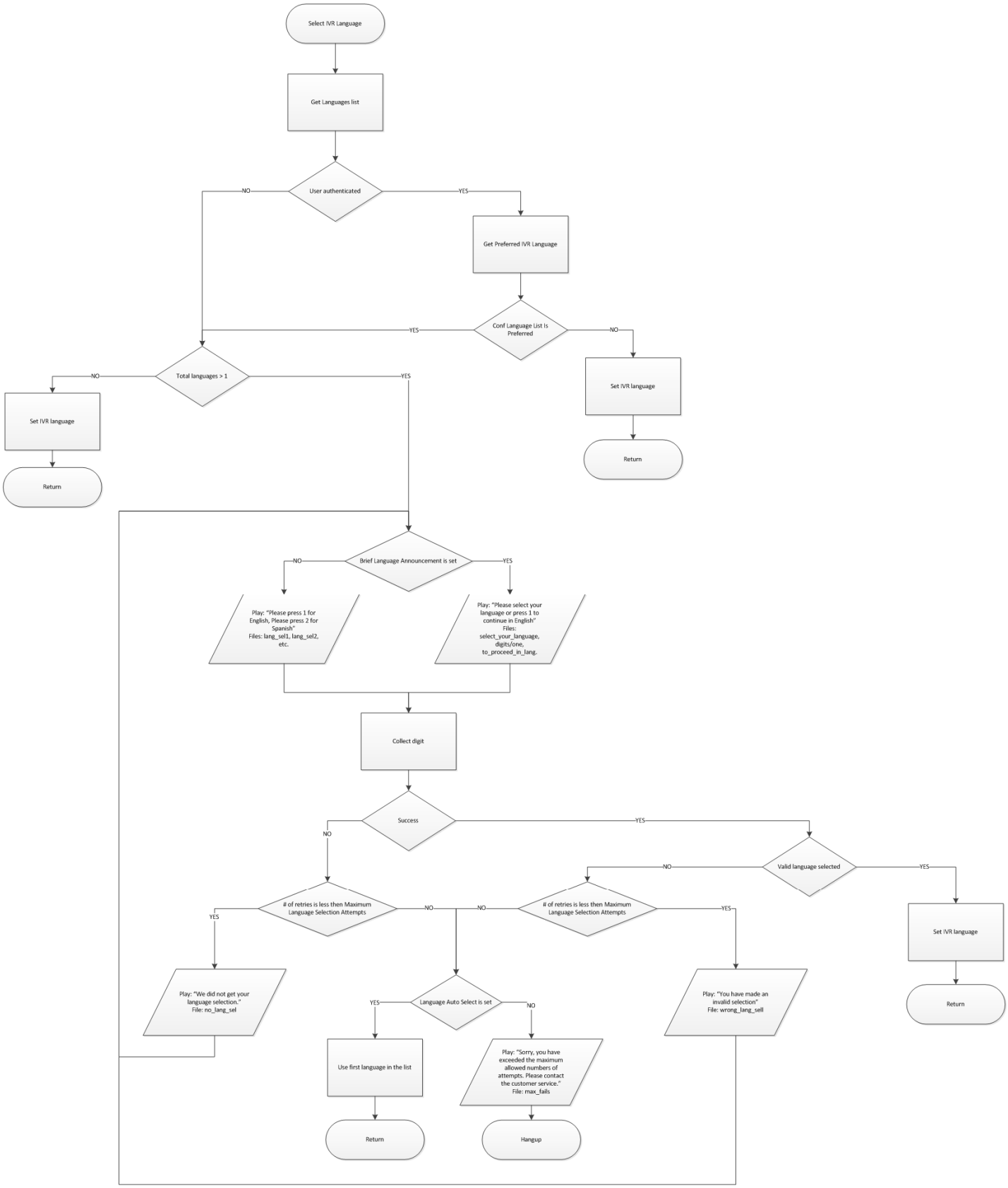
Prepaid card calling IVR. Select IVR language



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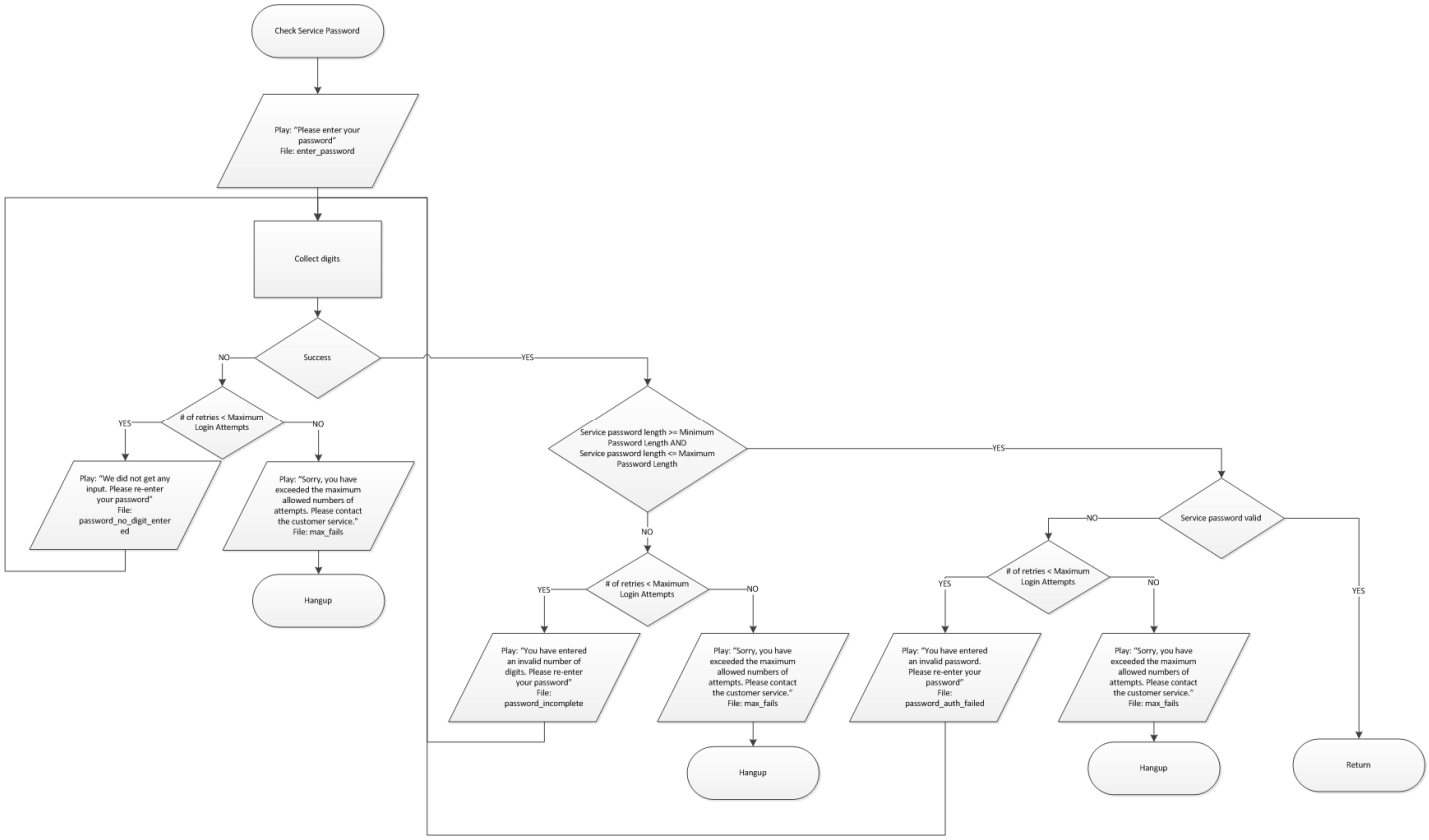
Prepaid card calling IVR. Collect card number



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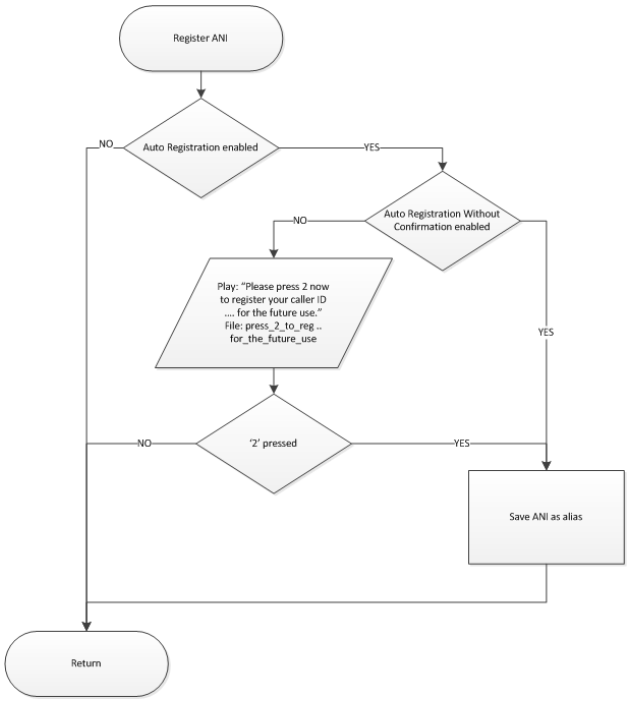
Prepaid card calling IVR. Register ANI



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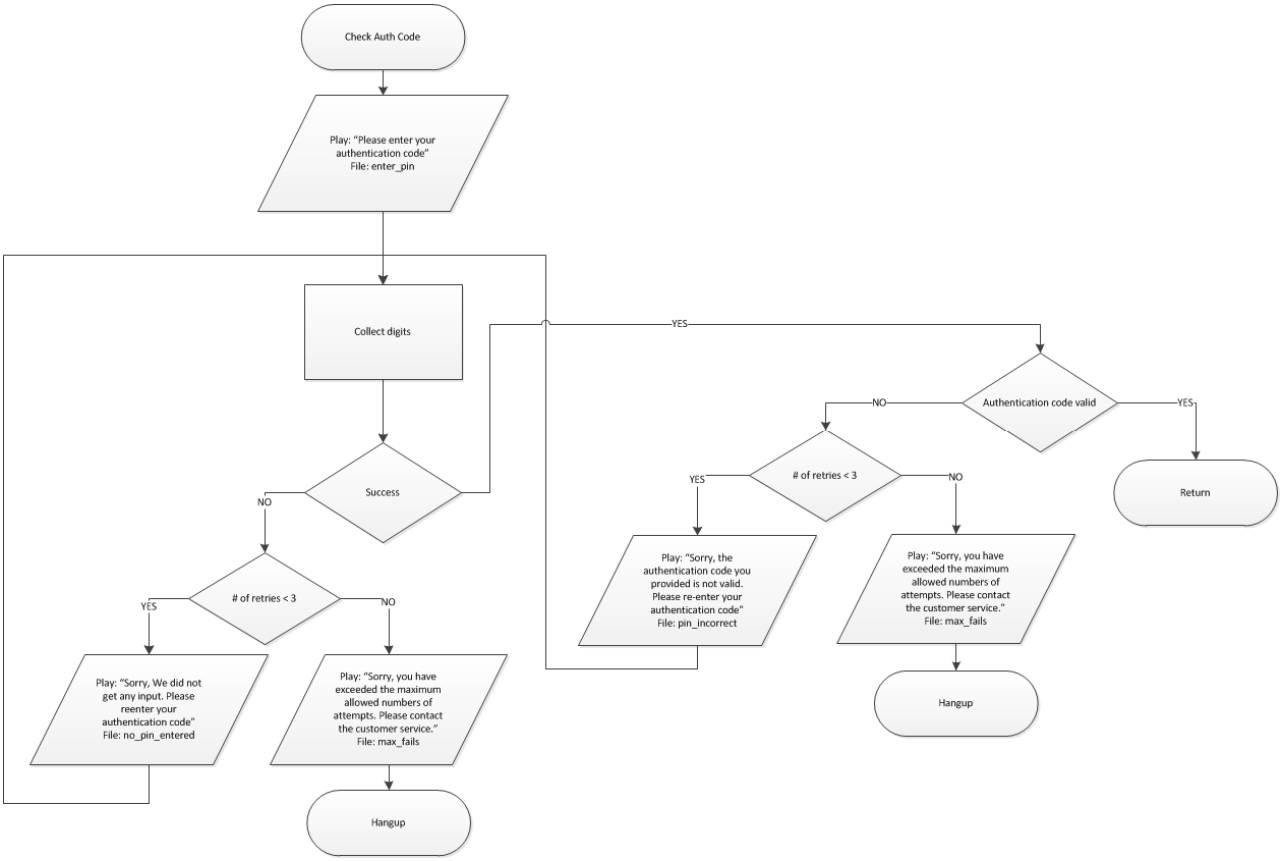
Prepaid card calling IVR. Check authentication code



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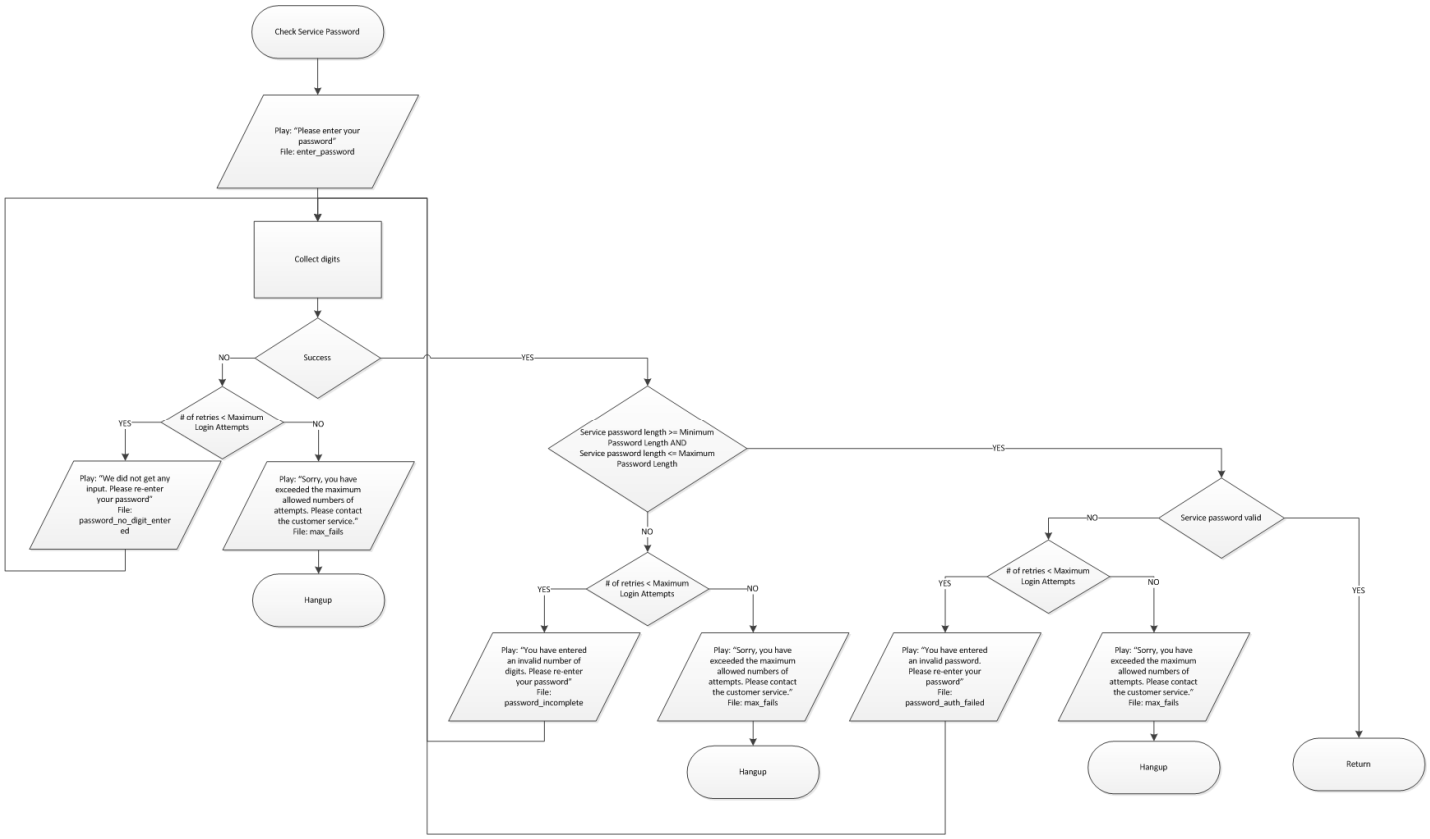
Prepaid card calling IVR. Check service password



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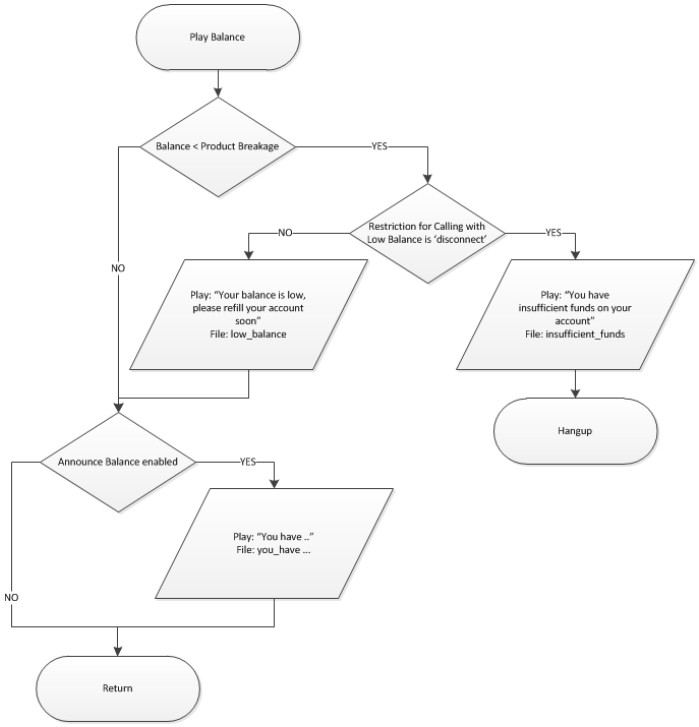
Prepaid card calling IVR. Play balance



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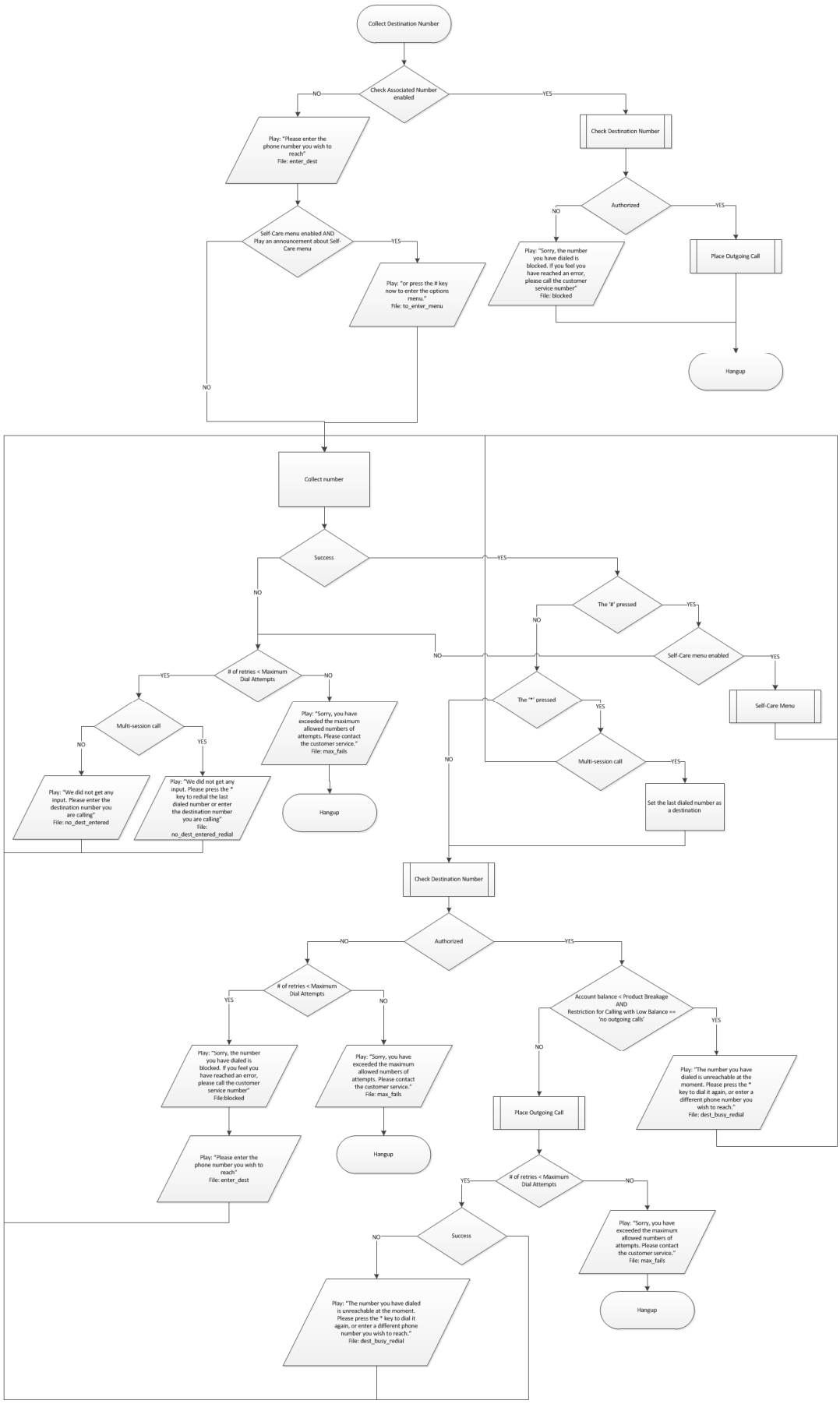
Prepaid card calling IVR. Collect destination number



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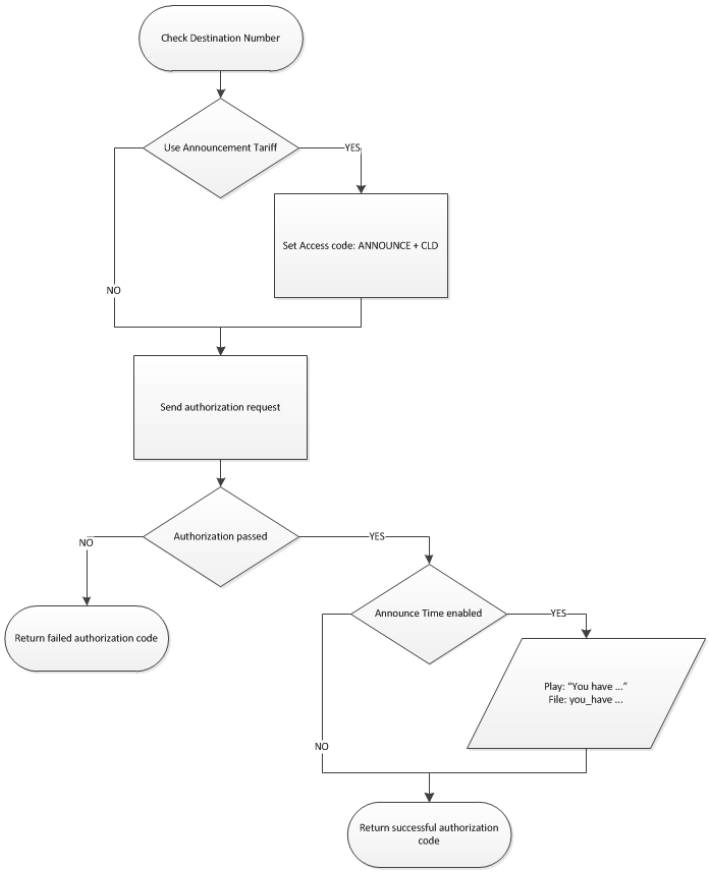
Prepaid card calling IVR. Check destination number



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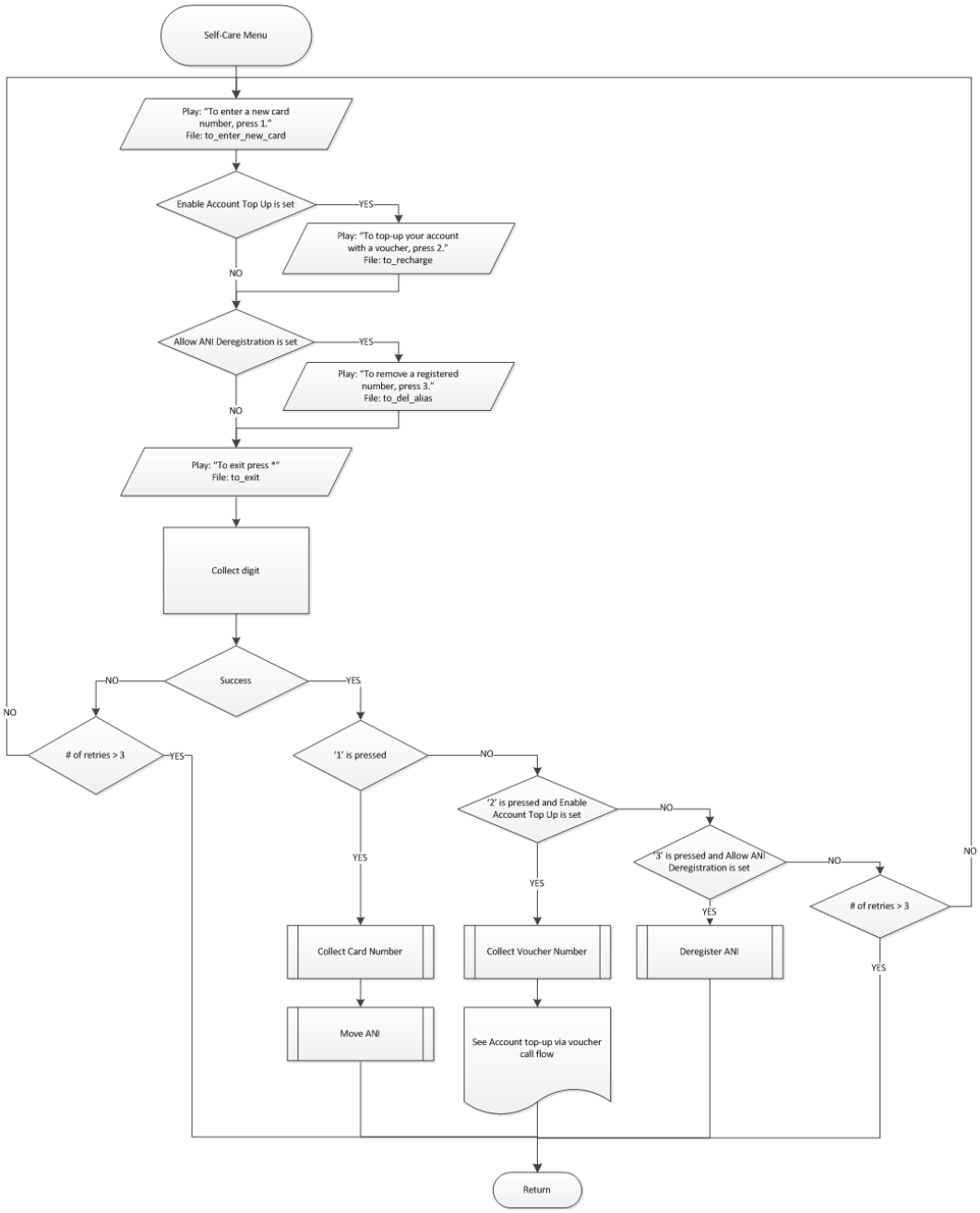
Prepaid card calling IVR. Self-care menu



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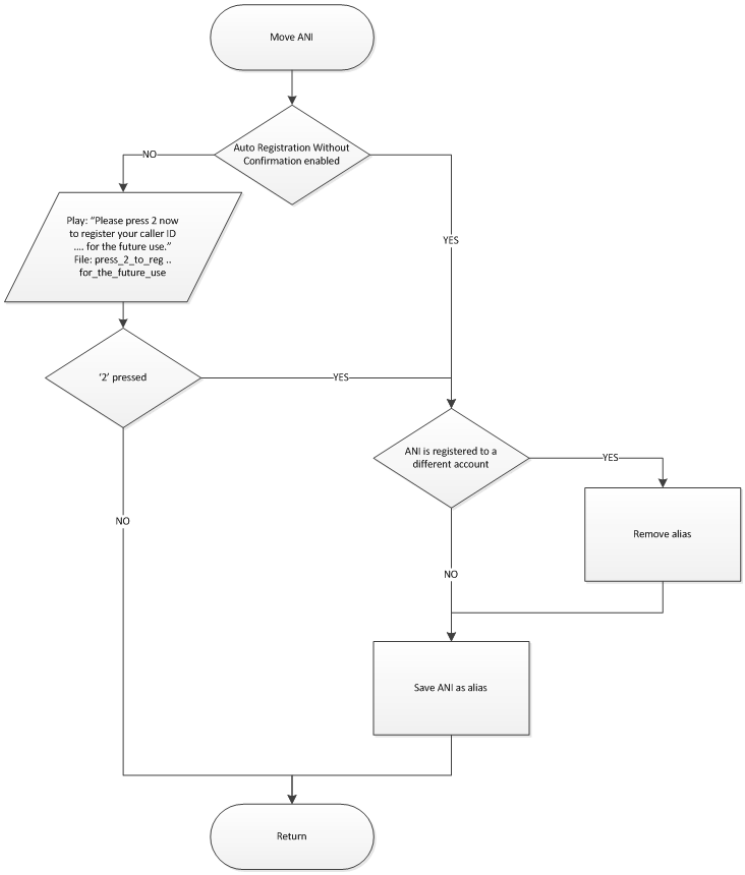
Prepaid card calling IVR. Move ANI



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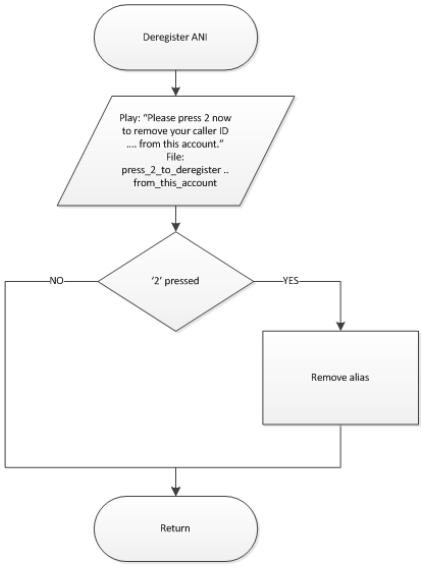
Prepaid card calling IVR. Deregister ANI



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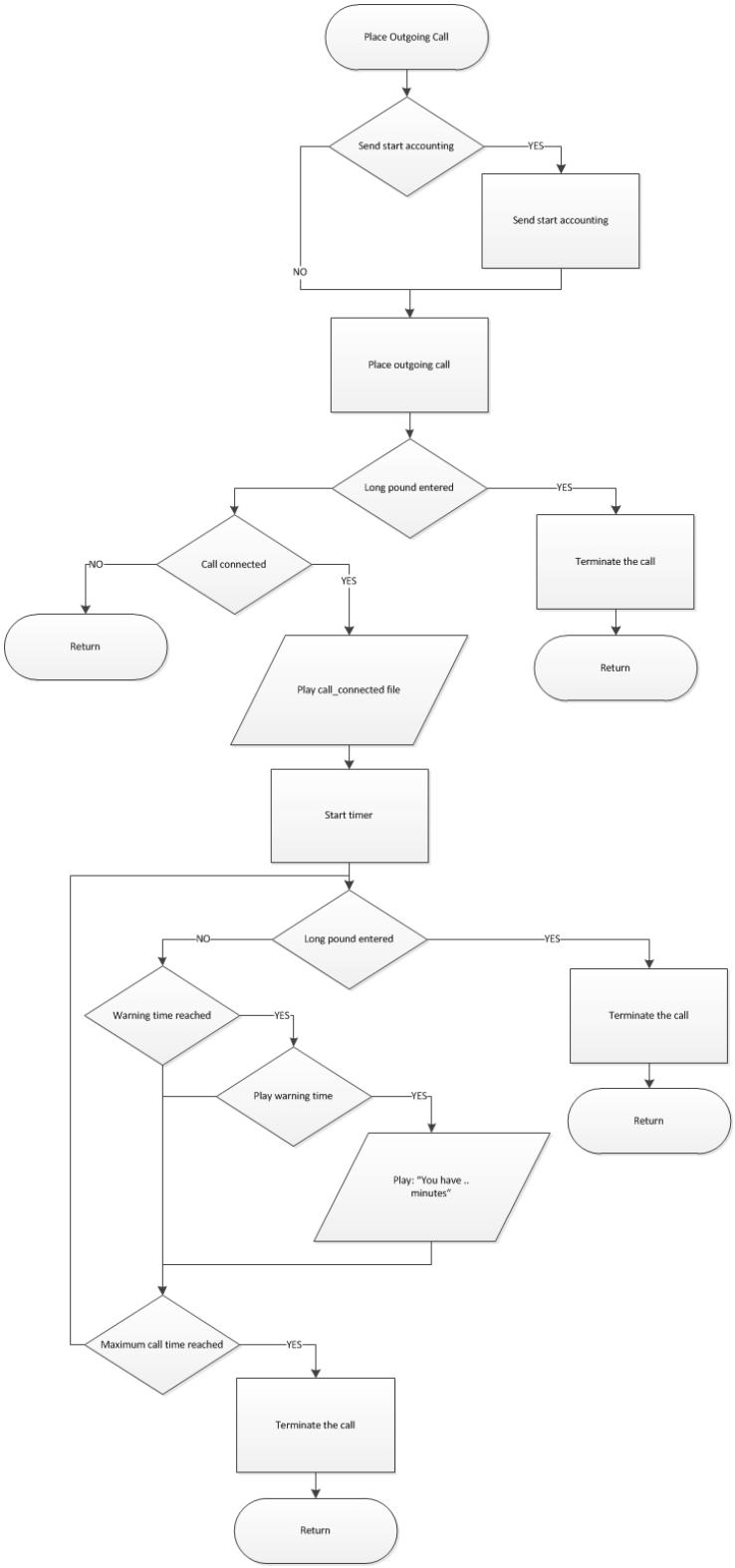
Prepaid card calling IVR. Place outgoing call



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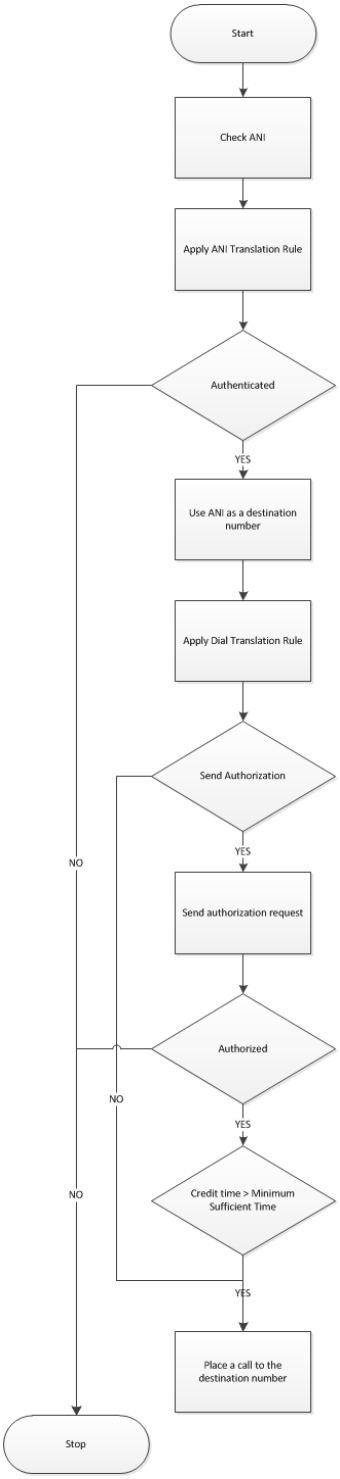
Callback IVRs

ANI callback IVR. Part 1



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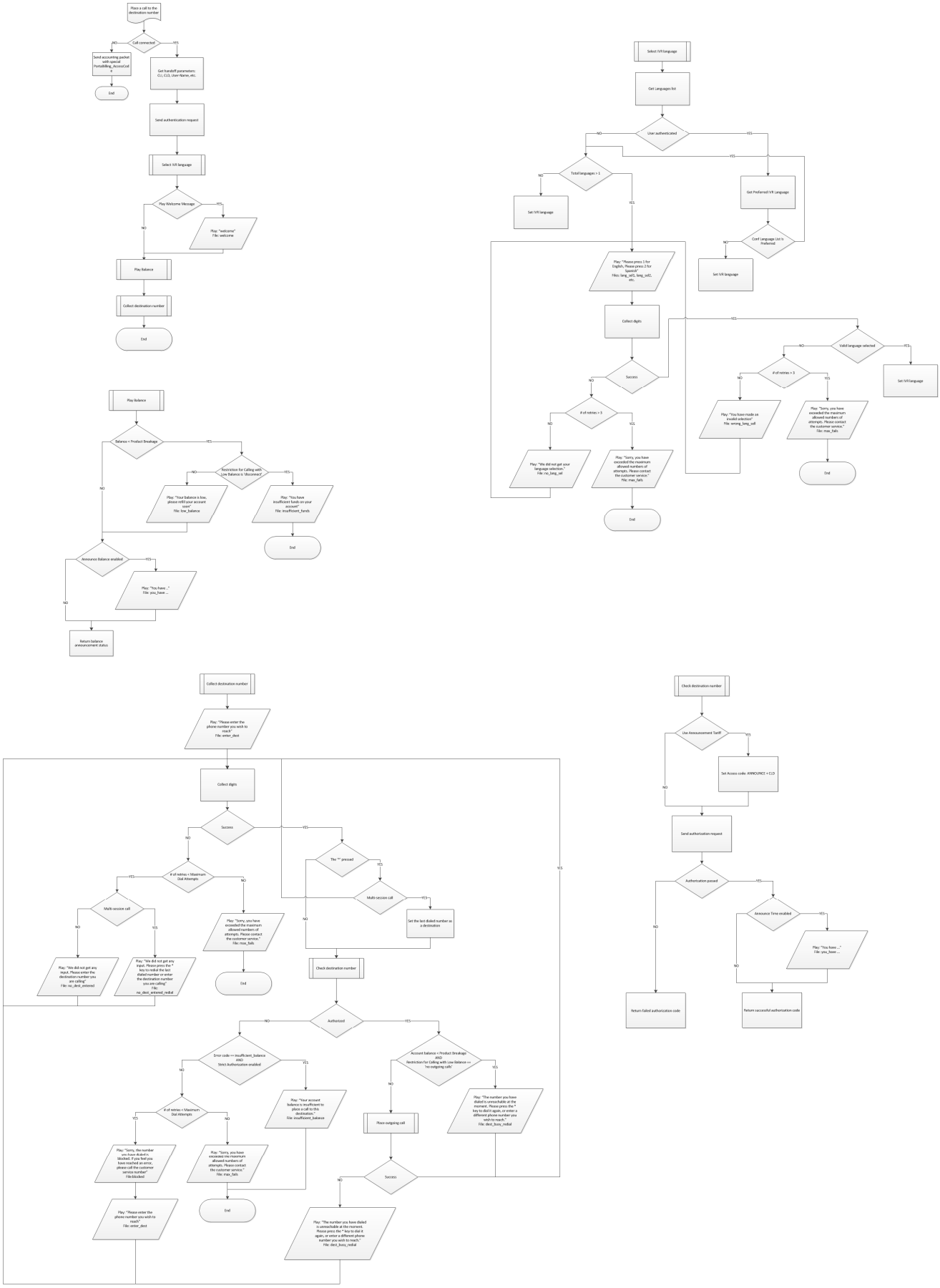
ANI callback IVR. Part 2



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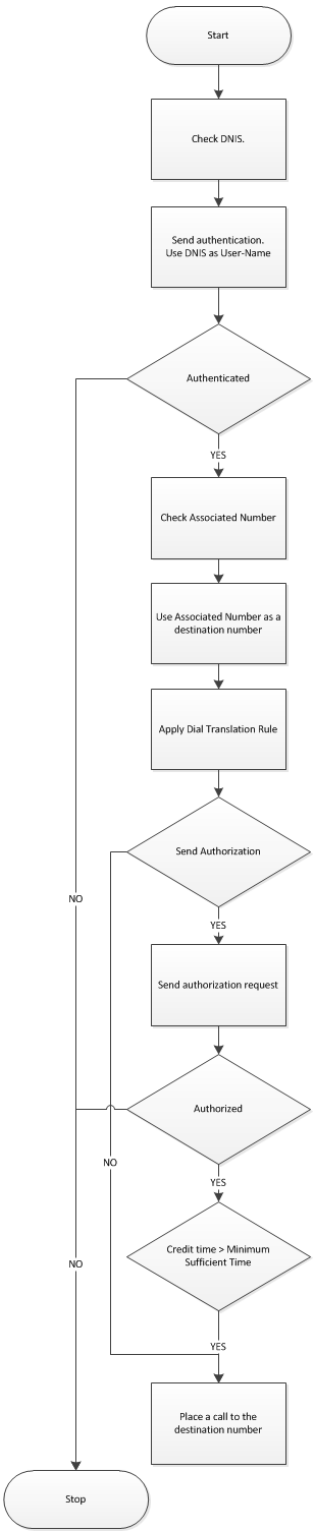
DNIS callback IVR. Part 1



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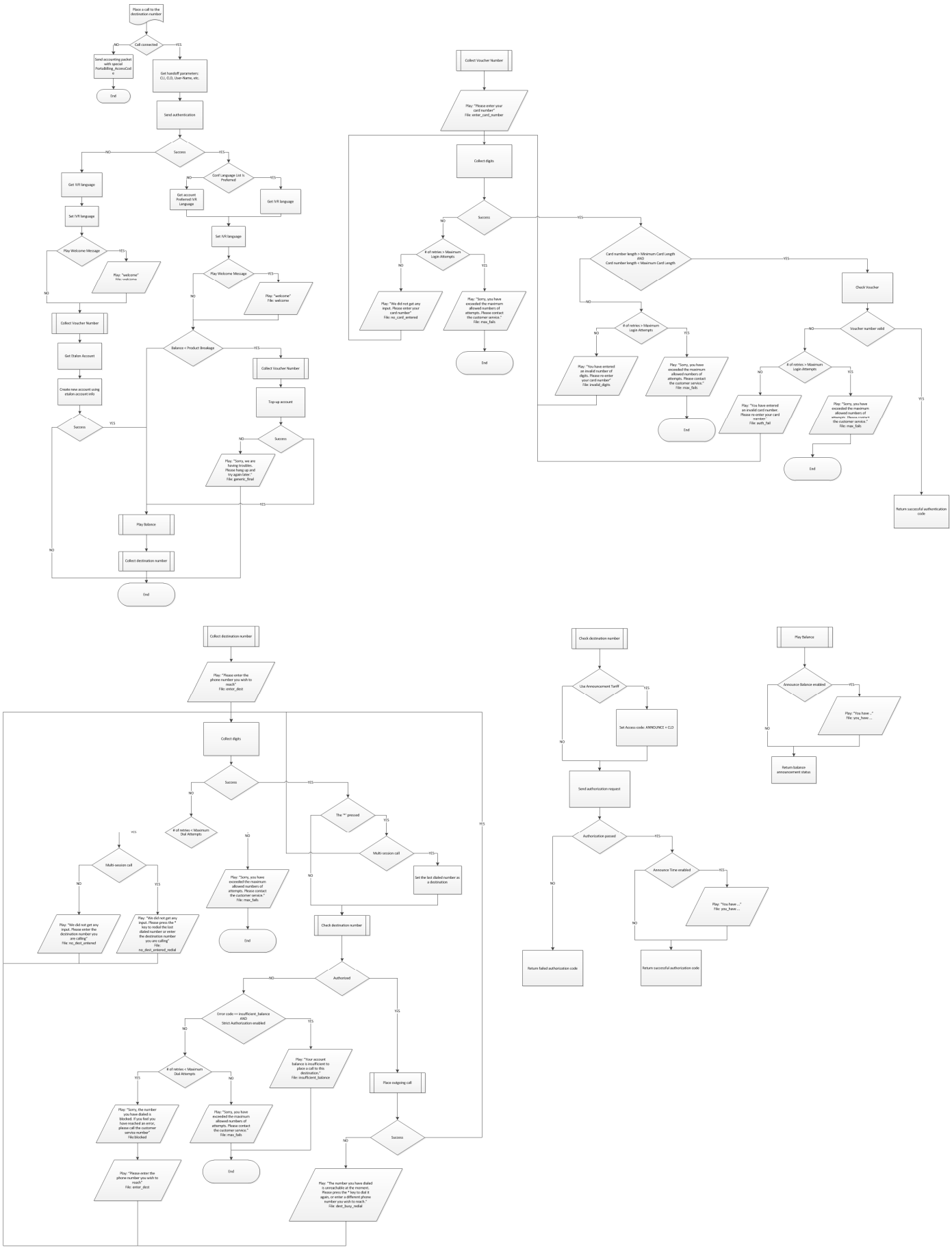
DNIS callback IVR. Part 2



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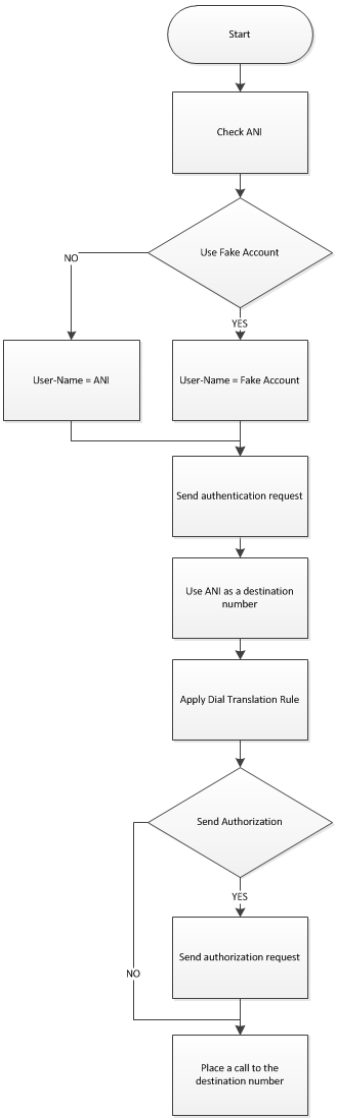
PIN callback IVR. Part 1



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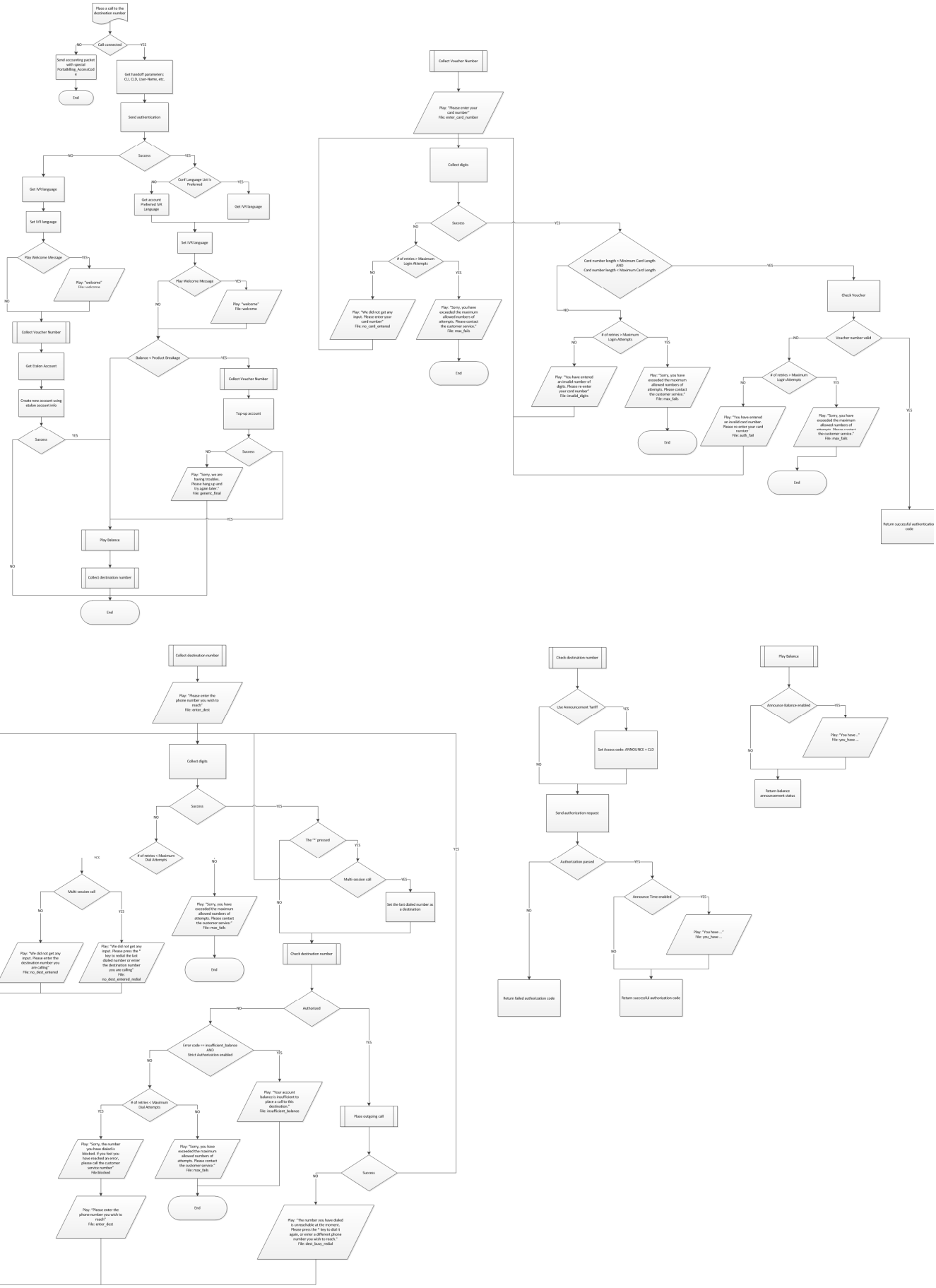
PIN callback IVR. Part 2



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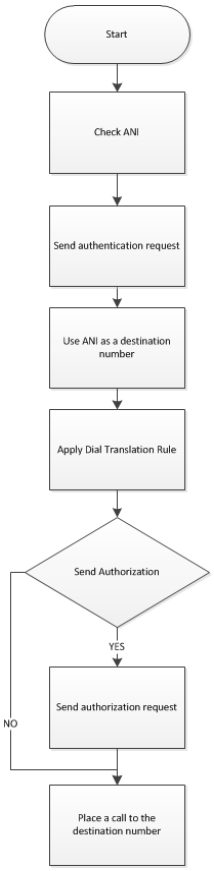
Callback with registration IVR. Part 1



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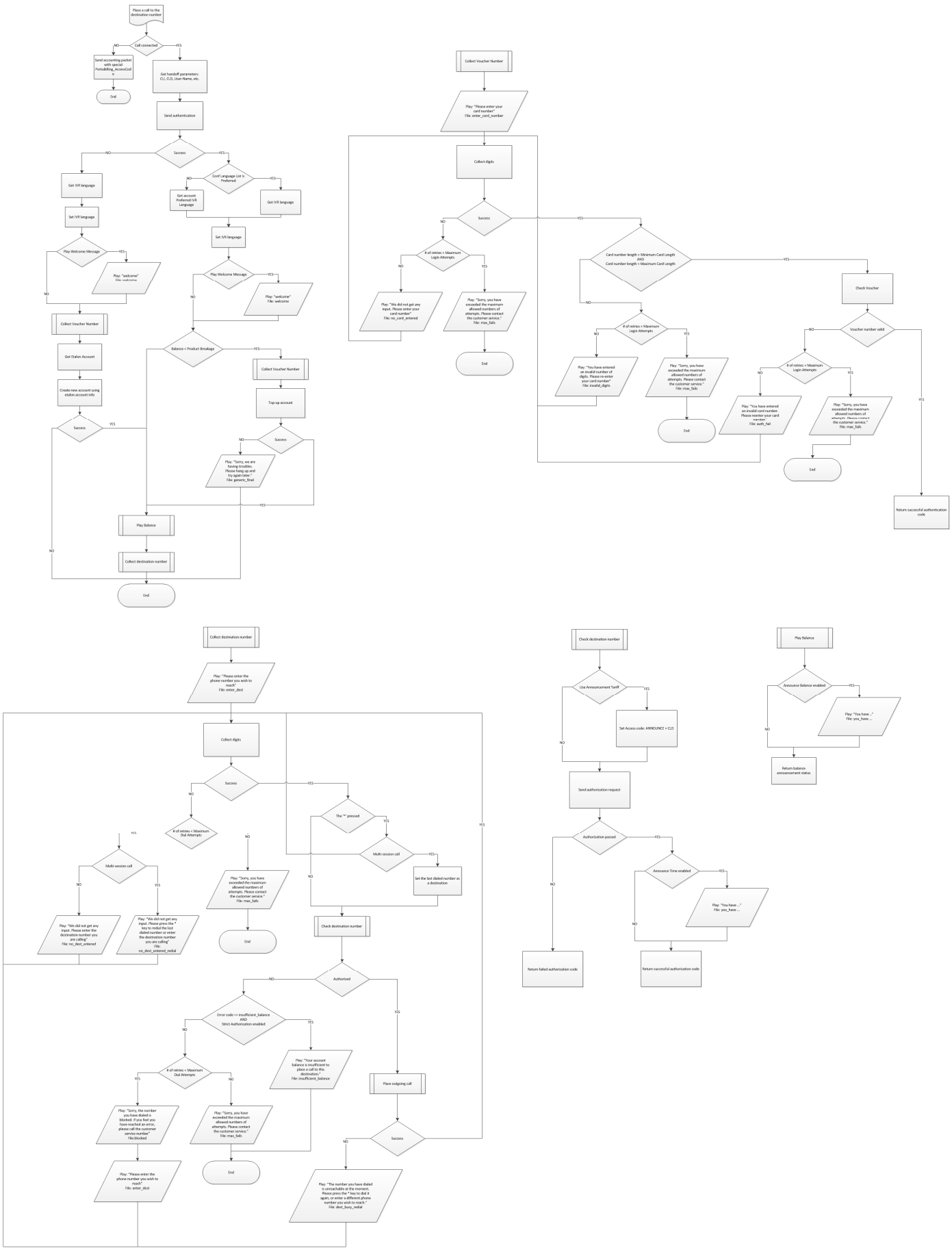
Callback with registration IVR. Part 2



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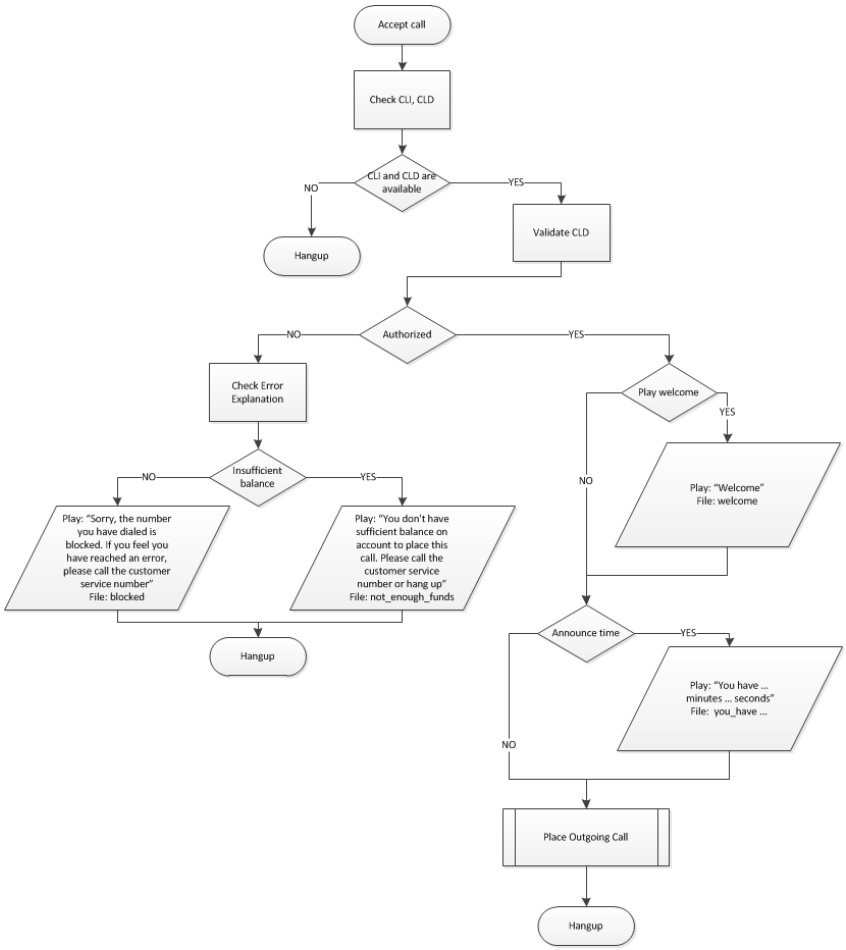
One-stage Calling IVR



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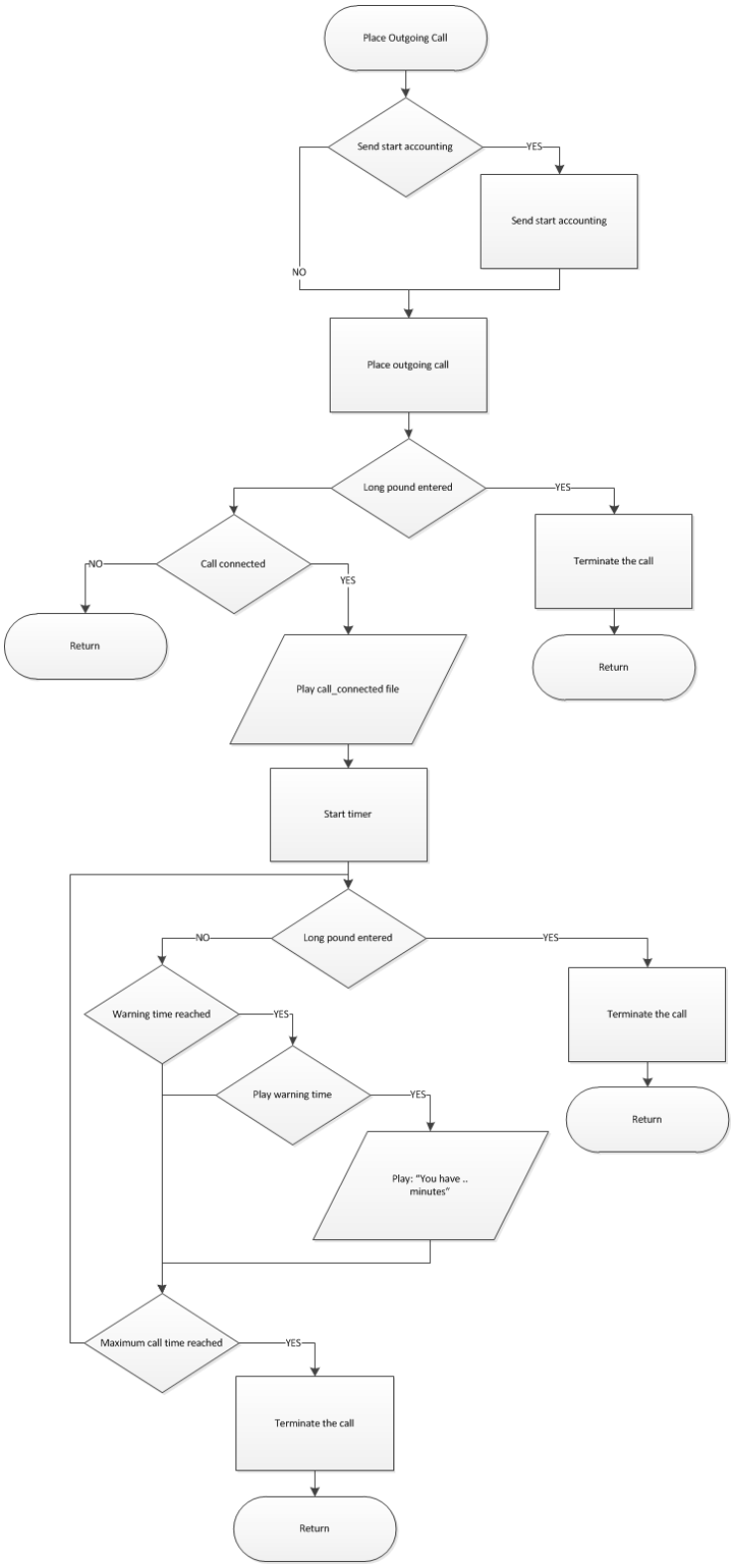
One-stage calling IVR. Place outgoing call



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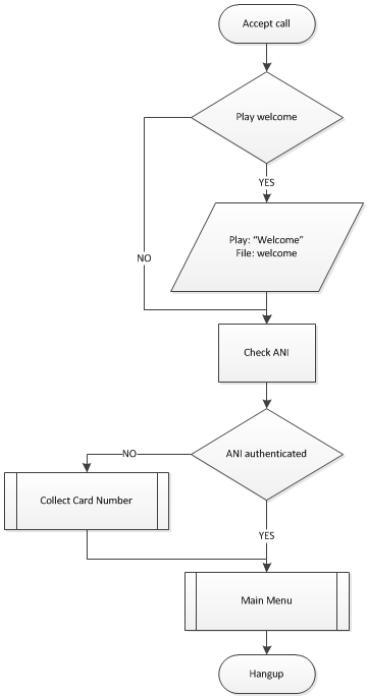
Payment Remittance. TransferTo IVR



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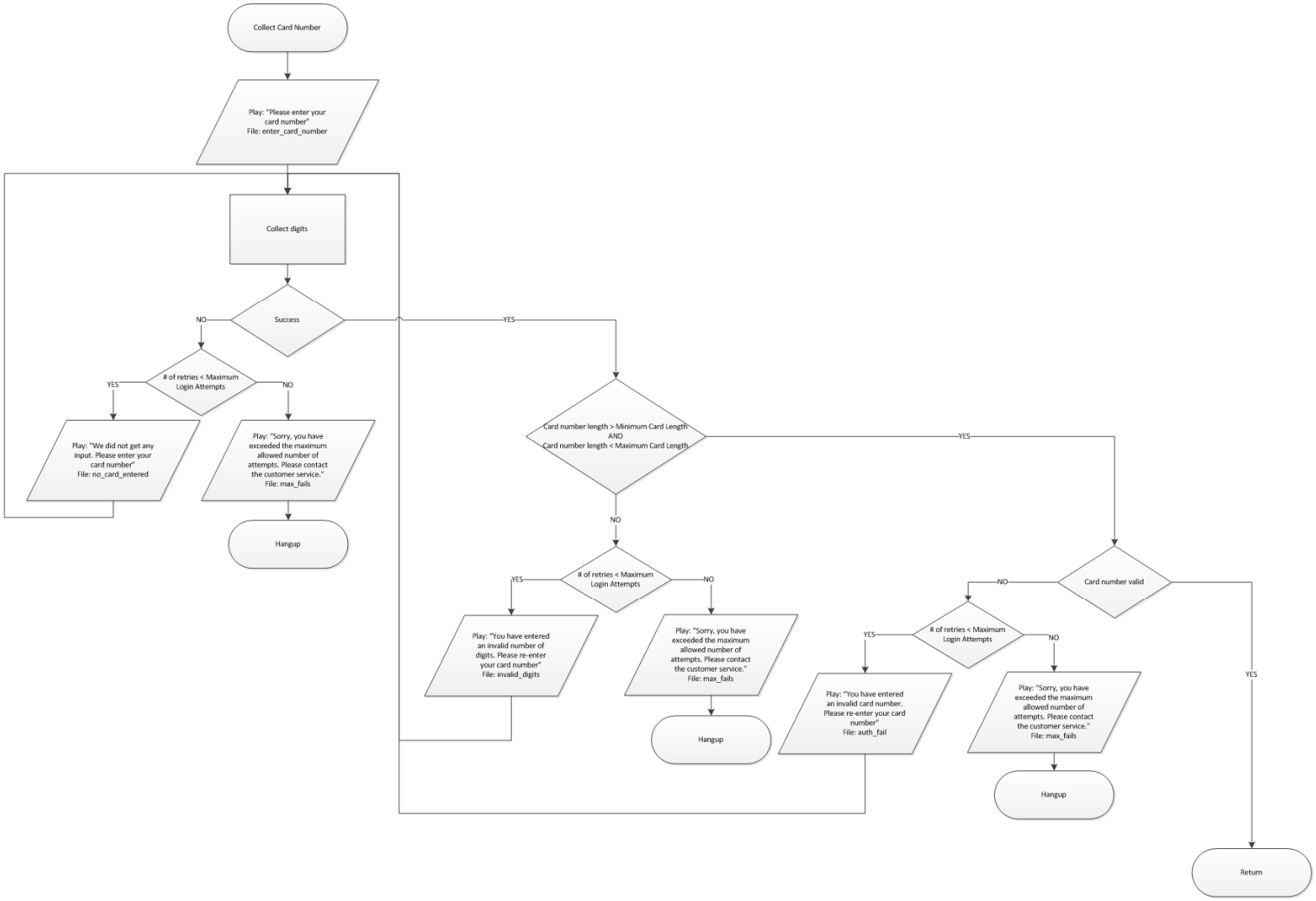
Payment remittance – TransferTo IVR. Collect card number



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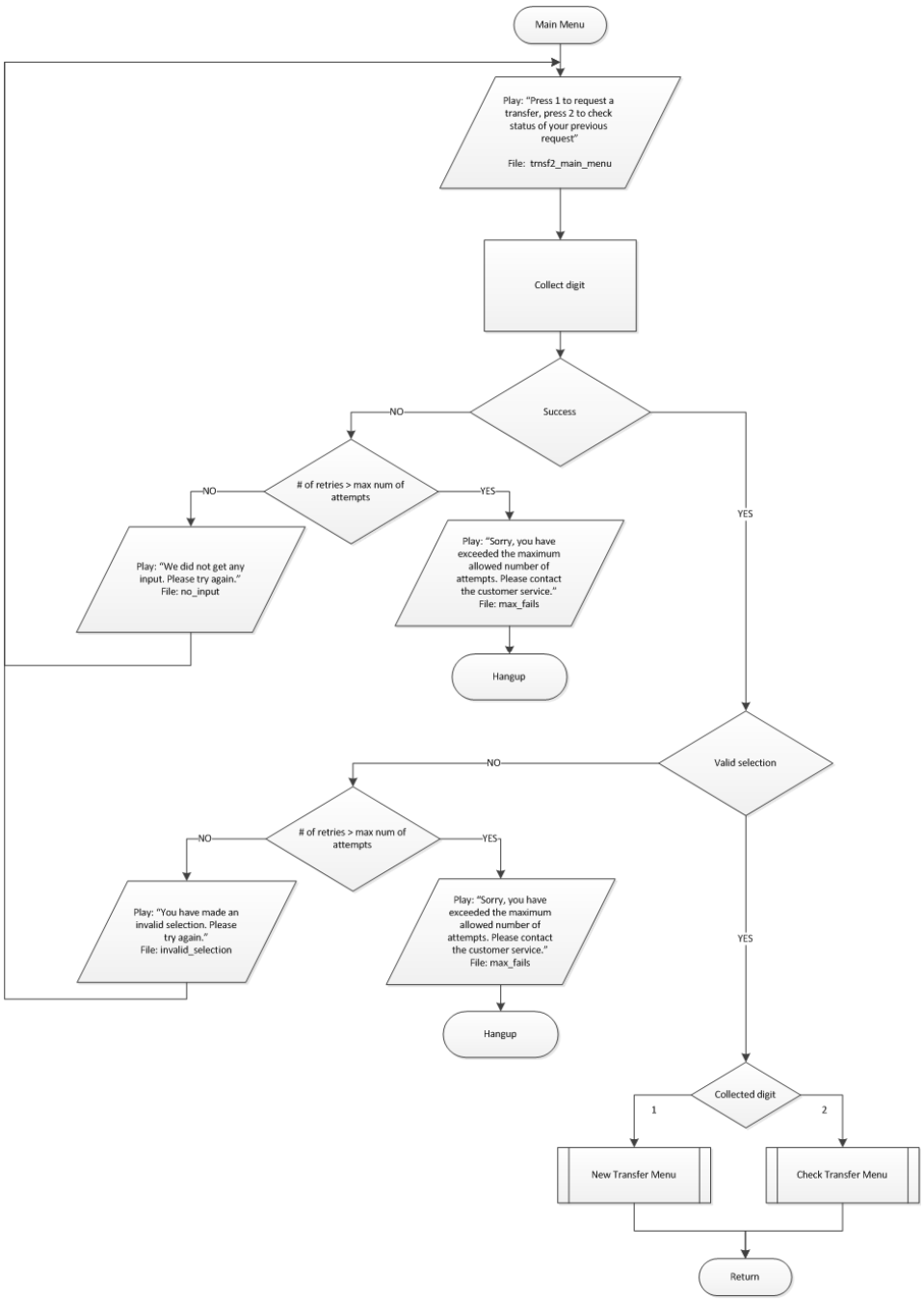
Payment remittance – TransferTo IVR. Main menu



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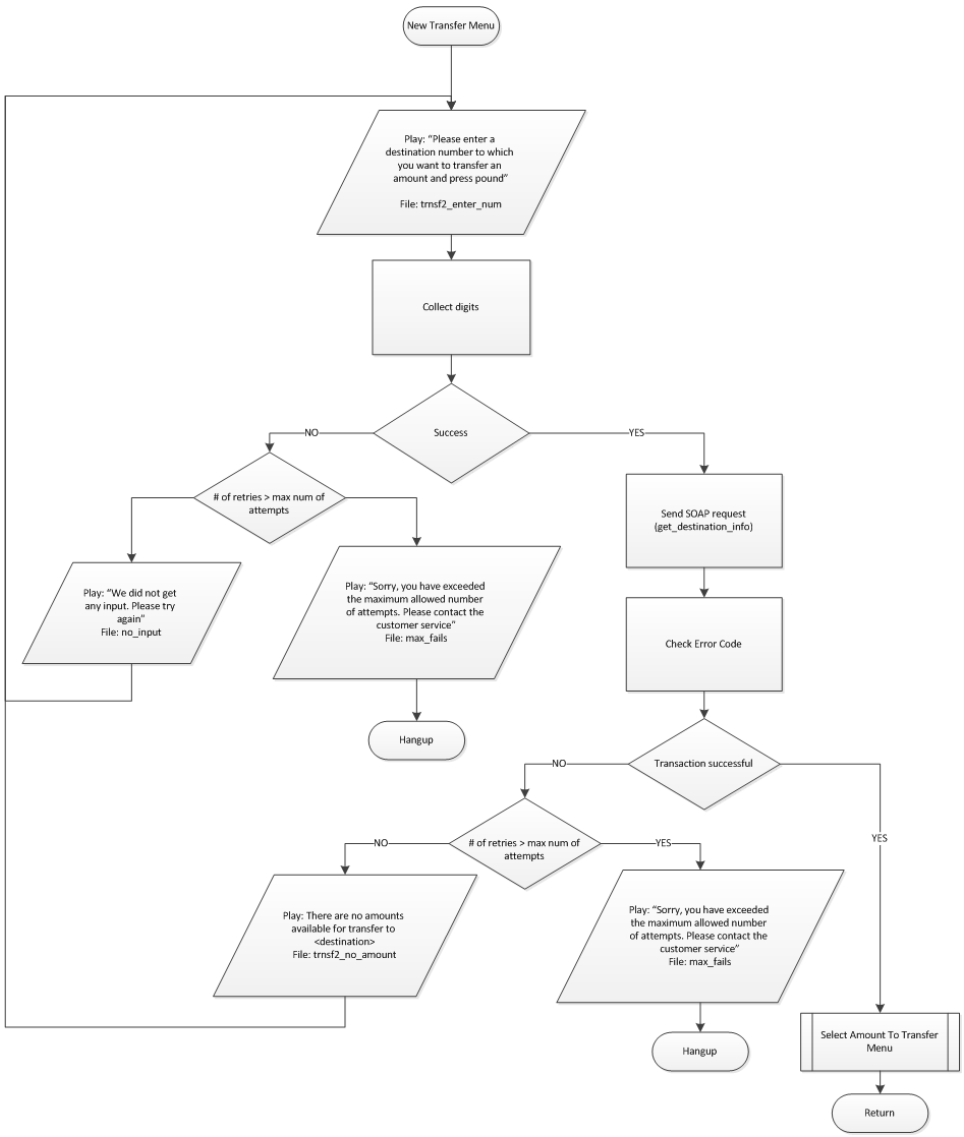
Payment remittance – TransferTo IVR. New transfer menu



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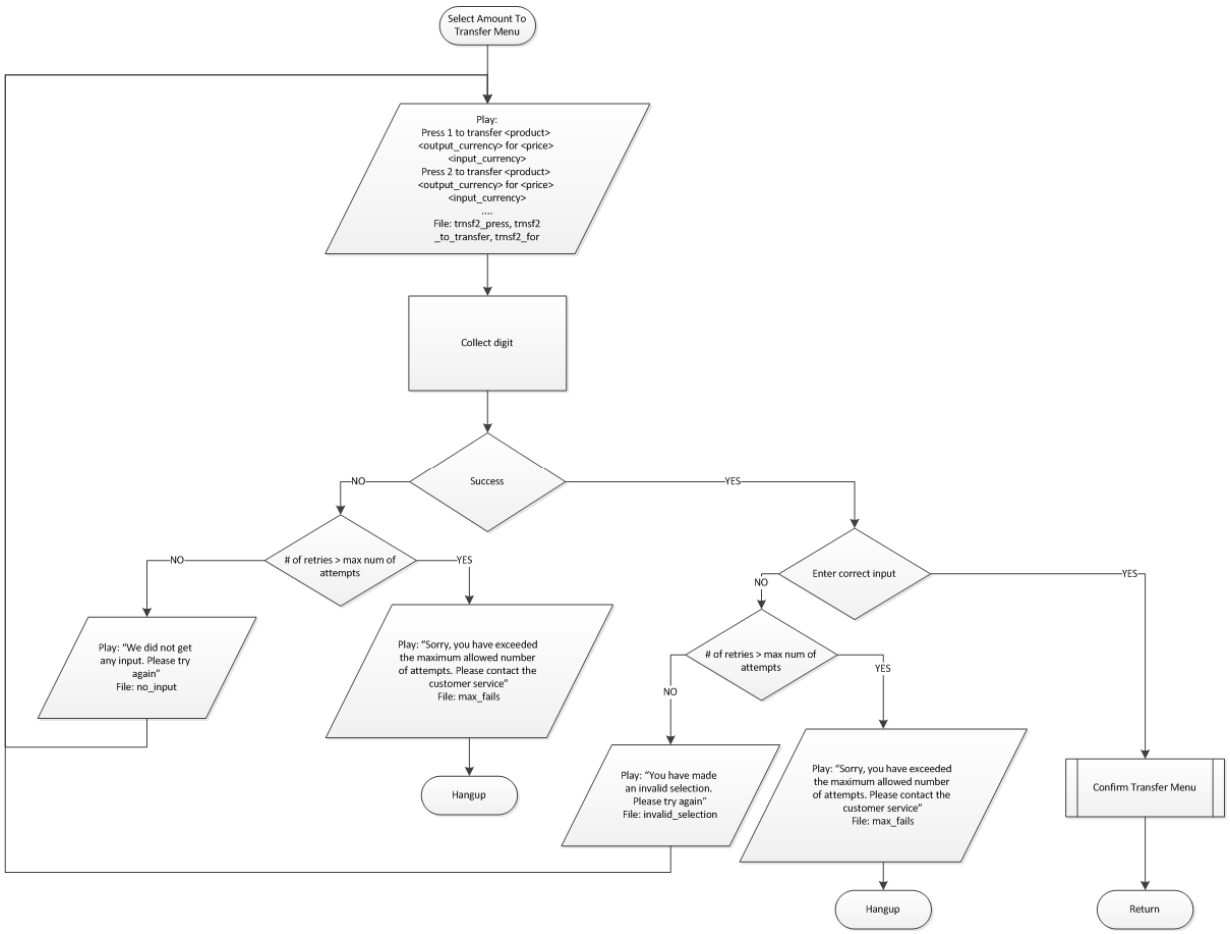
Payment remittance – TransferTo IVR. Select amount to transfer menu



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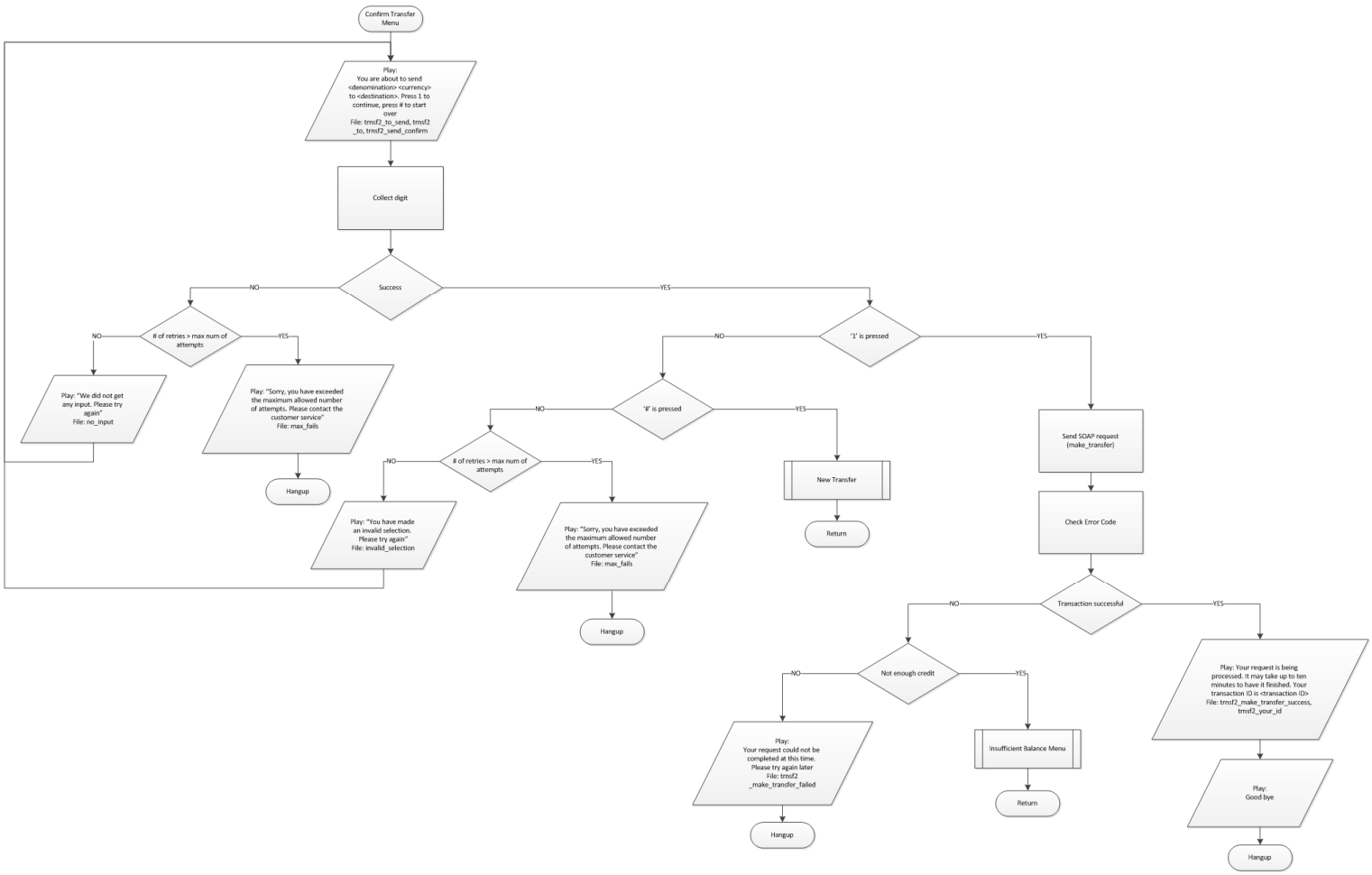
Payment remittance – TransferTo IVR. Confirm transfer menu



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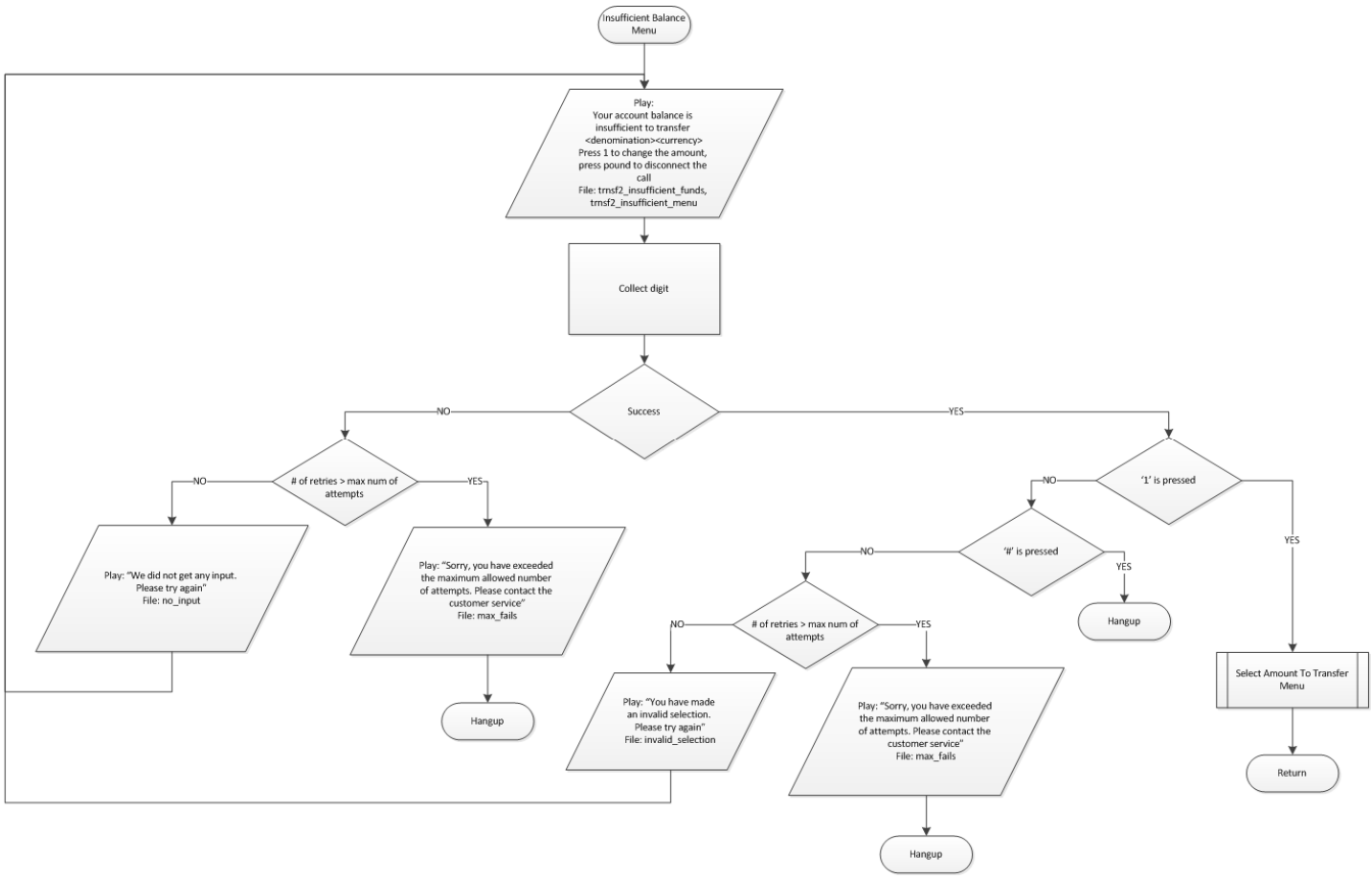
Payment remittance – TransferTo IVR. Insufficient balance menu



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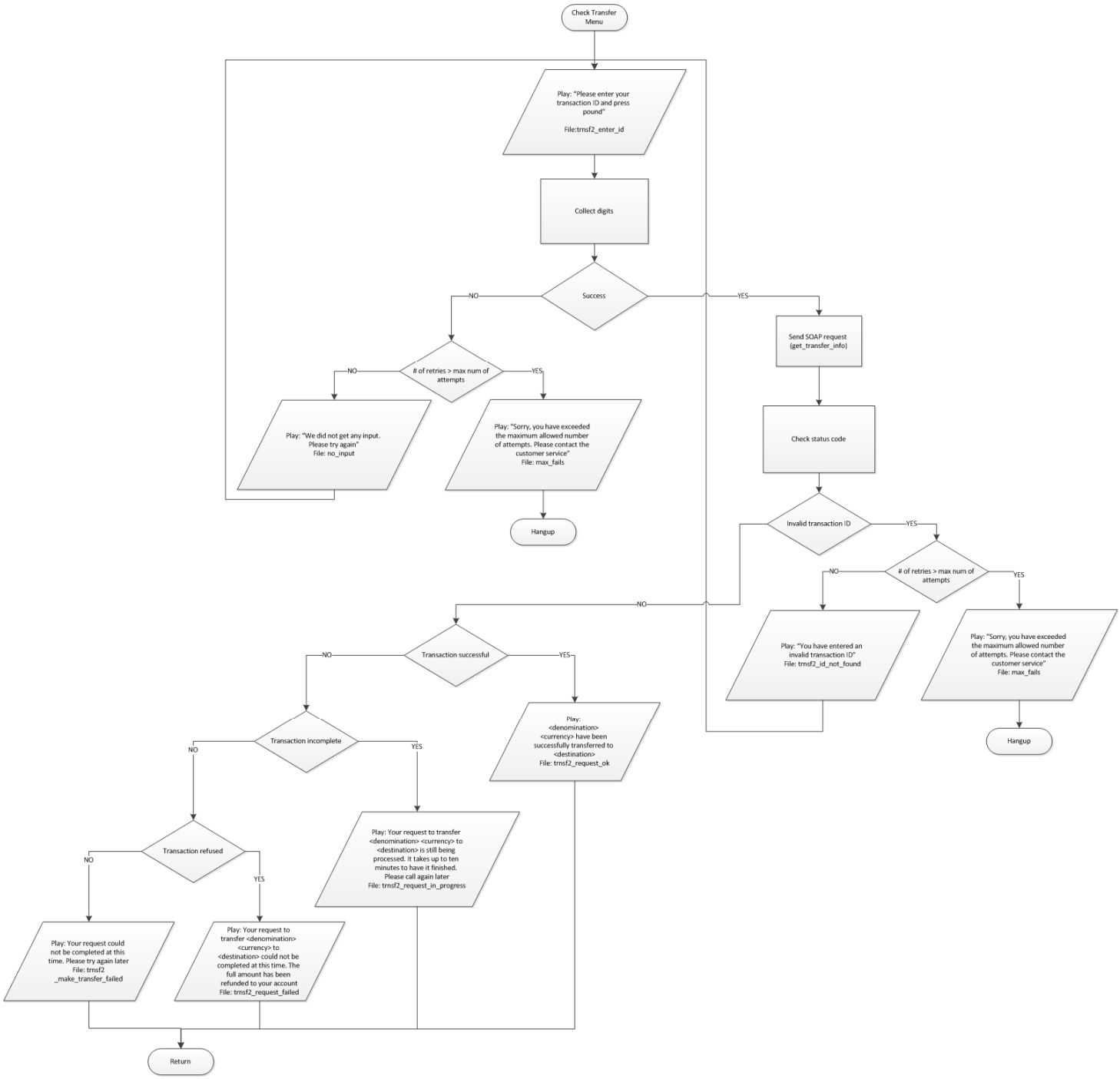
Payment remittance – TransferTo IVR. Check transfer menu



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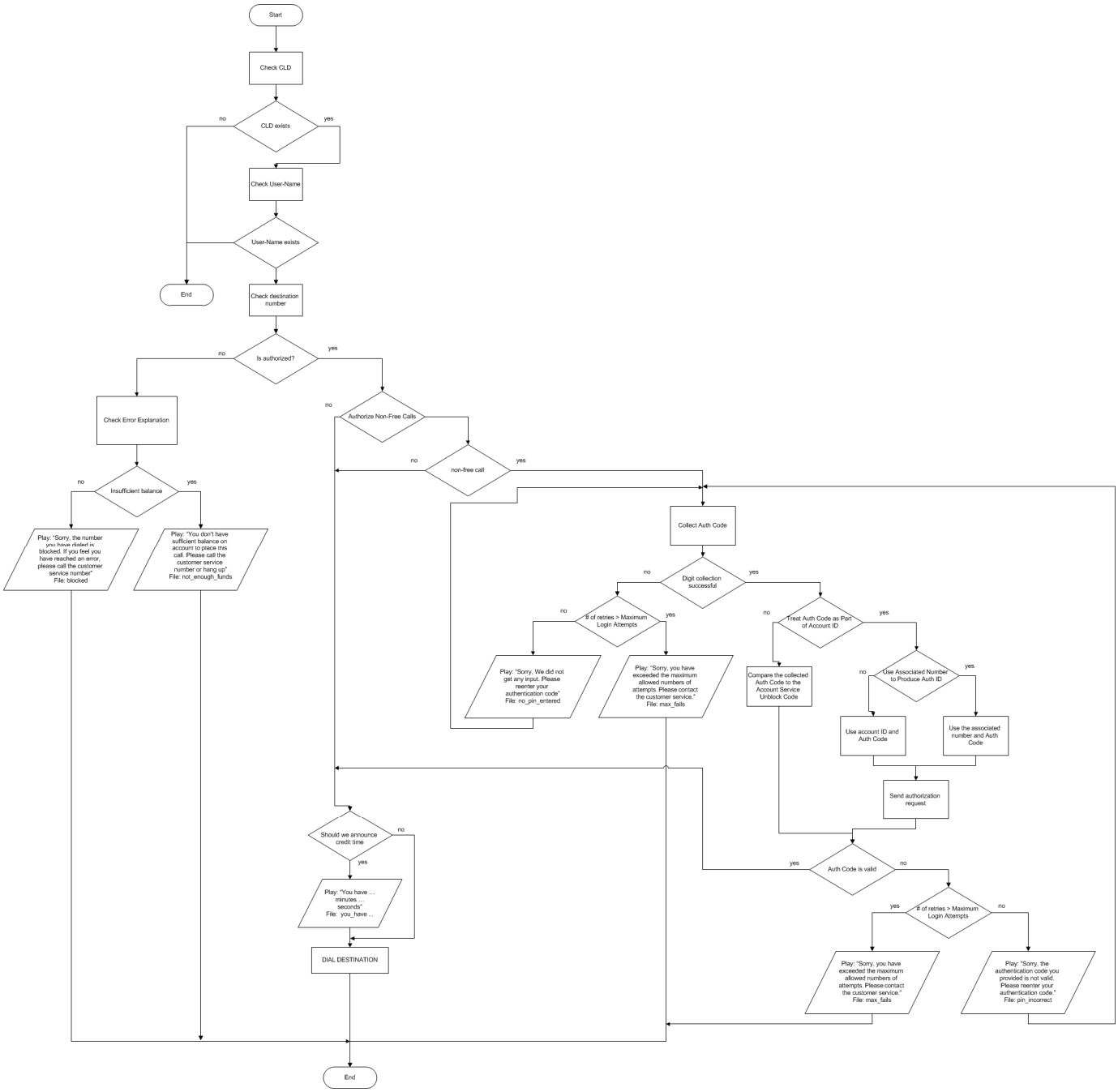
Pass-Through IVR



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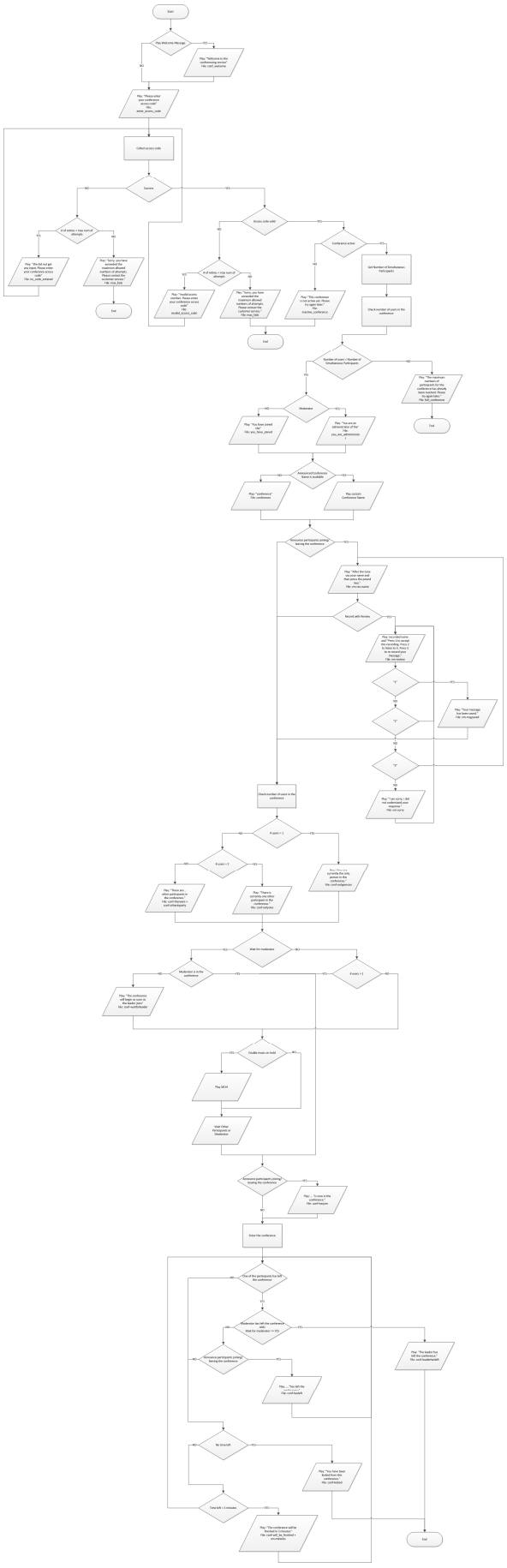
Conferencing IVR



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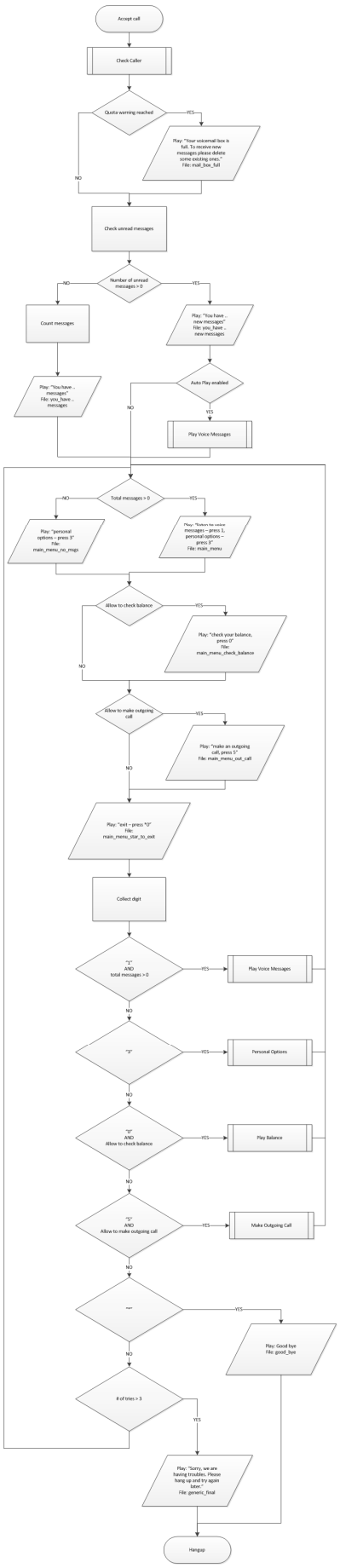
One's Own Voice Mailbox Access IVR



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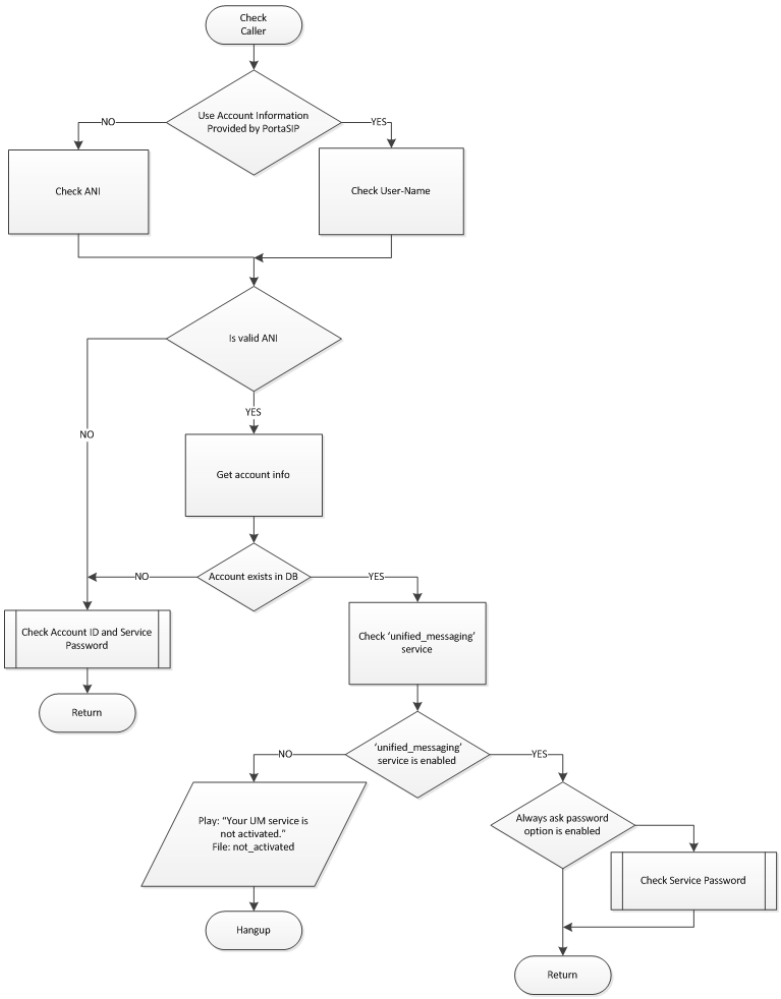
One's own voice mailbox access IVR. Check caller



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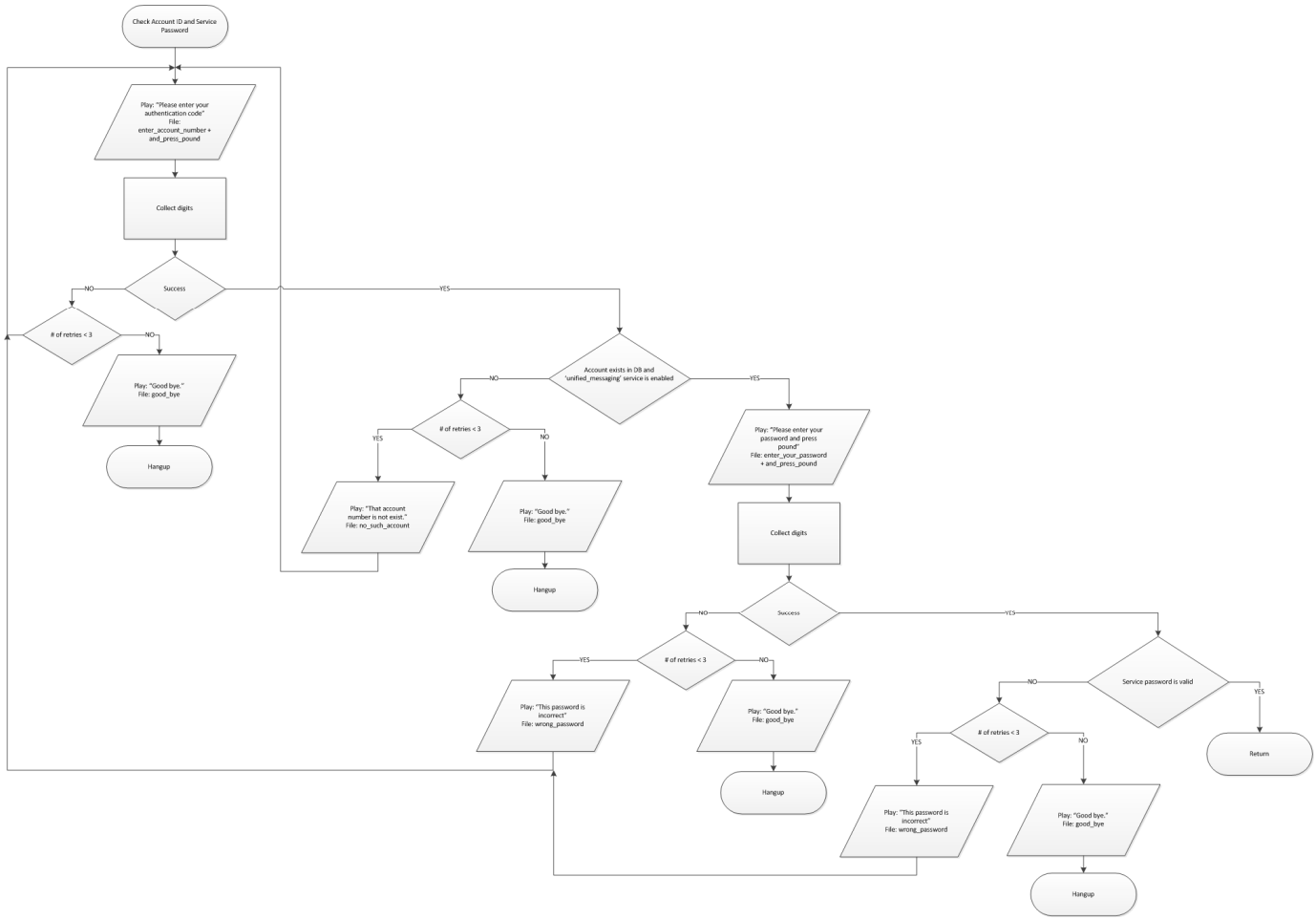
One's own voice mailbox access IVR. Check account ID and service password



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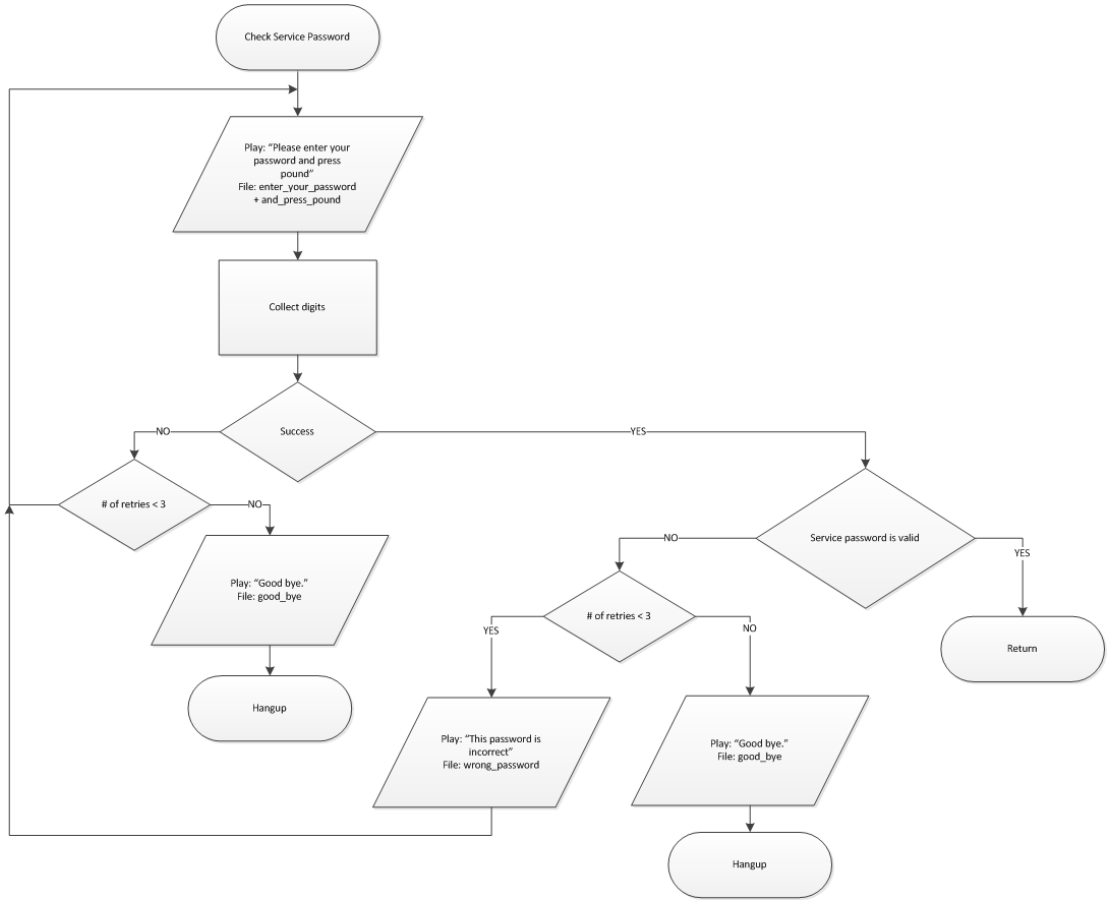
One's own voice mailbox access IVR. Check service password



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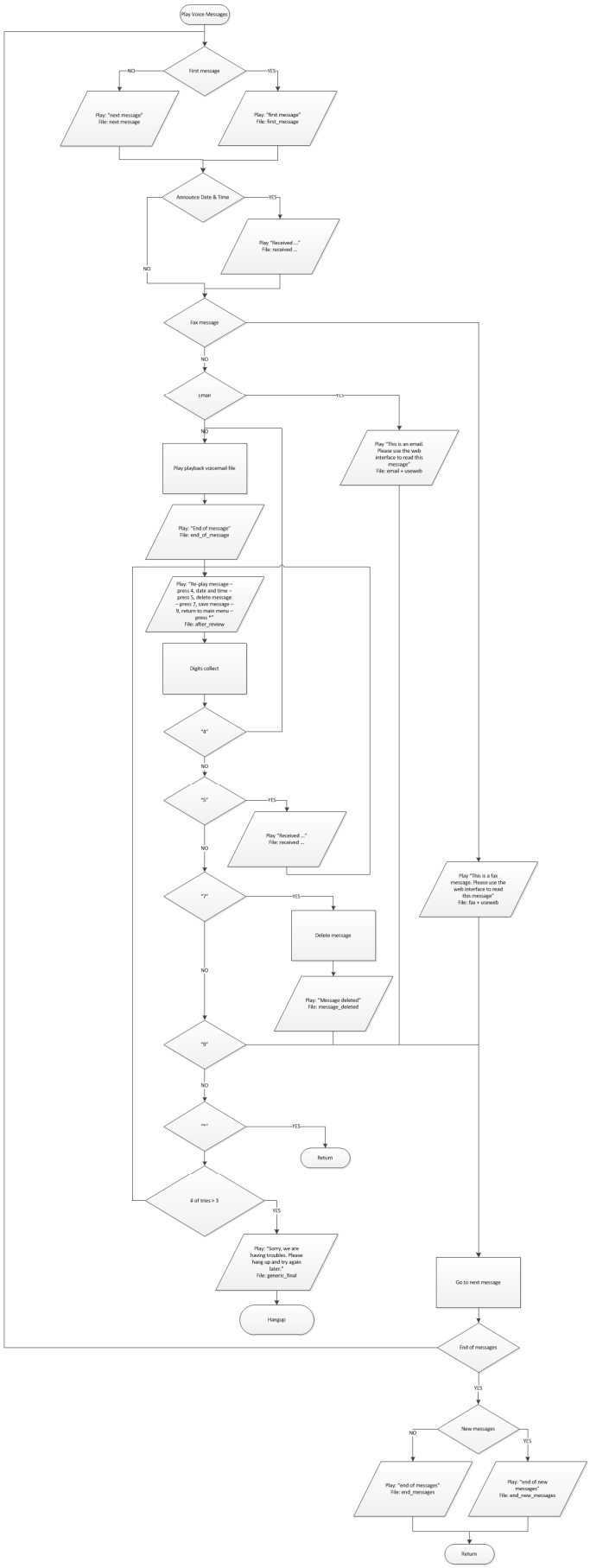
One's own voice mailbox access IVR. Play voice messages



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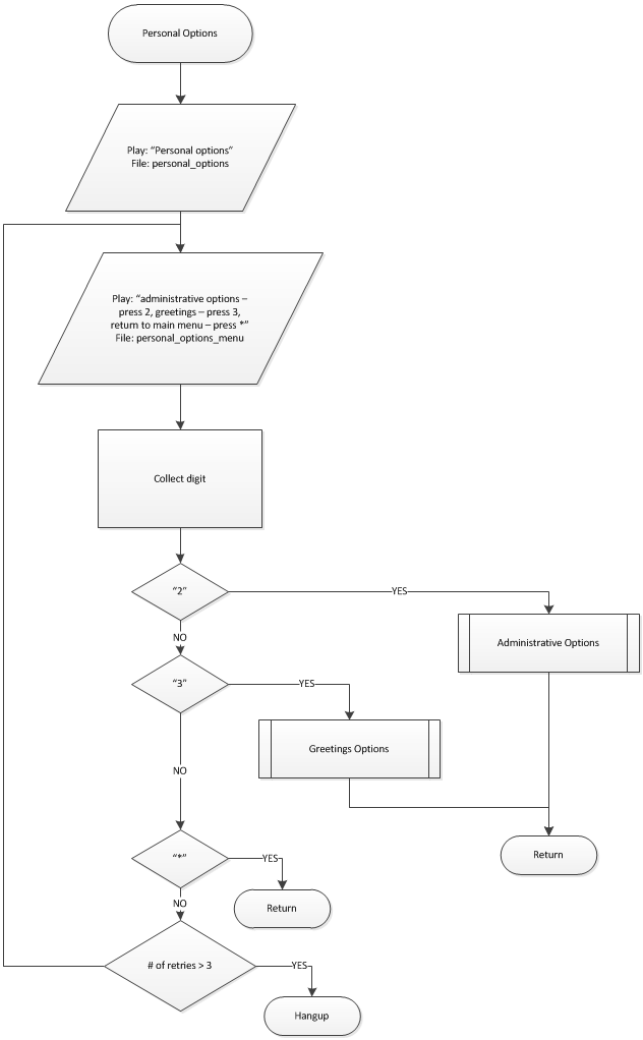
One's own voice mailbox access IVR. Personal options



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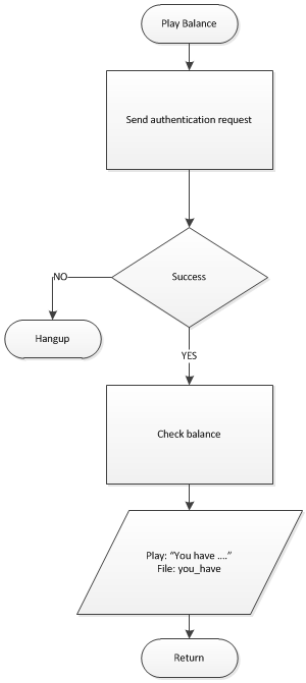
One's own voice mailbox access IVR. Play balance



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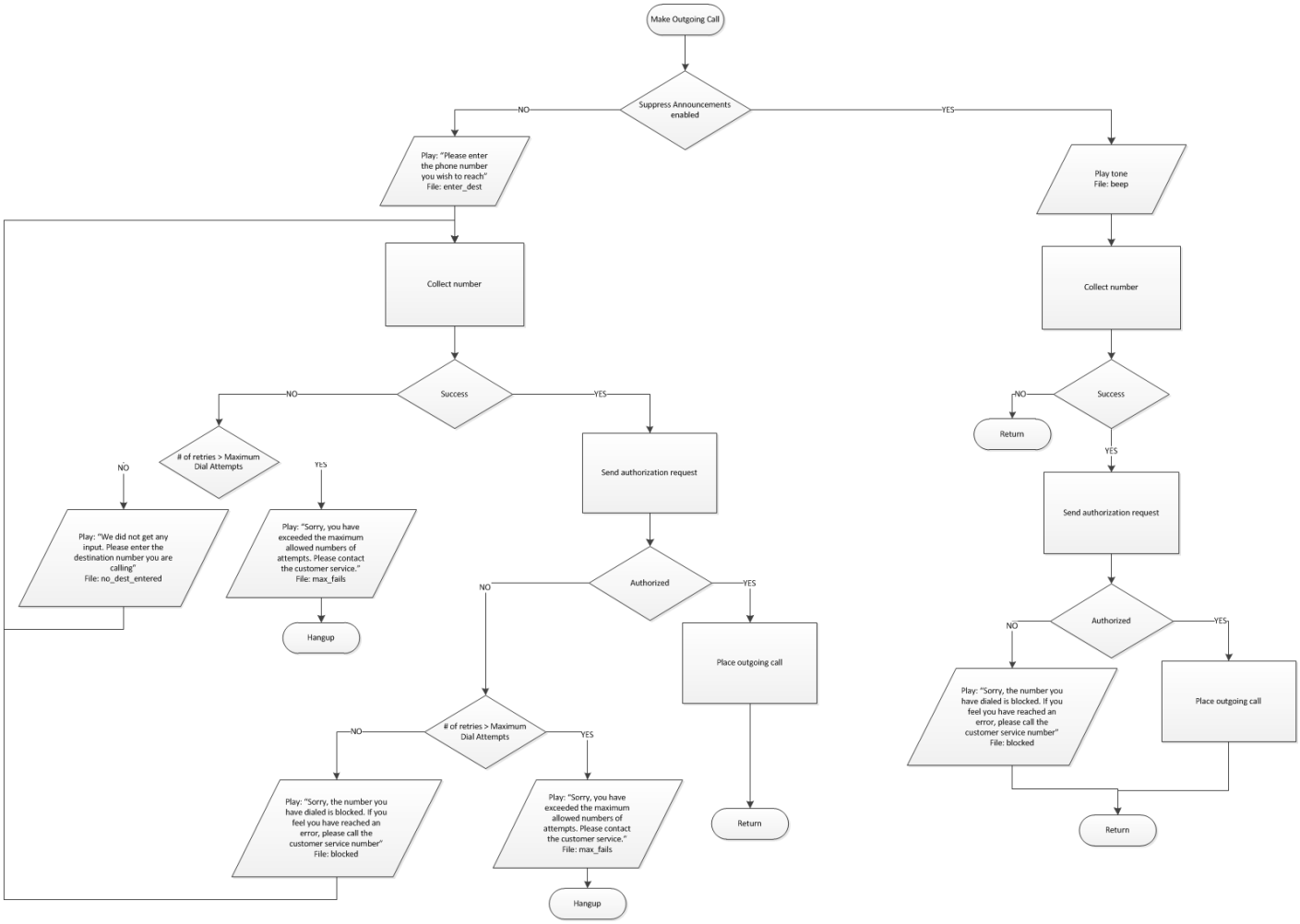
One's own voice mailbox access IVR. Make outgoing call



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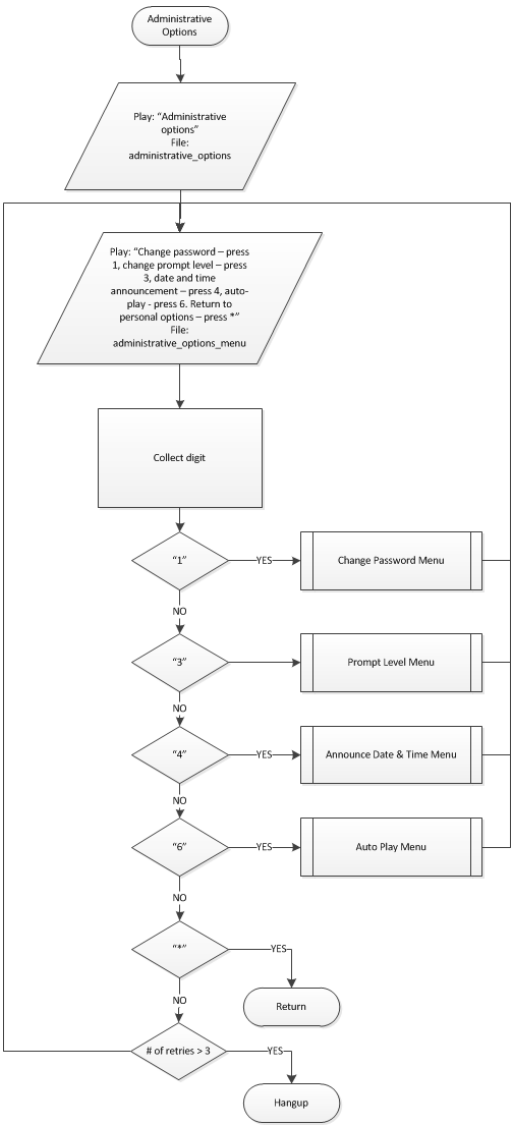
One's own voice mailbox access IVR. Administrative options



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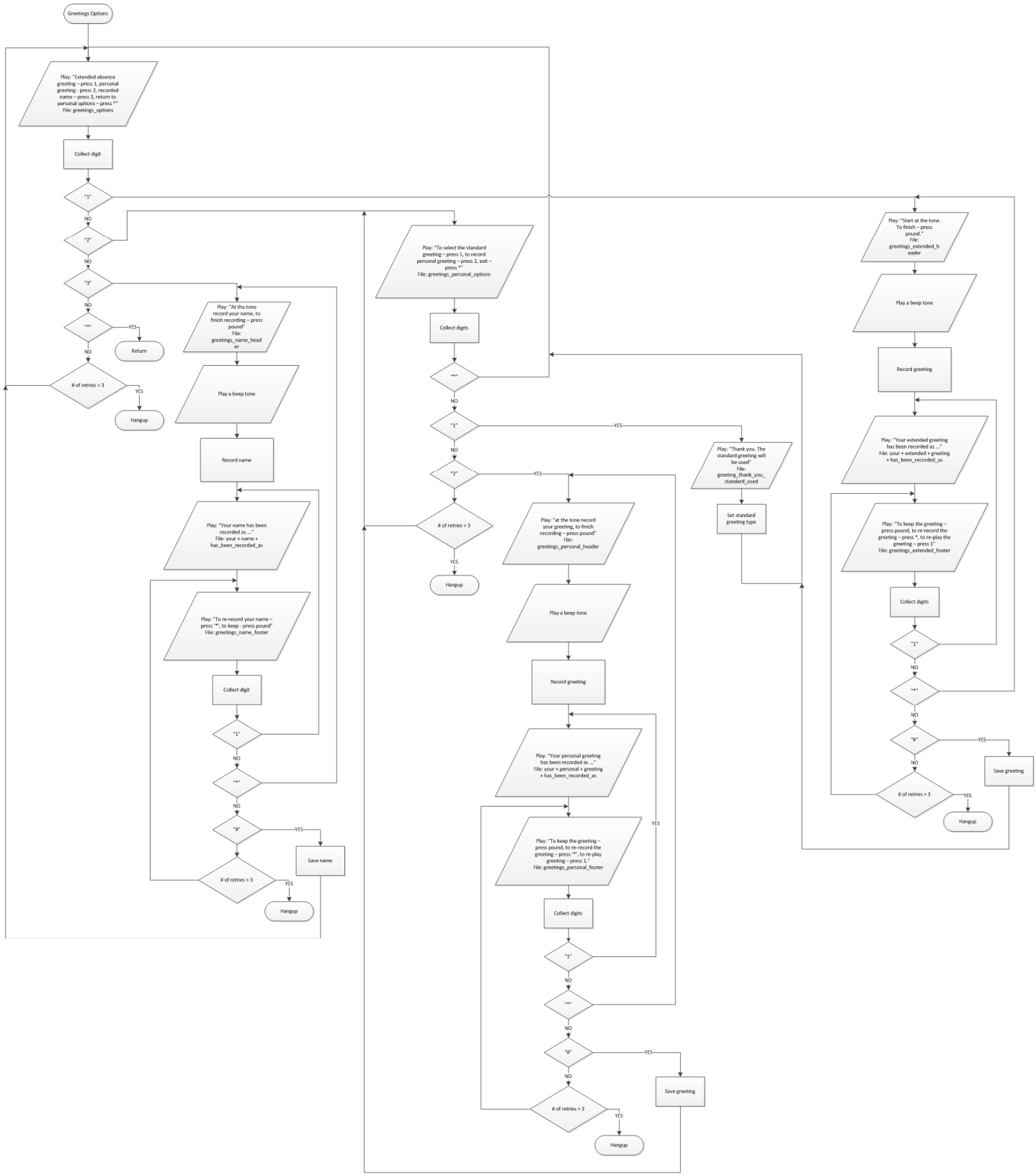
One's own voice mailbox access IVR. Greetings options



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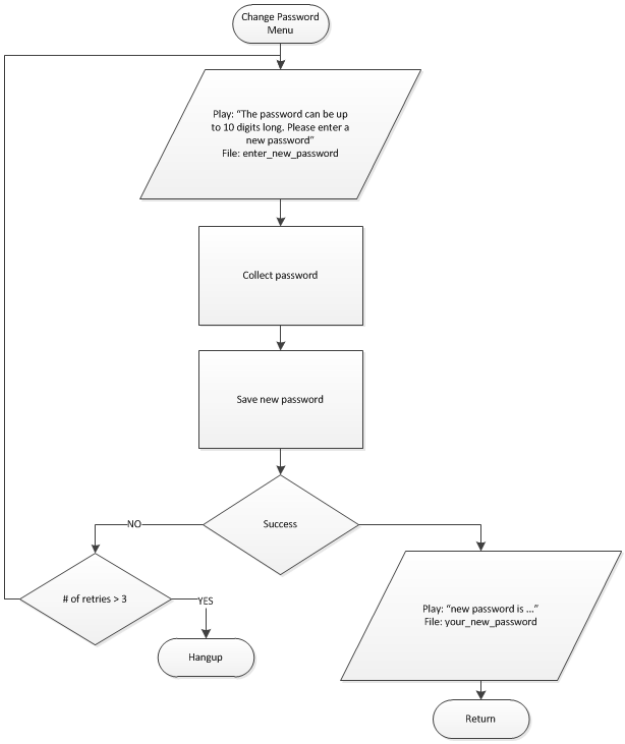
One's own voice mailbox access IVR. Change password menu



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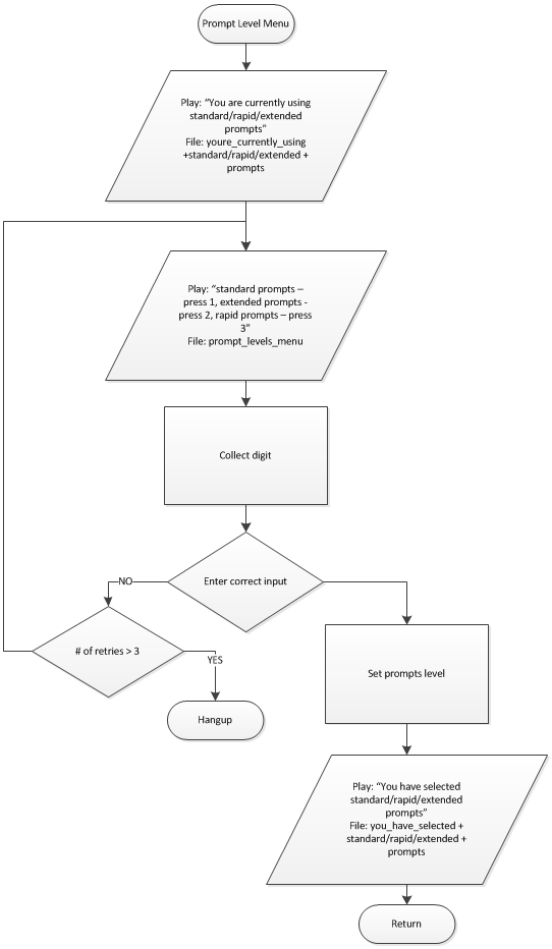
One's own voice mailbox access IVR. Prompt level menu



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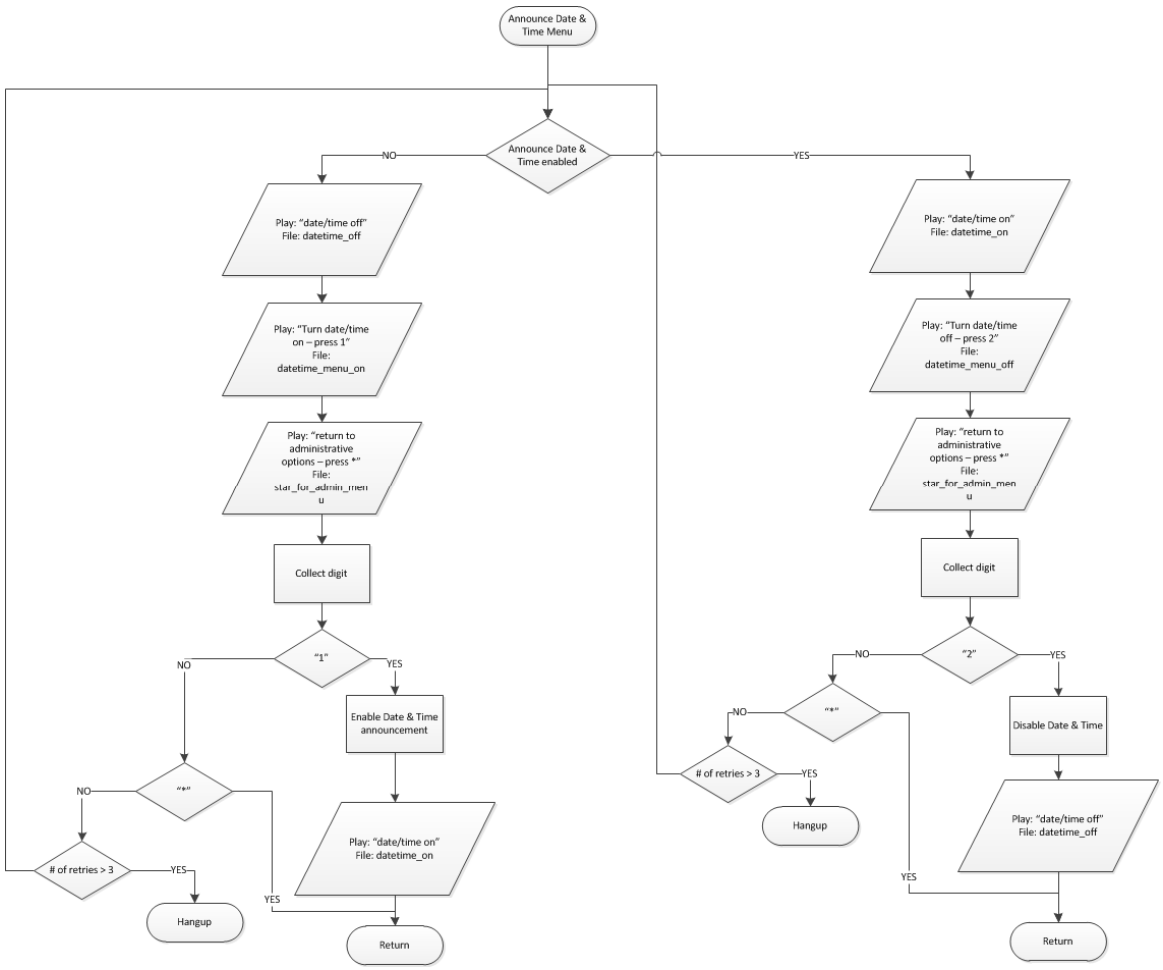
One's own voice mailbox access IVR. Announce date time menu



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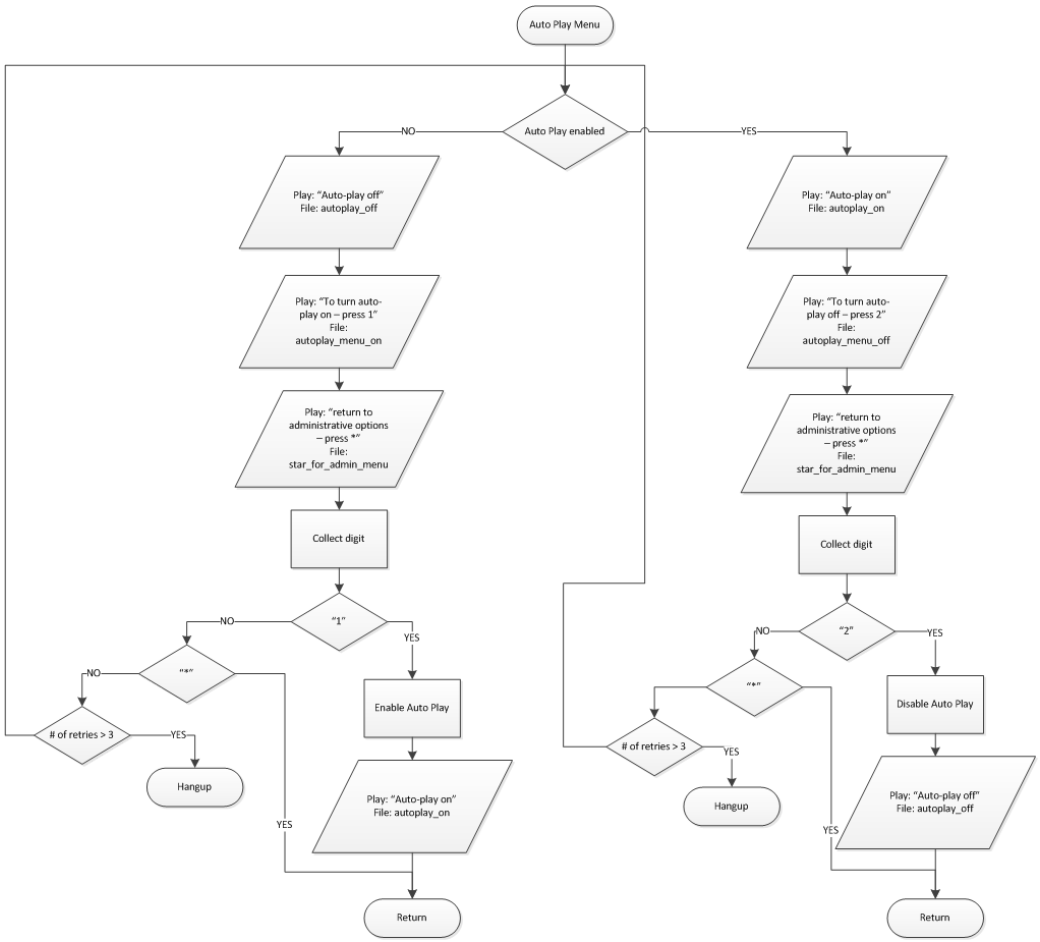
One's own voice mailbox access IVR. Auto-play menu



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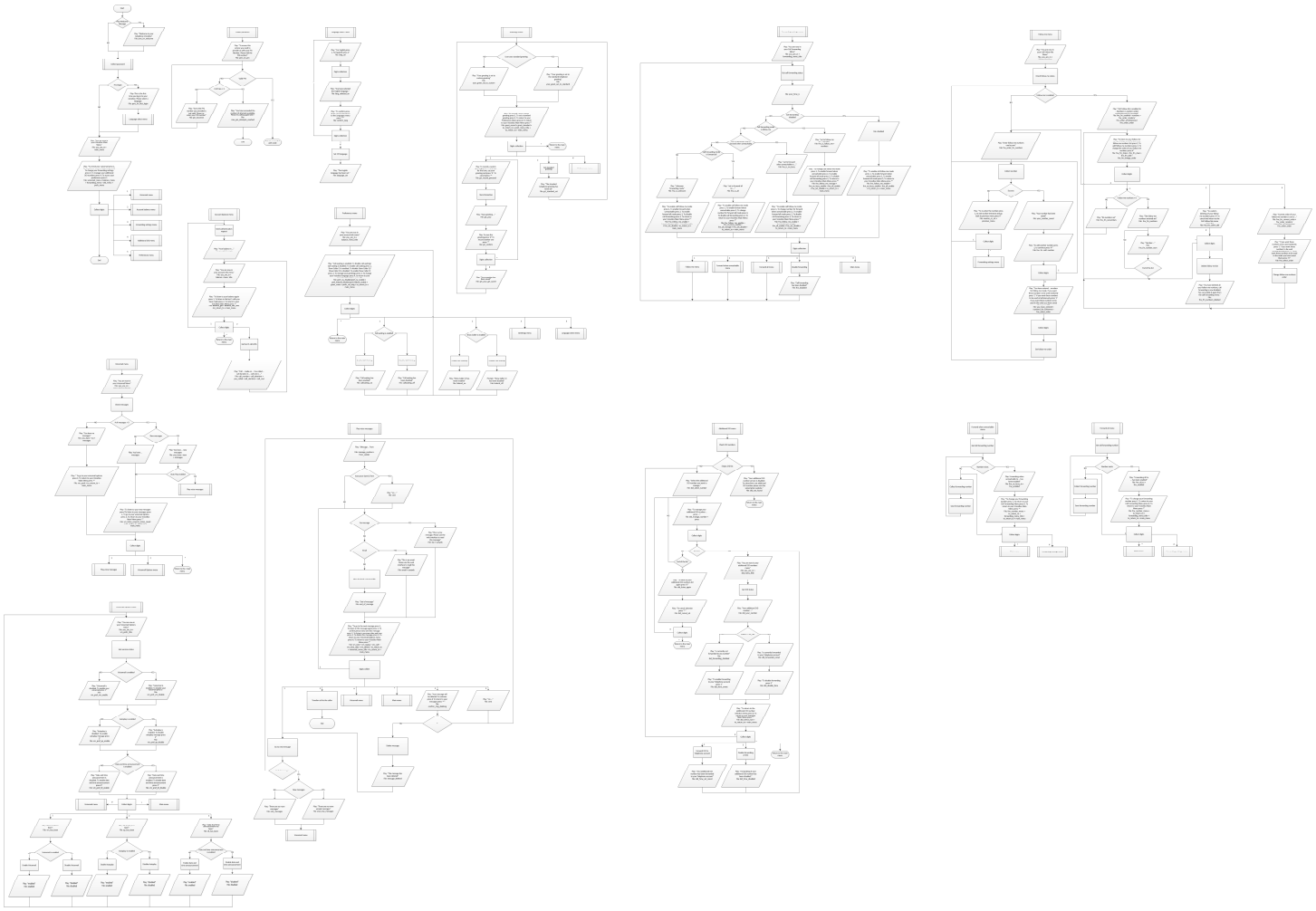
Account Self-care IVR



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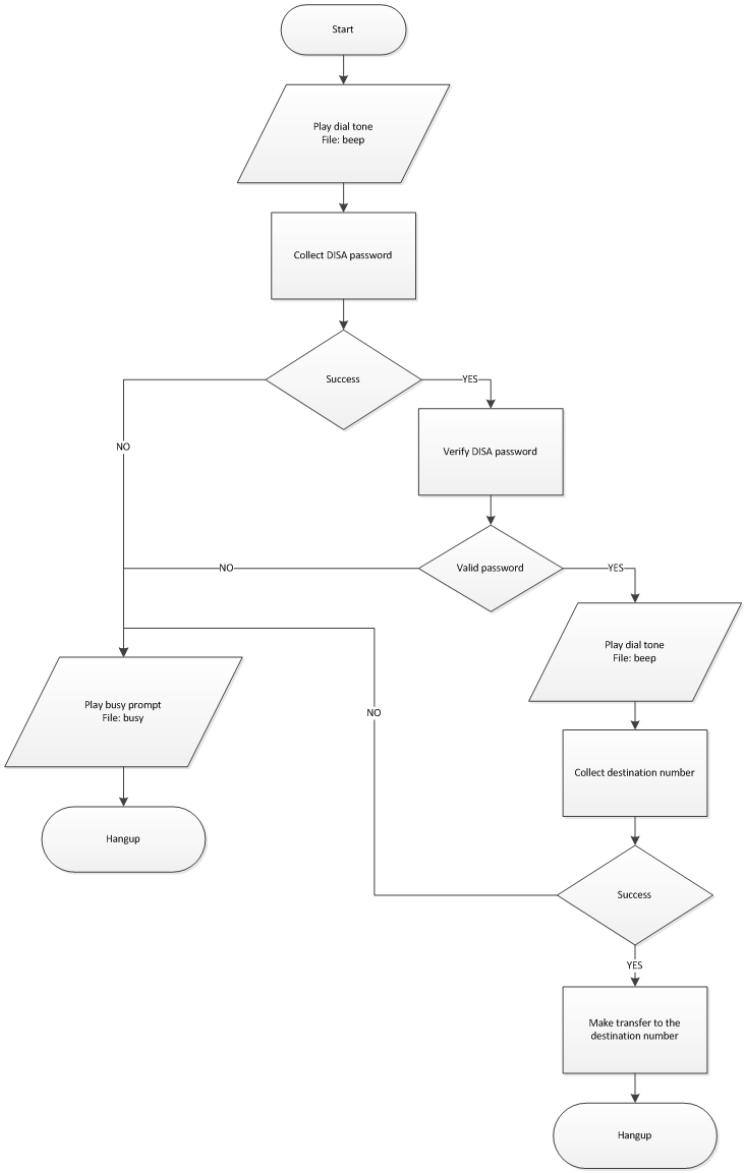
Auto Attendant IVR

Auto attendant. DISA



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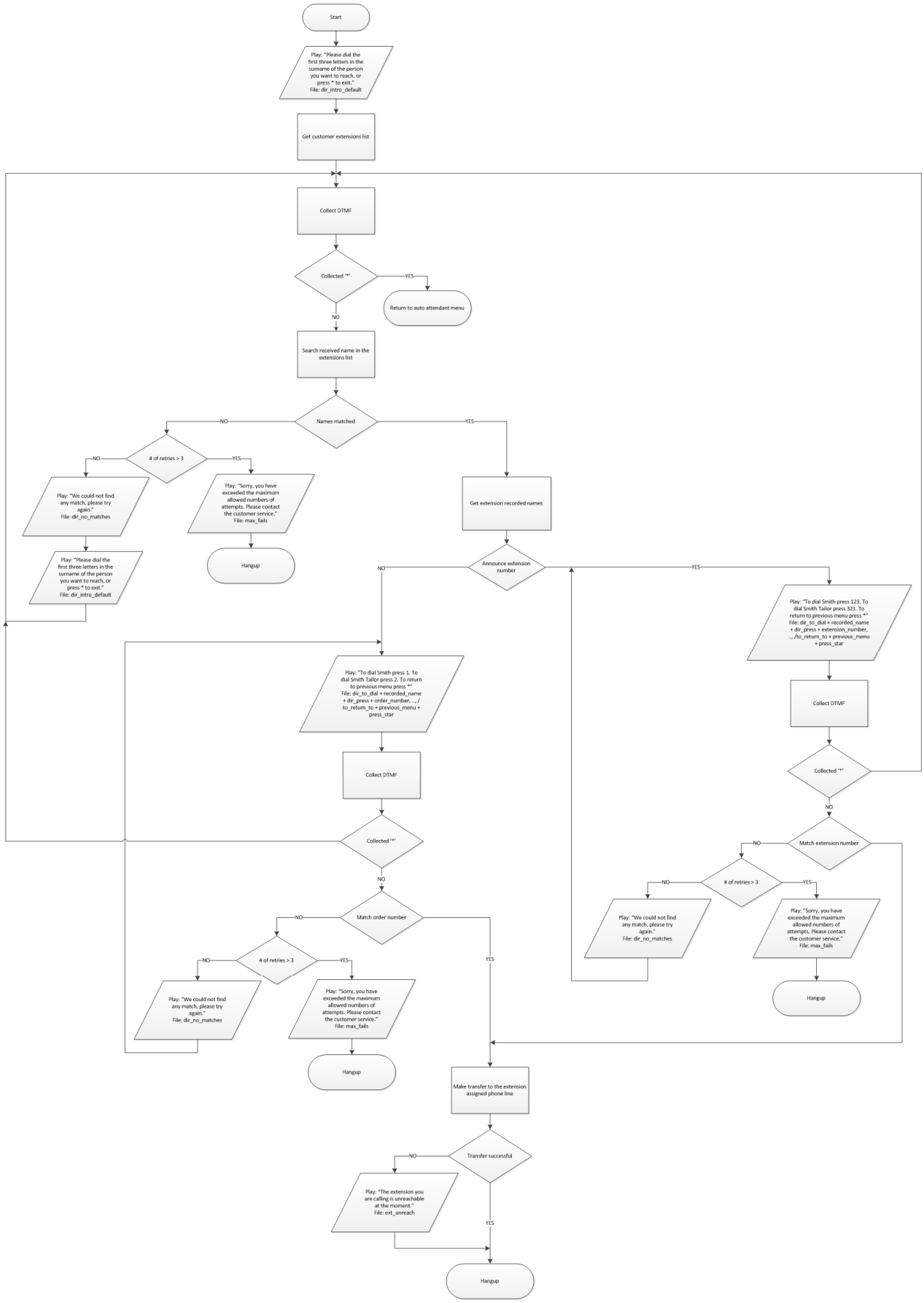
Auto attendant. Dial directory



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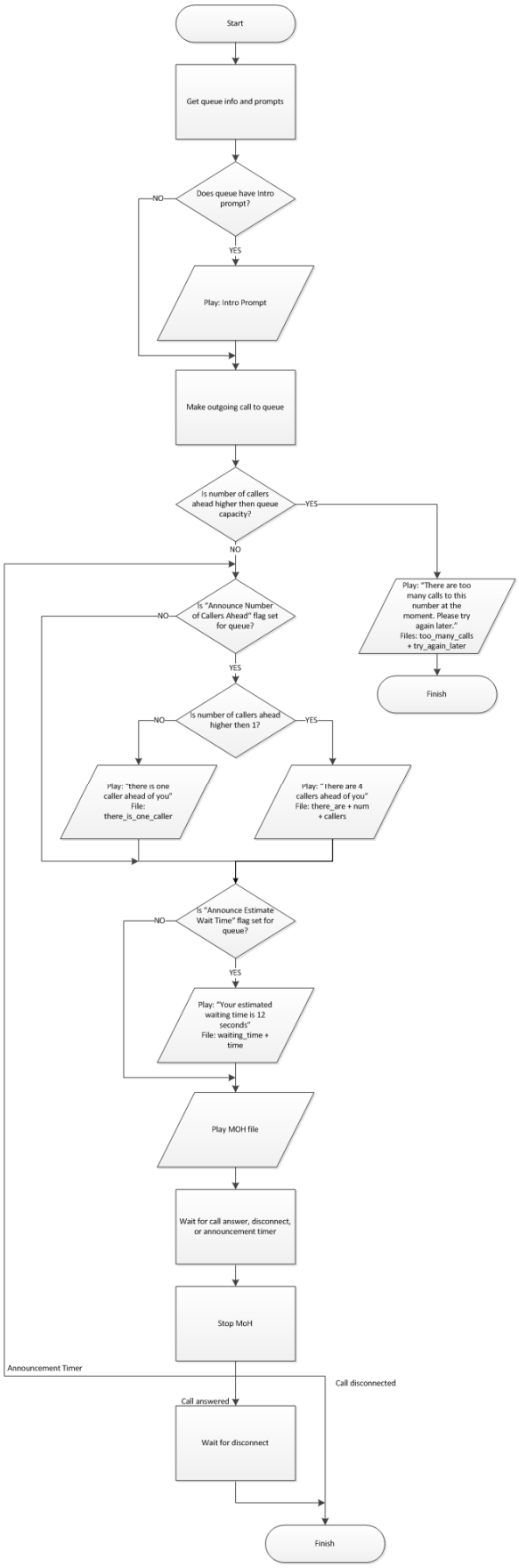
Auto attendant. Call Queues



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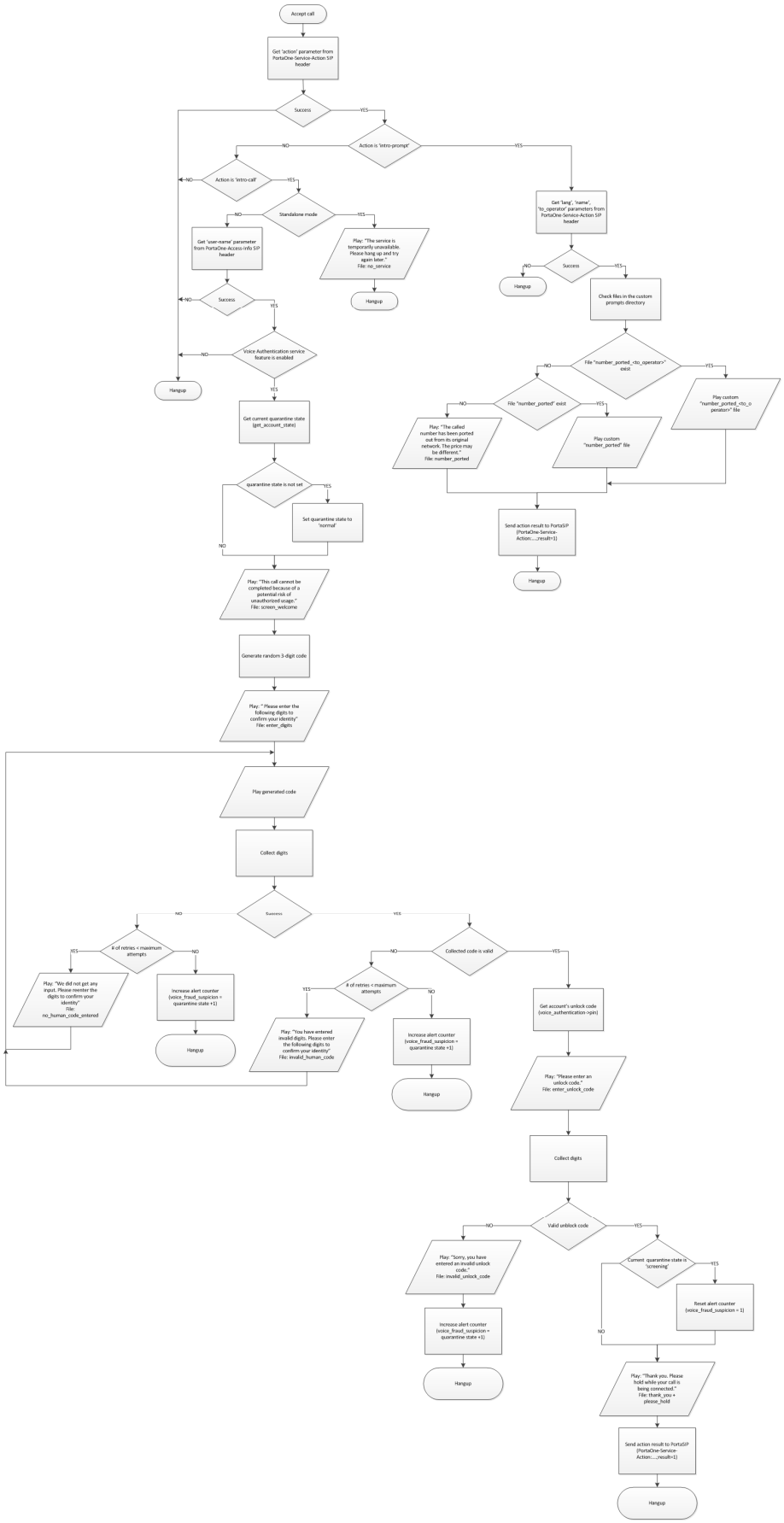
Intro IVR Application



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