UsVox Administrator Guide

Web Portal Administration

2012

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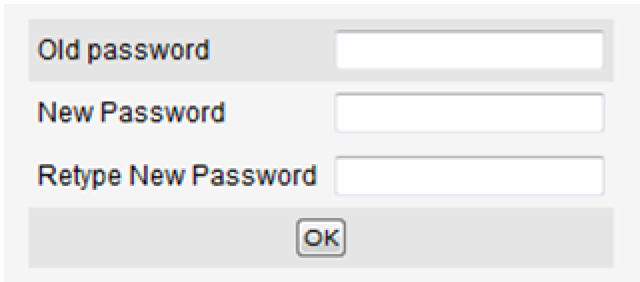
Customer

Login into web portal

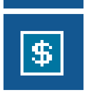
To login go to http://www.usvox.com/login the first time the administrator logs into the system, the portal will prompt to change the password.



Icons and Links


Home (First page)


Customer Info and Abbreviated Dialing

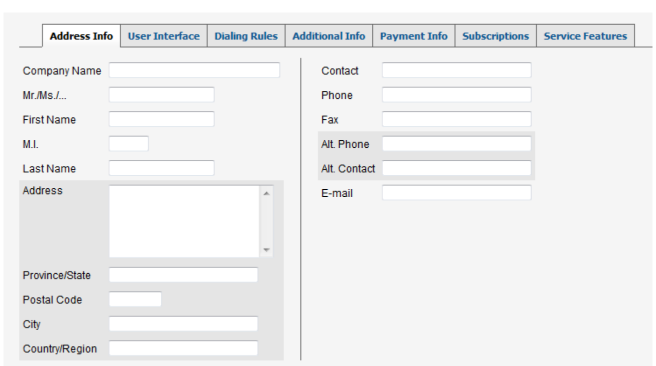

Accounts


xDR Browser, Reports, and Invoices

Customer Info

A customer is a company.
The customer's contact information is used to distribute account usage information, call statistics, invoices, and so on.

Address info

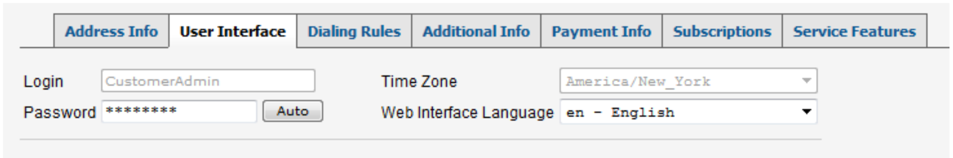


- Customer Name – Short name for the customer object; this will be used on the web interface.
- Email – An email address for the distribution of accounting information. After the billing period is over, a list of xDRs and other statistics will be sent to this address

- Bcc – Blind carbon copy in email; may be used for debug and archiving purposes.

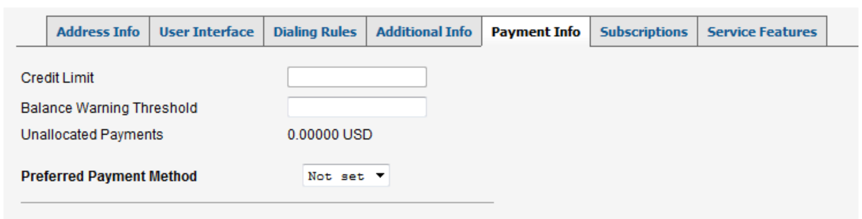
User Interface

- Password- Input the new Password or used the ‘Auto’ button to generate one.
- Web Interface Language – Language to be used on the customer self-care web interface.



Payment info

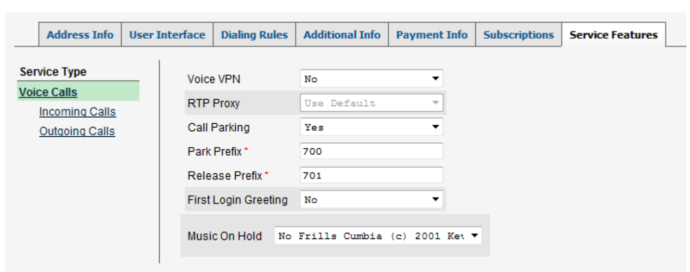
- Credit Limit – For prepaid traffic exchange, enter 0 in this field, so that the customer can only send you calls after he makes a payment (deposit).
- Balance Warning Threshold – The customer can be notified by email when his balance is dangerously close to the credit limit and service will soon be blocked. Here you can enter the value for such a warning threshold. This can be entered as a percentage (e.g. 90%). The warning will be sent when the customer’s balance exceeds that percentage of his credit limit. So, if the credit limit is USD 1000.00 and the threshold is 90%, a warning will be sent as soon as the balance exceeds USD 900.00. This is only applicable when the customer has a positive credit limit as an absolute value. The warning will be sent as soon as the balance goes over the specified value. For prepaid traffic exchange, where the credit limit is zero, you can enter 100.00 here.



Service Features

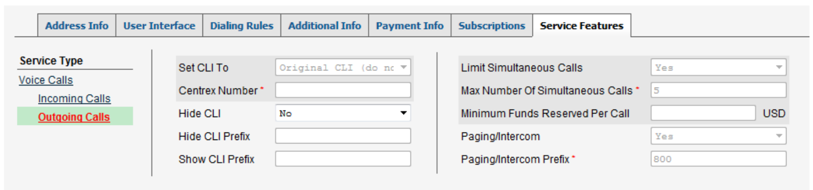
This tab allows you to define global call parameters for this customer’s environment. Most of these are related to the IP Centrex, you will find:

- Voice calls:
  Music on Hold
  Call Parking



- Incoming Calls:
  Group Pickup

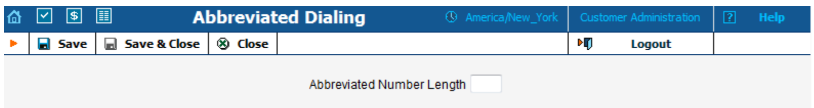
- Outgoing Calls:
  Company Outbound DID (Centrex Number)
  Paging/Intercom (set as default)



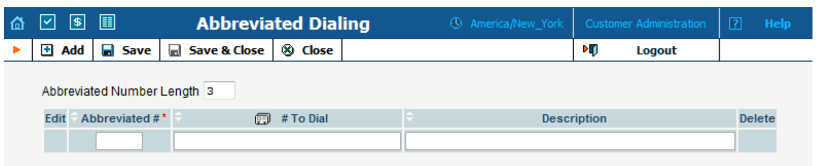
Abbreviated Dialing

If your customer has multiple SIP accounts, and plans to make calls between them, it would be very inconvenient to have to dial a complete E.164 number each time. Therefore, you can create abbreviated dialing rules, so that it will suffice to dial, for example, 120 to reach a Jeff Smith from any SIP phone using the customer's account.

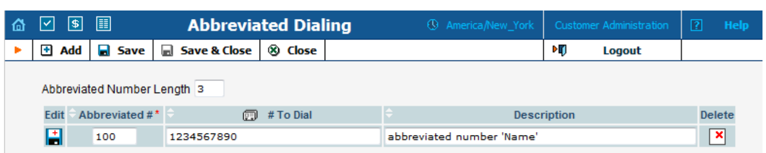
1. In Abbreviated Number Length enter the maximum number of digits in the abbreviated number (e.g. if you plan to have extension numbers 401, 402 and so forth, the length will be 3). Click Save.



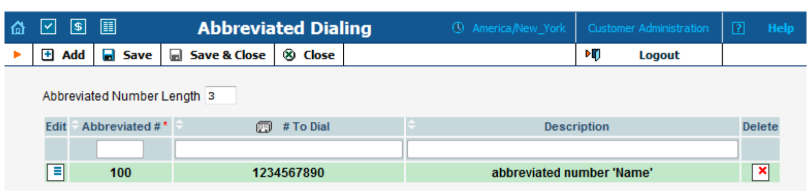
2. Now a table of abbreviated numbers will appear. Click on Add to add a new extension.



3. Enter the abbreviated number and the actual phone number the call will be forwarded to. You may use a popup window to search for a specific account. Also note that # To Dial may contain any phone number (e.g. your partner's mobile phone number), and not just one of the SIP account IDs.
NOTE: If you enter an off-net PSTN number in # To Dial; it must be in the E.164 format, (i.e. you cannot enter the number in the customer's dialing format).

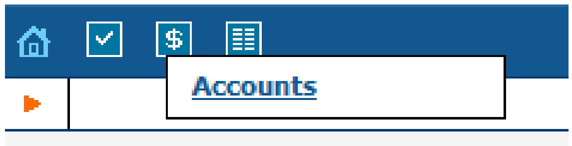


4. Click the Save button in the toolbar, or the icon on the left side of the row.



5. Repeat steps 2-4 to add all the required abbreviated numbers.

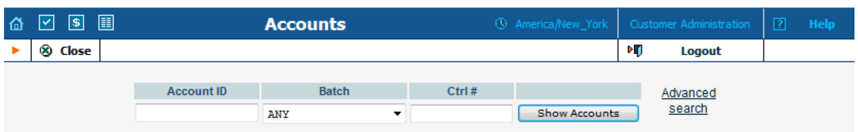
Accounts



Search for accounts

For searching press the icon  and click on the link Accounts

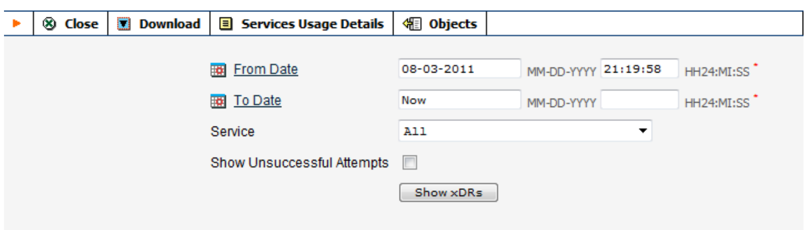
Press 'Show Account' for a full list of customer accounts or type the account number on the filed 'Account ID' and press 'Show Account'.



Account xDRs

To show Account xDRs press the icon 

Change the date by pressing the icon 

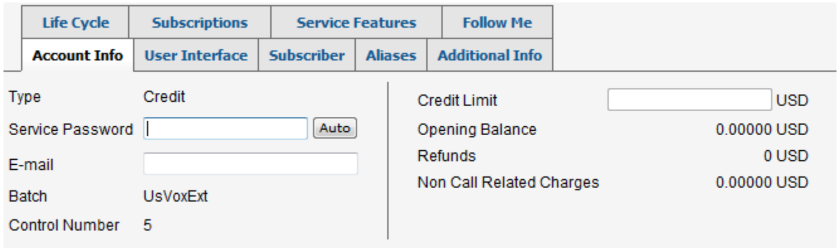


Manage account

Account Info

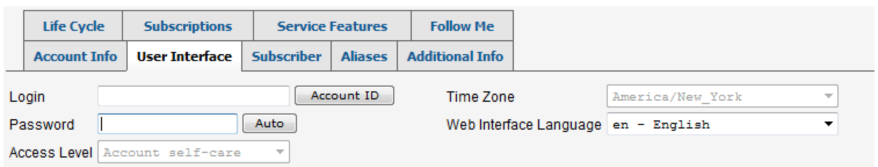
- Service Password (VoIP Password) – The account ID and this password will be used to authenticate SIP server login.
- Email – Enter the account owner's email address here. If he ever forgets his password for the web self-care pages, he will be able to reset it, and a new password will be sent to this email address.

You can also just leave this field empty

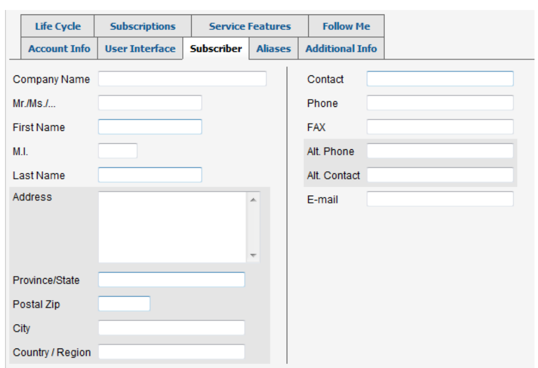


User Interface

- Login – Account login to web self-care pages. Can be the same as account ID.
- Password – Password for the web self-care pages.
- Web Interface Language – The language to be used on the customer self-care web interface.



Subscriber



- Subscriber Name – First Name and Last Name will be used to show the Account caller ID wend SIP phone device is calling to others.

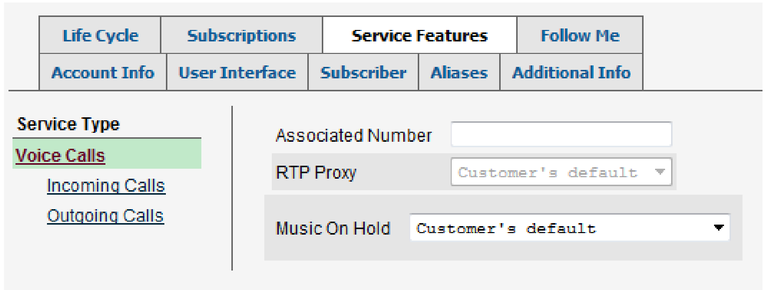
- Address info tab – General information

- Email – General information

Service Features

Music on Hold

If the account user need a specific Music on hold not set by the system administrators it can be uploaded (files under 5 mb)
This music will only be used by this user and not by the system.



Incoming Call

Set the Default Answering Mode correctly to your usage:
Immediate fallow me or forward, set to: forward only.
To get voice mail immediately if you used the account for faxing or auto attended, set to: Voice Mail.

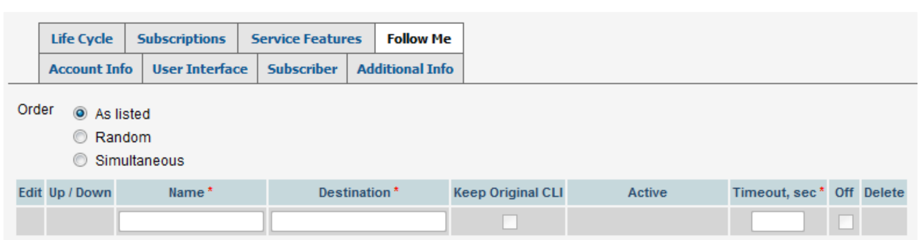
Timeout, Sec: To ring at the phone device before fallowing to the Default Answering Mode

Fallow Me

Press Add in top rigth of the page (not show in picture)
Order: Specifies the order for redirecting a call.

As listed: call every active follow-me number from the first (topmost) number to the last, until the call is answered.
Simultaneous: call every active follow-me number from the list at the same time until the call is answered.
Random: use a random order.

Edit: Click the Edit icon to edit the follow-me number details. To add a new number to the list, click the Add button .
Up, Down: Click these buttons to move a row before the previous one or after the next one in the list.
Name: Type name, Group name, or any information for the number to be forwarded.
Destination: Type the abbreviate number (Station), Account number, or the numbers to be forward used 11 digit diling for out bound numbers.
Timeout, Sec: The time for the device to ring.
Press save or save and close in top rigth of the page (not show in picture)
Off: Check this option to temporarily disable forwarding to a follow-me number.



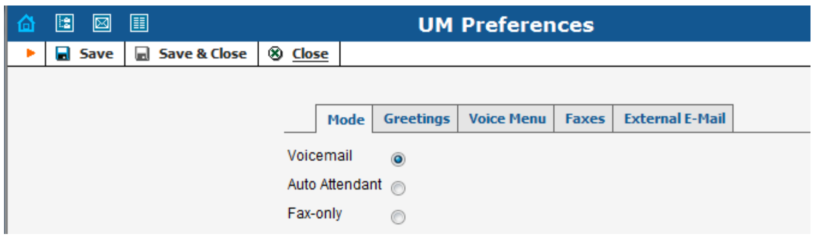
Manage UM Account

UM Preferences

UsVox UM also offers you a number of special options for accessing your email via phone.

Mode

You can choose between two basic UM options: voicemail or auto attendant.



Voicemail
Maintain the account as a regular voicemail box for a SIP phone device.

Auto Attendant
UsVox UM Auto Attendant is a flexible utility designed to greet callers and transfer them either to an existing UsVox Switch accounts, or to your current phone system.

Basic concept
UsVox UM Auto Attendant (AA) is composed of a set of menus.

All the menus are the same in every respect, except for the ROOT menu, which is always present and cannot be deleted, and whose name cannot be changed.

When a caller dials the system, AA will answer the call with the Intro prompt from the ROOT menu.

After this, the Menu prompt will be played, and AA will listen to the user input.
The user input will trigger the execution of the following available actions:

- Default – plays the default prompt from the current menu
- Transfer – transfers the call to a given telephone number or extension
- Menu – starts interpreting (executing) the selected menu; the user can choose from any of the available menus

The user may select whether the corresponding prompt is to be played prior to the action.

Please read more under the Auto Attended section Page-

Greetings

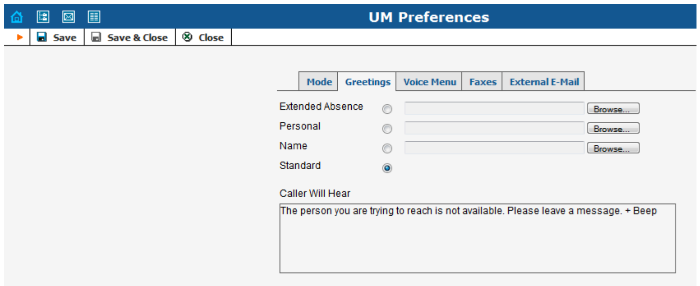
Using the Voice Menu tab, you can record, play back, and change the type of greeting message callers will hear when you are not available.

First choose between Extended Absence for a message providing your full name and number Standard: for a message giving your name only or Personal: for a message of your own choice.

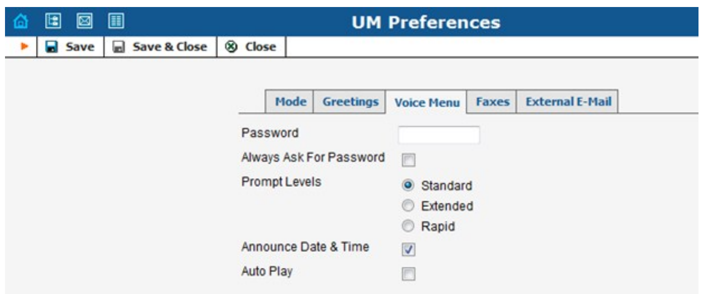
You can either upload an existing sound file using the Browse buttons on the right, or record a new one with a microphone attached to your computer. A green play button next to the message type indicates that a sound file already exists, while a red play button means that none is currently available.

To record a message, click the red record button. When you click record, all other buttons are disabled. Alternatively, you can use the radio buttons to choose and then record a given message type.
A blinking play button indicates that a message has been successfully recorded. To save the recorded message, click Save in the action panel above. To re-record a message, click the undo button next to the record button.

Two buttons are active when playing back messages: play and pause.



Voice Menu



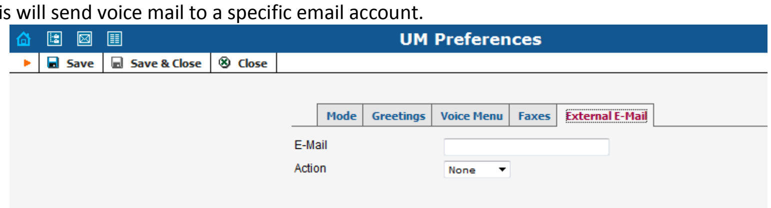
- PIN (Password) You can change the PIN you use to access your mailbox via phone.
- Always ask for PIN in some cases, the system may recognize you when you call from your home number, and will not request your PIN (ask your system administrator if this feature is supported). If you want to be asked for your PIN anyway, check this box.
- Prompt levels UsVox UM offers three voice prompt settings in each supported language: Standard, Extended, and Rapid. When entering an account number, the voice prompt for each setting is as follows:

Standard: "Please enter your account"
Extended: "Please enter your account number and press pound"
Rapid: "Account"

- Announce Date & Time You can switch this function on or off.
- Auto Play You can have new messages played automatically by checking this box.

External E-Mail

External E-Mail forward, this will send voice mail to a specific email account.



Auto Attendant

Basic Concept
UsVox UM's auto attendant is composed of a set of menus.
All the menus are identical in every respect, except for the ROOT menu, which is always present and cannot be deleted, and whose name cannot be changed.
When a caller dials the system, auto attendant will answer (connect) the call and proceed to the ROOT menu.
If a user tries to access a menu which is not currently active, the action specified in the Not Active configuration parameter will be performed; for instance, the user may be automatically forwarded to an "after hours" menu.
The Intro prompt (e.g. "Welcome to PortaOne, a VoIP solutions company!") is played when a user enters a menu for the first time.
After this, the Menu prompt will be played, listing all the available options (e.g. "Press 1 for sales, press 2 for technical support"), and auto attendant will collect the digits dialed by the user on his phone touchpad.
If no input is received (timeout), the Default prompt is played and the dialog reverts to the previous step (i.e. plays the Menu prompt and collects the user's input).
The user's input will be matched with the corresponding menu items, and the action associated with this item will be performed. The following actions are possible:

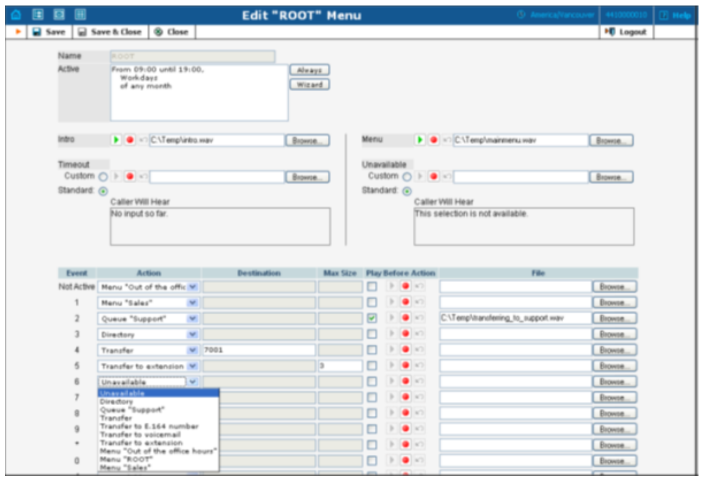
- Default – Plays the Default prompt from the current menu and returns to the "Play Menu prompt" step (this is the action used for all menu items where the initial value has not been modified).
- Transfer – Transfers the call to a given telephone number or extension. The phone number should be entered 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).
- Transfer to Extension – Transfers the call to an extension number entered by the caller from his phone. To prevent abuse (e.g. someone attempting to enter a long-distance number in this way), you can specify the maximum allowed number of digits in an extension (Max Size). Use the abbreviated dialing quantity of number of digits, if you are using 100, 101, 102 and so on, specify the maximum allowed to: 3
- 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. Also used the 'UM Preferences', External E-Mail with forward, this will send voice mail to a specific email account.
- Menu – Transfers the user to the selected menu.
- Directory – Launches this company's dial-by-name directory.
- Queue – Transfers the call to the specified call queue.

You may select whether the corresponding Before Action prompt is to be played prior to the action.
A call menu flow chart is shown in the diagram below.

Menu List

Auto attendant can be selected from the Options menu. The main screen lists all the menus available in the system.
To modify one of the existing menus, select its name from the list.
To add a new menu, select Add from the action panel.

Menu Mode



After selecting one of the existing menus, please allow all the prompts to load in your browser; this may be viewed on the status indicator in the action pane.

The fields of the Menu Edit screen are explained below:

Name:
A logical name for the menu (i.e. Sales for a sales department)

Active:
Time definition for when the current menu is active. To set the menu as always active, select the Always button on the right. [PortaUM also provides users with a Period Wizard]

Period Definition Wizard:
Via a series of screens, the user may select a time interval, day of the week, day of the month, and month; multiple selections are allowed.

The following example shows how to create a period starting at 6 pm every day and lasting until 6 am the next morning. Another interval is used on weekends. We will also include some holidays, e.g. January 1 and December 24-26.
In the first screen, select 6 pm in the ‘From’ column and 6 am in the ‘Until’ column. Now select the Next button. The two text areas on the right side of the screen provide the user with a display of the current period definition. The top text area displays a verbal definition of the period: From 6:00pm until 6:00am, while the bottom one contains the same information in a format which can be parsed by The PBX: hr{6pm-6am}.
These sets up the first period; in order to continue, skip the following screens by pressing the Skip or Next button, until the Period definition completed message is displayed. Click Add to create another period definition, and the wizard will return to the first screen.
Now for weekends: by pressing the Skip or Next button, go to the second screen and select Weekend, or hold the <Ctrl> key and select Saturday and Sunday from the list. Now use the Next button, skipping forward until the Period definition completed message is displayed. Click Add to create another period definition.
To include January 1st in the period definition, skip to the Day of the Month screen and select 1. Now click the Next button. Select January and click Next, skipping forward until the Period definition completed message is displayed. Follow the same steps to select the December 24-26 interval. Hold the <Ctrl> key to select multiple entries.

To review your work, look at the top text area. The following should be displayed:

From 6:00pm until 6:00am

any day of any month

OR Sunday and Saturday
of any month
OR 1
of January
OR 24-26
of December
If the definition is correct, select Finish.

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

- Record. Select to start recording your voice prompt. (You will need to connect a microphone to your computer's sound card to use this feature.)
After the existing prompt has been recorded over, the Undo icon becomes available, allowing rollback to the previous state. The blinking Play icon indicates that the existing prompt is being overwritten, but changes have not been saved yet.

- Stop. Select this to stop recording or play back the recorded message.

- Play. Select this to play back the recorded prompt. When selected, this icon will turn into - Pause.
Each of the icons above may appear in grayscale, meaning it cannot be accessed because some other task is active.
To give your auto attendant a professional sound, we recommend using a professional speaker and a digital recording studio when recording voice prompts.
To upload a prompt, select the Browse... button on the right side. The native audio file format for the system is the following:
Type: NeXT/Sun (Java) file .au
Format: G.711 u-Law
Attributes: 8,000 Hz, 8-bit, Mono

UsVox UM uses SOX - Sound eXchange, a universal sound sample translator for prompts uploaded into native UM format.
Here is a short list of supported audio file formats

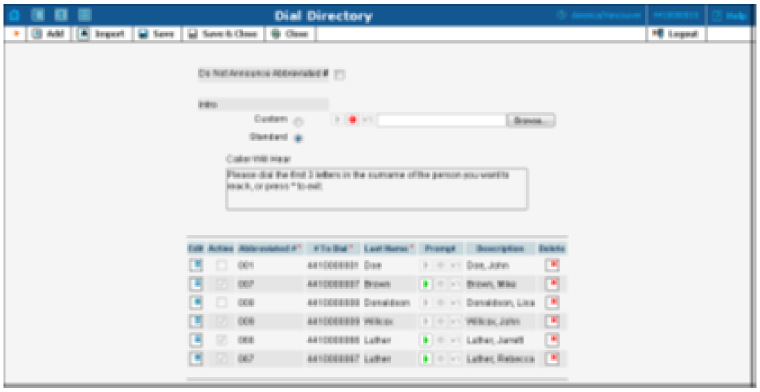
Here is a short list of supported audio file formats: Type	Description
.aiff	AIFF files used on Apple IIc/IIgs and SGI.
.au	SUN Microsystems AU files.
.gsm	GSM 06.10 Lossy Speech Compression
.mp3	MP3 Compressed Audio
.ogg	Ogg Vorbis Compressed Audio.
.raw	Raw files (no header).
.wav	Microsoft .WAV RIFF files.

Event Table

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

Dial Directory

This is another element of the auto attendant IVR functionality. If a caller does not know the extension number of the person he is trying to reach, he may look up the called party using the first three letters of his surname.



Every UM account has its own dial-by-name directory; however, an initial list of extensions may be imported from the main list in The PBX to save time and effort. You can upload a voice prompt with the actual person's name for each extension. You may also exclude certain extensions from being accessible via dial-by-name (e.g. you do not want tele-marketers to directly reach your CEO or CFO because their names are publicly accessible).

The dial-by-name directory can be assigned as an "action" item to any element in the ROOT menu or sub-menu. When a user reaches the dial-by-name dialog, he will be prompted to enter the three first letters in the called party's surname. Standard phone mapping is used, i.e. 2 is ABC, 3 is DEF, and so on. If no matching person is found, the user is informed of this, and may then re-enter the name or press * to exit. If more than one match is found (e.g. there are two persons with the "same" surname in the company, e.g. 276 will match both Brown and Asok), the user will hear a list of matching names and their extensions, and may then enter the correct extension.

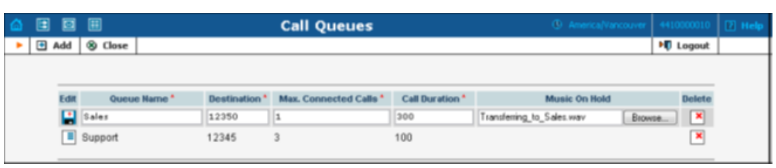
Two modes of selecting the call transfer destination are available:

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

Call Queues

This feature allows you to provide a "call center" functionality to your IP Centrex customers. When a large number of incoming calls from customers arrive to the auto attendant, PortaUM can forward these calls to the actual agents (customer service representatives) in a regulated fashion.

Every call queue contains several configuration parameters:



- Destination – A phone number where an outgoing call may be forwarded to. The fact that this is a single number does not, of course, mean that you can only have one agent answering calls. This is simply the number used to forward calls to The PBX. On the PBX side, you can use the call forwarding feature to direct an incoming call to multiple IP phones. Or, alternatively, you may forward this call to an external IP PBX or a gateway with multiple FXS ports; then "hunting" for an available agent will be done on that side.
- Max Connected Calls – This defines the "bottleneck" of your queue (i.e. the maximum number of concurrently connected outgoing calls).
- Call Duration – The average expected processing time for each call (used to calculate the estimated waiting time).
- Music on Hold – A melody (or announcement) which is played to users waiting to be connected.

Each call queue contains a pool of incoming calls (users trying to get connected) and a number of connected outgoing calls (calls that have already been connected to agents). When a new incoming call arrives, it is assigned a position in the queue. The caller will hear an announcement about his position in the queue and the estimated waiting time, which is calculated as (average call duration) / (maximum number of connected calls) * (total number of users before him in the queue). After that, the specified "music on hold" is played, and every minute the caller is updated as to his current position in the queue and the estimated waiting time.

If there are callers on hold and the number of connected outgoing calls is lower than the specified threshold, UsVox UM will attempt to connect the first person in the queue. A call invitation is sent to the destination number in The PBX, and UsVox UM waits until the call is answered by the other side. If the call is not connected on the first attempt (some representatives may not be available at the moment), UsVox UM will make another attempt, then another one, and so on. This will continue until either all of the incoming calls are connected, or the maximum threshold for outgoing calls is reached. In the latter case, UsVox UM will simply wait until one of the agents finishes serving his current customer, thereby disconnecting one call and making “room” for a new outgoing call.

Glossary

Call Queuing:
Call Queuing is a sophisticated queuing system that allows you to accept more calls into your telephone system than you have extensions or employees capable of answering them. It allows you to deal efficiently with calling peaks without losing valued customer's calls and projects a professional image of your business. With Call Queuing, instead of getting an engaged tone your customers are answered automatically and held in a queue. While they are waiting for a representative they receive personal messages about how many calls are in front of them followed by music while they are waiting.

Centrex: (central office exchange service)
Centrex is a service from local telephone companies in the United States in which up-to-date phone facilities at the phone company's central (local) office are offered to business users so that they don't need to purchase their own facilities.

DID: (direct inward dialing)
An Internet Engineering Task Force (IETF) standard for initiating, maintaining, and terminating an interactive user session involving video, voice, chat, gaming, virtual reality, and more.
Most businesses have several incoming telephone numbers used for specific purposes. For example customer service, sales, etc. Some have an individual telephone number for each user in the system. In a home setting on the other hand, each telephone number comes in on a different pair of wires typically. This is not practical in a business environment that has many telephone numbers.

E.164:
Is an ITU-T recommendation which defines the international public telecommunication numbering plan used in the PSTN and some other data networks. It also defines the format of telephone numbers. E.164 numbers can have a maximum of fifteen digits and are usually written with a + prefix. To actually dial such numbers from a normal fixed line phone, the appropriate international call prefix must be used.

E911:
E911 is the short form of the term Enhanced 911, and is used for providing emergency service on cellular and Internet voice calls.

Find-me/follow-me:
A feature that allows calls to find you wherever you are, ringing multiple phones (such as your cell phone, home phone, and work phone) all at once.

FXS and FXO:
These are the names of ports used by Analog phone lines (also known as POTS - Plain Old Telephone Service) or phones.
FXS - Foreign eXchange Subscriber interface is the port that actually delivers the analog line to the subscriber. In other words it is the 'plug on the wall' that delivers a dialtone, battery current and ring voltage.

FXO - Foreign eXchange Office interface is the port that receives the analog line. It is the plug on the phone or fax machine, or the plug(s) on your analog phone system. It delivers an on-hook/off-hook indication (loop closure). Since the FXO port is attached to a device, such as a fax or phone, the device is often called the ‘FXO device’.

Intercom: (intercommunication device)
A talkback or doorphone is a stand-alone voice communications system for use within a building or small collection of buildings, functioning independently of the public telephone network. Intercoms are generally mounted permanently in buildings and vehicles. Intercoms can incorporate connections to public address loudspeaker systems, walkie talkies, telephones, and to other intercom systems. Some intercom systems incorporate control of devices such as signal lights and door latches.

IP: (internet protocol)
This defines the way data packets, also called datagrams, and should be moved between the destination and the source. More technically, it can be defined as the network layer protocol in the TCP/IP communications protocol suite.

IVR: (interactive voice response)
In computer telephony, Interactive Voice Response is a horizontal application wherein computer-based information is accessed over the phone - with a telephone versus a computer. An IVR platform uses computer telephony components to translate callers' touch-tones or voice commands into computer queries after the callers hear an audio menu. For example: "Please enter your account number using the touch-tones on your telephone." These queries are then "fetched" by the IVR platform from the host computer. In some cases, the information resides in the same platform (self-hosted). The information is then converted into voice commands and then spoken over the phone to the caller. These spoken prompts can be pre-recorded, digitized speech messages that are then concatenated to form whole sentences. For example: "Your bank balance is five hundred and sixty-three dollars". The responses to the caller an also take the form of text-to-speech prompts. IVR systems can also be used for callers to change the information in a database instead of just "listen" to the information.

MOH:
Music on Hold

PBX: (private branch exchange)
In telephony, a PBX system behaves as a customer's premises over trunk lines (thus the term "branch"). At first, PBXs mimicked a small telephone company switchboard. Users would use an operator to take and make telephone calls to and from the PSTN (Public Switched Telephone Network). Over time, users were able to dial directly, without the use of an operator. Today, computer telephony platforms such as automated attendants are able to route incoming calls automatically, too.

PSTN: (public switched telephone network)
This refers to the telephone system that transmits analog voice data. Till recently, PSTN was the heart of all phone systems worldwide. However, most of the developed world is now switching to or has switched to telephone networks that are based on digital technologies, such as ISDN and FDDI. RJ45: RJ45, which is the acronym of Registered Jack-45, is a telephone connector that is used in Ethernet and Token Ring Type 3 devices. It has eight "pins" or electrical connections.

SIP: (session initiation protocol)
SIP can establish multimedia sessions or Internet telephony calls, and modifies, or terminates them. The protocol can also invite participants to unicast or multicast sessions that do not necessarily involve the initiator. Because the SIP supports name mapping and redirection services, it makes it possible for users to initiate and receive communications and services from any location and for networks to identify the user's wherever they are.

Telephony:
Taken from Greek root words meaning "far sound", telephony is the discipline of converting or transmitting voice or other signals over a distance, and then re-converting them to an audible sound at the far end.

UM: (sometimes referred to as the unified messaging system)
Is the handling of voice, fax, and regular text messages as objects in a single mailbox that a user can access either with a regular e-mail client or by telephone. The PC user can open and play back voice messages, assuming their PC has multimedia capabilities. Fax images can be saved or printed.

VoIP: (Voice over IP)
The process of making and receiving voice transmissions over any IP network. IP networks include the Internet, office LANs, and private data networks between corporate offices. The main advantage of VoIP is that users can connect from anywhere and make phone calls without incurring typical analog telephone charges, such as for long-distance calls.

Web Interface: (User interface)
In the industrial design field of human–machine interaction, the user interface is the space where interaction between humans and machines occurs. The goal of interaction between a human and a machine at the user interface is effective operation and control of the machine, and feedback from the machine which aids the operator in making operational decisions

xDR:
Call report, xDR Reports allow you to search for individual calls and analyze the technical message details. It is a fast and flexible tool for analyzing and a great way to do error tracking. xDR Reports are based on historical xDR Repositories (SQL databases).