



PBX Manager Portal v2.0

Introduction

Virtel's PBX Manager is a powerful tool that allows to configure a PBX very quickly and makes day-to-day PBX maintenance accessible to less technical users, while allowing addition of practically unlimited features by extending the system using Feature Codes and Script Library.

Management screens overview

Management screens of PBX Manager GUI are divided in the following categories:

PBX Settings

Schedules - time intervals (schedules) that are used to manage time based call handling. These schedules are currently used in Inbound Routes.

Media Files

Voice Recordings - recording of announcements and voice prompts used throughout the system and, in particular, by the Voice Menus (or auto-attendant).

Music-on-Hold - management of music-on-hold play lists and upload of sound files.

DIDs & Trunks

DIDs - management of an inventory of DIDs (phone numbers) that can be assigned to tenants.

Extensions & Directory

Bulk Generator - quick generation of multiple users, extensions, directory entries and mailboxes and creation of configuration files for phones' auto-provisioning. Used primarily during initial system configuration. Source file is a comma delimited file easily generated with Excel or a similar tool.

User Extensions , primary tool for the creation and management of user extensions. When defining a new user a phone model and MAC address may be specified to create the required files to automatically provision an IP phone when it's connected to the network.

Directory , management of directory entries which can include both extensions and external phone numbers. This directory is used for a ,click-to-call,Ä feature in the User Portal.[File:Clear.gif](#)

Routes

Inbound Routes ,, rules for how the inbound calls are handled and to route specific DID to specific applications (voice menus, voicemail, conference bridge, etc)

Outbound Routes - rules on how the outbound calls are routed.

PBX Features

IVR / Voice Menus - auto attendant and unlimited multi-level voice menus.

Hunt Lists ,, management of lists of ring groups that are dialed in sequence.

Feature Codes - extending PBX functionality by associating scripts that can contain groups of PBX commands with internal numbers.

Special Lines ,, SIP and IAX registrations not associated with users in addition to the primary registrations that are created automatically when creating User Extensions.

Special Mailboxes ,, mailboxes in addition to those that are created automatically when creating User Extensions.

Conference Rooms ,, conference rooms with real-time conference monitoring and management.

ACD / Call Queues

Agents ,, management of agents servicing call queues.

Queues ,, management of call queues.

Call History

Recorded Calls allows to view, search and manage recorded calls.

Call Detail Records allows to view and search call history.

User Portal screens overview

When PBX Manager creates a User Extension, it automatically creates a username with which the user can login and manage his/her extension using User Portal (call forwarding, listen/manage voicemail, etc.).

These screens are available to all end-users created using PBX Manager GUI.

Administrators also have access to these screens (and can manage user options on end-users' behalf) through **User Portal** links on the **User Extensions** screen.

Voicemail Settings - allows end-users to change their personal settings for voice mail handling.

Call Forwarding - allows end-users to configure call forwarding and follow-me.

Call Screening ,, allows end-users to configure call screening rules.

Call Blocking - allows end-users to configure call blocking rules.

Call Recording - allows end-users configure call recording rules.

Conferences - allows end-users to configure assigned conference rooms and monitor them in real-time.

Contacts & Directory - gives end-users an ability to access office directory, manage their personal contacts as well as initiate calls through ,click-to-call,Äù feature.

CRM Screen Pops ,, reverse caller id lookup in a CRM or search engines on incoming calls to the user's extension.

Voicemail - gives end-users an ability to access and manage voice mail messages.

Recorded Calls - provides end-users with access to recorded calls.

Call History - provides end-users with a Call Detail Records viewer, with selection, sorting and click-to-call capabilities.

Preferences - allows end-users to set personal web interface options.

The details of using these screens are provided in the User Portal Guide.

Configuring basic PBX

Creating extensions

The fastest way to create User Extensions (extensions associated with phones and users) is to upload a CSV file in **Extensions & Directory -> Bulk Generator**. Alternatively you can use a table in the **Bulk Generator** screen or create extensions individually by using **Extensions & Directory -> User Extensions**.

Since **Bulk Generator** relies on various default values, go to **PBX Settings -> Default Values** screen and review or modify the default settings for your devices (codecs, DTMF mode, NAT, etc) in the Phone Template and voicemail settings in the Voicemail Template.

You should also review Extension creation options , User Portal creation options , Conference Rooms' creation options (if creating personal conference rooms for your users).

If you are going to use auto-provisioning make sure that you selected a default phone model in Extension creation options.

For auto-provisioning you also have to select an auto-provisioning method and a directory for creation of configuration files.

Go to **System Settings -> Provisioning Settings** and verify that your provisioning settings are specified correctly. PBX Manager web server can be used to provision devices using HTTP or HTTPS, otherwise you must ensure that your provisioning server (FTP or TFTP) is running and serving files from the directory you specified (where PBX Manager will put the configuration files for each device).

Enter users' information into a CSV file in the format specified on the **Bulk Generator** screen (you can download the sample file) and upload it using **Bulk Generator** (Note that a MAC address is required for each user's phone if you would like to auto-provision them at this time).

This step will create user information, extensions, mailboxes, directory entries, personal conference bridge and optionally (if DIDs were specified) Inbound Routes.

When using Multi-Tenant PBX Manager user web login for the User Portal is xxx-extension where xxx is a tenant name, otherwise it is the same as extension. Default passwords for web login are the same as extensions.

At this point end-users can login into the User Portal and manage their own settings.

Configuring auto-attendant(s)

If you would like one or more of the incoming lines to be answered by an auto-attendant you can configure that next. You can set up an auto-attendant using **PBX Features -> IVR / Voice Menus**. Multiple auto-attendants can be also configured - each for a different external phone number (DID).

Each Voice Menu requires an Announcement , which you can record using a PBX phone as your are defining the VoiceMenu or from **Media Files -> Voice Recordings**. For professional recording (matching the rest of Asterisk sounds) we recommend Allison Smith at <http://www.theivrvoice.com>. Once the recordings are completed, go to **PBX Features -> IVR / Voice Menus** screen and create or edit a Voice Menu to suit your needs.

Once the Voice Menu is created, go to **Routes -> Inbound Routes** and create an inbound route for each external phone number (DID) you are going to direct to an auto-attendant. When creating Inbound Routes, you can select an existing Schedule or create a new Schedule using **PBX Settings -> Schedules** screen. Select a schedule, pick ,Go to Menu,À script and select your Voice Menu from the list.

System Settings

Schedules

Schedules screen allows you to define groups of date/time ranges used for time dependent call handling. These time intervals are currently used in Inbound Routes.

Existing Schedules are presented in a list. Clicking on the **Name** column displays **Edit Schedule** screen.

To create a Schedule, click on the Create Schedule button.

To delete a Schedule, select it using the check box on the left and click on Delete Selected.

Edit/Create Schedule

Name. Alphanumeric unique schedule name.

Descriptive name. Short user friendly name of the schedule.

Description. Schedule description

Using Add Time Range button user can add up to 10 time ranges. Note that an '*' specifies 'any', so for example leaving month and Day-of-Month as '*' and selecting from Monday to Friday would mean that the time range will be true Monday through Friday regardless of the month. Specifying the same Day-of-Week range and also selecting January in the from Month would mean 'Monday through Friday in January', etc.

You can remove time ranges by clicking Remove button.

You can save the schedule you are editing by clicking Save/Create, duplicate the schedule you are editing by clicking Clone, delete the existing schedule by clicking Delete, or go back to the list of schedules by clicking Cancel.

Script Library

Script Library section allows you to create scripts, which are groups of Asterisk commands. Scripts are fundamental building blocks in Virtel Voice PBX Manager and are used in Bulk Generator, User Extensions, Feature Codes, Voice Menus, Inbound Rules, Outbound Routes, Hunt Lists, etc where they are associated with extensions, feature codes and called numbers. Dialing an extension/feature code executes an associated script.

Virtel Voice PBX Manager comes with a number of scripts which can not be modified. Users can clone these scripts or create their own scripts to extend PBX functionality as necessary.

PBX Manager displays an installed version of the Virtel Voice scripts shipped with PBX Manager in the upper right corner of the screen. Clicking on ,Check for Script Updates,À allows you to check if the newer scripts are available and install the new scripts. Scripts created by PBX Manager users are preserved during the update.

Installed Virtel Voice and user scripts) are presented in a list.

Clicking on the Name column displays **Edit Script** screen.

Select scripts that can be used for drop down allows you to limit the list to the scripts that are intended for use in inbound routes, user extensions, feature codes, etc.

To create a script, click on the **Create Script** button.

To delete a script, select it using the check box on the left and click on Delete Selected. Note that Virtel Voice scripts shipped with PBX Manager are protected and cannot be modified or deleted.

Edit/Create Script

Name. Alphanumeric name of the script, no spaces allowed. We recommend you use a common prefix for your scripts. Virtel Voice scripts shipped with PBX Manager start with tl- prefix. You can use Virtel Voice scripts directly or create your own scripts either from scratch or by cloning and changing Virtel Voice scripts. Please do not use tl- prefix for your scripts.

Example:

ae-stdexten

Script can be used for:. Select, by checking the checkbox, of each category that is appropriate for this script. This will determine whether the script will be presented as an option when creating inbound routes, outbound routes, feature codes, etc

Descriptive name. Short user friendly name of the script that will be shown while assigning this script to extensions, feature codes, etc.

Example:

Standard Extension

Description: Description of the script

Example:

Rings phone(s) and leaves voice mail.

Asterisk Commands: Asterisk commands that will be executed when this script is invoked. Asterisk rules for macros apply, so scripts arguments will be referred to as `${ARG1}`, `#{ARG2}`, etc. Refer to www.voip-info.org for a description of the Asterisk commands.

Example:

```
exten => s,1,Wait(1)
```

```
exten => s,n,Set(CALLERID(name)=${ARG3}${CALLERID(name)})
```

```
exten => s,n,MeetMe(${ARG1},${ARG2})
```

```
exten => s,n,Playback(vm-goodbye)
```

```
exten => s,n,Hangup
```

You can add script arguments by clicking Add Argument button. The information you specify here will be used/shown when scripts are used with/assigned to/attached to routes, extensions, feature codes, etc. To delete a script argument, click the Remove button.

Description. Description of the argument

Example:

Phones to ring

Type. Select from the list of argument types. Most of the types will result in the list of that type to be presented when using/attaching the script. To allow any value to be entered when using/attaching the script select the Text type.

You can save the script you are editing by clicking Save or Create, create a duplicate of the existing script by clicking Clone, delete the existing script by clicking Delete, or go back to the list of scripts by clicking Cancel. *Note that you can not modify Virtel Voice scripts shipped with PBX Manager and Save button will not be shown when viewing scripts with tl- prefix.*

Media Files

Voice Recordings

Voice Recordings screen allows you to record announcements and voice prompts used throughout the system, and in particular in the Voice Menus.

Existing voice recordings are presented in a list. Clicking on the Recording column displays the **Edit Voice Recording** screen, where you can rerecord the message if necessary.

To create a Voice Recording, click on the **Create Voice Recording** button.

To delete a voice a Voice Recording, select it using the check box on the left and click on Delete Selected.

To listen to the existing Voice Recording click on Play, and the sound file will be downloaded to your computer and played (provided that you have an appropriate player on your computer). Typically a Windows system will be able to play messages recorded in the wav format, to play gsm messages you may need to install additional software.

Edit/Create Voice Recording

File name. File name for the sound file to be recorded, no spaces allowed.

Format. Sound file format used for recording.

Phone to use for recording. Select the phone to be used for recording.

Clicking Start recording will ring the specified phone, and recording can be performed by following simple instructions.

You can upload gsm and wav files (with .gsm or .wav extensions) by clicking the Upload button. Files without extensions are assumed to be in .wav format.

Music-on-hold

Music-on-hold allows you to manage music on hold playlists (groups of files).

Existing Music-on-Hold play lists are presented in a list. Clicking on the Playlist column displays the **Edit Playlists** screen.

To create a new Playlist, click on the Create Playlist button.

To delete a music-on-hold Playlist, select it using the check box on the left and click on Delete Selected.

Edit/Create Playlist

Name. Unique identifier for this Playlist.

Example:

jazz

Mode. Mode of music-on-hold. Valid modes include files, mp3, quietmp3, mp3nb, quietmp3nb, and custom. We suggest using ,files, mode when possible. See Asterisk documentation for details.

Format. Codec to be used when playing this Playlist. See Asterisk documentation for details.

Application. Specify a command that will play music-on-hold. Only available if ,custom, Mode is selected.

Random. Specifies that the files in the Playlist should be played in random order.

Options. Additional options for this Playlist. See Asterisk documentation for the list of available options.

The bottom part of the screen allows you to upload files into this Playlist.

You can save the Music-on-Hold Playlist definition you are editing by clicking Save/Create, delete the existing Playlist by clicking Delete, or go back to the list of Music-on-Hold Playlists by clicking Cancel.

Extensions & Directory

Bulk Generator

Bulk Generator lets you quickly create multiple user extensions. When using Bulk Generator, you first set default values using PBX Settings -> Default Values and then either upload a CSV file containing information for each user or enter data in a simple list like form.

When Bulk Generator creates user extensions it also creates corresponding phones, mailboxes, conference bridges and web users with permissions limited to accessing their personal information through the User Portal. Web users are created with a web login equal to their extension and default password based on the options specified for password generation (in PBX Settings -> Default Values or Tenant settings for Multi-Tenant version). Note that Multi-Tenant Manager **prepends** current tenant's name to the generated user names, and **appends** current tenant's name to the generated phone names.

The **Default script for extensions** as specified in PBX Settings -> Default Values will be associated with each generated user extension.

Generated phones will have the User Name, Caller ID, password and associated mailbox fields set based on the data entered on the form or the CSV file, and mailboxes will have the owner's name, email and pager set accordingly. If DID (external phone number) is specified, then an Inbound Route with appropriate inbound call handler will be also created by the Bulk Generator.

Specifying MAC address allows you to also create configuration files for auto-provisioning of the generated phone line. Note that you must select a default Phone Model first. This is done in Bulk Generator section of PBX Settings -> Default Values (for Single-Tenant) and Bulk Generator section of Tenant screen (for Multi-Tenant). You also have to insure specify server address and provisioning directory (in System Settings -> Provisioning Settings) and insure that a configuration server is running. If server address or provisioning directory are not set, PBX Manager will ignore specified MACs and not generate files for auto-provisioning.

User Extensions

User Extensions allows you to manage phone system users, their extensions, mailboxes, and web based User Portal access permissions.

For a new system the preferred way of entering user information is Bulk Generator, as it allows you to reduce manual data entry. **Edit User Extension** screen can be used to refine data for individual User Extensions or enter additional User Extensions.

Existing User Extensions are presented in a list. Clicking on the **Ext.** column displays **Edit User Extension** screen.

To create a new User Extension, click on the Create SIP Extension, Create IAX Extension or Create ZAP Extension (not available in Multi-Tenant PBX Manager) button.

To delete a User Extension, select it using the check box on the left and click on Delete Selected.

Edit/Create SIP/IAX Extension

This screen is used to create new and modify existing User Extensions.

Note that creation of a User Extension results in creation of extension (in extensions.include), appropriate entry in sip.conf or iax.conf, (optional) mailbox (in voicemail.conf), (optional) personal conference room (in meetme.conf), user (in users.txt), creation of a directory entry, creation of a Webmin user with User Portal access permissions, and population of Asterisk database with user information, and optionally creation of an Inbound Route (if DID has been specified) and configuration files for auto-provisioning (if MAC was specified).

Note that Multi-Tenant PBX Manager prepends current tenant's name followed by a '-' to the generated user's login and appends it to the generated Primary Registration, so that creating extension 100 for a tenant Virtel Voice results in creation of a user Virtel Voice-100 and a Primary Registration.

Note that there are only a few fields that have to be entered on this screen and there are many fields that are optional and where the default values can be used. To simplify data entry, this screen can be shown in two modes ,, as a Basic or Advanced form, with an ability to toggle between the two. You can specify which form is to be displayed initially in System Settings -> Preferences -> Initial forms' display mode.

The basic form contains the following fields:

First Name / Last Name . User's first and last name

Extension. User's extension

Example:

100

Phone Model. The model of the phone this user will be using. If the phone's MAC address is also entered, this field is used to select appropriate template when generating configuration files for auto-provisioning.

MAC address. MAC address of the phone this user will be using ,, entering the MAC address results in generation of appropriate configuration files for auto-provisioning. You must also properly set auto-provisioning method, directory and server address in System Settings -> Provisioning Settings.

The rest of the fields are part of the Advanced form:

DID. Optional external phone number (DID) for this User Extension. Entering a DID will result in creation of Inbound Route routing inbound calls to the DID to this User Extension.

Example:

14154441212

External Caller ID Number. This field is presented differently depending on whether you are using Multi-Tenant or Single-Tenant PBX Manager.

Caller ID precedence: if specified the Trunk CallerID will be used first, then the extension CallerID and lastly the PBX wide default external CallerID.

In Single-Tenant version it can be entered directly, in Multi-Tenant version user is given an option (provided that the tenant is configured with an option to set caller id) of using an assigned default or selecting from a list of assigned DIDs. If tenant is not allowed to set the caller id, then the assigned caller id is used.

External Caller ID Name. This field is presented differently depending on whether you are using Multi-Tenant or Single-Tenant PBX Manager.

In Single-Tenant version it can be entered directly, in Multi-Tenant version user is given an option (provided that the tenant is configured with an option to set caller id) of using an assigned default or entering a name. If tenant is not allowed to set the caller id, then the assigned caller id name is used.

Internal Caller ID Name. This field defaults to First Name + Last Name separated by a space. The default value can be overridden by entering data in this field.

Account code. Account code to be associated with the calls originated from this User Extension.

Script. Script that will handle calls to this extension. In order to support User's ability to manage their call handling options, this script has to be aware of internal variables managed in the User Portal. Another restriction is on the script's arguments ,, the first argument must be a phone, and the second a mailbox (these arguments are not shown and will default to the phone and mailbox generated when the User Extension is created). The rest of the arguments don't have any restrictions, neither on their type nor on the number of the arguments. Vtel Voice PBX Manager's default scripts for User Extensions are tl-userxten and tl-stdexten (these are the same and exist under 2 names for historical reasons and backward compatibility) which both internally use tl-stdexten-base script. This script handles call forwarding, follow-me, call screening, and call recording and should be sufficient in most cases. It also allows to dial an optional ,operator extension, by pressing 0 during voicemail announcement.

Phone settings.

When creating Users Extensions the values of fields in this section are populated based on the defaults specified in the Phone Template in **PBX Settings -> Default Values.**

Dialing permissions. Determines whether the phone can dial all, unrestricted or only internal and emergency Outbound Routes.

Password. Password for the SIP or IAX registration ,, PBX Manager generates random password but you can enter another value.

Call groups. Determines the group membership that this phone belongs to for call pickup purposes .

Pickup groups. Determines the list of groups that this phone is permitted to pickup. You can dial *8# (This is an Asterisk default that can be changed) and pickup a ringing phone if you have a group in your "pickupgroup" that is in the ringing phone's "callgroup".

Authentication method. (IAX) Authentication method this phone will use when registering with Asterisk. This should match authentication method specified in phone configuration.

Enabled codecs and disabled codecs. Codecs to be used that must match codecs in phone configuration. You can use drag-and-drop to move codecs around.

DTMF mode. Should match DTMF mode in phone configuration (applies to SIP phones only).

NAT. Check this if your phone is behind NAT firewall.

Can reinvite. (SIP) Is used to allow sending of a SIP reinvite which allows the phones to send media stream directly to each other. Turning this off will keep Asterisk in the media path.

Qualify. You can select ,yes, (defaults to 2 seconds), ,no, or a time interval (in milliseconds) to specify how frequently Asterisk will check if the device is reachable.

Call-limit. Number of simultaneous calls for this users phone. Set to 1 to disable call waiting.

Other options. Specify any additional phone options in the key=value form. Each pair should be entered on a separate line. Please consult Asterisk documentation for the list of available options.

Voicemail Settings.

Mailbox. A mailbox associated with this user. User Extensions can ,own, a mailbox, use some other mailbox or not have voicemail at all.

Example:

100

PIN. PIN for accessing mailbox ,, defaults to extension.

Notify by email. Determines whether user will be notified about new voice mail messages via email. If this is checked, then Email address is also required.

Email address. User's email address.

Attach messages to email. Determines whether message sound files will be attached to the notification e-mail.

Delete after delivery. Determines whether voice mail message will be deleted after it is delivered via email.

Notify by pager. Determines whether user will be notified about new voice mail messages via pager. If this is checked, then Pager is also required.

Pager. User's pager address.

Personal Conference Room.

Create personal Conference Room. If set PBX Manager will create a conference room with the same number as the User Extension and give the User Extension's owner the rights to manage this conference room in the User Portal using Conferences -> Manage Conference screen. **Note that by default PBX Manager will set conference room's user pin number to be the same as extension, and administrator's pin to be the next higher number.**

Call Recording.

Record calls. Specify whether calls to/from this extension will be recorded. The options are to Do not Record, Record all calls, Record calls selectively by pressing a key sequence. This key sequence is defined in features.conf (automon) and is #9 by default.

Web Settings.

Here you can specify extension owner's permissions for accessing User Portal as well as the User Portal options.

Language. User Portal language for this user.

Color scheme. User Portal color scheme for this user.

Password. User's password for accessing User Portal web interface. By default it is the same as extension .

Directory Information.

Here you can specify options for extension owner's inclusion in a dial-by-name directory and web based company directory.

Include in dial-by-name directory. Specify whether the user will be included and allows to enter a different name than the default based on Last and First name.

Add to Web based directory. By default PBX Manager adds users to the directory when User Extensions are created. You can turn this off if necessary.

Home phone. User's home phone .

Mobile phone. User's mobile phone.

Other phone. User's other phone.

Department. User's department.

You can save User Extension you are editing by clicking Save/Create, delete an existing User Extension by clicking Delete, or go back to the list of User Extensions users by clicking Cancel. Deleting a User Extension will also delete an associated Webmin user, Primary Registration, mailbox, and Inbound Route associated with a DID if it was specified.

Directory

Directory allows you to create or edit directory entries both for the extensions and external phone numbers and is available for a ,click-to-call,Ä in the User's Portal. Note that directory entries are created automatically (by default) when creating User Extensions.

Selection Filter allows limiting your selection of directory entries based on Last Name, First Name or Department. For each of these fields you can specify whether you are looking for an exact match or entries starting with or containing the specified string. If multiple fields are specified all of them are combined in the filter.

Existing directory entries are presented in a list.

Clicking on the Last Name column will present **Edit Contact** screen.

To create a new Contact (directory entry), click on the Create Contact button.

To delete a directory entry select it using the check box on the left and click on Delete Selected.

Edit/Create Contact

This screen allows you to enter Contact information for a directory entry. Note that directory entries are created automatically (by default) when creating User Extensions.

Routes

Inbound Routes

Inbound Routes allows you to associate external phone numbers (DIDs/DDIs) with schedules and scripts thus defining how the inbound calls to each number will be handled. Note that the Single-Tenant PBX Manager allows you to use patterns to match a group of numbers as well as directly enter DIDs when creating Inbound Routes, while in the Multi-Tenant PBX Manager DIDs are restricted to those assigned to the current tenant and patterns can not be used.

Further, a schedule can be associated with the DID such that different actions are taken based on time of day, day of week and specific dates. A typical use of Inbound Routes would be to specify that the calls to a main number would execute a daytime Auto Attendant during business hours, a different Auto Attendant during off hours and possibly a third Auto Attendant during holidays.

Existing patterns/DIDs are presented in a list. Note that in Multi-Tenant PBX Manager the DIDs with assigned Inbound Routes have a checkbox next to them and can be edited, and DIDs without a corresponding Inbound Route do not have the checkbox and can be assigned when creating a new Inbound Route.

Clicking on the **DID** column for a DID with assigned Inbound Route will present **Edit Inbound Route** screen.

To create a new Inbound Route, click on the Create Inbound Route button.

To delete an Inbound Route select it using the check box on the left and click on Delete Selected.

Note that creating an Inbound Route in Multi-Tenant PBX Manager changes corresponding DID status from ,Assigned, to ,In Use, and deleting an Inbound Route does the opposite.

Edit/Create Inbound Route

DID. DID (external phone number) or pattern. This can be entered directly (Single-Tenant) or selected from a list of DIDs assigned to the tenant (Multi-Tenant).

Description. Description of this Inbound Route.

By clicking **Add time based handler** you will be able to add up to 10 time based handlers - script/schedule pairs that define the Inbound Route (typically 1 or 2 is sufficient, with the first one covering a special time condition and the second covering the rest by using tl-all-hours schedule or an equivalent). PBX Manager will use these handlers for handling inbound calls and process them in sequence ,, if date/time of the call matches first ,When, condition, the first script will be executed, if the second ,When, is matched, the second script, etc.

Important: insure the all-hours schedule is last in the list.

When. Name of the schedule that will be used to select a time range to this pattern/number.

Run script. Name of the script that will be used to handle calls to this DID/pattern if the call time falls within the time range of the corresponding schedule.

Except when this variable is set. Name of a global variable which can be set to any non empty value to prevent this script from executing. This can be used to temporarily change Inbound Route.

Depending on the selected script, you will be presented with a list of parameters that were defined for the script. Depending on the type of these parameters you will be able to select from the lists of possible values or enter them directly.

You can save the Inbound Route you are editing by clicking Save/Create, delete the existing Inbound Route by clicking Delete, or go back to the list of Inbound Routes by clicking Cancel.

Outbound Routes

Outbound Routes allows you to manage outbound dialing and associate patterns with scripts to specify how the calls to these numbers will be handled. PBX Manager allows you to assign Outbound Routes to 3 categories (Emergency, Unrestricted, Restricted) in order to control outbound dialing permissions from your phones.

Existing routes are presented in a list. Clicking on the **Outbound Route** column displays **Edit Route** screen.

To create an Outbound Route, click on the Create Route button.

To delete an Outbound Route, select it using the check box on the left and click on Delete Selected.

Edit/Create Route

Route. A number or a pattern for outbound dialing.

If the first character of the route is _ it means that whatever follows is to be treated as a pattern as follows:

X matching any digit 0-9

Z matching any digit 1-9

N matching any digit 2-9

bracketed expressions like [13-5] matching any digit in the brackets (in this case 1,3,4,5)

. matching one or more characters.

See Asterisk documentation for the details and for the available formats for pattern definition.

Example:

_1NXXNXXXXXX

Description. Description of this route

Category. Specify whether the route is in Restricted, Unrestricted or Emergency category. Routes in Emergency category can be dialed from all phones, ,Unrestricted,À routes are not available to phones configured to make only internal calls, and ,Restricted,À are not available to phones that are not allowed to make all calls.

For example you can put pattern for making international calls into ,Restricted,À category and make some specific international destinations and all local calls ,Unrestricted,À.

Script. Name of the script that will be used to handle calls that match the pattern for this route. You should also fill in the parameters expected by the script. ,Trunk,À parameter is required, but the rest of the parameters for the default Virtel Voice supplied scripts are optional.

Note that the default Virtel Voice supplied scripts used in Outbound Routes references global variables `$_DIALOUT` and `$_INTERNATIONAL-PREFIX`. `$_DIALOUT` can be set so that all the calls to outside numbers will require preceding them with the specified digit, and `$_INTERNATIONAL-PREFIX` is a country and provider specific prefix for making international calls ,, you can change these according to your requirements. If you don't use `$_DIALOUT` , you can remove all the Virtel Voice supplied routes that reference it.

You can save the Outbound Route you are editing by clicking Save/Create, delete the existing Outbound Route by clicking Delete, or go back to the list of Outbound Routes by clicking Cancel.

PBX Features

IVR / Voice Menus

IVR / Voice Menus (also referred to as an Auto Attendant) allows you to create a basic or multi level auto-attendant system with multiple menus that can be linked as necessary.

Existing voice menus are presented in a list. Clicking on the Name column displays the **Edit Voice Menu** screen.

To create a voice menu, click on the **Create Voice Menu** button.

To delete a voice menu, select it using the check box on the left and click on Delete Selected.

Edit/Create Voice Menu

Name. Unique alphanumeric name for the menu, no spaces allowed.

Description. Menu description, optional.

What to play?. Allows you to choose whether the menu will play a prerecorded sound file (Prerecorded announcement) or a sound file dynamically generated by a custom command (Announcement generated using command). You can select existing announcements from the list or record (or upload) new announcement by clicking Record new.

Wait before playing (sec). Total time Asterisk will wait before playing the announcement.

Ring while waiting. If selected will simulate ringing to the caller before the announcement is played.

Wait for response (sec). Total time Asterisk will wait for the response from the caller before it executes the action specified for ,No Input,Äü.

Wait for key press (sec). The time Asterisk will wait for a key press by the caller before it executes the action specified for ,No Input,Äü.

Allow dialing extensions. If set, will allow user to dial any User Extension.

Allow dialing Feature Codes. If set, will allow user to dial any of the Feature Codes defined in **PBX Features - > Feature Codes**.

Additional allowed dialing pattern. Used to allow the caller to enter digits other than extensions or feature codes. Control will be passed to the call handling script for further processing.

Authentication. You can make menus ,protected,Äü by specifying a code that user will be prompted to enter before accessing this menu. An example of using this would be a voice portal accessible only to the company employees.

You can assign actions to user selections (key presses) by specifying options in the **What to do?** and **Select** columns.

You can save the Voice Menu you are editing by clicking Save/Create, create a duplicate of the existing voice menu by clicking Clone, delete the existing voice menu by clicking Delete, or go back to the list of voice menus by clicking Cancel.

Hunt Lists

Hunt Lists allows you to create lists of ring groups that are processed sequentially. In order to use Hunt Lists they must be associated with/attached to Feature Codes.

Existing Hunt Lists are presented in a list. Clicking on the Name column displays the **Edit Hunt List** screen.

To create a Hunt List, click on the **Create Hunt List** button.

To delete a Hunt List, select it using the check box on the left and click on Delete Selected.

Edit/Create Hunt List

Name. Unique alphanumeric name for the Hunt List, no spaces allowed.

Description. Hunt List description, optional.

Ring Groups. Up to 10 ring groups can be associated with a Hunt List. To add a Ring Group to the list click on Add Ring Group to Hunt List button. Each Ring Group allows you to specify the following:

Announcement to caller before ringing. A voice announcement played to callers before ringing the phones in this ring group.

User extensions to ring. A list of user extensions in this ring group. Hold the Control key down while selecting each extension. At least one of either the User Extensions to ring or External phone numbers to dial has to be selected.

External phone numbers to dial. External phone numbers to dial as part of this ring group. Separate phone numbers by spaces or commas.

How long to ring (sec). The time Asterisk will continue to ring this ring group. Note that voicemail and all the call forwarding and screening options for the User Extension are disabled when User Extension is dialed in a ring group, while Call Recording options remain in effect.

Description. Hunt List description, optional.

The **If no one answers run the script** field is used to specify an action taken if none of the ring groups answers. You can select a script to be executed from a list of available scripts and fill the scripts parameters as per script definition. See PBX Settings -> Script Library for details on Scripts.

You can save the Hunt List you are editing by clicking Save/Create, delete the existing Hunt List by clicking Delete, or go back to the list of Hunt Lists by clicking Cancel.

Feature Codes

Feature Codes allow you to add functionality to your PBX and execute scripts when these codes are dialed either directly or using selections in Voice Menus. Virtel Voice PBX Manager comes with a number of Feature Codes for basic PBX functions and you should freely add feature code as needed.

Existing Feature Codes are presented in a list. Clicking on the **Feature Code** column displays **Edit Feature Codes** screen.

To create a Feature Code, click on the Create Feature Code button.

To delete a Feature Code, select it using the check box on the left and click on Delete Selected.

Edit/Create Feature Code

Feature Code. Code that will invoke an associated script. Frequently these codes start with an * and are called star codes, but this is not a requirement. You have to be careful choosing these codes so that they don't interfere with either User Extensions nor Outbound Routes.

Example:

*85

Description. Description of this Feature Code

Example:

Check Voice Mail

Script. Name of a script that will be associated with this Feature Code.

Depending on the script selected for handling calls to this Feature Code, you will be presented with a list of arguments that were defined for the script. Depending on the type of arguments you will be able select from the lists of possible values or enter the values for the argument directly.

You can save the Feature Code you are editing by clicking Save/Create, delete the existing Feature Code by clicking Delete, or go back to the list of Feature Codes by clicking Cancel.

Special Lines

Special Lines allows you to create SIP or IAX lines (registrations) that are not associated with a User Extension. A common use for this feature is for softphones where you don't want to associate and extension or require a voicemail box. Phones associated with a special line can place calls and they can be part of a hunt group or simultaneous ring scenario.

Existing Special Lines are presented in a list. Clicking on the Name column displays the **Edit Line** screen.

To create a Special Line, click on the Create SIP Line or Create IAX Line button.

To delete a Special Line, select it using the check box on the left and click on Delete Selected.

Edit/Create Line

Fields on this screen are a subset of the fields available on the Edit User Extension screen. Please use Edit User Extension screen section as a reference.

Special Mailboxes

Special Mailboxes allows you to manage voice mail mailboxes that are not associated with a User Extension.

Existing mailboxes are presented in a list. Clicking on the Name column displays the **Edit Mailbox** screen.

To create a Mailbox, click on the Create Mailbox button.

To delete a Mailbox, select it using the check box on the left and click on Delete Selected.

Edit/Create Mailbox

Name. Unique identifier for this mailbox, typically a number.

Example:

100

Owner's name. Name of the mailbox user; used by the Asterisk Directory application.

PIN. PIN (password) for accessing this mailbox.

Example:

100

Email Address. Email address to forward voice mail messages. Specifying email address will result in email notifications being sent every time a new voice mail message is left in the mailbox.

Example

john@doe.com

Attach messages to email. Determines whether the sound file will be attached to the notification e-mail.

Delete after delivery. Determines whether voice mail message will be deleted after it is delivered via email.

Pager. Pager address to send notice about the new voice mail. Specifying pager address will result in pager notifications being sent every time a new voice mail message is left in the mailbox.

Example:

johnspager@doe.com

Other mailbox options. Additional options for this mailbox. See Asterisk documentation for the list of available options.

You can save the Mailbox you are editing by clicking Save/Create, delete the existing Mailbox by clicking Delete, or go back to the list of Special Mailboxes by clicking Cancel.

Conference Rooms

Conference Rooms allows you to create and manage conference rooms.

Existing conference rooms are presented in a list. Clicking on the Conference Room column displays the **Edit Conference Room** screen.

To access **Conference Manager** screen to manage a conference in progress click **Manage Conference** link.

To create a Conference Room, click on the **Create Conference Room** button.

To delete a Conference Room, select it using the check box on the left and click on Delete Selected.

Edit/Create Conference Room

Conference Room. Conference room number. Note that you will also have to create a Feature Code associated provided script Dial Conference to access conference rooms.

Example:

8600

Description. Short description.

Enabled. If checked the conference room is enabled.

Maximum number of participants. Maximum number of participants that are allowed to join the conference.

PIN. Password for regular user's access to the conference room.

Administrator's PIN. Password for an administrator's access the conference room. PBX Manager makes administrator a ,marked,Ä user.

Owners . User's who have the rights to manage the conference either through Conference Rooms -> Manage Conference or User Portal -> Conferences -> Manage Conference.

Wait until the marked user enters the conference. If set will make users to wait for a marked user (In PBX Manager a user who enters administrators password is considered to be the marked user).

Enable music on hold when single caller. If set will enable music-on-hold even when conference has a single caller.

Music-on-Hold. Music-on-hold associated with the conference which plays when users are waiting for a marked user.

Present menu (user or admin) when '*' is received. If set will allow a voice menu to play to a user who presses an * key. The options presented are dependent on whether it is a regular user or an administrator who presses the key and include muting and unmuting, increasing or decreasing volume, etc

Announce user(s) count on joining a conference. If set will play an announcement on a user count to users joining the conference.

Announce user join/leave. If set will play an announcement every time someone joins or leaves the conference.

Set talker detection. If set will send information about the current talkers to the Asterisk Manager Interface (required if you would like to see who is talking in the PBX Manager -> Manage Conference screens)

Set talker optimization. Improves conference quality and reduces transcoding overhead by muting participants who are not currently speaking .

Record conference. If set then the conference will be recorded. Use this with caution as the recordings can be quite large.

Other options . Other options for Asterisk MeetMe application. See Asterisk MeetMe application documentation for details.

Conference Manager

Conference Manager screen allows authorized users (conference owners) to manage the conference in real-time.

Conference Manager screen requires that the address of the user's workstation be included in the list of ,allow,Ä addresses in the manager.conf file (see Asterisk documentation for the description of manager.conf).

Access to the Asterisk Manager Interface should only be allowed from a secure internal network.

Each conference participant is displayed on the screen with the indicators of the status and whether the participant is talking.

Commands are available to mute/unmute participants, raise and lower volume, lock the conference (so that no more participants can join), and ,kick,Ä participants off the conference.

ACD / Call Queues

Agents

Agents allows you to manage call queue agents.

Existing agents are presented in a list. Clicking on the Agent column displays the **Edit Agent** screen.

To create an agent, click on the **Create Agent** button.

To delete an agent, select it using the check box on the left and click on Delete Selected.

Edit/Create Agent

Agent. Unique identifier for this agent.

Example:

100

Name. Descriptive name for this agent

Example:

John Smith

Password. This agent's password

You can save the agent you are editing by clicking Save/Create, delete the existing agent by clicking Delete, or go back to the list of agents by clicking Cancel.

Queues

Queues allows you to manage call queues.

Queues consist of:

- Incoming calls being placed in the queue
- Members that answer the queue (phones or users that login as agents)
- A strategy for how to handle the queue and divide calls between members
- Music played while waiting in the queue
- Announcements for members and callers

Existing queues are presented in a list. Clicking on the Name column displays the **Edit Queue** screen.

To create a queue, click on the Create Queue button.

To delete a queue, select it using the check box on the left and click on Delete Selected.

Edit/Create Queue

Name. Unique identifier for this queue.

Example:

sales

Description. Queue description

Maximum Callers. Maximum queue size (0 is unlimited)

Announce Hold Time. Specify whether the hold time is to be announced and when.

Announce Frequency. Specify how frequently the hold time will be announced.

Periodic Announcement. Specify whether the periodic announcements are played to the caller.

Periodic Announcement Frequency. Specify frequency of periodic announcements.

Rounding of announced hold time. How long can the agents phone ring before a timeout.

Agent timeout. How long can the agents phone ring before a timeout.

Agent retry. Retry timer between attempts to call queue members.

Report hold time to agent. Specify whether the caller's hold time will be announced to the agent when the call is connected.

Announcement to agent. Specify whether an announcement will be played to the agent when they answer a call.

Call Recording. Specifies whether the call recording will be enabled for this queue, as well as the recording format.

Combine recorded files. Specifies whether files containing recorded conversation should be joined together.

Ring Strategy. A strategy may be specified and includes:

- ringall - ring all available members until one answers (default)
- roundrobin - take turns ringing each available member
- leastrcent - ring a member which was least recently called by queue
- fewestcalls - ring the one with fewest completed calls from this queue
- random - ring randomly
- rrmemory - next call will go to the agent after the last one who answered.

Music-on-Hold. Specify music-on-hold for the queue.

Other queue options. Additional options for the queue. See Asterisk documentation for the list of available options.

Queue members. Specify members of the queue. Members can be selected from the list of available members by clicking the > button.

You can save the queue you are editing by clicking Save/Create, delete the existing queue by clicking Delete, or go back to the list of queues by clicking Cancel.

Call History

Recorded Calls

Recorded Calls allows you to manage files containing calls recorded during PBX operation.

You view a list of the files and listen to them on your computer by clicking PLAY. You can also send files as attachments via email.

To delete a file, select it using the check box on the left and click on Delete Selected.

Call Detail Records

Call Detail Records allows you to view Call Detail Records created during PBX operation.

You can specify a selection filter to be applied. You can filter by a range of dates, caller id, and source and destination channels.

Note that the users of the Multi-Tenant PBX Manager can only see the CDR for the tenants they are allowed to manage. Note that PBX Manager stores tenant information in so called ,userfield,.

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