Thirdlane PBX Manager 5.0

Users Guide

02/27/2007

http://www.thirdlane.com

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Revision History

02/27/2007 5.0, major changes, auto-provisioning
06/24/2006 User management, User Portal
06/02/2006 System, Generator and MOH changes
04/09/2006 Generator options
03/28/2006 Generator changes
03/27/2006 Multi tenant features
01/25/2006 Minor corrections
12/27/2005 Name changes, Trunks, PBX Options changes, minor corrections
11/25/2005 Agents, minor corrections
11/08/2005 Auto reload option, minor corrections

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Introduction

Thirdlane PBX Manager is a web GUI for management of Asterisk based phone systems.

PBX Manager is available in a **Single-Tenant** or **Multi-Tenant** versions.

The **Multi-Tenant** version of PBX Manager allows for multiple virtualized PBXs to be set up on the same Asterisk box. These PBXs are completely independent and have their own administrators, users, extensions, routes, call queues, media files, etc.

PBX Manager is a powerful tool that allows allows you to configure your base 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 scripts.

Management screens overview

Management screens are divided in the following categories:

Tenant Management (Multi-Tenant PBX Manager only)

Tenant Management allows to manage tenants and perform basic branding.

System Settings

General Settings allows to specify Asterisk directories, ports and run time options, as well as (for Single-Tenant) perform basic branding.

Network allows to specify hostname , IP address, subnet and DNS servers.

Provisioning Settings allows to specify the server address and directory for TFTP (or other supported protocol) based phone configuration.

Administrators allows to create and manage access permissions for additional administrators.

Menu / Link Manager allows to create and manage custom links for addition to PBX Manager menu.

Preferences allows setting of web GUI preferences and specifying when the changes made through the GUI will be applied.

License shows license information and allows to request or install a new license.

PBX Settings

Default Values allows to set default values that will be used by various PBX Manager screens.

Global Variables allows to set dialplan variables for use in PBX Manager scripts.

Schedules allows to define date/time ranges that are used to manage time based call handling. These schedules are currently used by Inbound Routes.

Scripts / Macros allows to create scripts, which are groups of PBX commands used as the building blocks for extending PBX functionality.

Media Files

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

Music-on-Hold allows to manage music-on-hold play lists and upload sound files.

DIDs & Trunks

DIDs allows to manage an inventory of external phone numbers that can be assigned to tenants. (Multi-Tenant PBX Manager only).

Trunks allows to manage service providers and peers.

Extensions & Directory

Bulk Generator allows quick generation multiple users, extensions, Primary Registrations, directory entries and mailboxes as well as create configuration files for phones' auto-provisioning.

User Extensions allows performing all-in-one generation of individual users, extensions, Primary Registrations, directory entries and mailboxes as well as create configuration files for phone auto-provisioning.

Directory allows to create or edit directory entries both for extensions and external phone numbers. This directory can be used for a "click-to-call" feature in the User's Portal.

Routes

Inbound Routes allows to specify how the inbound calls are handled.

Outbound Routes allows to specify how the outbound calls are handled.

PBX Features

IVR / Voice Menus allows to create unlimited multi-level menus for voice response and link them as necessary.

Feature Codes allows to extend PBX functionality by associating scripts that can contain groups of PBX commands with internal numbers.

Registrations allows to create and manage SIP and IAX registrations in addition to the Primary Registrations that are created automatically when creating User Extensions.

Special Mailboxes allows to create and manage mailboxes in addition to those that are created automatically when creating User Extensions.

Conference Rooms allows to manage conference rooms.

ACD / Call Queues

Agents allows to create agents for servicing call queues.

Queues allows to manage call queues.

Tools

Auto-Provisioning allows to auto-provision phones using TFTP or other supported

protocols.

Configuration Editor allows direct editing of Asterisk configuration files. Use with caution.

Backup / Restore allows to set up scheduled backups, selectively archive PBX related files (configuration files, as well as media files) and then upload and deploy these archives to other boxes with Thirdlane PBX Manager.

Command Shell allows to execute Asterisk commands as well as to start and stop PBX.

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 user that can login and manage his/her extension using User Portal.

These screens are available to all end-users created using PBX Manager. PBX Manager 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 the rules for call forwarding.

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

Call Blocking allows end-users to configure the rules for blocking incoming calls.

Call Recording allows end-users to configure the rules for call recording.

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

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

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

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

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

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

Compatibility

PBX Manager has been tested with Webmin 1.070 and later, and Perl 5.8 and later.

Prerequisites

Asterisk PBX

You have to install Asterisk prior to installing PBX Manager. Asterisk is available at http://www.asterisk.org/ where you can also find installation instructions. PBX Manager works with Asterisk 1.2.x as well as Asterisk 1.4.x.

Webmin

PBX Manager is a Webmin module and therefore requires Webmin to be installed on the same computer.

Webmin is a web-based interface for system administration for Unix and contains a web server. Webmin is free and is distributed under a BSD-like license.

It can be downloaded from http://www.webmin.com, where you can also find installation instructions.

Installation

Please make sure that you backup your system prior to installing PBX Manager.

To install PBX Manager Webmin module go to Webmin -> Webmin Configuration -> Webmin Modules, select radio button labeled "From ftp or http URL", enter the URL provided by Third Lane Technologies, LLC and click "Install Module" button. If you have PBX Manager module archive on your computer (you downloaded it or received it via email), then select "From uploaded file", browse to the archive location and click the

"Install Module" button. This will install Thirdlane PBX Manager module in the Servers category of Webmin and provide you with a temporary license.

To start using Thirdlane PBX Manager go to Servers -> Thirdlane PBX Manager.

Note that the latest version of Webmin uses frames and PBX Manager screens will appear "framed". If you prefer a full screen view, you can change Webmin theme to "Old Webmin Theme". You can also set an option to go directly to PBX Manager after login in Webmin Configuration -> Index Page Options.

These options can be set on a system wide basis or for individual users.

Adding administrators and setting permissions

After the installation, the only user with access to PBX Manager is the "original" Webmin user which is either "root" or "admin" (this depends on the default you selected during Webmin installation).

You can add other administrators and manage their permissions using **System Settings -> Administrators** screen.

For example, you can add an administrator with access rights to manage User Extensions and nothing else (by disabling everything except "Can manage User Extensions?", or just prevent an administrator from directly editing configuration files and using command shell by disabling "Can edit configuration files?" and "Can use command shell?" options.

Obtaining permanent license

PBX Manager includes a temporary evaluation license.

If you purchased PBX Manager and decided that you will be permanently using it on the computer where it is installed, you should request a permanent license. To request a permanent license, select **System Settings -> License** on the PBX Manager menu and click on the email address next to the **Request License**. Alternatively you can just send your contact information, the name you would like to appear on the license, as well as the MAC address of any interface cards installed on the computer you will be using to run PBX Manager to license@thirdlane.com.

See **License** section for the instructions on installing your license.

Getting Started

Terminology

While we tried not to introduce any new terms to the general   and Asterisk specific telephony terminology, the term "Registration" used in this document requires some explanation.

"Registration" refers to a SIP or IAX account, an entity that can receive and initiate calls, and can be associated with a Line/Button of an IP phone. When creating User Extensions PBX Manager creates a SIP or IAX account which is referred to as Primary Registration.

Registrations can be created and managed using **PBX Features -> Registrations**.

PBX Manager configuration files' structure

Thirdlane PBX Manager configuration files are installed automatically. Default configuration includes the following:

1. Sample schedules for defining company business hours
2. Sample IVR / Voice Menus (auto-attendant) based on the company business hours
3. Various Feature Codes including:
    1. Access to voice mail
    2. Company directory
    3. Codes for call forwarding activation and deactivation
4. Various scripts (see **Scripts** section).
5. Basic Outbound Routes (see **Outbound Routes** section)

While not required, it may be useful to study the installed configuration files in order to understand how the PBX is organized.

Basic organization is as follows:

1. Outbound or internal calls from registered phones start in one of the "from-inside" contexts depending on the access permissions.
2. Inbound calls start in the "from-outside" context.
3. Extensions and Feature Codes are placed in "local-extensions" and "feature-extensions" contexts respectively and are stored in the extensions.include file.
4. Outbound Routes are placed in one of the "outgoing" contexts (depending on Category) and stored in the outbound.include file.
5. Contexts are included as appropriate using include => context-to-include.
6. Files are included as appropriate using #include xxx.include

Making and applying changes

Note that using **System Settings -> Preferences** you can set an option to reload configuration files after every change made by the PBX Manager.

Alternatively you can make multiple changes and then reload PBX configuration. PBX Manager informs you that reload or restart is required by displaying "Reload" or "Restart" link that allows you to apply changes below the menu.

Note that "Reload" is not disruptive while "Restart" will disconnect all the calls in progress so you should use it with caution.

Creating basic PBX

What you do next depends on your specific requirements, here is a typical scenario:

Verifying system configuration

Go to **Settings -> General Settings** and verify that the information there is correct for your system, also check or set **System Settings -> Network** and (if you plan to auto-provision your phones) **System Settings -> Provisioning Settings.**

Configuring trunks

If you have any hardware that needs to be installed and configured, do that following the manufacturers instructions. Make sure that all the necessary drivers are loaded and that the hardware is operational.

If you are using FXO connection to PSTN, make sure that the context you specify in zapata.conf is **from-outside**. This is required in order to make PBX Manager generated code properly route inbound calls.
The same applies to ISDN trunks.

If you use IP trunks, go to **Trunks** screen and create trunk(s) based on the information from your provider.

If you are using Digium or other zaptel compatible hardware to connect to PSTN create a ZAP trunk for each channel group you specified in zapata.conf. Naming **must** follow these rules - for group1 in zapata.conf, name the trunk g1, for group2 name it g2, etc.

Configuring Outbound Routes

PBX Manager ships with a few sample Outbound Routes, they need to be edited as they don't have any real trunks associated with them. Go to **Routes -> Outbound Routes** screen, edit the routes you are going to use (or add new routes if needed) and select your trunk(s) as the Trunk for these routes.

Creating extensions and users

The fastest way to create extensions 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, first go to **PBX Settings -> Default Values** screen and review the defaults or enter whatever may be appropriate for your devices (codecs, DTMF mode, etc) in the Phone Template and voicemail settings in the Voicemail Template.

If you are going to use auto-provisioning feature for your phones' configuration, go to **System Settings -> Provisioning Settings** and make sure that your provisioning settings are specified correctly. You should also verify that the provisioning directory is accessible and that TFTP (or other provisioning server if you use FTP or HTTP) is running.

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 you would like to auto-provision).

This step will create user information, extensions, Primary Registrations, mailboxes, directory entries, and optionally (if DIDs were specified) Inbound Routes. At this point end-users can login into the User Portal (by default Webmin user name and password are the same as extension) and manage their own settings.

See the **Bulk Generator** section for more details.

Configuring phones

You can either configure your phones using tools provided by the phone manufacturer or use PBX Manager for auto-provisioning of supported SIP phone models (see **Provisioning** section for details). If you are using auto-provisioning, make sure that your **System Settings -> Provisioning Settings** are specified correctly and that TFTP (or other provisioning server) is running.

When configuring phones by hand, make sure that the registration information (name and password) matches those created by the **Bulk Generator**. Verify that the phones are working.

Configuring an auto-attendant

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 **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 done (or uploaded), 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" script and select your Voice Menu from the list.

Management Screens

This section describes screens and features available to PBX Manager   administrators. Administrators are Webmin users with permissions to access Thirdlane PBX Manager module. User Portal screens are described in the User Portal Guide.

Tenant Management (Multi-Tenant PBX Manager only)

Tenant Management allows you to manage individual virtual PBXs (tenants) sharing the Asterisk box. Each tenants' users, extensions, mailboxes, phones as well as the menus, media files (announcements and music-on-hold) are managed independently.

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

To create a tenant, click on the Create Tenant button.

To select a tenant for further management (all the subsequent tenant specific operations will be performed on the selected tenant) use the dropdown list on the top of the screen.

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

Edit/Create Tenant

Tenant name. Unique alphanumeric name for the tenant (no spaces, underscores or dashes are allowed).

Example:

thirdlane

Tenant ID. This field is only shown when editing an existing tenant. Used internally by PBX Manager.

Description. Description, optional.

Status. Tenant status (enabled/disabled). This allows a system administrator to temporarily disable the tenant (all calls originating from the phones associated with the tenant will be disabled).

Maximum number of extensions. By default the number of user extensions for a tenant is unlimited. Entering a number in this field will limit the number of user extensions.

Maximum number of Voice Mailboxes. By default the number of voicemail mailboxes for a tenant is unlimited. Entering a number in this field will limit the number of voicemail mailboxes.

Maximum number of Registrations. By default the number of Registrations for a tenant is unlimited. Entering a number in this field will limit the number of Registrations.

Maximum number of Managed Phones. By default the number of Managed Phones for a tenant is unlimited. Entering a number in this field will limit the number of Managed Phones.

Maximum number of Conference Rooms. By default the number of Conference Rooms for a tenant is unlimited. Entering a number in this field will limit the number of Conference Rooms.

Allow tenant to manage Caller IDs. You can use this field to allow a tenant level administrator to set up their own caller id. Alternatively you can set the caller id for the tenant and prevent them from modifying it. This caller id will appear when outbound calls are made by the phones associated with the tenant. Note that whether this caller id will be available to the called party is dependent on your service provider.

Default External Caller ID. You can set a default caller id for the tenant by selecting

one of the DIDs assigned to the tenant. Note that whether this caller id will be available to the called party is dependent on your service provider.

Default External Caller ID Name. Default caller id name for the tenant. Note that whether this caller id will be available to the called party is dependent on your service provider.

Administrators

Administrators. This field is only shown when editing an existing tenant. List of the Webmin users who are allowed to manage this tenant. Typically you would create a tenant level administrator (Webmin user) for each tenant. If necessary, you can further limit their access rights, for example you may not allow these users to manage Trunks, Routes, Inbound Numbers. Note that the root user or any user with the permission "to manage all tenants" will be able to manage any tenant.

Add an Administrator. This field is only shown when editing an existing tenant. You can add tenant level administrator here.

Use existing tenant as prototype. This filed is shown only when creating a new tenant and allows you to "clone" an existing tenant's configuration. If you select a tenant to "clone" you will be offered a list of options for the further selection of which parts of the configuration will be "cloned".

Bulk Generator options

Phone Model. Allows to select a default phone model for auto-provisioning. This will be used in Bulk Generator or User Extensions when creating an extension (entering a MAC address in Bulk Generator is required for the auto-provisioning to take place).

Length of randomly generated phone passwords. Passwords (for phone Registration) are generated randomly and this field allows to specify their length. Note that these passwords can be changed after they are generated and the length of new passwords can be different.

Create web passwords and mailbox PINs. Web passwords and mailbox PINs can be generated based on extension or the default value specified here.

Append to mailbox PINs. If mailbox PINs are based on the extension number this field will be appended to the PIN generated for the corresponding extension. For example if the extension is 100 and this field contains 175, then the generated PIN will be 100175.

Append to web passwords. If web passwords for users based on the extension number, this field will be appended to the password generated for the corresponding user. For example if the extension is 100 and this field contains xyz, then the generated web password will be 100xyz.

System Settings

General Settings

Command to start PBX allows you to specify a command that PBX Manager uses to start your PBX.

Ports/Bindings section allows you to specify bind addresses and ports for your PBX.

Defaults are generally don't need to be changed.

Manager interface access section allows you to configure Asterisk Manager interface and specify manager port and user password.
The default values are set during PBX Manager installation, but can be changed for security reasons.

Note that Manager Interface access is required for normal PBX Manager operation.

Branding (This section is only available in Single-Tenant PBX Manager – Multi-Tenant version offers branding on a tenant level).
You can specify whether a custom logo is going to be used, upload your logo image and specify a URL that the logo links to (by default it links to Third Lane Technologies web site).

Network

The **Network** screen allows system administrator to configure networking .

For each of the interfaces you can configure the following:

Configure interface. Select an interface for configuration.

IP Address. Specify whether the IP address will be assigned by DHCP or specify static IP address

Example: 192.168.1.200

Netmask. Specify netmask

Example: 255.255.255.0

Broadcast address. Specify broadcast address

Example: 192.168.1.255

In **Global network configuration** section you can specify the following:

System hostname. Specify fully qualified name for your PBX

Example: pbx.yourcompany.com

Default getaway. Specify default getaway

DNS domain. Specify DNS domain

Example: yourcompany.com

DNS nameservers. Specify addresses of DNS name servers

Example: 255.255.255.0

Provisioning Settings

The **Provisioning Settings** screen allows system administrator to specify settings for auto-provisioning of SIP phones.

This server's address. This is the address that will be placed in the generated phone configuration files as server or proxy address. The default for this field is the fully qualified name of your server.

Example: pbx.yourcompany.com

TFTP directory. Directory where generated files will be placed for retrieval. If you are using TFTP you have to specify this directory as TFTP server default directory. If you choose to provision your phones using a protocol other than TFTP (HTTP or FTP), or prefer to run TFTP server on a different computer, make sure that this directory is accessible by your provisioning server or that the generated files are moved to the appropriate location. In either case you have to make sure that the provisioning server is running and has access to the generated configuration files.

Administrators

The **Administrators** screen allows system administrator to create and manage administrators and their permissions.

Existing administrators are presented in a list. Clicking on the **Login Name** column displays **Edit Administrator** screen.

To create a new administrator, click on the Add a new administrator button.

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

Edit/Create Administrator

This screen is used to create new and modify existing PBX administrators.

Login name. Administrator's login

Login password. Password

Description. Short description

Allowed capabilities. Check boxes corresponding to PBX Manager screens/functions. Checking a "capability" will make the corresponding functionality available for the administrator. All the unselected items will not be present on PBX Manager menus for the administrator .

Menu / Link Manager

The **Menu / Link Manager** screen allows you to add custom menu items to the PBX Manager menu.

Existing links are presented in a list. Clicking on the **id** column displays **Edit Link** screen.

To create a Link, click on the Create link button.

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

Edit/Create Link

Link text. Text to appear on the menu.

Example:

Operator Panel

Link URL. Associated URL.

Example:

http://iporname/panel

Show in management menu. Specify whether this link should appear on the PBX Manager management/adminstration menu.

Show in end-user menu. Specify whether this link should appear in the end-user menu in User Portal.

Weight (menu position). Specify a relative position of this link in relation to other links (higher wieight positions the link lower in the menu.

Preferences

Preferences screen allows you to set your preferences for using PBX Manager.

The following can be set:

Operational preferences

> **Issue reload after every change**. If set to yes, a reload command will be issued after PBX Manager applies changes to the configuration files, and changes will take effect immediately without the need to reload manually.

Display preferences

> **Language.** Select user interface language.

> **Color scheme.** Select user interface color scheme.

> **Number of rows to display per page.** Allows you to specify the number of rows displayed on pages presenting lists of data.

> **Initial forms display mode.** Allows you to specify how the forms that have "Advanced" or "Basic" options will be displayed initially.

License

License screen shows license information and allows to request or install a new license.

> **License status**. PBX Manager performs license file validation and displays the status here.

> **Select license file to install**. Allows you to upload and install a license file from your computer.

To request a new license from Third Lane Technologies follow the instructions in **Obtaining permanent license** in this document.

PBX Settings

Default Values

Default Values screen is used for creation of default values used by PBX Manager when creating various items.

The **Caller ID** section allows to set default caller id for outbound calls. (Single-Tenant only, for Multi-Tenant PBX Manager this is managed in Edit Tenant screen).

The **Outbound Dialing** section allows you to select a default script used by the Outbound Routes when creating new routes.

The **Extensions** section allows you to specify a default script that will be used by the Bulk Generator and User Extensions when creating new user extensions.

The **Voicemail Template** section allows you to set the defaults used in the Bulk Generator, User Extensions or Special Mailboxes when creating voicemail mailboxes. You should consult Asterisk documentation for the available options.

The **Phone Template** section allows you to set the defaults used by the Bulk Generator, User Extensions and Registrations when creating SIP or IAX Primary Registrations. You should consult Asterisk documentation for the available options.

To summarize: User Extensions, and Outbound Routes and the Bulk Generator will use the value specified in Default Script of respective sections when creating new user extensions and routes, User Extensions, Registrations and Bulk Generator will use the default values specified in Phone Template when creating new Registrations, and User Extensions, Mailboxes and the Bulk Generator will use the default values specified in Voicemail Template when creating new mailboxes.

Specific fields on the screen are described below.

Caller ID (Single-Tenant only, for Multi-Tenant PBX Manager this is managed in Edit Tenant screen)

Default External Caller ID. You can set a default caller id for calls made outside of your company by entering it here. Note that whether this caller id will be available to the called party is dependent on your service provider. This can be overridden by specifying it for an individual extension.

Default External Caller ID Name. Default caller id name for calls outside of your company. Note that whether this caller id will be available to the called party is dependent on your service provider. This can be overridden by specifying it for an individual extension.

Outbound Dialing

Default Script for outbound dialing. Name of the script that will be used by default when creating new routes in Outbound Routes.

Extensions

Default Script for extensions. Name of the script that will be used to handle calls

when creating new user extensions.

Bulk Generator Options

The Bulk Generator Options section is available on this screen only for the Single-Tenant version of PBX Manager. In Multi-Tenant version these options are specified for each tenant.

Phone Model. Allows to select a default phone model for auto-provisioning. This will be used in Bulk Generator or User Extensions when creating an extension (entering a MAC address in Bulk Generator is required for the auto-provisioning to take place).

Length of randomly generated phone passwords. Passwords for Primary Registrations are generated randomly and this field allows to specify their length. Note that these passwords can be changed after they are generated and the length of new passwords can be different.

Create web passwords and mailbox PINs. Web passwords and mailbox PINs can be generated based on extension or the default value specified here.

Append to mailbox PINs. If mailbox PINs are based on the extension number this field will be appended to the PIN generated for the corresponding extension. For example if the extension is 100 and this field contains 175, then the generated PIN will be 100175.

Append to web passwords. If web passwords for users based on the extension number, this field will be appended to the password generated for the corresponding user. For example if the extension is 100 and this field contains xyz, then the generated web password will be 100xyz.

Default schedule for Inbound Routes. Bulk Generator generates an Inbound Route if a DID (external phone number) is specified for a User Extension. This field determines what schedule will be used when creating Inbound Route handler for this DID.

Voicemail Template

Voicemail Template allow you to set defaults for user voicemail mailboxes. These options will be used by the Bulk Generator, User Extensions and Special Mailboxes screens.

Notify by email. Determines whether user will be notified about new voice mail messages via email.

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.

Notify by pager. Determines whether user will be notified about new voice mail messages via pager.

Other options. Specify any additional options not provided in the GUI in the key=value form. Each pair must be entered on a separate line. Please consult Asterisk documentation for the list of available options.

Phone Template

While we refer to this section as "Phone Template", it could be more accurately called "Registration Template", as it allows you to set defaults for Registrations rather than phones. The default values specified here will be used by the Bulk Generator, User Extensions and Registrations screens.

Protocol. Default protocol (SIP or IAX).

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

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 order and disabled codecs. Codecs to be used that must match codecs in phone configuration. You can use drag-and-drop to select/unselect and reorder codecs order.

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 tell the server to issue a reinvite to the client. 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 still on line.

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.

Global Variables

Users can create variables for use in their scripts and elsewhere in the dialplan. The following variables are used in Thirdlane PBX Manager configuration files as shipped.

DIALOUT. Some organizations use a digit (typically 9) for dialing outside, you can specify it here. This variable is used when specifying patterns in outbound routes – can be changed.

INTERNATIONAL-PREFIX. Digits dialed by users when making international calls (011 in the US). This typically depends on your provider, and may not have to be specified at all. This variable is used when specifying patterns in outbound routes – can be changed.

RINGTIME. Default time for dialing in seconds – can be changed.

TL-DASH. Used internally – do not remove or change.

TL-MULTI. Used internally – do not remove or change.

OPERATOR. Default operator extension – can be changed.

RECORDING_FORMAT. Default recording format – can be changed.

Schedules

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

Scripts / Macros

Scripts / Macros section allows you to create scripts, which are groups of Asterisk commands. Scripts are a fundamental building block in Thirdlane PBX Manage and are used in the various parts of the application - Bulk Generator, User Extensions , Feature Codes, Voice Menus, Inbound Rules, Outbound Routes, etc where they are associated with extensions, feature codes and called numbers. Dialing an extension/feature code executes an associated script.

Thirdlane 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 Manager functionality as necessary.

PBX Manager displays an installed version of the Thirdlane scripts shipped with PBX Manager in the upper right corner of the screen. Clicking on "Check for Script Updates" 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 (Thirdlane and user scripts) are presented in a list – note that Thirdlane scripts do not have a selector checkbox for deletion).

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 to be used for a particular purpose: 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 Thirdlane 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. Thirdlane scripts shipped with PBX Manager include a number of useful scripts, which by convention start with tl- prefix. You can use Thirdlane scripts directly or create your own scripts either from scratch or by cloning and changing Thirdlane scripts. <u>Please do not use tl- prefix for your scripts.</u>

Example:

ae-stdexten

Script can be used for:. The intended purpose of the script. This will determine whether the script will be presented as an option when creating inbound routes, outbound routes, feature extensions, etc

Descriptive name. Short user friendly name of the script taht 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

Example:

```
exten => s,1,Dial(${ARG1},30,rt)
exten => s,2,Voicemail(${ARG2})
exten => s,3,Hangup
exten => s,102,Voicemail(${ARG2})
exten => s,103,Hangup
```

You can add script arguments by clicking Add Argument button. The information you specify here will be used in when scripts are assigned 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. Some of the types will result in the list of that type to be presented when used by the script user. To allow the user of the script to enter any value, 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 Thirdlane 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 and played (provided that you have an appropriate player). Typically windows workstation 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 playlists are presented in a list. Clicking on the Playlist column displays the **Edit Playlist** 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 if 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.

DIDs & Trunks

DIDs

The DID screen applies to the Multi-Tenant PBX Manager only and is used to manage your external phone numbers frequently called DIDs or DDIs. Single-Tenant version allows you to enter DIDs directly in Bulk Generator, and when creating Inbound Routes or User Extensions.

You can enter your DIDs into the system and then assign them to tenants as required. Once assigned to a tenant they can be used in creation of Inbound Routes or in User Extensions (both in Bulk Generator or when creating individual User Extensions) for that

tenant.

Existing DIDs are presented in a list. A selector on the top of the screen allows you to view all DIDs, DIDs that are assigned to tenants, unassigned DIDs, DIDs assigned to the currently selected tenant, or DIDs in use (assigned to User Extensions, Voice Menus, etc).

To add a range of DIDs enter them in the fields on the bottom of the screen (to enter just one DID repeat it in the "through" field) and click Add DIDs button.

To assign (unassigned) DIDs to the currently selected tenant select them and click Assign Selected, to unassign select and click Unassign Selected (Note that you can not unassign DIDs that are in use - you would have to delete corresponding Inbound Route first).

To delete unassigned DIDs select them and click Delete Selected.

Trunks

Trunks allows you to configure SIP, IAX, ZAP and custom trunks – your connections with VoIP service providers, PSTN or other Asterisk servers.

Existing trunks are presented in a list. A selector on the top of the screen allows to view all trunks or only the trunks for the current tenant.

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

To create a trunk, click on the Create <type> Trunk button. Currently SIP, IAX, ZAP and CUSTOM trunk types are supported.

To delete a trunk, select it using the check box on the left and click on Delete Selected. Note that deleting a trunk that is used by Outbound Routes will make these routes invalid.

Edit/Create Trunk

Type. Trunk's type or protocol (SIP, IAX, ZAP or CUSTOM).

Name. Unique alphanumeric name for the trunk (no spaces allowed). For ZAP trunks the name must match a group defined in zapata.conf following a convention where a trunk corresponding to the group named group1 in zapata.conf must be named g1, a trunk corresponding to group2 must be named g2, etc. In some cases the name of IP trunks will be provided to you by

your service provider.

Description. Description of the trunk, optional.

Can be used by. Trunks can be created for use by all tenants or only a particular tenant. If trunk is for a particular tenant it won't be available for use by other tenants and will not appear as an option when creating Outbound Routes for other tenants.

The rest of the fields are shown on the form depending on the trunk type and protocol.

Host. (SIP, IAX) Domain name or address of your VoIP service provider's gateway or proxy.

Direction. (SIP, IAX) Specifies whether this trunk is inbound, outbound or both.

Context. (SIP, IAX, ZAP, CUSTOM). Specifies the context for this trunk. For external inbound providers it should from-outside. Note that for ZAP or ISDN (and CUSTOM trunks the context in zapata.conf or relevant configuration files should be also specified as from-outside).

User Name. (SIP, IAX) Specify user name associated with the trunk, usually provided by your VoIP service provider.

Caller ID. (SIP, IAX) Caller id associated with the trunk. You can specify both the numeric caller id and a readable name.

Example:

John Doe <100>

Password. (SIP, IAX) Password for this trunk, usually provided by your VoIP service provider.

Example:

1y3ts89h

Authentication method. (IAX) This is supposed to match authentication method expected by your VoIP service provider.

Enabled codecs order. (SIP, IAX) This is supposed to match codec prererences specified by the service provider. You can drag-and-drop codecs to enable/disable them and set the order.

Disabled codecs. (SIP, IAX) Codecs to be disabled for this trunk.

DTMF mode. (SIP) Must match DTMF mode specified by your VoIP service provider.

Other Options. (SIP, IAX). This field allows you to specify additional options that don't have corresponding GUI fields. Specify any additional options in the key=value form. Each pair should be entered on a separate line. See Asterisk documentation for the list of available options.

Example:

inkeys=freeworlddialup

Registration. (SIP, IAX) The registration string as provided by your VoIP service provider (typically required for inbound trunks).

Example:

12345:passwordx@iax2.fwdnet.net

Trunk Dialing options allow you to specify how the calls routed through this trunk will be handled.

Status. (All trunk types) This sets a global dialplan variable TRUNK_STATUS for this trunk. This variable is referenced in the default Thirdlane supplied scripts used for Outbound Routes . This allows to disable trunk temporarily if required.

Dialstring creation. Allows to control how the dailstring for the Dial command will be generated from the dialed number. For custom trunks the only option is Custom string, for other trunk types the dialstring can be either generated based on the type (Generate) or specified directly (Use Custom).

If option Generate is selected the following fields are also available:

# of digits to strip – how many digits will be stripped from the front of a dialed number.

String to prepend - string that will be prepended to the

dialed number.

Note that digits can be also prepended or stripped when specifying Outbound Routes.

Custom string should conform to the format of Dial command. To specify the position of a dialed number in the custom string use $.

Examples:

ZAP/g1/$

CAPI/contr1/$

Provider's web site. This filed is for informational and convenience purposes only. If specified, will open this in a new browser window.

PBX Manager stores this information in the trunk specific global variables that can be used by user scripts.

Please see the source of tl-dialout-base script shipped with the default PBX Manager configuration for the details of using these variables.

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

Extensions & Directory

Bulk Generator

Bulk Generator lets you perform bulk add or update operations for phone users and the associated extensions, phones and mailboxes. When using Bulk Generator, you first set default values using PBX Options and then use CSV file containing information for each user.

When Bulk Generator creates phone users it also creates corresponding extensions, phones, mailboxes and web users with permissions limited to accessing their personal information through the User Portal. Phone users are created with a web login equal to their extension and default password based on the options specified for password generation (in PBX Options or Tenant 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 Options 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 set up 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 Options (for Single-Tenant) and Bulk Generator section of Tenant screen (for Multi-Tenant). You also have to insure that TFTP directory and server name are set properly in Provisioning Settings. If any of these are not set, PBX Manager will ignore specified MACs and not generate files for auto-provisioning.

If Bulk Generator is executed again with the same list of extensions, all the matching previously created users, extensions, phones and mailboxes will be updated based on the new information.

Repeating the generation process with a different list of extensions, will not affect previously created phone users, extensions, phones and mailboxes.

If the extensions overlap, then the corresponding items will be updated based on the options selected at the time of the last generation.

User Extensions

User Extensions allows you to manage phone system users, their extensions, Primary Registrations and mailboxes.

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

Note that creation of a User Extension results in creation of extension (in extensions.include), Primary Registration (in sip.conf or iax.conf), mailbox (in voicemail.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).

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

To create a new user, click on the Create User button.

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

Edit/Create User 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), Primary Registration (in sip.conf or iax.conf), mailbox (in voicemail.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 thirdlane results in creation of a user thirdlane-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 facilitate data entry, this screen can be shown in two modes – as a Basic or Advanced form, with an abiliity 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 TFTP 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.

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 but can be overriden by entering data in this field.

Account code. Account code to be associated with the calls originated form 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. Thirdlane PBX Manager's default script for User Extensions is tl-stdexten (Standard Extension (IP)). It 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.**

Registration name. SIP or IAX User ID. This is a name for the Primary Registration for this phone. By default PBX Manager will use extension as the registration name, but you can also enter a different name if needed. Multi-Tenant PBX Manager also prepends tenant's name to registration name.

Protocol. Protocol (SIP or IAX).

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

Password. Password for the 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 order and disabled codecs. Codecs to be used that must match

codecs in phone configuration. You can use drag-and-drop to select/unselect and reorder codecs order.

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 tell the server to issue a reinvite to the client. 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 still on line.

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. Name of a mailbox associated with this user. A simple convention is to use the extension number as a mailbox name, which is what is offered as a default in create mode (Same as Extension), alternatively a new name can be entered.

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.

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.

Allow user to manage settings. If set, will allow user to change call recording settings through the User Portal.

Web Settings.

Language. User Portal language for this user.

Password. 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.

All the fields in this section are optional.

Home phone. User's home phone .

Mobile phone. User's department.

Other phone. Other user's phone.

Department. User's department.

Add to directory. By default PBX Manager adds users to directory, you can turn this off if necessary.

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" 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.

A typical use of Inbound Routes would be to specify that the calls to different numbers at different times would be answered by different auto-attendants (Voice Menus), or a particular number will be answered by an operator during business hours and will be forwarded to an auto-attendant after hours, etc.

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 by "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.

  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" routes are not available to phones
configured to make only internal calls, and "Restricted" 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"
category and make some specific international destinations
and all local calls "Unrestricted".

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" parameter is required, but the rest of
the parameters for the default Thirdlane supplied scripts are
optional.

Note that the default Thirdlane 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 Thirdlane 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 allows you to create a basic auto-attendant or multi level IVR
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 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".

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".

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**.

Authentication. You can make menus "protected" 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.

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. Thirdlane 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 Code** 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 neither 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.

Registrations

Registrations allows you to create SIP or IAX registrations in addition to those that are created when creating User Extensions.

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

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

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

Edit/Create Registration

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 in addition to those that are created when creating User Extensions.

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:

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 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

> **Password.** Password to access the conference room, optional

Example:

8600

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
- **leastrecent** - 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 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.

Tools

Auto-Provisioning

To "provision" a phone is to supply it with all the information it needs to register with a server (Asterisk) and become operational. While registration information can be entered using phone manufacturer's supplied web interface, it is also possible (and generally more efficient) to use PBX Manager's auto-provisioning feature to create configuration files for the phone.

The phones' auto-provisioning files are generated automatically when User Extensions are created in Bulk Generator or individually, provided that Provisioning Settings are in place (see System Settings -> Provisioning Settings), and phone models and MAC addresses are specified. Note that if you use Bulk Generator you must specify the default

Phone Model (in PBX Settings when using Single-Tenant and Tenant Management if using Multi-Tenant PBX Manager).

By default the Primary Registration created when creating User Extension is assigned to all the phone's Lines/Buttons.

The Auto-Provisioning screens allow you to further refine auto-provisioning information and assign Registrations to phones' Lines/Buttons as needed.

All the phones for which PBX Manager generates auto-provisioning configuration files are considered to be Managed Phones as opposed to unmanaged devices whose configuration and provisioning is not handled by PBX Manager .

Auto-Provisioning screen presents all the Managed Phones in a list. Clicking on the MAC Address column displays the **Edit Managed Phone** screen.

To create a Managed Phone click on the Create Managed Phone button.

To generate configuration files for existing Managed Phones select them using the check box on the left and click on Provision Selected.

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

Edit/Create Managed Phone

MAC Address. Phone' s MAC address.

Description. Short description

Phone Model. Select phone model from the list of supported models.

Clicking add Line/Button creates a row where you can select a Line/Button of the phone being provisioned and specify a Registration that will be associated with this Line/Button. To associate a Registration with all available Lines/Buttons use "All Lines/Buttons" option.

You can generated configuration files for the by clicking Save and Provision or go back to the list of Managed Phones by clicking Cancel.

Configuration Editor

Configuration Editor allows direct editing of PBX configuration files.

   Edit File: Select one of the configuration files from the list

After the file is edited, you can save it by clicking the Save button.

Backup/Restore

Backup/Restore allows you to archive Asterisk related files (configuration files, as well as Media Files) and then upload and deploy these archives to other boxes running Asterisk. It is both a simple backup tool and a powerful tool for quickly "cloning" Asterisk configurations.

Backup/Restore screen contains two sections:

Replace PBX Manager configuration. This allows you to deploy previously saved configurations or reinstall the configuration files shipped with PBX Manager.

   We strongly encourage you to to backup your system before performing this step.

   Use PBX Manager configuration archive. To upload and deploy a previously created archive, type in the name of the archive or use the Browse button, and click Deploy New Configuration. PBX Manager will upload the archive and uncompress it into appropriate directories.

   Reinstall configuration shipped with PBX Manager. This will install default configuration files shipped with PBX Manager witch can be useful if your configuration files are damaged or you just want to start from a fresh installation. If you are trying to start fresh, you should delete your existing User Extensions before performing this step (deleting User Extensions will remove Webmin users and clean up user related data from the Asterisk database) . Note that in the earlier versions of PBX Manager installing files shipped with PBX Manager was a required step after the PBX Manager installation - this is no longer required as the default configuration files are installed automatically.

Backup now (download PBX Manager archive). This allows you to archive your Asterisk configuration, and save it on your local computer. To create an archive click the Download button. The archive will be created as a compressed .tar.gz file.

    Include Voicemail and recordings? Specify whether all Asterisk voicemail messages and media files you have recorded will be included in the backup.

Scheduled backup. This allows you to specify frequency, content and target location for a scheduled PBX backup.

Command Shell

Command Shell allows you to start and stop Asterisk, as well as issue Asterisk commands using a simple web interface.

Note that command interface screen elements will only be shown if PBX Manager detects that the Asterisk process is running.

    Command. Enter Asterisk commands here.

To execute the command, click the Execute button. All the executed   commands and their outputs (if any) will be shown in the Command History at the top of the screen.

To get a list of available commands enter the "help" command.

Example:

help

    Command History. Drop down the box that contains a list of commands previously executed by the same user. This list is maintained across the sessions, so commands will be available even after you log out.

To execute a previously executed command, select it in the Command History drop down box and click the Execute button on the right.

To edit a previously executed command, click the Edit button.

To clear the output of previously executed commands, click the Clear History button.

To clear the Command History drop down box, click the Clear Commands button.

Start / Stop PBX. The appearance and function of this button depends on whether PBX Manager detects that Asterisk is running or not.

Clicking on this button will either start Asterisk using the "Command to start PBX" as specified in System Settings -> General Settings or stop the running Asterisk.

Note that you can also restart Asterisk by issuing the "restart now" or "restart gracefully" commands.

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 you 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". To include this field in CDRs you have to enable it while compiling Asterisk (see cdrrelevant

You can sort records by clicking on the headers of any column; clicking on the same header again reverses the sort order.