

Thirdlane PBX and Thirdlane PBX Manager 6.0
Administrator's Guide

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Introduction

Thirdlane PBX is a complete Asterisk based PBX packaged as a Linux distribution that can be installed quickly from a downloaded ISO CD image and includes **CentOS Linux, Asterisk, Webmin, Munin, vtiger CRM, and PBX Manager GUI**.

Thirdlane PBX Manager GUI is a web based graphical user interface for PBX administration. It is a Webmin module and can be installed either as a component of Thirdlane PBX or separately.

Thirdlane PBX and Thirdlane PBX Manager GUI are available in a **Single-Tenant (PBX Manager)** or **Multi-Tenant (PBX Manager MTE)** versions.

The **Multi-Tenant** version allows to set up multiple PBXs on the same instance of Asterisk. This creates practically no overhead and allows these PBXs to be independent and have their own users, extensions, routes, call queues, media files, etc.

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:

Cluster Management

Managed PBXs - management of multiple Thirdlane PBX servers from the same web interface with a single sign-on.

Tenant Management (Multi-Tenant PBX Manager only)

Tenant Management - creation and management of multiple virtual PBXs (tenants).

System Settings

General Settings – Asterisk ports and run time options

Branding - upload of logo images and customization on a system level.

Network – network , hostname , IP address, subnet and DNS servers.

Provisioning Settings - specifying parameters related to automated provisioning of IP telephones. Parameters include the server's address and auto-provisioning directory.

CDR Settings - CDR (Call Detail Records) destinations and display options.

Administrators - creation and management of access permissions for additional administrators on a system level.

Menu / Link Manager – creation and management of custom links added to PBX Manager menu.

CRM Links - allows to configure URLs for your CRM reverse caller id lookup.

CRM Settings - allows to set the default values for CRM integration and screen pops for inbound calls.

Preferences - web GUI preferences.

License – license upload, license information and requesting to issue a license.

PBX Settings

Default Values - setting of default values used elsewhere in PBX Manager.

Global Variables – setting of global variables for use in PBX Manager scripts.

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

Script Library - management of scripts, which are groups of PBX commands used as the building blocks for extending PBX functionality (such as feature codes and call handling for phones and DIDs)

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. (Multi-Tenant PBX Manager only).

Trunks - configuration of incoming/outgoing telephone services. Service may be either/or VoIP or traditional (both analog and/or digital).

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” feature in the User Portal..

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)

Channel to DID Map – association of analog trunks (also known as ZAP channels) to a DID number. Then the DID can be defined in Inbound Route. (Not available in Multi-Tenant PBX Manager).

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.

Tools

PBX Information – display PBX status information..

Auto-Provisioning – define phones with the associated line, speed dial and Busy Lamp Field (BLF) buttons. The appropriate configuration files are created for the telephone and the telephone will retrieve these generated files as part of the phone's booting process. The phones may use TFTP, FTP or HTTP(s) to retrieve the configuration files..

Configuration Editor - direct editing of Asterisk configuration files.

Use with caution!

Backup / Restore – PBX backup and restore, set up of scheduled backups.

PBX Command Shell - execute Asterisk commands, start and stop PBX.

Manager Interface – connection to Asterisk Manager Interface – mostly for testing and educational purposes.

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” 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.

Installation

The installation process differs and depends on whether you are installing Thirdlane PBX from an ISO CD image or manually installing Thirdlane PBX Manager GUI via Webmin

Installing Thirdlane PBX from the .iso is the fastest and easiest way to set up a working PBX.

Installing Thirdlane PBX using ISO CD image

Note that Thirdlane PBX requires a dedicated computer and will erase the entire content of your hard drive.

- 1) Download ISO CD image from Thirdlane web site <http://www.thirdlane.com>
- 2) Burn a CD image on a CD
- 3) Boot your computer from CD and press enter to begin installation process.
- 4) When requested choose a root password (make sure you remember it :)), select a time zone and enter network configuration information.
- 5) After system reboots login as root
- 6) Follow instructions for accessing Thirdlane web GUI.
- 7) Configure your PBX using browser from another computer.

Thirdlane PBX Manager GUI standalone installation

You will only need this if you are installing Thirdlane PBX Manager GUI as a standalone application.

- 1) Install Asterisk 1.4.x. Asterisk is available at <http://www.asterisk.org/> where you can also find installation instructions. Make sure that you perform “make samples” step to install sample configuration files.
- 2) Install Webmin. PBX Manager is a Webmin module and therefore requires Webmin to be installed. 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.
- 3) Download PBX Manager GUI (Webmin module archive) from Third Lane Technologies web site <http://www.thirdlane.com>
 - 4) Install PBX Manager GUI. Please make sure that you backup your system prior to installing PBX Manager GUI. PBX Manager GUI installer will overwrite configuration files in your Asterisk installation directory. To install PBX Manager GUI go to Webmin -> Webmin Configuration -> Webmin Modules, 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.

Post installation system setup

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 default Webmin theme to the "Old Webmin Theme".

You can also set up Webmin to go directly to PBX Manager after login in Webmin Configuration -> Index Page Options.

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

PBX Manager GUI organizes its menus in categories. While you are getting familiar with the menus it may be convenient to configure PBX Manager to keep all the menus extended (rather than showing only the items for a selected category). You can set this option in PBX Manager -> System Settings -> Preferences.

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".

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

For example, you can add an administrator authorized to manage User Extensions and nothing else by disabling everything except “Can manage User Extensions?”, or prevent an administrator from directly editing configuration files and using PBX Command Shell by disabling “Can edit configuration files?” and “Can use PBX command shell?”

Note that you should not use System Settings -> Administrators to add tenant level administrators in Multi-Tenant PBX Manager – add them using Tenant Management screen.

Obtaining permanent license

Thirdlane PBX and Thirdlane PBX Manager GUI include a temporary evaluation license.

Permanent licenses are based on a MAC (network card hardware address) of your computer. Once you decided that you will be permanently using installed Thirdlane PBX, 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, and the MAC address of any of the network interface cards installed on the computer to license@thirdlane.com.

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

Getting Started

PBX Manager configuration files' structure

Thirdlane PBX Manager configuration files are installed automatically and include the following:

1. Sample schedules for defining company business hours
2. Sample IVR / Voice Menus (auto-attendant)
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 **Script Library** section).
5. Sample music-on-hold
6. Sample voice prompts
7. Sample 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 user's 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 in the extensions.include file.
4. Outbound Routes are placed in one of the “outgoing” contexts in the outbound.include file.
5. Contexts are included using include => context-to-include.
6. Configuration files are included using #include xxx.include

Making and applying changes

Note that in **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” indicator.

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

Configuring basic PBX

Verifying system configuration

Go to **Settings -> General Settings** and verify that the information 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.

Asterisk refers to traditional phone trunks as ZAP trunks (also known as TDM trunks). These trunks can either be delivered by the phone provider as individual trunks (analog) or bundled on a T1. A T1 may be made up of 24 voice channels or as an ISDN PRI (generally 23 channels). A fractional T1 service is delivered as a T1 but with fewer than the maximum 24 channels.

Analog trunks are referred to as FXO trunks. Analog station ports are referred to as FXS lines.

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

If you are using Digium, Sangoma or other zaptel compatible hardware for outbound dialing 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. Note: when creating the trunk group use a capital "G" if you wish the calls to hunt in reverse order for out going trunks (example: if a T1 has 24 channels and all are members of trunk group #1, then if you create trunk G1 and specify G1 for an outbound route then calls will begin at channel 24, then 23, etc. If you specify "g1" then calls will begin at channel 1, followed by channel 2. Recommendation: use "G" to avoid call collisions or glare.

If you use IP trunks, go to **Trunks** screen and create trunk(s) based

on the information from your provider.

Configuring Outbound Routes

PBX Manager ships with sample Outbound Routes which 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) and select your trunk(s) as the Trunk for these routes.

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 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 manually via the phone's web interface or creating your own configuration files, make sure that authorized user name and password match the values created by the **Bulk Generator** or **User Extensions** screens.

Note that in Multi-Tenant PBX Manager the username (authorized user) for the devices is extension-xxx (not xxx-extension) where xxx is a tenant name.

Passwords for device registration are created based on the rules specified in PBX Defaults and by default are randomly generated and are different from mailbox pins (which by default are equal to extensions) and web passwords (also equal to extensions).

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" 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.

Cluster Management

Managed PBXs

Managed PBXs allows you to register and manage multiple PBXs from the same web GUI.

Existing registered PBXs are presented in a list showing whether each one is reachable (whether PBX Manager on a remote PBX can be contacted) as well as the status of the remote PBX.

Clicking on a Stop/Start PBX button will start or stop a remote PBX, clicking **More Information** will display PBX Information screen for a remote PBX and clicking **Manage** link will seamlessly log you into PBX Manager on a remote PBX.

Clicking on the **Hostname** column displays **Edit PBX** screen.

To create a Registered PBX, click on the Register PBX button.

To remove a Registered PBX from the list, select it using the check box on the left and click on Delete Selected.

Register PBX

Server hostname. IP address or hostname of the remote PBX .

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

Port. PBX Manager (Webmin) port of the remote PBX. Typically 80 or 10000.

Short name. A short name to identify the remote PBX.

PBX Manager login. A user name to login into remote PBX.

PBX Manager password. A password of a PBX Manager user of the remote PBX.

PBX Manager connection type. Protocol used by Webmin on a remote PBX host (http or https).

Tenant Management (Multi-Tenant PBX Manager only)

Tenant Management allows you to manage individual virtual PBXs (tenants) sharing the Asterisk box. Each tenant's 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 drop down 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).

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

Description. Short 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 that can be created 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 that can be created for a tenant is unlimited. Entering a number in this field will limit the number of voicemail mailboxes.

Maximum number of Special Lines. By default the number of Special Lines (other than the User Extensions) that can be created for a tenant is unlimited. Entering a number in this field will limit the number of Special Lines.

Maximum number of Devices. By default the number of Devices (that can be created in auto-provisioning) for a tenant is unlimited.

Entering a number in this field will limit the number of Devices that can be auto-provisioned.

Maximum number of Conference Rooms. By default the number of Conference Rooms that can be created 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(s) here.

Use existing tenant as a prototype. This field 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” and a list of options for the further selection of which parts of the configuration will be “cloned”.

Extension creation options

These options are used in Bulk Generator and as defaults on the User Extensions screen.

Note that the values specified on the tenant level override system wide defaults specified in the **PBX Settings -> Default Values** screens.

Default script for extensions. Allows to select a default script for handling calls to User Extensions.

Create mailbox and enable voicemail. Allows to specify whether User Extensions will have voicemail enabled.

Phone Model. Allows to select a default phone model for selection of appropriate templates in auto-provisioning.

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.

Default schedule for inbound calls. Default schedule to be used when automatically generating Inbound Routes for created User Extensions – both used in

Bulk Generator (when DIDs are specified) or in User Extensions.

User Portal creation options

These options determine user permissions for access to User Portal and are used in Bulk Generator or as defaults on the User Extensions screen.

Note that the values specified on the tenant level override system wide defaults specified in PBX Settings -> Default Values.

Conference Rooms' creation options

Default values used when creating conferences in Conference Manager and Bulk Generator and as defaults on the User Extensions screen (if option of creating an associate conference is selected in Extension creation options).

Note that the values specified on a tenant level override system wide defaults specified in PBX Settings -> Default Values.

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

CRM Settings

Settings for reverse caller id lookup in CRM screen of the User Portal.

The values specified here are used in Bulk Generator and as defaults on the User Extensions screen.

Note that specifying this on a tenant level overrides system wide defaults specified in PBX Settings -> Default Values.

Shortest caller id length triggering screen pops. This can be used to eliminate screen pops for extension-to-extension calls. Specify a minimal length that should trigger a screen pop – calls with a caller id shorter than specified here will be ignored.

Number of digits to strip from caller id . Number of characters to remove from the beginning of caller id before processing. Note that the positions in `${callerid:firstdigit:lastdigit}` are computed after the characters are removed.

Link for reverse caller id lookup. Select a default link from the list specified in CRM Links.

Branding

Branding allows uploading a custom logo and setting of user interface preferences for each tenant.

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.

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

By default Manager interface is configured to allow access only from the local host. Tools -> Manager Interface and Manage Conference screens require manager interface access from the browser. This access can be enabled by adding "allow" for a network or host addresses in manager.conf file.

Branding

This screen allows you to upload a custom logo to be used in the page header and the Login screen, specify a URL that the logo links to (by default it links to Third Lane Technologies web site) and hide some Thirdlane related information if you don't want that to be shown to your customers or users.

Multi-Tenant PBX Manager also allows branding on a tenant level.

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 network 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 gateway. Specify default gateway

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

Auto provisioning method. Your phones may support different protocols for auto provisioning (HTPP(S), FTP, TFTP,etc). If using HTTP or HTTPS, you can use PBX Manager for serving configuration files. If using other methods you will have to configure and run an appropriate server and specify a directory where PBX Manager is supposed to place configuration files it generates. You have to make sure that this directory is accessible by your provisioning

server or that the generated files are moved to the appropriate location.

CDR Settings

The **CDR Settings** screen allows you to specify Asterisk CDR destinations and connection options as well as to specify where PBX Manager is going to look for CDR records to display. Note that for writing CDR to databases you must also install an appropriate database software and compile Asterisk with relevant libraries. Note that CDR records will be written to all the enabled destinations.

Call Detail Records destinations

Write CDRs to CSV text file allows you to enable/disable of Asterisk writing Call Detail Records to a text file. This is enabled by default.

Write CDRs to PostgreSQL allows you to enable/disable of Asterisk writing Call Detail Records to a PostgreSQL database and allows to enter connection options.

Write CDRs to MySQL allows you to enable/disable of Asterisk writing Call Detail Records to a MySQL database and allows to enter connection options.

Call Detail Records display

When displaying records select them from: allows to specify the default for selection of records for display in Call History -> Call Detail Records screen. Typically the records would be displayed from the same location as they are written by Asterisk but this feature allows you to show the data from a different source (where data could be loaded after some preprocessing).

Administrators

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

Note that this section is not supposed to be used for managing tenant level administrators, you should use Tenant Management screen for

that.

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 links/menus 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.

Link URL. Associated URL – this only applies to lower level links that are not parents of other links.

Description. A description of this link that will appear on an index page of a parent link.

Parent Link. A parent link of this link for multi-level menus.

Show in management menu. Specify whether this link should appear on the PBX Manager management/administration menu. You can also specify which users will be allowed to see this link.

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

Open in a new window. Specify whether this link should open in a new window.

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

CRM Links

The **CRM Links** screen allows you to add custom URLs that can be used for reverse caller id lookup in the CRM screen of the User Portal.

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 Description. Short description of this link.

Link URL. Associated URL. You can use parts of the caller id in the URL by using the following syntax:

`${callerid:firstdigit:lastdigit}`

For example:

`${callerid:1:10}` – caller id starting at the 1st digit up to (and including) 10th digit

`${callerid:1:3}` – caller id starting at the 1st digit up to (and including) 3rd digit.

`${callerid:4:10}` – caller id starting at the 4th digit up to (and including) 10th digit.

Available to tenants. (Multi-Tenant PBX Manager only). Specify which tenant will be able to use this link.

Editable by tenant administrators. (Multi-Tenant PBX Manager only). Specify whether tenant level administrators can modify this link.

CRM Settings

The **CRM Settings** screen allows system administrator to specify default values used in reverse caller id lookup in the CRM screen of the User Portal. Note that CRM Settings can be specified for each extension and in the Multi-Tenant PBX Manager these defaults can be overridden on a tenant level.

Shortest caller id length triggering screen pops. This can be used to eliminate screen pops for extension-to-extension calls. Specify a minimal length that should trigger a screen pop – calls with a caller id shorter than specified here will be ignored.

Number of digits to strip from caller id . Number of characters to remove from the beginning of caller id before processing. Note that the positions in

`#{callerid:firstdigit:lastdigit}` are computed after the characters are removed.

Link for reverse caller id lookup. Select a default link from the list specified in CRM Links.

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. User interface language.

Color scheme. 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.

Show expanded menus. Allows you to specify whether the menu items for all categories will be always shown. If “No” is selected then only the items for the currently selected category are shown.

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.

If you purchased PBX Manager from Third Lane Technologies, you can request a license by following 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. In 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 look at the related documentation for creating User Extensions and 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 look at the related documentation for creating User Extensions and 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 Teplate 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.

Extension creation options

These options are used in Bulk Generator and as defaults on the User

Extensions screen.

Note that the values specified on the tenant level override system wide defaults specified in the PBX Settings -> Default Values screens.

Default script for extensions. Allows to select a default script for handling calls to User Extensions.

Create mailbox and enable voicemail. Allows to specify whether User Extensions will have voicemail enabled.

Phone Model. Allows to select a default phone model for selection of appropriate templates in auto-provisioning.

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.

Default schedule for inbound calls. Default schedule to be used when automatically generating Inbound Routes for created User Extensions – both used in Bulk Generator (when DIDs are specified) or in User Extensions.

User Portal creation options

These options determine user permissions for access to User Portal and are used in Bulk Generator or as defaults on the User Extensions screen.

Note that the values specified on the tenant level override system wide defaults specified in PBX Settings -> Default Values.

Conference Rooms' creation options

Default values used when creating conferences in Conference Manager and Bulk Generator and as defaults on the User Extensions screen (if option of creating an associate conference is selected in Extension creation options).

Note that the values specified on a tenant level override system wide defaults specified in PBX Settings -> Default Values.

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

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. The text and subject line of the email sent can be edited in `/etc/asterisk/voicemail.conf`.

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. Sends a short email message to the specified email address. This is usefull for sending a text message via SMS.

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

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 in features.conf) 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.

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

Enabled codecs and disabled codecs. Codecs for this phone configured in Asterisk – these must match codecs available and configured in the actual phone. Recommendation: G711u and G729a

DTMF mode. Should match DTMF mode in phone configuration (applies to SIP phones only).
Recommendation: RFC2833.

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 send a message then watch for a response to determine if the device is reachable.

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 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 Thirdlane 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.

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 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.

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 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 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. 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 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 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 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 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.

DIDs & Trunks

DIDs

Single-Tenant version does not require to maintain the DIDs separately and allows you to enter DIDs directly in **Bulk Generator**, and when creating Inbound Routes or User Extensions.

The DID screen applies to the Multi-Tenant PBX Manager only and is used to manage phone numbers frequently called DIDs or DDIs.

You can enter 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.

When adding DIDs prepend field allows adding prefix to a range of DIDs you specify – this can be useful when DIDs may start with 0 or a +.

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, other Asterisk servers or soft switches.

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 be 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. Note that in cases where the traffic from the host is allowed based on the IP address you can omit both user name and password and specify insecure=very in other options field.

Password. (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. Note: it is not recommended to specify caller id as this will override any other caller id settings.

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. (SIP, IAX) This is supposed to match codec preferences specified by the service provider.

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

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

NAT. Check this if trunk 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. You should have this unchecked for trunks behind NAT.

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 trunk is reachable.

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. For example, in cases where the traffic from your provider is allowed based on the IP address you can omit both user name and password and specify insecure=very in this field.

Example:

```
insecure=very
```

Registration. (SIP, IAX) The registration string as provided by your VoIP service provider (may be 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 dialstring 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 trunk 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/\$

Caller ID options. Allows to alter Caller ID on the outbound calls using this trunk.

The following fields are also available:

of digits to strip – how many digits will be stripped from the front of th caller id.

String to prepend - string that will be prepended to the caller id.

Provider's web site. This field 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 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 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 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:

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. Thirdlane 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” 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 take 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 an 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 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.

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” 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 (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 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**.

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” 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. 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 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 to 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 an 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 the 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" 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" 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” 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

- `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

PBX Information

This screen display a summary of your PBX configuration and status including PBX version, Uptime, SIP Channels, SIP Peers, SIP Registry, IAX Channels, IAX Peers, IAX Registry, ZAP Channels, Conferences, and Voicemail Users. Please see Asterisk documentation for details.

Note that this and additional information can be obtain using individual Asterisk command using Tools -> PBX Command Shell.

Auto-Provisioning

While configuration of the devices can be performed by entering data into manufacturer's supplied web interface, it is more efficient to use PBX Manager's auto-provisioning feature to create configuration files that a device downloads using one of the supported methods (HTTP, FTP, TFTP, etc).

For device auto-provisioning PBX Manager uses templates which are combined with device specific information to produce device configuration files.

A directory `/etc/asterisk/provisioning` contains a file called `models.txt` and various template files.

If you wish to customize existing template or create new templates then use the directory `/etc/asterisk/user_provisioning`. The directory `/etc/asterisk/provisioning` will be over written whenever PBX Manager is updated. Copy the `models.txt` from the primary directory to the `user_provisioning` directory and delete any phones not defined in `user_provisioning`. If a phone exists in both directories the one in `user_provisioning` will take precedence.

`models.txt` contains entries for all the supported devices and provides references to the templates for each device type.

Templates contain text and variables which are replaced with appropriate data from PBX Manager during provisioning.

Variables are specified as `${VARIABLE_NAME}` and are case sensitive.

PBX Manager processes all the templates, replaces variables, and places the resulting files in the directory specified in System Settings - > Provisioning Settings.

Here is a `models.txt` entry for a snom 320:

```
[snom-320]
label=Snom 320
```



```
lines=12
phone_template=snom3xx_phone.cfg
line_template=snom3xx_line.cfg
blf_template=snom3xx_blf.cfg
speeddial_template=snom3xx_speeddial.cfg
output=snom3xx- $\{MAC\}$ .cfg
input_1=snom3xx_settings.cfg
output_1=snom3xx.cfg
```

Fields in the models.txt entry are as follows:

label. Phone model (used to select this device model PBX Manager GUI).

lines. Number of lines this device supports

phone_template. contains all of the general phone information other than the line specific entries.

line_template. contains the configuration parameters for a line. For example, if 3 lines are defined, then 3 copies of the entries in the line_template will be included in the final output file.

blf_template. The template processed for each button designated as a BLF (Busy Lamp Field)

speeddial_template. The template processed for each button designated as a Speed Dial

output. The name of the generated (output) configuration file for each provisioned device. Note that variables can be used as part of the file name.

input_x. Any additional files that need to be processed.

output_x. The name of the file resulting from processing input_x. Note that variables can be used as part of the file name.

required_x. Specifies any files that PBX Manager will simply move in the output directory without any substitutions. Typically contains system wide configuration settings

command_x. Command to be executed at the end of the provisioning for each device. This can be used to run special applications required by some phone manufacturers (like Grandstream) to load the configuration files. Note that variables can be used as part of the command.

Here is an entry for Polycom 501:

```
[polycom-501]
label=Polycom 501
lines=3
phone_template=polycom_phone.cfg
line_template=polycom_line.cfg
output=${mac}-registration.cfg
input_1=polycom_mac.cfg
output_1=${mac}.cfg
input_2=polycom_local.cfg
output_2=local-settings.cfg
required_1=sip.cfg
```

Note that the `${MAC}` variable will be substituted by the MAC address in uppercase, while `${mac}` will be in lowercase.

Here is the `snom3xx_phone.cfg`

```
<html>
<pre>
```

```

${LINES}
${BLFS}
${SPEEDDIALS}

```

```

</pre>
</html>

```

Note the `${LINES}`, `${BLFS}` and `${SPEEDDIALS}` variables. These will be replaced by the combined content of processed templates - `line_template`, `blf_template`, and `speeddial_template` for each button configured.

Customers can add support for any devices not currently supported by Thirdlane or customize the templates as follows:

1. If adding a new device model create a `models.txt` file in `user_provisioning` directory under the configuration directory (typically `/etc/asterisk/user_provisioning`) and add an entry for the device model you want to add
2. Create the templates (or change the existing templates) and place them in the same directory. Note that you are not required to create `blf_template` and `speeddial_template` files unless you use them.
3. For the existing device models the information specified in `models.txt` in `user_provisioning` directory will take precedence over the same in provisioning directory, and any new models will be added to the list of models listed in PBX Manager GUI.

Here is a list of variables that can be specified in `models.txt` and the provisioning templates.

Variable	Value
<code>\${MAC}</code>	Device MAC address in uppercase
<code>\${mac}</code>	Device MAC address in lowercase

<code>\${SERVER}</code>	PBX server address
<code>\${TENANT}</code>	Tenant name (Multi-Tenant only)
<code>\${TENANT_ID}</code>	Internal tenant id (Multi-Tenant only)
<code>\${LABEL}</code>	<code>\$device{'description'};</code>
<code>\${COMPANY_NAME}</code>	Name on the license (Licensed to)
<code>\${LINE_KEYS}</code>	Number of lines specified in models.txt
<code>\${LINES}</code>	Content of processed line_templates
<code>\${BLFS}</code>	Content of processed blf_templates
<code>\${SPEEDDIALS}</code>	Content of processed speeddial templates
<code>\${STATION_NAME}</code>	Caller id name
<code>\${VOICEMAIL_NUMBER}</code>	Extension
<code>\${USERID}</code>	Line name as specified in User Extensions
<code>\${PASSWORD}</code>	Phone password as specified in User Extensions
<code>\${DISPLAY_NAME}</code>	Caller id name
<code>\${EXTENSION}</code>	Extension
<code>\${NAT_MAPPING}</code>	Yes/'No' depending on NAT setting in User Extensions
<code>\${NAT_KEEPALIVE}</code>	'Yes/'No' depending on NAT setting in User Extensions
<code>\${BUTTON}</code>	Button number as specified in Tools -> Auto-Provisioning
<code>\${LINE}</code>	Button number as specified in Tools -> Auto-Provisioning (it is the same as <code>\${BUTTON}</code> and is kept for backward compatibility. You should use <code>\${BUTTON}</code> for new templates.

<code>\${BUTTON_ZERO_BASED}</code>	Button number - 1. This is useful when buttons are zero based as in snom phones
<code>\${DISPLAY_NAME}</code>	Caller id name
<code>\${LINE_LABEL}</code>	Line label as specified in Tools -> Auto-Provisioning
<code>\${LABEL}</code>	Line label if non blank (if specified in Tools -> Auto-Provisioning) or caller id number
<code>\${BLF_LABEL}</code>	Label for a BLF button as specified in Tools -> Auto-Provisioning
<code>\${BLF_TARGET}</code>	BLF target as specified in Tools -> Auto-Provisioning
<code>\${SPEEDDIAL_LABEL}</code>	Label for a Speed Dial button as specified in Tools -> Auto-Provisioning
<code>\${SPEEDDIAL_TARGET}</code>	Speed Dial target as specified in Tools -> Auto-Provisioning

If MAC address is specified the configuration files are generated automatically when User Extensions are created either in **Bulk Generator** or individually (provided that Provisioning Settings are in place (see System Settings -> Provisioning Settings)).

Note that if you use Bulk Generator you must specify the default Phone Model (in PBX Settings when using Single-Tenant or in Tenant Management if using Multi-Tenant PBX Manager).

By default sip lines created when User Extensions are created is assigned to all buttons.

The **Auto-Provisioning** screens allow you to further refine auto-provisioning information and assign behavior to phones' 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 Button creates a row where you can select a button of the phone being provisioned and specify its behavior.

Currently you can configure each button as a Line, BLF or a Speed Dial. To associate a Line with all available Buttons use “All Buttons” option.

You can generate configuration files for the phone 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 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 backup and restore of Thirdlane PBX.

Backup/Restore screen contains 3 sections:

Restore PBX. This allows you to restore from a previously created backup which can be:

Uploaded file. The backup file will be uploaded from a local computer and deployed on a server.

File on the server. The restore process will use a file already on the Thirdlane PBX server.

Recent backup. PBX Manager will use one of the recent backups – either a local file on the server or a file on a remote FTP server.

Backup now. This allows you to perform an immediate backup.

The destination of the backup file can be selected as follows:

Download in browser. Backup file will be downloaded to the users computer.

File on server. If selected, provide the filename to use for the backup file to be created on the Thirdlane PBX server (absolute path is required).

FTP server. If selected, provide a host name or ip address of an FTP server, the name of the file to be placed on the FTP server, and FTP login and password. The backup will be sent to the specified server using FTP protocol.

Recent backup. Allows to select the same file (or ftp configuration) as was used in one of the previous backups.

What to include. Allows to specify whether voicemail and recorded calls are to be included in the backup.

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

When to backup. Allows to enable or disable scheduled backup.

File on server. If selected you need to also provide the name to use for the backup file to be created on the Thirdlane PBX server (absolute path is required).

FTP server. If selected you also need to provide a host name or ip address of an FTP server, the name of the file on FTP server, and FTP login and password. The backup will be sent to the specified server using FTP protocol.

Recent backup. Allows to select the same file (or ftp configuration) as was used in one of the previous backups.

What to include. Allows to specify whether voicemail and recorded calls are to be included in the backup.

PBX Command Shell

PBX 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 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.

Manager Interface

Manager Interface screen can be used for capturing Asterisk Manager Interface (AMI) output for diagnostic purposes and testing of Asterisk Manager Interface commands.

Manager Interface screen requires that the address of the user's workstation be included in the list of “allow” 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.

Manager Interface screen contains 2 windows. The window on the left simply displays Asterisk Manager events, the window on the right can be used as a scratch pad and allows sending commands to the Manager. Pressing Send button sends the whole content of the window (which may contain multiple commands) to the Manager, pressing Send Selected button sends only selected text.

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".