Your First PBX: A Beginner's Guide to Issabel

A Guide to Installing, Configuring, and Managing Issabel For Your Needs

Version 1.0

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1. This guide was created for an educational assignment and is not affiliated with or endorsed by Issabel LLC. Any use of this book is for educational purposes only.

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Preface: Purpose of this guide

This guide intends to help users of all backgrounds successfully install, configure, and manage Issabel PBX. The instructions and reference material in this guide are written with new users in mind to provide definitions and practical instructions for you to understand how to use Issabel PBX. Whether you are deploying Issabel for the first time, integrating it with existing systems, or handling ongoing maintenance, this guide will show you how to get the most out of Issabel PBX.

Who this guide is for

This guide is made to support a wide range of users involved with the Issabel PBX system, from installation and configuration to daily operation and troubleshooting. The primary audience of this guide includes:

- IT Administrators responsible for telephony systems and infrastructure
- Technical Support Staff providing assistance or troubleshooting
- Call Center Managers managing operations and call flow

- Business Users requiring guidance on daily features like voicemail and call forwarding
- Consultants and Integrators extending or customizing Issabel with third-party solutions

No prior experience with Issabel is required, although basic networking and telephony knowledge will be helpful.

Using Issabel for your business

Issabel PBX enables businesses to manage internal and external communications — including voice, video, and messaging — in one system. Issabel is suitable for companies of all sizes that are looking to improve collaboration and reduce telephony costs.

Your business can benefit from Issabel's cost-effective open source model and the extensive ecosystem of supported features. Issabel's modular design allows you to tailor the system to your specific communication needs, leveraging its capabilities to improve productivity and customer engagement.

Setting up Issabel on your device

To use Issabel, install the software on supported hardware or virtual a virtual machine. Then, configure basic system settings such as network access and user extensions. This guide has step-by-step instructions to help you complete the setup with ease.

This guide will cover:

- Device and software specifications
- Hardware and network requirements
- Configuration settings to handle calls, voicemail, and device registration

Additional guidance is provided on connecting SIP trunks, configuring security settings, and planning user roles for complex deployments.

Operating with Issabel

After Issabel has been installed on your device, this guide will walk you through how it is used in managing daily operations. These chapters will cover the following major features:

- Call handling
- Using interactive voice response (IVR) menus
- Managing voicemail
- Monitoring system status
- Campaign and queue management

This guide also provides instructions for common system maintenance and support operations such as:

- Troubleshooting common issues
- Maintaining system health
- Performing routine administrative task

Chapter 1: Overview of the Issabel PBX System

This chapter explores the key features of PBX systems and how they are used in businesses. You will learn about the different types of PBX systems, key features and components, VoIP technology, and requirements to use PBX systems. By the end of this chapter, you will understand how organizations use PBX systems to structure and manage their telephone network and the technology used to facilitate their communications.

This chapter at a glance

Tom has a development company and has recently extended to offer travel services. He wants to set up a phone system for his development and travel company to help him direct callers to his various services and manage his customer communications. He prefers to minimize the amount of hardware required as much as possible and is looking for a cost-effective option that would allow him to have control over its features. After doing some research, Tom chooses Issabel PBX for its cost-effective open-source platform and flexible features with active support.

Introduction to Issabel

Issabel PBX (often referred to as just Issabel) is an open-source unified communications platform created to help businesses manage their telephony and contact center operations as a cost-effective alternative to traditional telephony management systems.

Issabel was created in 2017 as a branch off Elastix, another unified communications platform, after it was acquired by 3CX. It aims to continue providing users with the same functionality and flexibility while maintaining its open-source availability.

Issabel uses Asterisk — another open-source PBX — to run its core functions and builds upon it by adding an accessible management user interface and a unified platform that integrates other communication tools.

Understanding PBX systems

Many of us have probably used a PBX system as a caller in some form or another throughout our lives. However, we probably don't spend a lot of time understanding how the systems that our calls take place in are set up, or how the business customizes the automated menu to provide us with the options we need to navigate their services. In order to understand what Issabel is and what it is capable of, we have to first understand what a PBX is, and how it works in the realm of telephony.

What is a PBX?

A private branch exchange (PBX) is a private telephone network that is used within an organization. It provides general telephone services along with additional features to manage both internal and external calls. It is the foundation that enables organizations to direct calls, create call menus, and organize them in a queue until a representative is available.

Comparing types of PBX systems

Table 1.1 shows that there are four main types of PBX systems that differ by system type and deployment location:

System Type	Deployment Model	Location
Traditional (Analog/Digital)	On-premise	Located at the business
	Hosted	Located off-site
IP-based (VoIP)	On-premise	Located at the business
	Hosted	Located off-site

Table 1.1: Types of PBX Systems

Issabel is an IP-based PBX system that can function on-premise or be hosted by a third party. *Table 1.2* highlights the benefits and tradeoffs for both options, and ultimately businesses can choose the one that fits their needs and priorities.

Deployment Type	**Benefits**	**Tradeoffs**
On-Premise	Full control over hardware and system configuration. No recurring hosting fees. Higher level of security if properly maintained	Higher upfront costs for hardware and setup Requires in-house IT expertise for maintenance and troubleshooting Limited scalability
Hosted (Third-Party)	Lower upfront costs Minimal IT involvement Easier to scale and update Access from multiple locations	Ongoing monthly or annual fees Less control over infrastructure Dependent on Internet connection and third-party reliability

Table 1.2: Comparison of On-Premise and Hosted PBX Systems

Core components of a PBX System

A PBX system has a few key components that it needs to function:

- **PBX Server** - acts as the "brain" and manages call routing, extensions, voicemail, and more.
- **Telephony endpoints** - include IP phones, analog phones, softphones, and mobile extensions.
- **Gateways** - devices that bridge analog/digital phone systems with IP networks, allowing legacy device integration and PSTN connectivity (only necessary if using a hybrid of analog devices).
- **Network infrastructure** - relies on reliable local area network (LAN) with sufficient bandwidth for voice traffic.

Comparing softphones or deskphones

Softphones and deskphones are the endpoints that communicate voice data. Deskphones are hardware devices like the traditional telephone. Softphones are software applications that have been installed on computers or mobile devices.

Softphones provide flexibility for remote work and mobility, while desk phones offer a traditional, tactile experience suited to office needs. Many organizations deploy a mix of both to meet different user needs.

Using softphones

Softphones are software applications that replicate the functionality of traditional phones. The SIP connects them to the PBX and allows users to make voice and video calls, send messages, an access voicemail from a computer or smartphone.

Popular softphones include:

- Zoiper - known for cross-platform compatibility and ease of use
- MicroSIP - lightweight and open-source, ideal for Windows users
- Bria (formerly X-lite) - a feature-rich softphone with advanced calling features
- Linphone - Free and open-source with support for encrypted calls

Each softphone varies in features, licensing, and ease of integration with PBX systems like Issabel.

Using deskphones

Deskphones, or physical IP phones, connect directly to your network and communicate with the PBX using SIP protocols.

SIP phones are standalone hardware devices that plug into an Ethernet port or connect via Wi-Fi. They handle voice encoding, decoding, and signaling internally, without needing a computer or mobile device.

SIP phone models range from basic units with simple call functions to advanced models offering high-definition audio, video calling, an touchscreens. Popular brands include Yealink, Cisco, Polycom, and Grandstream.

Key Features of PBX and Issabel

Issabel offers many of the same features as common modern PBX systems, with greater flexibility and affordability for additional features, as shown in *Table 1.3*. These features equip your organization with the fundamental qualities needed for your internal phone system. While you can use Issabel without any licensing fees, it is recommended that organizations register for a license with Issabel to leverage its advanced features. You can learn more by visiting *https://issabel.guru/*

Feature	Issabel PBX	Common PBX Systems
Open source	Yes	No
Free to use (No licensing fees)	Yes	No
Web-based user interface	Yes	Varies by vendor
IP telephony support (VoIP)	Yes	Yes
Analog device support	Yes	Yes
Call routing & IVR	Yes	Yes
Voicemail to email integration	Yes	Yes
Call recording	Yes	Yes

Table 1.3: Comparison of Issabel and other PBX Systems

Feature	Issabel PBX	Common PBX Systems
CRM integration	Yes	Varies by vendor
Multi-language support	Yes	Varies by vendor
Customization via plugins	Yes	Limited
Third-party SIP trunking	Yes	Varies by vendor
Commercial support	Yes	Yes
Scalable	Yes	Yes
Built-in call center features	Yes	May require add ons

Table 1.3: Comparison of Issabel and other PBX Systems (Continued)

Call center tools

In addition to call management features, Issabel offers a wide arrange of specifically helpful for call centers.

- Campaign management
- Omnichannel communication
- Agent and supervisor console
- Queue and ring group control
- Dialer and agent workflow tools
- Reporting and analytics
- Call control and recording

Managing customer data

Organizations can manage customer data and alternative communication channels by using Issabel's Customer Relation Management (CRM) module or integrating an external CRM for more advanced features. *Table 1.4* shows these features below.

Issabel's Built-in CRM		**External CRM Integration**	
Feature	**Description**	**Feature**	**Description**
Customer database	store basic contact details (name, phone, email address)	Detailed customer profiles	track tickets, emails, and social interactions
Call logging	automatically link inbound and outbound calls to customer records	Sales and marketing tools	lead scoring, campaign management, and email tracking
Notes and reminders	can be added during or after calls with follow-up reminders	Workflow automation	automatically assign tasks, escalate issues, or trigger emails based on call events
Call history	all calls and interactions with each customer	Advanced reporting	analytics to provide customer insights and forecasting
Search and filter	customers by name or number and filter by recent activity	Multi-channel support	integrates SMS, chat, social media, and email alongside voice calls
Basic reporting	summaries and data of customer interactions that can be exported	Team collaboration tools	share customer info, assign follow-ups, and track team performance

Table 1.4: Built-in and External CRM Features

Using VoIP for PBX Systems

Now we understand how businesses use PBX systems to create a customized network to handle external communications. But how are calls actually connected? How do words travel from the caller to the business? Now that we understand what PBX systems are, let's explore how they use VoIP to operate.

What is VoIP?

Voice over Internet Protocol (VoIP) is a technology that allows you to make voice calls using an Internet connection instead of a traditional phone line. Instead of using physical wires, VoIP converts your voice into digital data packets that are transmitted over an Internet connection. This is what allows phone calls to be made using computers, smartphones, and even traditional phones through an adapter.

Implementing VoIP in PBX systems

All IP-based PBX systems use VoIP technology to transmit calls. In summary, analog voice is converted to digital data by an Analog Telephone Adapter (ATA) that is compressed by codecs to reduce the size. This data is then packaged into packages and transmitted over the Internet using VoIP. At its destination, another codec decompresses the data and, if necessary, converted to analog by an ATA if the receiving phone is traditional.

Here is an overview of how calls are transmitted using VoIP:

1. An Analog Telephone Adapter (ATA) converts the analog voice spoken into the receiver into digital data.

2. A codec compresses this digital data to reduce its size and make it more efficient during transmission.

3. VoIP packages the compressed data into packets that are transmitted over the Internet.

4. Another codec decompresses the packets at its destination. If the receiving device is a traditional phone, an ATA converts the digital data back to analog.

Exploring SIP basics

Session Initiation Protocol (SIP) is the most common type of protocol used by VoIP to handle calls. What SIP does:

1. **Establish sessions** - finds and connects endpoints.

2. **Manage sessions** - handles call transfers, holds, and adding/removing participants.

3. **Terminate sessions** - ends calls when either party hangs up.

SIP can be thought of as a “traffic director,” coordinating how and when calls start, what routes they take, and when they end.

Considering codecs and bandwidth requirements

A codec, which is short for “coder-decoder,” is an algorithm that compresses and decompresses voice data for transmission. They determine:

- Audio quality (how clear the call sounds)

- Bandwidth usage (how much data is sent over the network)

- CPU load (how demanding it is on the processor)

Issabel uses Asterisk for all its codec management. Issabel updates with every iteration of Asterisk to keep up with current telephony specifications. *Table 1.5* highlights the common types of codecs supported by Issabel. You can prioritize codecs in **Asterisk SIP settings** in Issabel's web GUI.

Codec	Bandwidth per call (approx.)	Notes
G.711	~64 kbps (plus overhead ~87 kbps)	Uncompressed, high quality, uses more bandwidth
G.729	~8 kbps (plus overhead ~31 kbps)	Compressed, good for low-bandwidth networks, requires licensing
G.722	~64 kbps	High-definition (wideband) audio
Opus	Variable (6-510 kbps)	Adaptive, high quality, efficient

Table 1.5: Common Issabel-supported SIP Codes

Each active call consumes bandwidth. It is important to select an Internet plan that has enough bandwidth to suit your needs.

Example:

- 10 simultaneous G.711 calls ~ 870 kbps upstream and downstream
- 10 simultaneous G.729 calls ~ 310 kbps upstream and downstream

Meeting Issabel's requirements

Before deployment, it's important to understand the system and environmental needs to ensure smooth installation and optimal performance. By meeting these requirements, organizations can take advantage of Issabel's full capabilities while maintaining reliability and scalability.

Assessing system requirements and deployment options

Evaluate your organization's computing environment and usage goals before deploying Issabel PBX.

Defining hardware and software specifications

These are the minimum hardware specifications required to operate Issabel:

- **Processor**: single-core x86 processor (at least 1.6 GHz)
- **RAM**: At least 1 GB (4 GB or higher recommended for >60 users or high call volumes)
- **Storage**: 10 GB of hard disk space (SSD preferred for performance and call recordings), consider 500 GB+ for call recording or enterprise environments.
- **Network**: Gigabit Ethernet recommended

Recommended hardware example: 4-core Intel Celeron (or better) CPU, 4 GB RAM, 500 GB HDD/SSD, hardware RAID option, dual Gigabit NICs.

Note: To support more users and record multiple calls at once, use a faster processor and larger storage. The number of extensions or trunk lines does not affect system performance.

Software requirements:

- Issabel runs on 64-bit platforms (Linux based, commonly CentOS/Rocky Linux base)
- Virtualization support for VMware ESXi, VMware Workstation, Oracle VirtualBox, or equivalent environments

Deployment choices

There are flexible deployment options that accommodate everything from small office setups to complex, multi-site enterprises. Select the deployment model that aligns with your organization's scale, budget, and operational requirements.

There are a few options provided for deployment:

- **Physical server**: ideal for organization needing dedicated resources and total hardware control. This can involve small office rackmounts or enterprise-grade servers with hardware RAID, redundant power, and robust cooling.
- **Virtual machine**: Supports flexible resource allocation and easy migration using platforms like VMware ESXi, VirtualBox, or cloud-based virtualization services. Virtualization allows for efficient scaling as your user base or call volume grows.
- **Cloud-based deployment**: suitable for distributed or remote teams, or organization that want to reduce on-site hardware. Issabel can run in a cloud infrastructure or be purchased as a managed PBX-as-a-Service.
- **Hybrid architectures**: combining on-premise servers with cloud-based PBX enables multi-site organizations to maintain local resilience and centralized management.

Network and Connectivity Requirements

Issabel offers flexible options for businesses to configure their network, including local (LAN), wide area (WAN), and cloud-based deployments. This section will cover the network fundamentals and best practices needed to ensure call quality, seamless device registration, and secure, dependable connections to providers and endpoints.

IP addressing and DNS options

- Static IP is recommended for Issabel servers to ensure consistent connectivity with SIP/VoIP providers an remote devices.
- If DHCP is used, configure DHCP reservations to 'lock' the server to a specific address.
- Reliable DNS service is required to resolve external SIP trunk provider and remote endpoint hostnames.
- Configure reverse DNS (PTR records) for enterprise deployments to aid in troubleshooting and security.

Internet connectivity

- High quality, stable broadband with sufficient bandwidth for peak concurrent call volume.
- Low latency and jitter for voice quality; avoid consumer-grade networks for high call volumes.
- Consider redundant Internet connections or secondary WAN for critical business continuity.

- Public IP may be required for remote SIP device registration or cloud trunking.
- Proper NAT/firewall configuration to easily ensure ports (typically UDP 5060 for SIP, 10,000-20,000 for RTP) are accessible as needed.

User and Role Planning

Your organization likely has a variety of roles that have their own set of responsibilities and authority. When you are setting up your Issabel PBX system, it is important to consider who will have access to administrative settings and the tasks that users can carry out within the system.

Administrator login and access credentials

Here are some things to consider when setting up Issabel and administrator accounts:

- Create strong, unique passwords for all admin accounts.
- Consider setting up multiple admin accounts (with named users) for auditing and accountability.
- Secure admin access to Issabel GUI with HTTPS (restrict by IP if possible) and enable two-factor authentication if offered.

User roles and access permissions

Define roles based on business needs:

- Administrator: Full system and configuration access
- Operator/Supervisor: Access to call monitoring, call center, limited settings
- User/Extension Owner: Individual voicemail, call recording access, and user portal

Assign permissions to restrict sensitive operations to only trusted roles. Periodically review role assignments and rotate credentials for security.

Chapter 2: Setting Up the Issabel Environment

This chapter explains the steps needed to set up Issabel on your system. You will learn how to install a virtual machine, create a virtual environment, configure Issabel settings, and sign in to Issabel's web UI. By the end of this chapter, you will understand how to gain access to Issabel's interface and navigate its modules.

This chapter at a glance

Now that Tom has chosen a PBX system, he needs to set up an environment to run it from. Rather than installing it on a physical server, Tom decides to use a virtual machine (VM) to simplify deployment and maintenance and reduce the amount of physical hardware he'll have to manage.
He chooses Oracle VirtualBox because it doesn't require extensive knowledge on testing and deploying software and is cost-effective for his small island business. Through VirtualBox, Tom is able to customize the storage and network settings for the environment that Issabel will run from. He can easily launch and manage the Issabel interface whenever he'd like to from a single laptop. He can now begin to tailor Issabel's features to his business needs.

Preparing a virtual machine for Issabel

Issabel can be installed to and run on a single physical server. However, it is more practical to run Issabel from a virtual machine, because it reduces the physical resources and costs required and simplifies maintenance operations. In this guide, we will be using Oracle Virtualbox, however, you may choose another option that suits your needs.

Installing Oracle VirtualBox

Oracle VirtualBox is a free open-source platform that has flexible network configurations and is easy to use, making it an ideal choice for testing and small businesses.

Before installing your virtual machine, ensure that you have the right specifications, *See "Defining hardware and software specifications" on page 12*.

To install Oracle VirtualBox on your computer

1. Go to https://www.virtualbox.org/

2. Click **Download**.

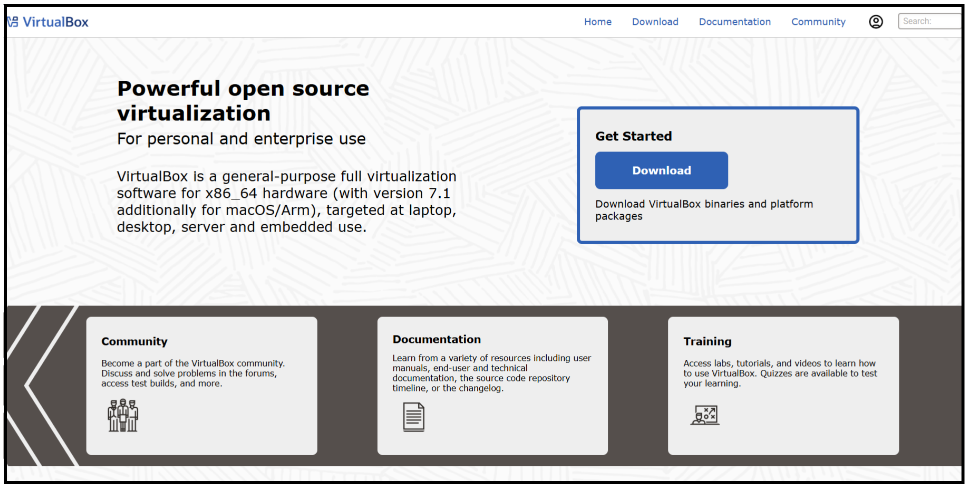

Figure 2.1: VirtualBox Homepage

3. Click **Windows hosts** from VirtualBox Platform Packages.

   Downloads window displays.

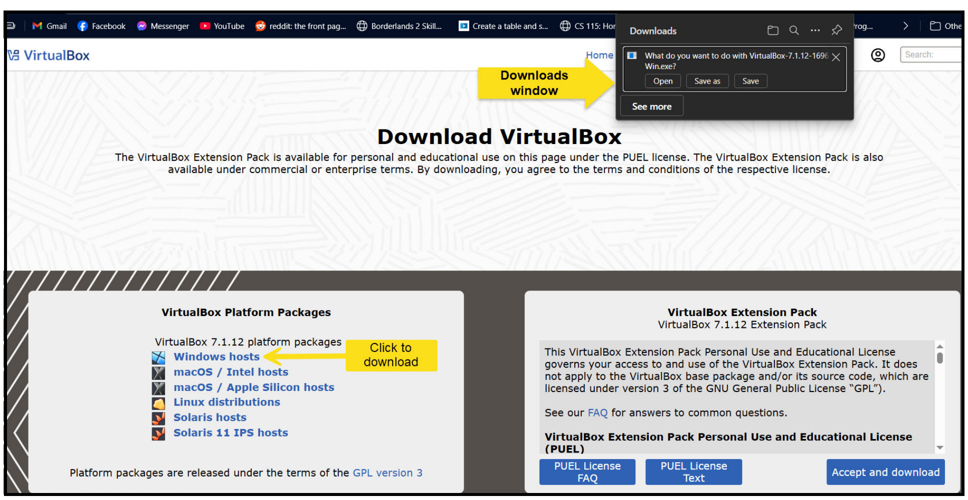

Figure 2.2: VirtualBox Download Page

4. Click **Save** and open the file location.

5. Double-click the file name to launch the Setup Wizard.

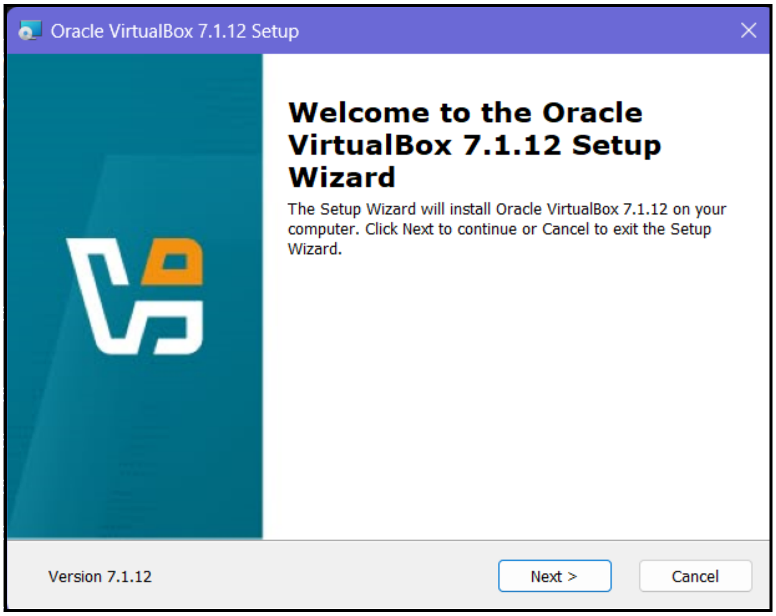

Figure 2.3: Oracle VirtualBox Setup Wizard

6. Follow the on-screen instructions. Accept the default settings.

7. Click **Install**.

   Installation begins.

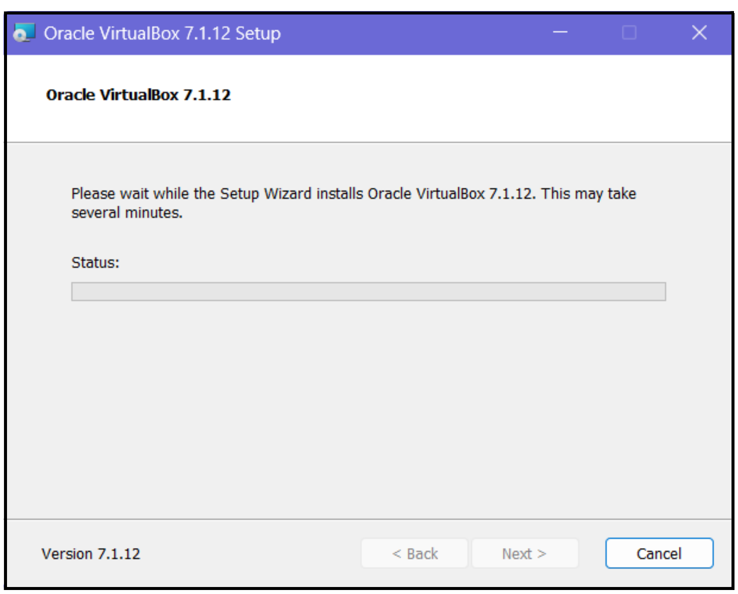

Figure 2.4: Oracle VirtualBox Setup Wizard Installation

8. Once the installation is complete, click **Finish**.

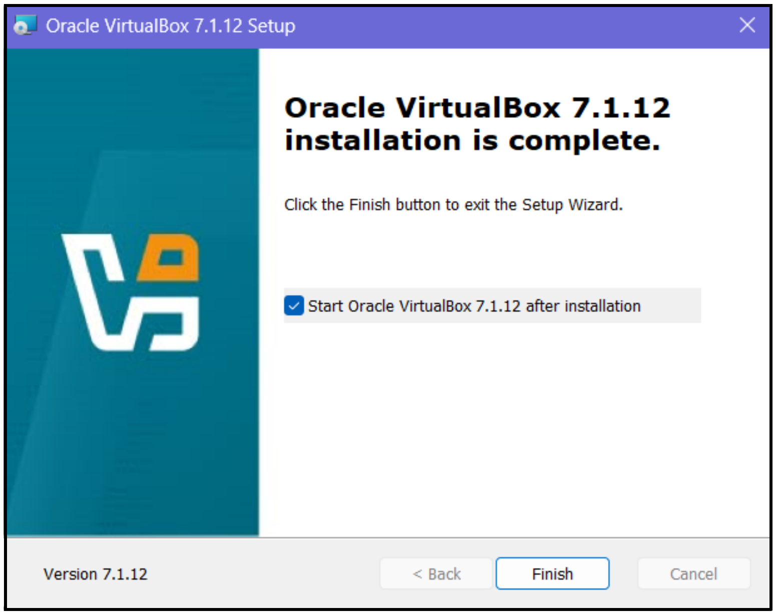

Figure 2.5: Completed Oracle VirtualBox Setup Wizard

Alternatives to VirtualBox

Here are some alternative virtual machines that are better suited to businesses that operate at a larger scale:

- Proxmox Virtual Environment: free, best for advanced server management
- Kernel-based Virtual Machine (KVM): free, best for servers, enterprise
- Citrix Hypervisor (Xen/XenServer): free/paid, best for cloud, large scale use

Setting up Issabel

To set up Issabel on your system, you need to download the file, create an environment for Issabel in VirtualBox (or your chosen virtual machine), and install Issabel from there. Then, you can choose your language and timezone, and configure your network settings and create a root password.

To download Issabel on your device

1. Go to https://sourceforge.net/projects/issabelpbx/.

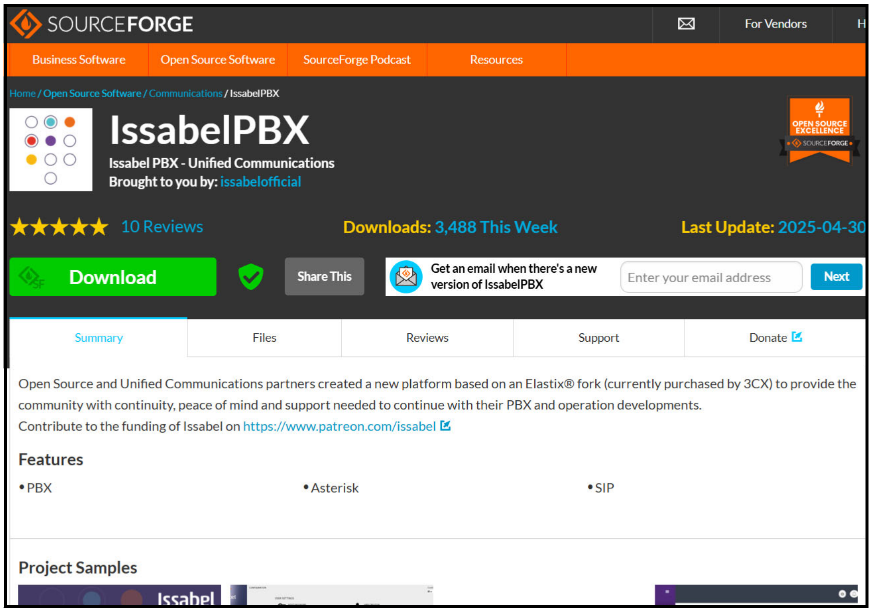

Figure 2.6: SourceForge IssabelPBX Page

2. Click **Download**.

   Download page displays.

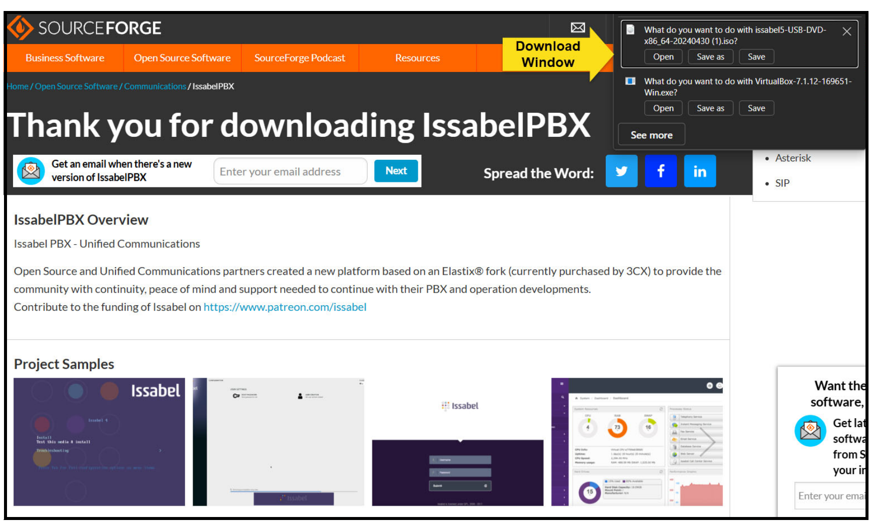

Figure 2.7: IssabelPBX Download Page

3. Click **Save**.

   File download begins.

Configuring virtual machine settings

Once the Issabel file has been downloaded, you can install it through your virtual machine. Before the installation, you will have to create a virtual machine for Issabel and configure its settings in the VirtualBox Manager shown in *Figure 2.8*.

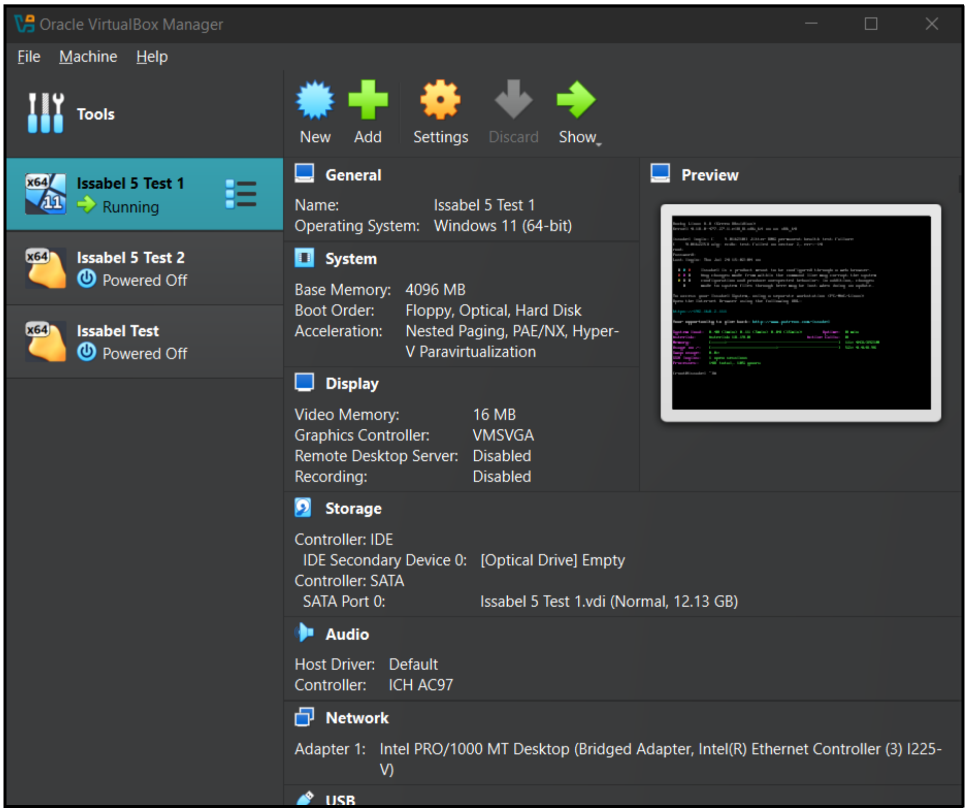

Figure 2.8: Oracle VirtualBox Manager Homepage

To create a virtual machine

1. Open Oracle VirtualBox and click **New.**

   Create Virtual Machine window displays.

2. Enter a name for your machine in the Name field.

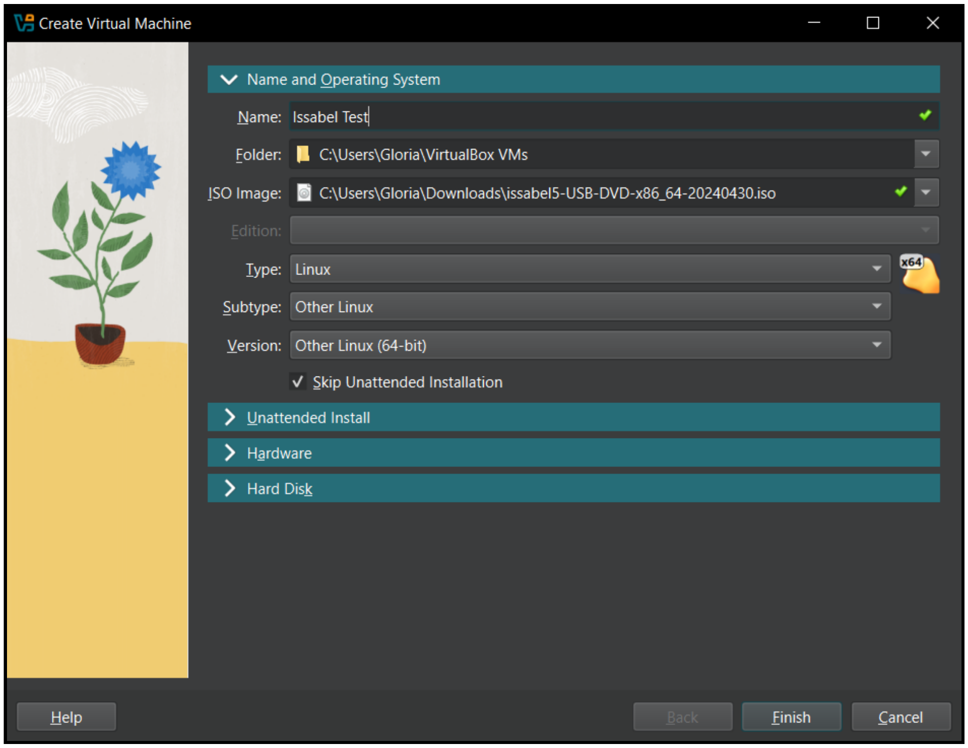

Figure 2.9: Create VM Name and OS System

3. Select the Issabel file from the ISO Image dropdown list.

Note: If you do not see it in the list, click **Other** and locate the file from the pop-up window.

4. Click **Skip Unattended Installation**.

5. Select **Linux** from the Type menu.

6. Select **Other Linux (64-bit)** from the Version menu.

7. Click **Hardware**.

   - Move the Base Memory slider to 4096 MB (4 GB).
   - Move the Processors slider to 1 CPU.

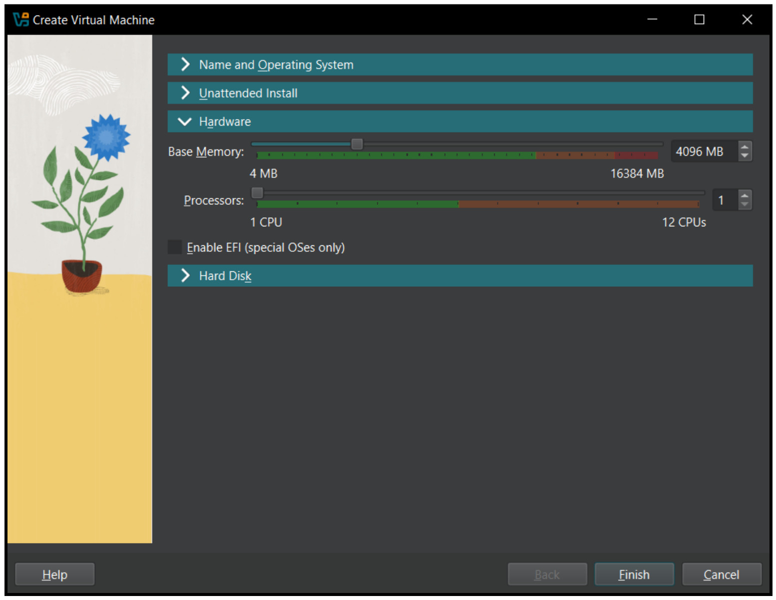

Figure 2.10: Create VM Hardware

8. Click **Hard Disk**.

   - Enter 12 GB in the Hard Disk File Location and Size field.

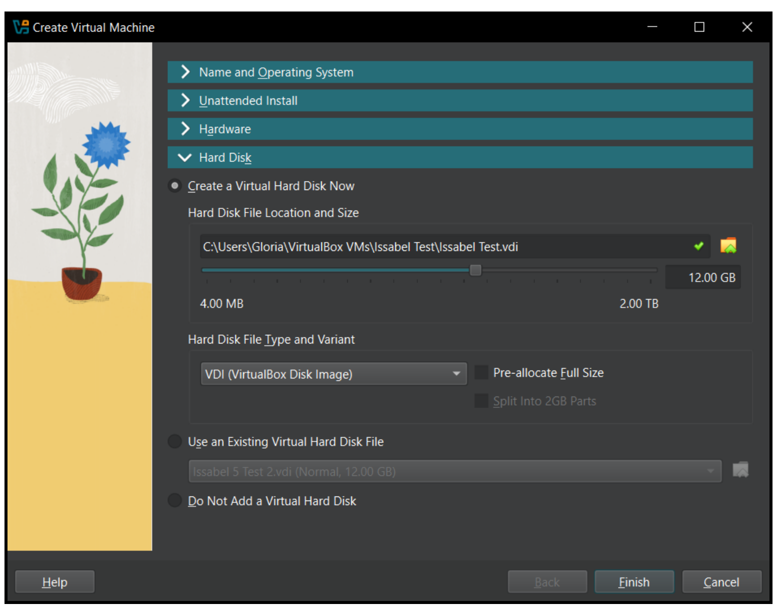

Figure 2.11: Create VM Hard Disk

9. Click **Finish**.

   Create Virtual Machine window closes.

To configure the network interface

1. Click **Settings**.

   Machine Settings displays.

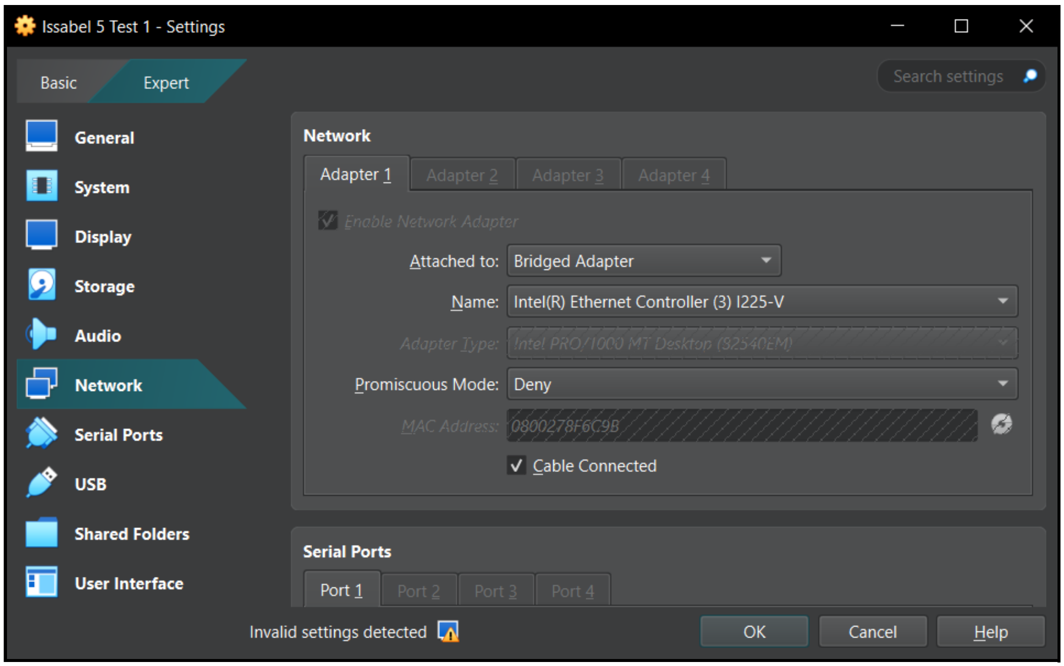

Figure 2.12: Virtual Machine Settings

2. Click **Network**.

3. Select **Bridged Adapter** from the Attached to dropdown menu.

4. Click **OK**.

To launch the Issabel PBX installer

1. Click **Start**.

   Starting virtual machine displays.

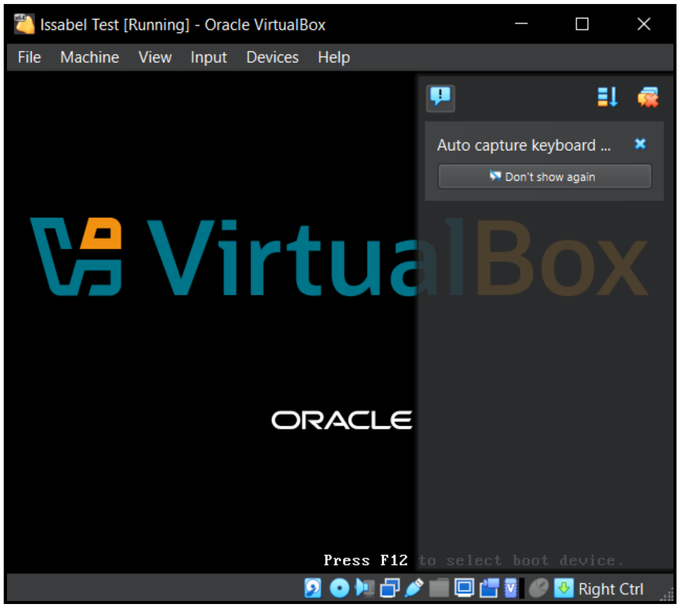

Figure 2.13: Virtual Machine Launch Screen

2. Press **Enter** on your keyboard to select Test this media & install.

   Server test begins. Issabel installer displays.

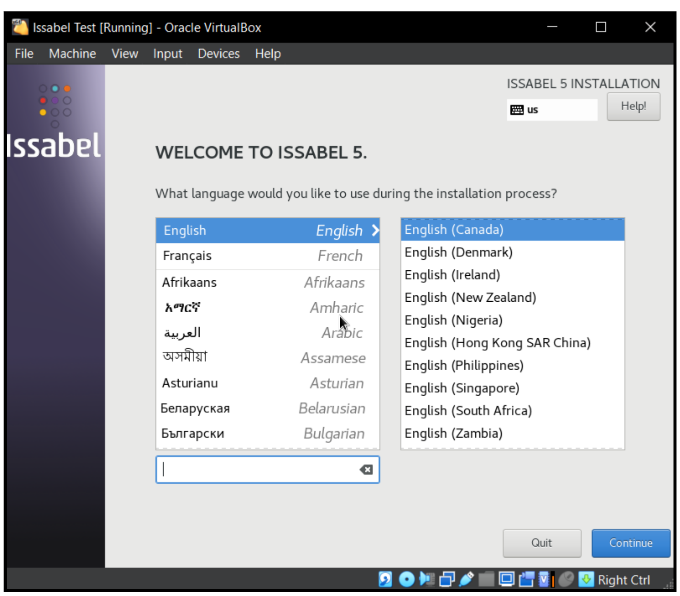

Figure 2.14: IssabelPBX Installer Language Selection

3. Select the language that you want to complete the installation in. Additional languages can be added later.

Preparing the Issabel installer

Now that you have created the virtual machine that Issabel will run from, you can install Issabel in the environment. Once you have launched the installer, you can customize settings for your location, storage, and network. You will also create a root password and administrator account that will be used to sign in to the Issabel web interface for the first time (*Figure 2.15*).

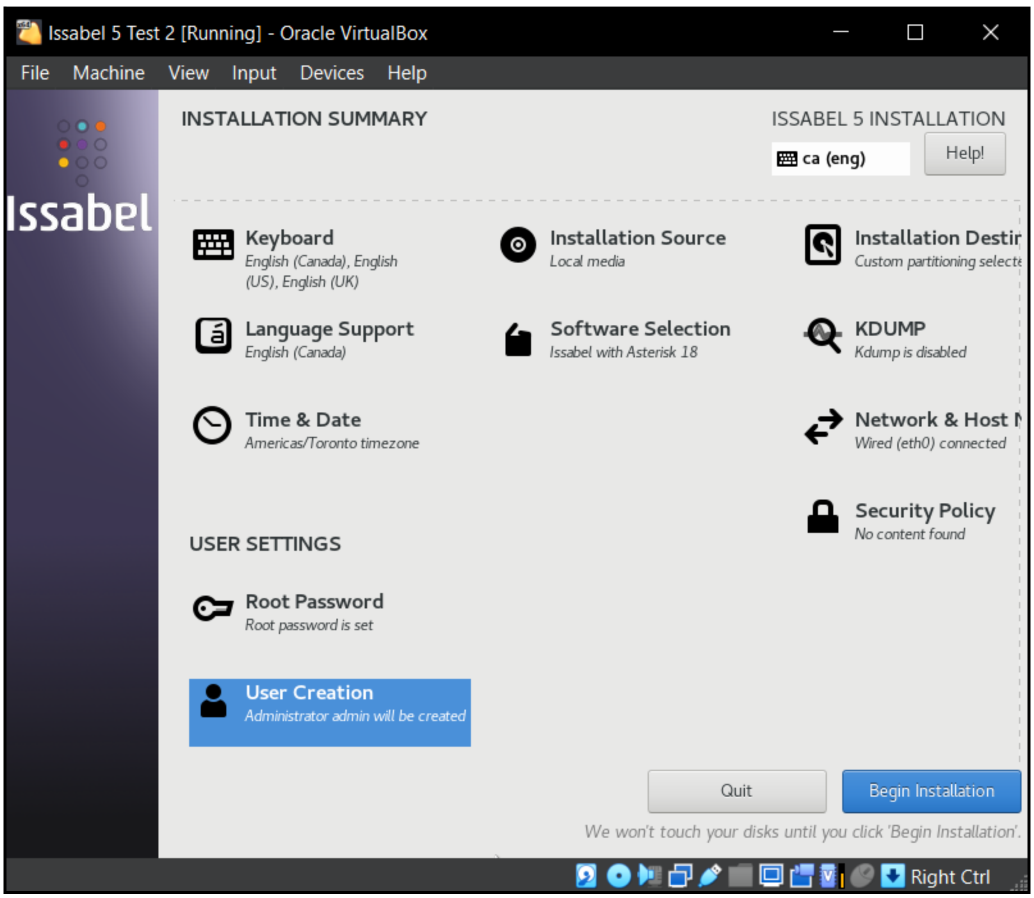

Figure 2.15: IssabelPBX Installer Summary Page

Choosing localization settings

The Localization section in the Installation Summary allows you to customize your keyboard layout, languages, and timezone. The keyboard and language support settings default to your choice from the language selection. Time & date are determined automatically from your network. These settings can be manually adjusted.

To set keyboard layouts

1. Click **Keyboard**.

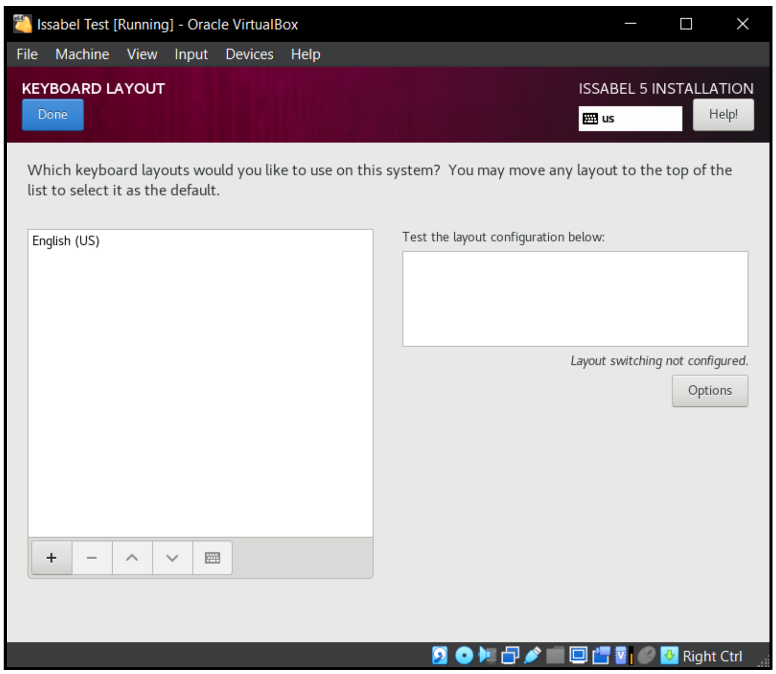

Figure 2.16: Issabel Installer Keyboard Layout

2. Click **+** to add a keyboard layout from the list.

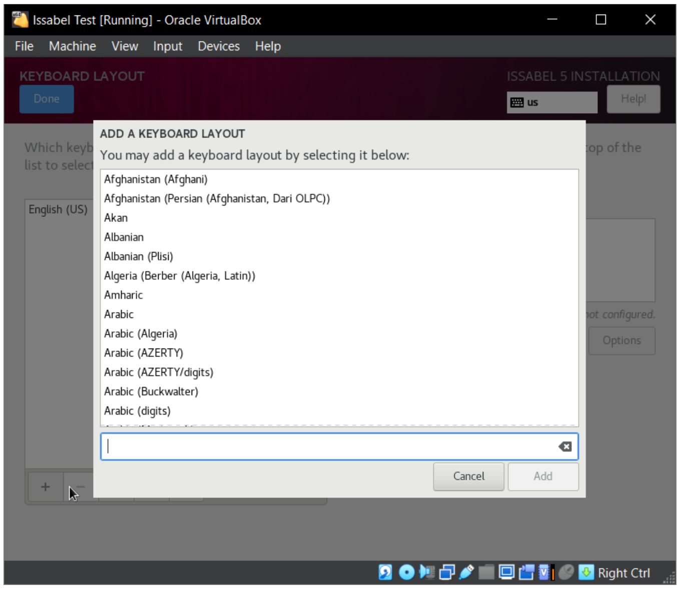

Figure 2.17: Add a Keyboard Layout

3. Click **Add**.

4. Click **Done**.

To add additional languages

1. Click **Language Support**.

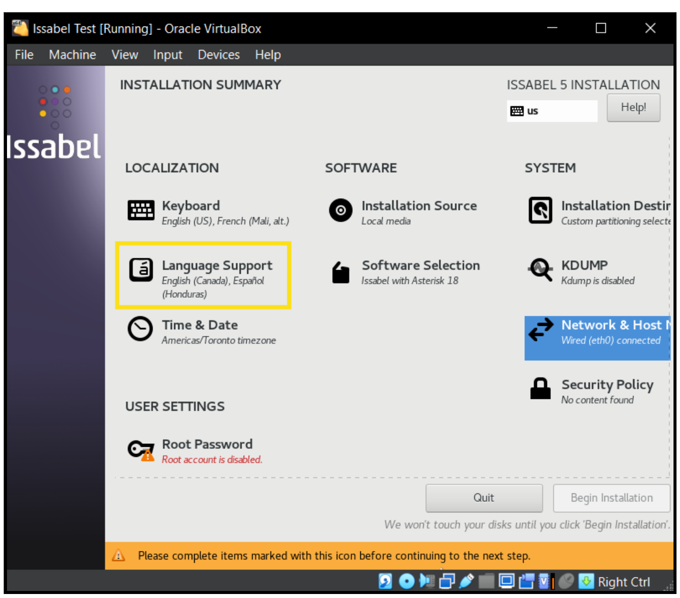

Figure 2.18: Language Support Settings

2. Search for your desired language.

3. Select your specific language variation.

4. Click **Done**.

To set a timezone manually

1. Click **Time & Date**.

2. Click the Network Time switch to manually select your region and time.

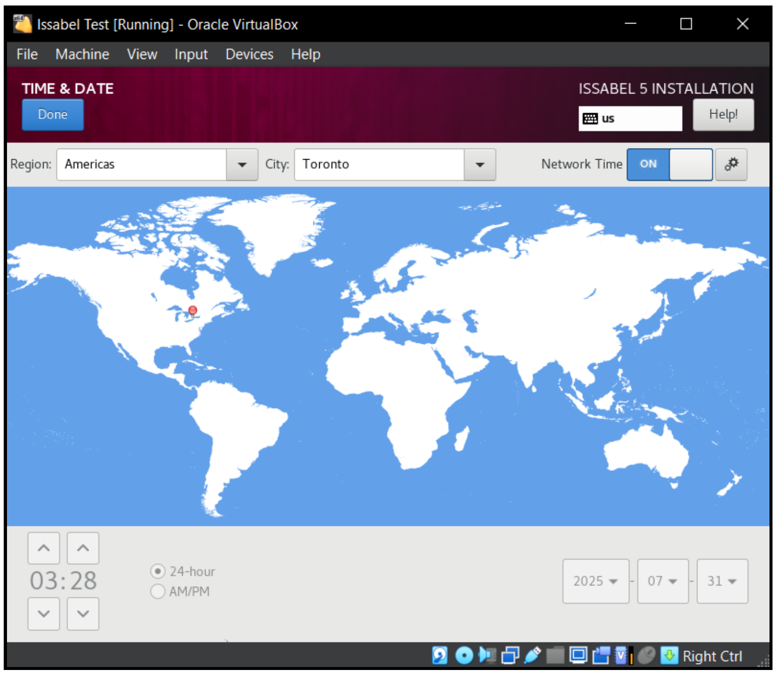

Figure 2.19: Time and Date Page

3. Select your Region and City from the dropdown list.

4. Click the arrows to adjust the time and choose between the 24-hour and AM/PM options.

5. Click **Done**.

Customizing system settings

To customize storage configuration

1. Click **Installation Destination**.

2. Click **Custom** under Storage Configuration.

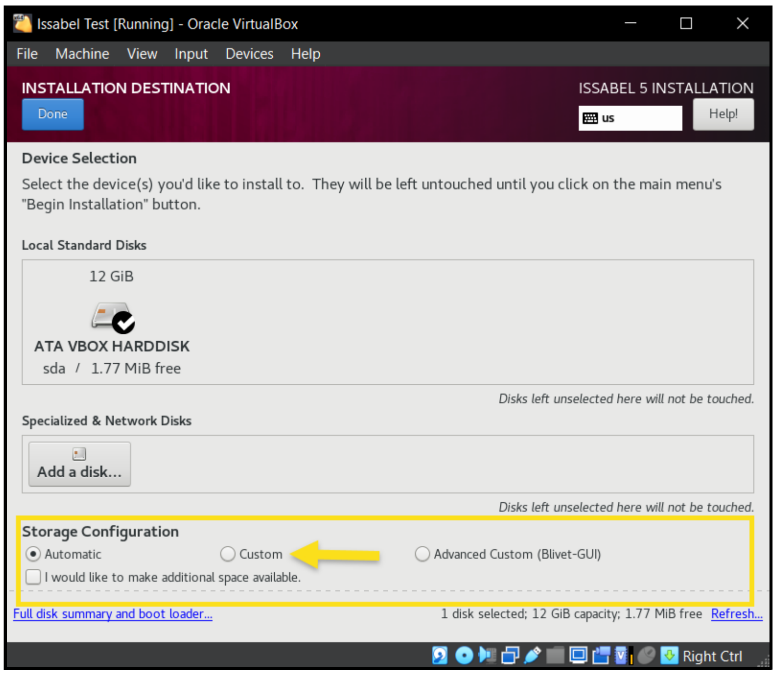

Figure 2.20: Storage Configuration within Installation Destination

3. Click **Done**.

   Manual Partitioning page displays.

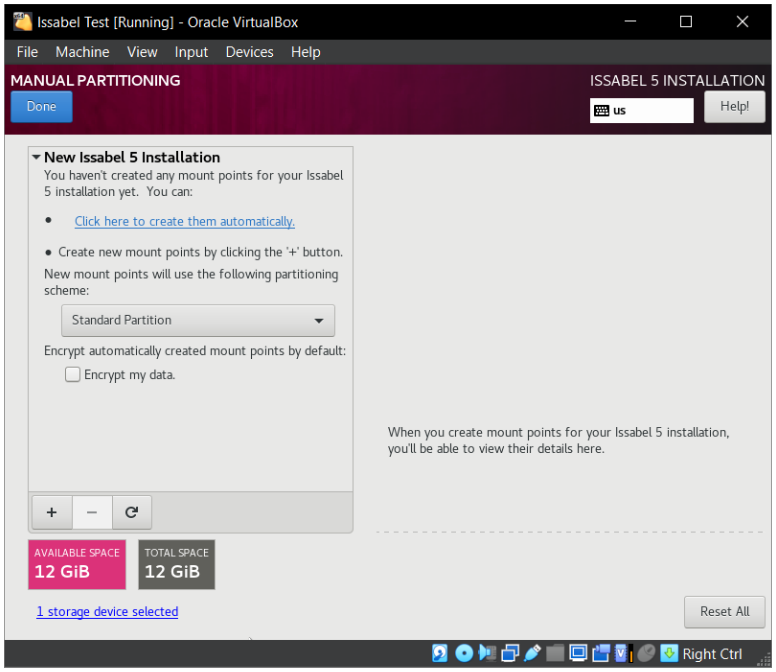

Figure 2.21: Manual Partitioning Page

4. Select **Click here to create them automatically**.

   Memory mount point displays.

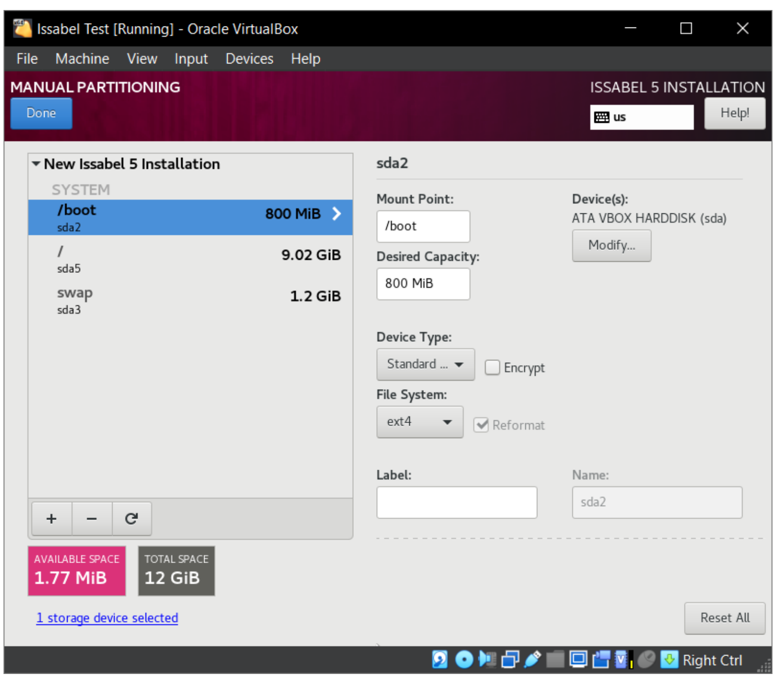

Figure 2.22: Memory Mount Point

5. Click **Done**.

   Summary of changes displays.

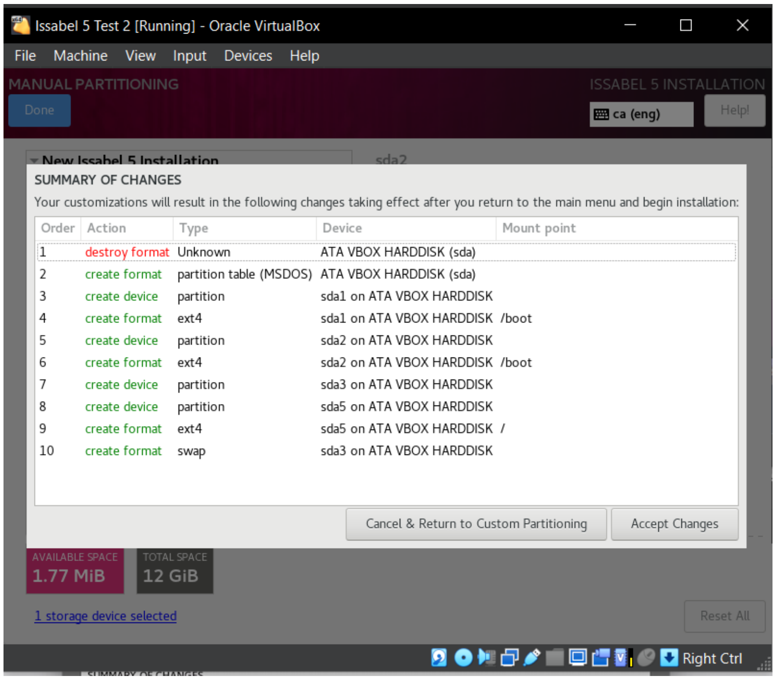

Figure 2.23: Summary of Changes

6. Click **Accept Changes**.

Note: The automatic partitions created by the installer ensures that your storage allocates enough space for your required files (for example, call recordings).

To change your host name

1. Click **Network & Host Name**.

2. Enter your desired Host Name and click **Apply**.

3. Click **Done**.

Customizing software selection

This section allows you to choose which version of Issabel and Asterisk you want to run your program with, as well as additional software (for example, Virtual Fax) that will be included in the graphical user interface (GUI).

1. Click **Software Selection**.
   Software Selection displays.

2. Choose a version of Issabel in Base Environment.

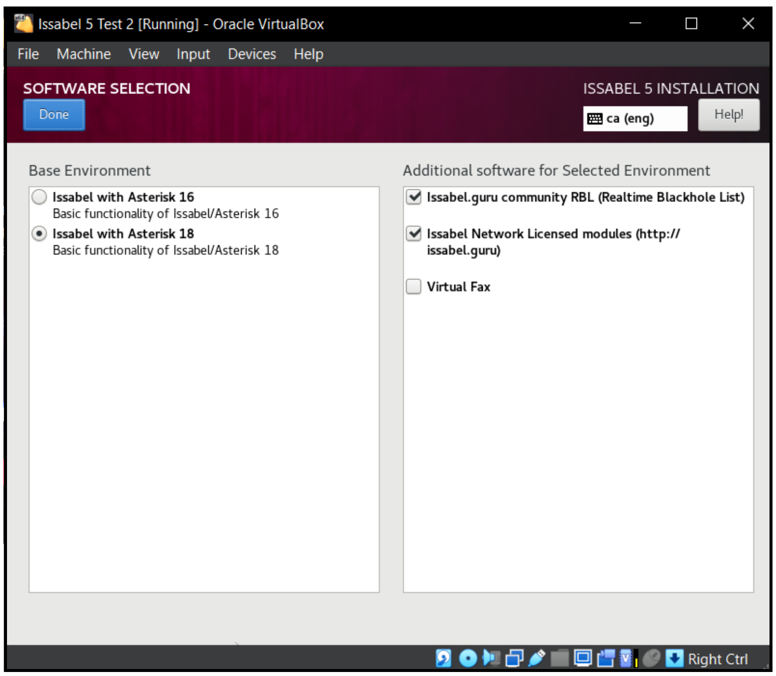

Figure 2.24: Software Selection

3. Select any additional software you want to be included.

4. Click **Done**.

Creating a root password and user account

Having an administrator account in addition to a root user is a recommended practice for recovery access.

To create a root password

1. Click **Root Password**.
   Root password window displays.

2. Enter a strong password.

3. Confirm the password.

4. Click **Done**.

> **Note**: A strong password should be secure but easy to remember.

To create an administrator user

1. Click **User Creation.**

2. Enter name in Full name.

3. Change username (optional).

4. Select **Make this user administrator**.

5. Select **Require a password to use this account**.

6. Create password.

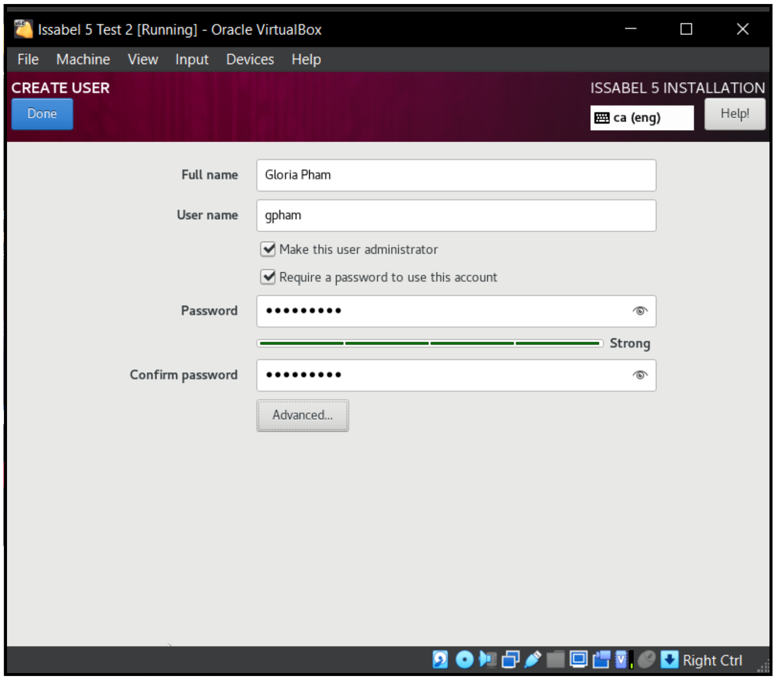

Figure 2.25: Create User

7. Click **Done**.

8. After you have finished configuring all settings, click **Begin Installation**.

Note: This process can take between 20 and 30 minutes. When the installation is completed, the system will automatically reboot.

Finalizing Issabel installation

In the final steps of the installation, Issabel asks you to create additional passwords for administrative security, set a language for the web UI, and to select a default SIP that will handle your calls.

Creating database and web passwords

After the system is rebooted, Issabel requires you to set up two passwords:

- Web administrator password: provides access to the web interface
- MariaDB root password: provides database access for Issabel internals

Issabel PBX uses an open-source database software called MariaDB to handle all of the data in Issabel, from extension settings to call recording files. The root password is meant to protect access to your database, and should be restricted to only top-level users.

To create a root password for MariaDB

1. Enter a password.

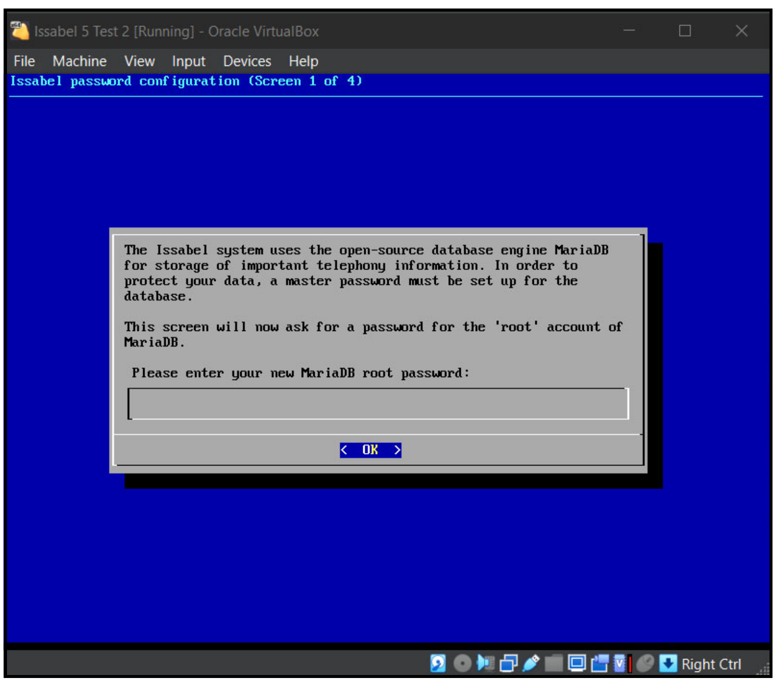

Figure 2.26: MariaDB Root Password

2. Click **OK**.

3. Re-enter the password to confirm.

4. Click **OK**.

   Message “The password for MariaDB and Cyrus admin were successfully changed!” displays.

You will need to set a password for an ‘admin’ account that will be used for Issabel Web Login and IssabelPBX. This is because some Issabel components have administrative interfaces that can be used through the web, so you have to prevent unauthorized access to these interfaces.

To set a password for IssabelPBX ‘admin’

1. Enter a password for “admin.”

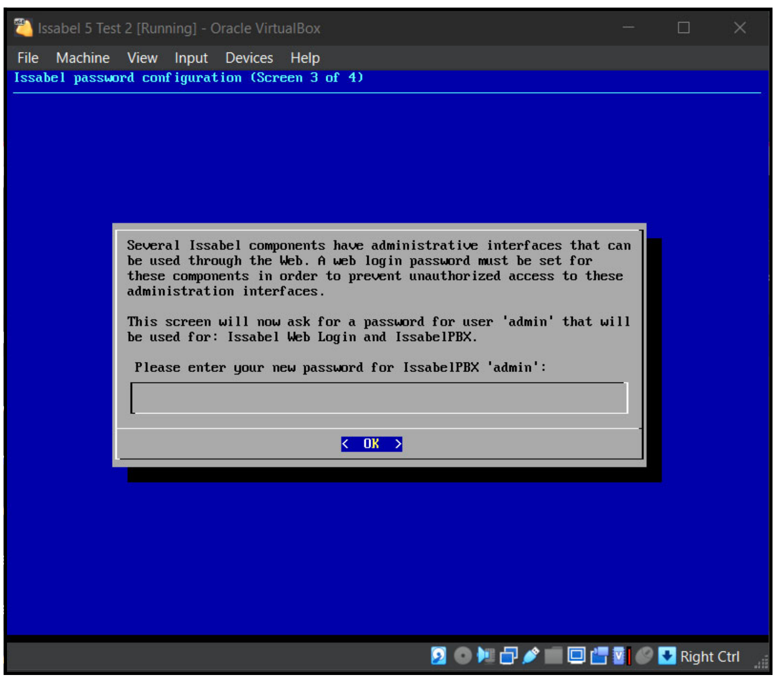

Figure 2.27: Admin Password Creation

2. Click **OK**.

3. Re-enter the password to confirm.

4. Click **OK**.
   Message "Updating IssabelPBX admin password" is displayed.

Table 2.1 shows all of the passwords that have been created.

Credentials	Created in	Used For
Root user/password	Issabel Installer Setup under User Settings	Launching Issabel system from console
Admin user/password	Issabel Installer Setup under User Settings	Recovery alternative to root user
Maria DB root user/ password	Reboot after Issabel is installed	Accessing database used for Issabel
Web admin user/ password	Reboot after Issabel is installed	Signing into Issabel's web UI

Table 2.1: Passwords Created During Issabel Setup

Configuring language and default SIP

You can change the language of the web interface later in Preferences. The default SIP refers to the particular protocol that will handle your voice communication. It is recommended that you select chan_pjsip, as it has improved features and security compared to the older chan_sip.

To select a language for the PBX

1. Use the arrow keys to select a language.

2. Click **OK**.

To select a default SIP for the PBX

1. Use the arrow keys to select **chan_pjsip**.

2. Click **OK**.

Using the console to launch web interface

After the initial installation, the Issabel console will launch automatically. In the future, you can launch the Issabel system from the Oracle VirtualBox Manager window by clicking your machine name and clicking Start.

To use the Issabel console

1. Enter `root` in the console. Press **Enter**.

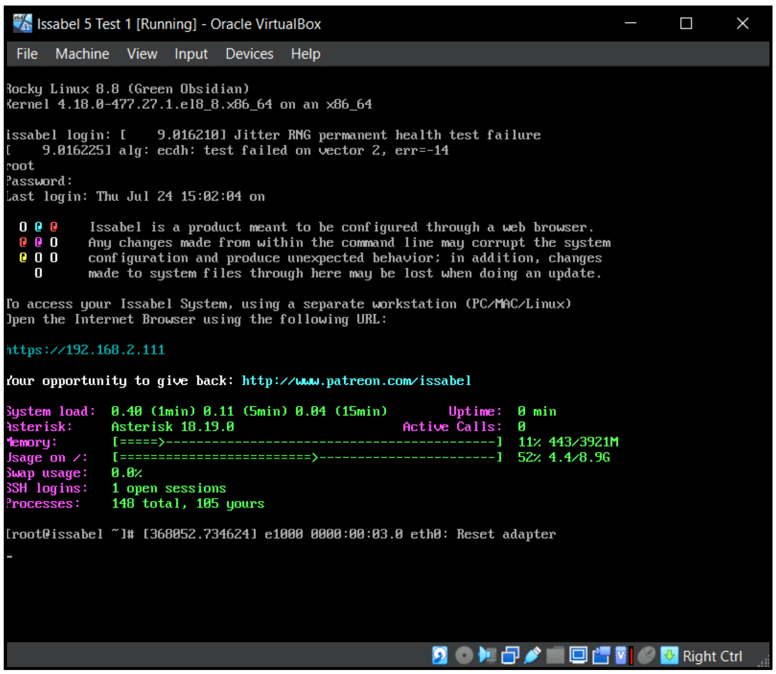

Figure 2.28: Issabel Console

2. Enter the root password. Press **Enter**.

3. Enter the URL from the console into a browser window.

   Issabel web login page displays.

Note: You may see a security risk warning in your browser when trying to load the URL. Click proceed anyway to continue.

Navigating the Issabel dashboard

After Issabel has been installed and set up in your virtual machine, you can access the user interface from the web using the URL provided in the console.

To sign into Issabel's web interface

1. Enter the URL from the console into a web browser.
   Issabel login page displays.

2. Sign in with the web administrator password that was created after the installation. (See table for reference). Click **Submit**.
   Issabel dashboard page displays.

Exploring the administration dashboard

When you are logged into the Issabel PBX web interface, the administration dashboard serves as your homepage to give you real-time monitoring of system status and processes (*Figure 2.29*). You can customize the dashboard to display specific information with 'Applets.' By default, your dashboard will display your System Resources, Processes Status, Hard Drives, Performance Graphic, News, and Issabel Network. You can customize this from the Dashboard Applet Admin module.

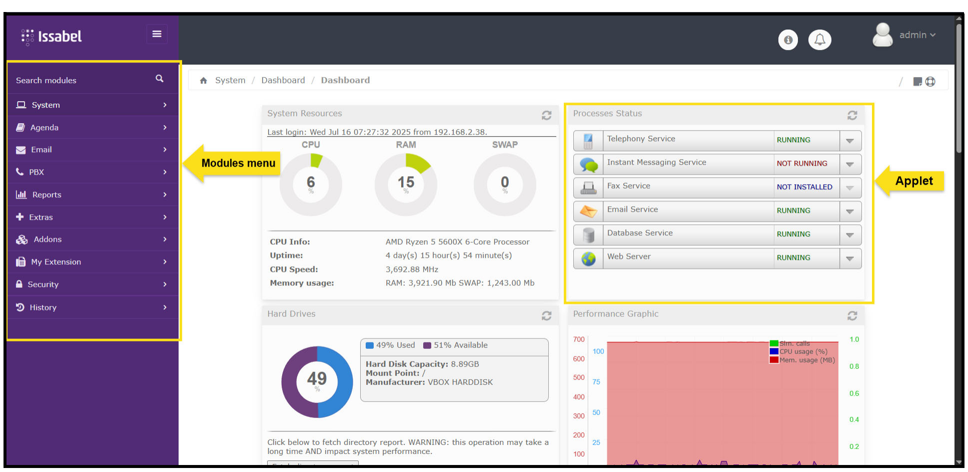

Figure 2.29: Issabel Dashboard

Menu layout and modules

The Issabel navigation menu contains all of the modules Issabel offers to set up and manage your system. Each main module contains additional features that you can configure. *Table 2.2* shows the modules included:

Module Name	Features
System	Administrative configuration tools for network, backups, users, and preferences.
Agenda	Calendar and address book to use as internal directory and planner.
Email	Accounts, domains, email list, stats, vacation messages can be added and modified.
PBX	PBX, voicemail, conference, call recordings can be configured and viewed.
Reports	Call detail records (CDR), billing, summary of channel usage, calls, and statistics.
Extras	Issabel Meet video conference module.
Addons	Additional modules that can be installed to Issabel UI.
My Extension	Settings to configure user extension.
Security	Firewall, Two factor authentication, Fail2Ban, HTTPS certificate to configure security and view activity.
History	Recent modules visited.

Table 2.2: Issabel Modules Menu

Chapter 3: Connecting to External Services

This chapter focuses on connecting your PBX system to the public network. You will learn about trunk types, service providers, route types, and dial patterns. By the end of this chapter, you will understand how SIP trunks enable outside calling, how to choose a service provider, and how to customize routes for your incoming and outgoing calls.

This chapter at a glance

Now that Tom has access to the Issabel interface, he needs to connect to the public telephone network so that he can make and receive calls. He chooses two service providers, Twilio and VoIP.ms, who will let him use their communication channels to make outbound calls and receive inbound calls. He signs up for both services and sets up trunks in Issabel using the SIP credentials provided by each vendor.

Tom creates two outbound routes in Issabel:

Local Outbound Route: Uses a VoIP.ms trunk for local numbers.

International Outbound Route: Uses a Twilio trunk for international numbers due to Twilio’s lower global rates.

Tom then creates two inbound routes in Issabel. He requested two phone numbers from VoIP.ms so that he could dedicate one two his development services, and the other to his travel services.

+1-555-123-4567 routes to his development team ring group (ext. 201-205)

+1-987-654-3210 routes to an IVR for his travel services, letting callers press 1 to speak to an agent, or 2 to leave a voicemail.

This setup allows Tom to manage outbound and inbound calls efficiently and affordably. His Issabel PBX is now fully connected to the public telephone network.

Understanding trunks

A trunk is a communication line that allows many calls to travel between two systems. Trunks are essential for conducting calls and serve as the backbone of call routing and connectivity.

In Issabel, a trunk connects the PBX system to an external network such as the Public Switched Telephone Network (PSTN), VoIP providers, or another PBX system. SIP trunks are virtual phone lines that allow businesses to communicate over the Internet instead of using traditional, physical phone lines. This guide will focus on configuring SIP trunks, as they are the most common type used in modern VoIP systems.

Differentiating SIP and trunks

While SIP trunks are a type of trunk, it can be helpful to distinguish between the two individually.

- SIP is a set of rules (protocol) that is used to start, manage, and end voice calls over the Internet.
- A trunk is a connection/channel that allows calls to travel between systems.

You can think about trunks like a highway and the SIP as the rules of the road.

Types of trunks in Issabel PBX

Issabel supports several types of trunks for different communication methods shown in *Table 3.1*:

Trunk Type	Uses	Notes
SIP Trunks	Most common trunk type used in VoIP systems	Highly scalable and cost-effective
IAX Trunks	SIP alternative, used in Asterisk based systems Ideal for connecting two Asterisk-based PBX systems	Efficient for navigating through routers and firewalls (NAT) can send many calls together in one data stream instead of separately
DAHDI/ Analog Trunks	Commonly used with traditional phone lines (PSTN or ISSDN) Uses physical analog lines or Digital Access Hybrid Device Interface (DAHDI) for connectivity	Requires compatible hardware cards to be installed on PBX server
Custom Trunks	For advanced/non-standard setups	Custom configuration allows integration with specialized systems or protocols

Table 3.1: Supported Trunk Types in Issabel

Enabling outside calling with SIP trunks

SIP trunks connect your PBX to the public phone network (PSTN) through an Internet Telephony Service Provider (ITSP). Calls can then travel between your PBX to the SIP provider, and then worldwide as Internet data (VoIP). Without SIP trunks, you're limited to internal calls only.

You can configure a SIP trunk in the PBX Configuration section under the PBX module of Issabel. *Table 3.2* explains the two types of outside calling.

Call Type	Direction	Pathway
Inbound	From the public to your office	Travel from SIP provider's server to Issabel PBX
Outbound	From your office to the public	Leave from Issabel through the SIP trunk to SIP provider who delivers via PSTN

Table 3.2: Outside Call Types

Understanding the role of ITSP and PSTN

Internet Telephony Service Providers (ITSP), such as SIP providers, are the bridge between your PBX and the Public Switched Telephone Network (PSTN). The PSTN uses traditional phone technology to connect calls which is different from VoIP systems. ITSPs are licensed carriers with the necessary legal, regulatory, and technical arrangements to convert between the packet-based protocols of VoIP systems and the circuit-switched protocols of the PSTN.

Using routing and gateways

Routing is the pathway or directions that calls take to arrive at the right destination.

- Outbound routes tell Issabel which trunk (i.e., provider) to use for which type of outgoing call (for example, sales calls or follow-up calls).
- Inbound routes dictate what happens when an external call arrives such as which extension or voicemail should answer.

A gateway is a hardware device or software that acts as a conversion point where communications enter or leave the PBX, typically translating between analog and digital technologies and protocols.

Usually, IP PBX systems like Issabel do not need a physical gateway, as the SIP trunk acts as the bridge between your PBX and the PSTN. However, if you want to connect analog phones or fax machines, you will need a gateway such as a DAHDI card.

You can configure routing and gateway options under PBX Configuration > Trunks and Outbound/Inbound Routes.

Porting existing numbers

Number portability is a major advantage of SIP trunking. It allows you to keep your existing phone numbers when switching to Issabel so there is no need to change your business number when you upgrade your system.

A few things to note about porting numbers:

- You can request your new provider to port/transfer your numbers from the old provider or landline service.
- Not all numbers are portable in all regions, please check with your ITSP.
- During porting, service may be interrupted for a short period of time.
- Ported numbers can be assigned to extensions, queues, or ring group.

Selecting a SIP trunk provider

Choosing the right SIP trunk provider is essential for ensuring call quality, security, scalability, and compatibility with Issabel PBX (*Table 3.3*).

Consideration	What to look for
Issabel compatibility	Supports Asterisk-based systems or Issabel directly
Call quality and reliability	99.99% uptime, low jitter, low latency, nearby data centers
Pricing and billing	Transparent rates, no hidden fees, scalable channel pricing
Concurrent call capacity	Ability to increase/decrease SIP channels as needed
International support	Availability of global numbers and international call support
Number porting	Easy process to move existing phone numbers
Security features	SIP over TLS, SRTP, fraud detection, IP whitelisting
Support and documentation	24/7 tech support, clear Issabel setup guidelines
Extra services	SMS, fax support, caller ID, etc.

Table 3.3: SIP Trunk Provider Checklist

Setting up an account with a SIP provider

Once you've selected a compatible provider, you will need an active account before configuring SIP trunking in Issabel. This will allow you to establish your business telephone numbers and access your SIP credentials.

To set up an account

1. Choose a SIP trunk plan (*Table 3.4*).

Considerations	Options
Number of SIP channels	Number of concurrent calls possible
Payment plan	Flat-rate or metered (pay-as-you go)
Calling features	Regional availability, direct inward dialing (DID), international calling

Table 3.4: SIP Trunk Plan Considerations

2. Register for an account. Be prepared to provide:

    • Business name and contact information

    • Billing details

    • Account administrator email

    • Business ID or verification

3. Choose new telephone numbers or port existing ones.

4. Access SIP credentials (*Table 3.5*) or trunk settings.

Note: These credentials will be used in Issabel's trunk configuration.

    a. Sign in to your provider's web portal.

    b. Document your:

SIP Credentials
SIP username/ password
SIP server address or IP

Table 3.5: SIP Credentials

SIP Credentials
Port number (usually 5060 or 5061)
Outbound/inbound DID settings

Table 3.5: SIP Credentials

Creating outbound routes

Outbound routes determine how your calls are directed from your internal system to the public network. Setting up different outbound routes is helpful for distinguishing how different calls should be handled. When you create an outbound route, you choose which trunk or connection to use. You might have a different outbound route for local numbers, long distance, or emergency services.

To create an outbound route

1. Click **PBX > PBX Configuration** in the modules menu.
   IssabelPBX System Status page displays.

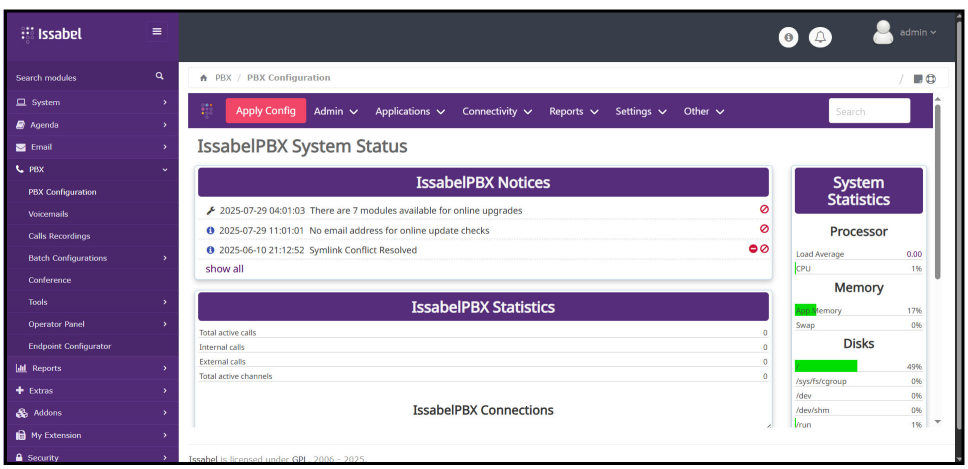

Figure 3.1: PBX System Status

2. Click **Connectivity > Outbound Routes**.

   Create Outbound Routes page displays.

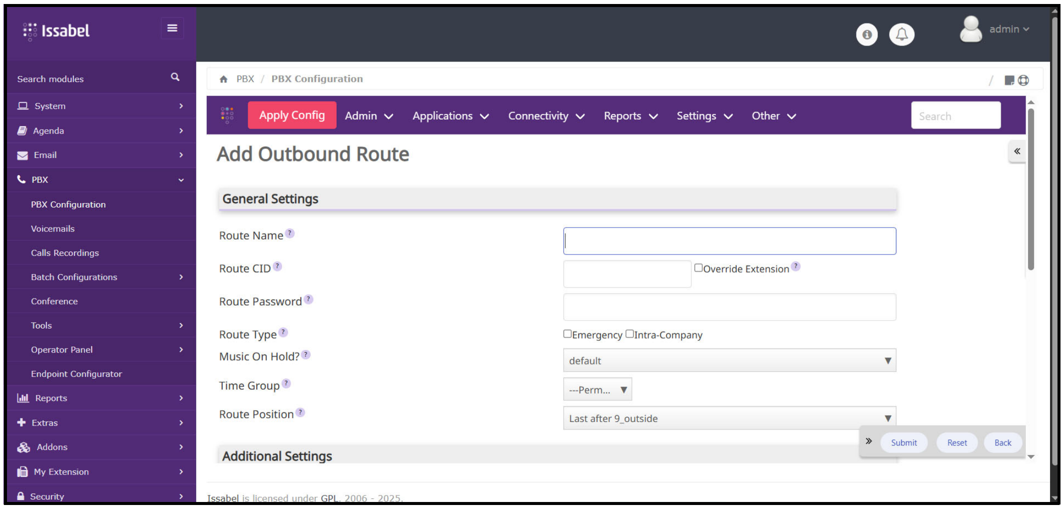

Figure 3.2: Outbound Routes Page

Defining dial patterns and rules

Dial patterns and rules are the components that outbound routes use to determine what numbers can be dialed, how they are formatted or processed, and where they travel to. Dial patterns act as filters so that only valid numbers can be dialed out, while rules route calls to specific trunks that serve different purposes (for example, using one for local calls and another for international calls). This helps businesses manage the calls permitted, lower costs, and ensure quality.

Dial pattern syntax

Pattern syntax defines the format that the PBX uses to match and process dialed numbers. The syntax acts like a code to filter accepted numbers and formats before they leave the PBX. This is illustrated in *Table 3.6*.

Symbol	Meaning	Example	Result
X	Any digit (0-9)	NXXNXXXXXX	4162121234
Z	Any digit **except 0** (1-9)	ZXXX	1000, 2999 (not 0000)
N	Any digit **except 0 or 1** (2-9)	NXX	200, 555 (not 011, 100)
[]	Match a character from set	[9, 1, 2, 5]	91235, 91245
.	Matches any trailing digits	9.	911, 912345, 90000000
!	No further digits, exact match	911!	911 only

Table 3.6: Pattern Syntax Symbols

Example: Local calls for North American numbers will often use the pattern **NXXNXXXXXX** that follows a 10 digit phone number that excludes invalid area codes

Stripping and prepending numbers

Strip and prepend are functions that can be set in your PBX's outbound route configuration.

- **Strip**: removes digits from the beginning of the dialed number
  - Example: if users must dial '9' to make outside calls, your syntax would be 9|XXXXXX, and Strip = 1 so dialing 9123456 becomes 123456
- **Prepend**: adds digits to the beginning of a number
  - Example: if providers require a country code, prepend can be set to '+1' (for Canadian numbers) to convert 4162345678 to +14162345678

Choosing a trunk sequence

A trunk sequence refers to the priority list of trunks you want to use to conduct your call for the corresponding dial pattern. For example, for dial patterns that match long distance numbers, you might set the cheapest channel to carry long distance calls (for example, VoIP trunks). Additional trunks can be added to the list if the primary trunk is unusable.

The two most common dialing rules for North American numbers are the 10/11 Digit Dialing Rule and the 911/922 Dialing Rule. You can set your own custom rules to suit your business needs.

To create a basic 10/11 digit dialing rule

1. Enter the following to their corresponding fields in General Settings from *Table 3.7*:

Field Name	Field Value
Route Name	10_and_11_Digit_Dialing_Rule
Route CID	Caller ID to display
match pattern	NXXNXXXXXX

Table 3.7: 10/11 digit dialing rule field values

2. Click **+ Add More Dial Patterns**.

3. Enter **`1NXXNXXXXXX`** in the second match pattern field.

4. Select your preferred default SIP trunk in Trunk Sequence for Matched Routes (highest priority starts at 0).

5. Click **Submit**.

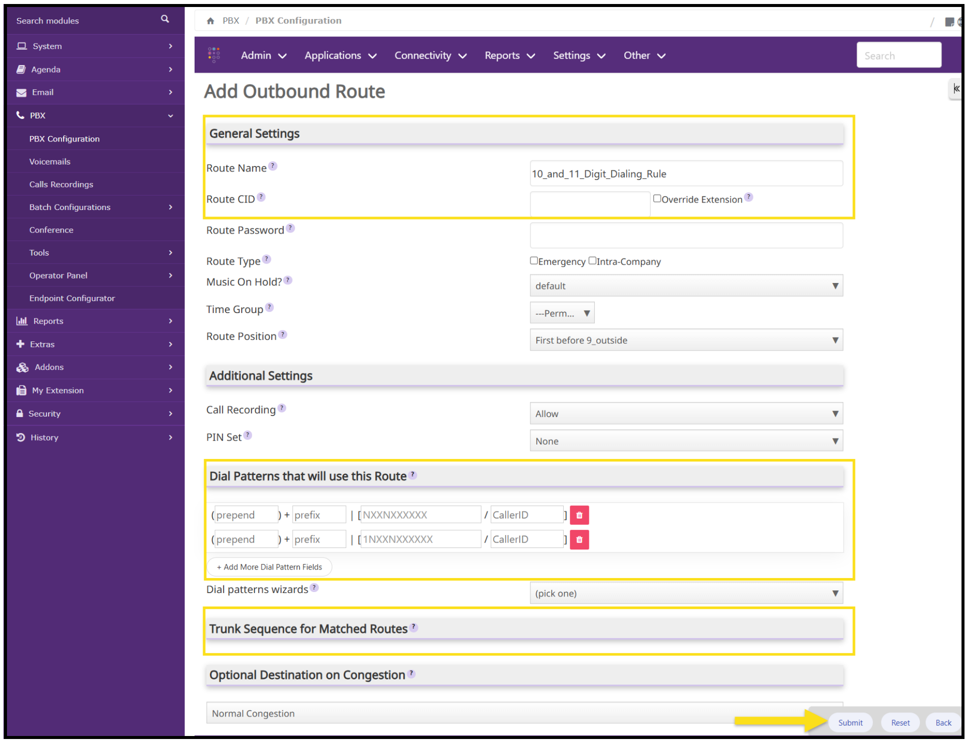

Figure 3.3: Outbound Routes Fields

To create a basic 911/922 dialing rule

1. Enter the following to their corresponding fields in General Settings:

Field Name	Field Value
Route Name	Emergency
Route CID	Caller ID to display
match pattern	911

Table 3.8: 911/922 dialing rule field values

2. Click **+ Add More Dial Pattern**.

3. Enter **922** in the second match pattern field.

4. Select your preferred default SIP trunk in Trunk Sequence for Matched Routes (highest priority starts at 0).

5. Click **Submit**.

6. Make sure to submit your changes after each rule. Once all the rules are built, click **Apply Config**.

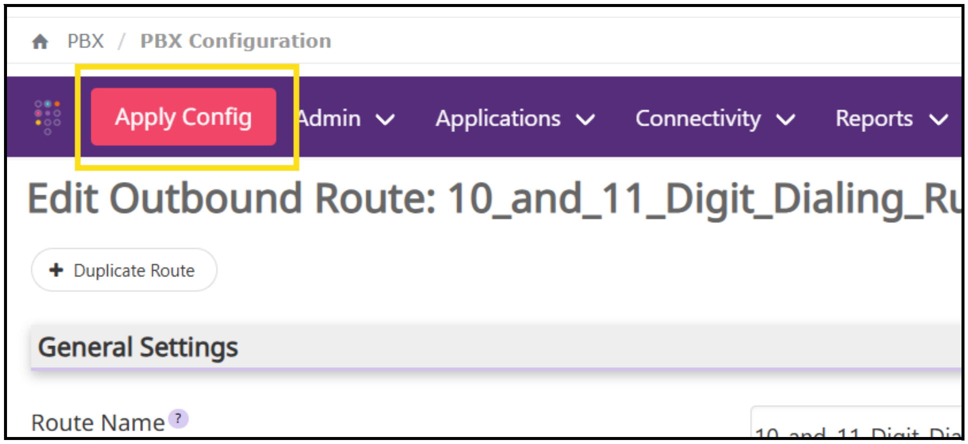

Figure 3.4: Apply Outbound Route

Creating inbound routes

Inbound routes control how incoming calls are directed using your defined rules. Establishing inbound routes delivers different types of calls to their appropriate destination efficiently. Inbound routes typically match the number used to reach the business (DID) and the caller’s ID to determine where to direct the call to.

For example, your business might have a list of phone numbers associated with their most important customers as well as a toll-free number that answers calls during business hours and redirects them to voicemail after hours. If someone on the list of important customers called the toll-free number after hours, you might have an inbound route set up that directs the call to an emergency contact, whereas any other number would be sent to voicemail.

To create an inbound route

1. Click **PBX > PBX Configuration** in the module menu.
   IssabelPBX System Status displays.

2. Click **Connectivity > Inbound Routes**.
   Add Incoming Route page displays.

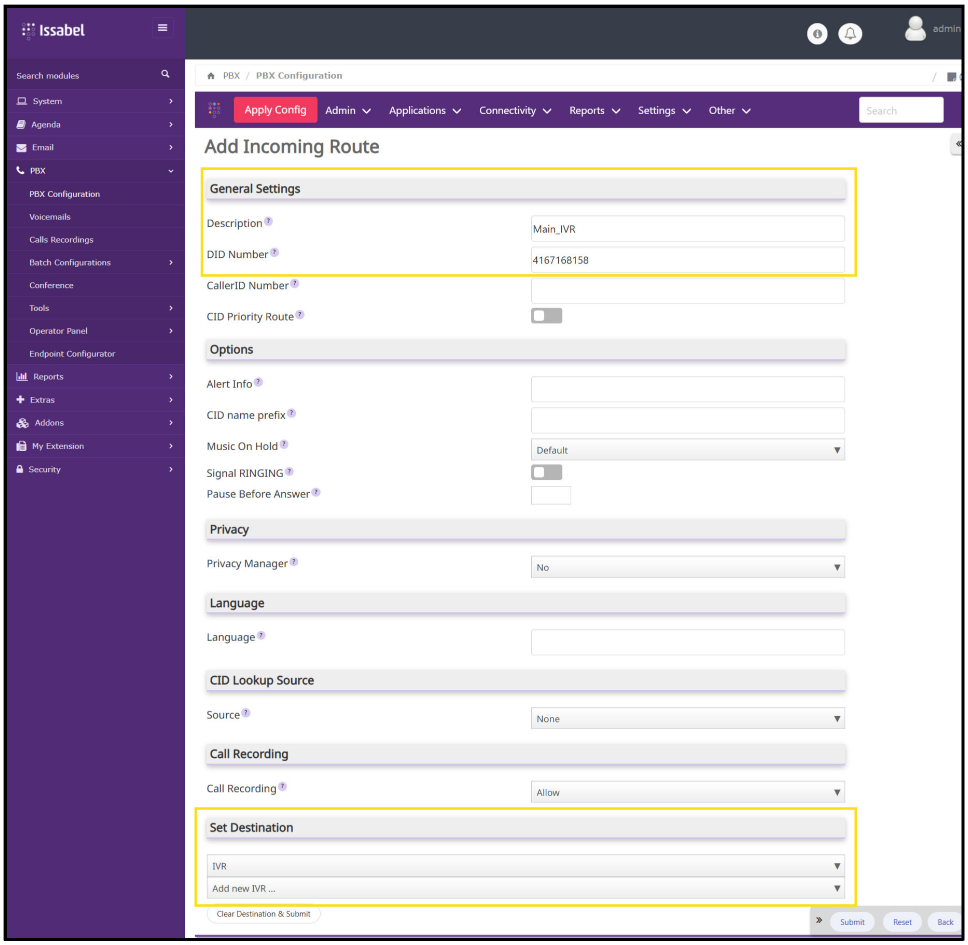

Figure 3.5: Inbound Route Fields

3. Enter the following in the corresponding field under General Settings:

Field Name	Field Value
Description	Rule Name (for example, Main_IVR)
DID number	ITSP provided number

Table 3.9: Inbound route rule field values

4. Select the destination of the call from the Set Destination dropdown menu.

5. Click **Submit**.

6. Remember to submit changes after each rule. After adding all rules, click **Apply config**.

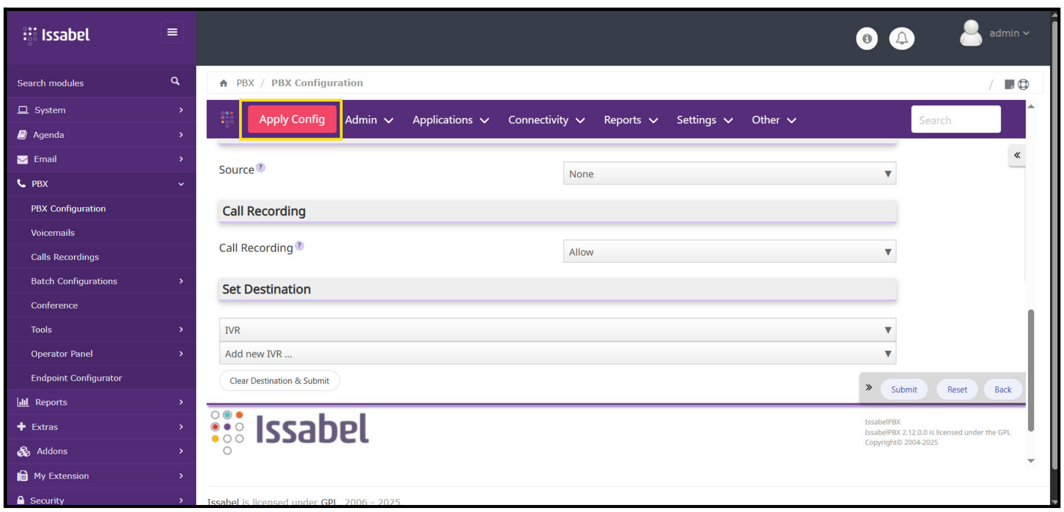

Figure 3.6: Apply Inbound Route

Assigning DID numbers

You can request multiple Direct Inward Dialing (DID) numbers from your SIP trunk provider or ITSP. A DID is a telephone number that the public can call to reach someone within your PBX system, such as your main line, department, or specific user.

- It can be a local number or a toll-free number
- A business may have one DID for the main office and separate DIDs for individual employees or departments, instead of extensions

Formatting and using DIDs

Table 3.10 illustrates the different DIDs that can be assigned for different inbound routes.

Purpose of Inbound Route	DID Entry
Rules for a specific DID number	Enter expected DID Number
General, for all calls	Leave blank
For a range of DIDs	Enter dial pattern to match range

Table 3.10: Using DIDs in Inbound Routes

Routing calls to modules

You can choose to direct your incoming calls to a wide range of modules when you create your inbound routes. These modules help facilitate efficient and professional call-handling workflows.

Later, we will explore how to create and customize modules like IVR menus and voicemail, *See "Creating an IVR menu" on page 76*.

Chapter 4: Configuring Extensions and Devices

This chapter explains how extensions and devices can be added and used with IssabelPBX. By the end of this chapter, you will know how to add extensions, connect them to softphones and deskphones, and make test calls.

This chapter at a glance

With his phone lines set up, Tom can now create extensions for his department and staff and connect them to his phones. In Issabel, he adds internal extensions such as 100 for himself, 101 for Timmy in Development, 102 for Tommy in Support, and 103 for Gulliver in Travel. Each extension is linked to a deskphone or softphone, depending on the setup of the employee.

Tom shares each user’s SIP credentials so they can register their devices and start making and receiving calls through the system. He also organizes the extensions to match departmental needs, such as enabling call waiting for Gulliver, who often receives a high volume of calls.

Using extensions and phones, Tom can now handle a variety of calls at once, and allow callers to directly reach their intended recipient efficiently.

Using extensions for efficiency

Extensions are an integral part of internal telephone systems. They allow organizations to direct callers to their intended recipient efficiently and organize communication directories within the company. An extension is usually a three or four digit number that acts as an identifier within a telephone network.

Issabel PBX allows you to create extensions and connect them to devices. Extensions can be assigned to individual users or departments with custom permission and security options. Extensions can support standard features such as voicemail, call forwarding, and call recording. Because of their identifiable nature, extensions can be used to filter call detail records, activity, and monitor performance.

Each extension requires a unique set of SIP credentials to function. These credentials authenticate the user or device to the PBX system and must be carefully managed.

Adding, editing, and deleting extensions

You can add, edit, and delete extensions from the Extensions page. You can access this page from the module menu by going to PBX > PBX Configurations > Applications > Extensions.

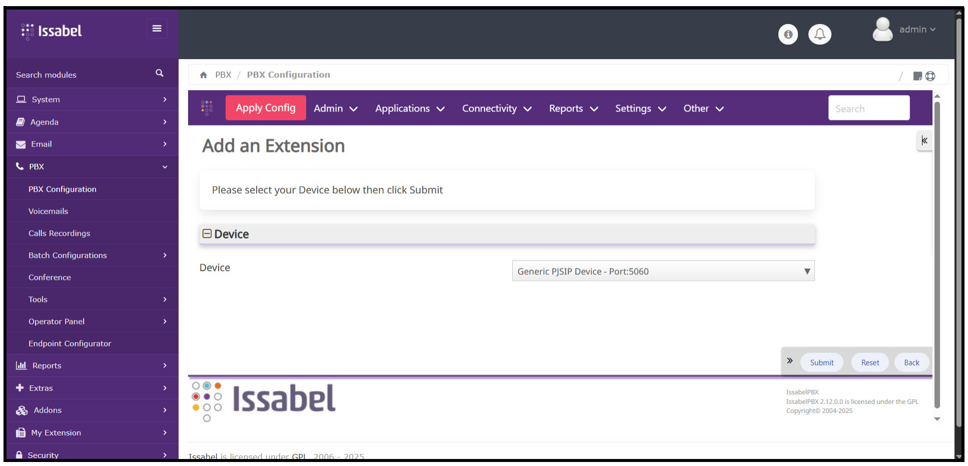

Figure 4.1: Add an Extension Page

To add an extension

1. Select **Generic SIP Device - Port:5066** from the dropdown menu.

2. Enter the following in the corresponding fields from *Table 4.1*:

Field Name	Field Value
User Extension	Extension number (for example, 1001)
Display Name	User’s Name (for example, Leslie Knope)
Outbound CID	Caller ID for this user
Emergency CID	Emergency caller ID for user

Table 4.1: Adding extension field values

3. Click **Submit**.

4. Submit changes after each extension. Once all extensions are added, click **Apply config**.

To edit an extension

1. Click the extensions tab.

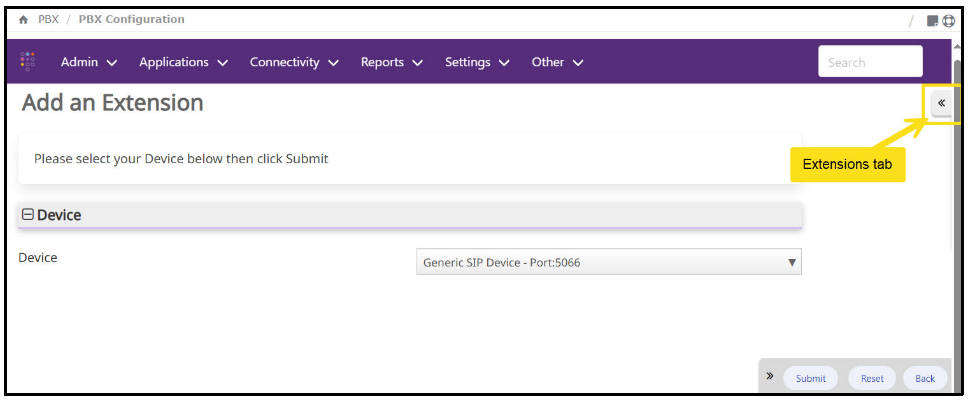

Figure 4.2: Extensions Tab

2. Select extension.
   Edit extension page displays.

3. Modify desired values.

4. Click **Submit**.

5. Click **Apply config** once all edits have been made.

To delete an extension

1. Click the extensions tab.

2. Select extension.
   Edit extension page displays.

3. Click **Delete**.

   Delete confirmation pop-up displays.

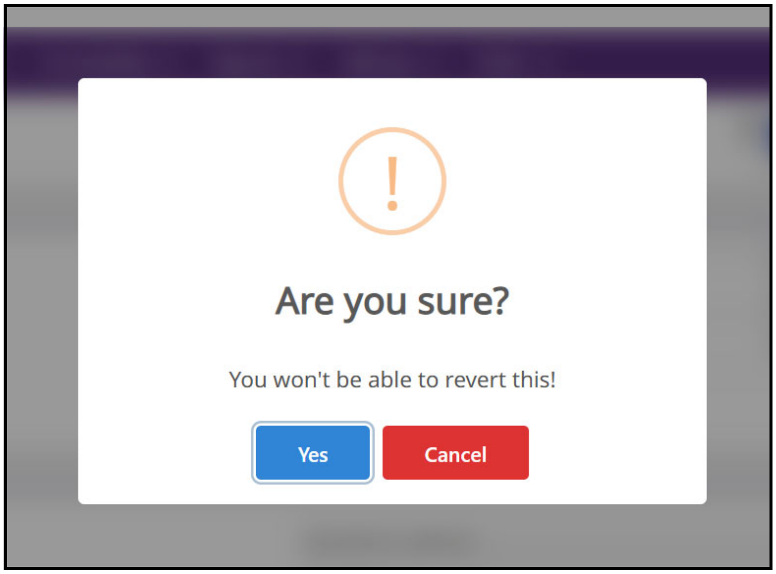

Figure 4.3: Delete Extension Confirmation

4. Click **Yes**.

   Extension deletes.

5. Click **Apply config** once all changes have been made.

What are SIP credentials?

Session Initiation Protocol (SIP) credentials can be thought of as the keys needed for a user and device to make phone calls through a network.

SIP credentials provide user with access to the communication line shared between your PBX, your SIP provider, and the public network. This is different from the system login credentials that users need to access the internal Issabel PBX ecosystem.

Authenticating your extension

Each extension acts as a username that is paired with a password (secret) that is created when the extension is registered. When you sign in for the first time with that device, you must manually enter the password to authenticate it.

After that, your device automatically re-connects with the server and authenticates itself using the password/secret that was associated with it. The only times that SIP credentials need to be re-entered is:

- During a password reset
- Factory reset of the phone/device
- Security issue (for example, account compromise)

Connecting softphones to Issabel

Softphones are an increasingly popular choice for businesses to conduct voice communication. Their software platform allows you to make calls over the Internet without requiring physical hardware. This streamlines the connection process between the phone and your PBX system.

Choosing a softphone application

Here is a table summarizing the considerations in choosing a softphone (*Table 4.2*).

Consideration	What to look for
PBX Compatibility	SIP support, works with Issabel
Platform support	Windows, macOS, Linux, Android, or web-based
Core features	Call hold/transfer, voicemail, call recording, multiple accounts

Table 4.2: Considerations for Softphones

Consideration	What to look for
Audio quality	HD codecs (for example, G.722), echo cancellation, noise reduction
Security	TLS/SRTP encryption, password protection, push notifications for mobile
Ease of use	Simple configuration, localized language support, clean UI
Integration	Contacts, calendars, CRM (if needed)
Network compatibility	Firewall-friendly settings
Cost and licensing	Free, premium, or paid based on features
Support and updates	Active development, vendor support, regular updates

Table 4.2: (Continued)Considerations for Softphones

Registering the softphone with SIP credentials

Once you've created an extension and have chosen a softphone, you can connect your phone to the PBX system and give it the assigned extension. You will need the extension number and secret that you created when adding the extension. You will also need the IP address or server domain that Issabel is running from.

To register and authenticate your device

1. Open your softphone application.
2. Click **Connect device**.
   SIP credentials page displays.
3. Enter the following in the corresponding field:
   - **SIP Username**: extension number from PBX
   - **SIP Password**: secret from Issabel
   - **SIP Server**: IP address or domain of Issabel server
4. Click **Connect**.

Troubleshooting registration issues

Sometimes there can be issues with registering the softphone to the ITSP. *See “Troubleshooting common setup issues” on page 95* for common troubleshooting topics and resources.

Connecting deskphones and analog devices

Using physical phones allows you to make and receive VoIP calls using dedicated hardware. Desk phones offer a few advantages over softphones such as:

- **Better audio quality and reliability** (overall): wired connections reduce Wi-Fi instability issues or system crashes; hardware is built for noise cancellation, reduction, and HD voice
- **Compatibility with analog infrastructure**: using ATAs allows businesses to retain investment in existing analog phones or fax machines while transitioning to VoIP
- **Physical keypads and function buttons**: the tactile interface of deskphones feel more intuitive than navigating a softphone UI

Provisioning an Internet protocol (IP) phone

IP phones connect directly to your LAN and communicate with Issabel over the SIP. Most IP phones support both manual setup and automatic configuration using settings files provided by a local or cloud-based computer.

To set up an IP phone

1. Create a SIP extension in Issabel.
2. Access the phone's settings using the screen or logging into its web interface on the computer.
3. Enter the SIP credentials and IP address of your Issabel server.

Supported manufacturers of IP phones

Issabel PBX is compatible with a variety of SIP-based IP phones from major vendors. Here are some common options:

- Yealink
- Grandstream
- Cisco (SIP-enabled models)
- Poly

For optimal environments, use phones with open SIP support and consult the manufacturer's documentation for guidance.

Comparing manual and auto configurations

- **Manual configurations**: requires accessing the phone's interface and entering SIP account settings manually
  - Can be time-consuming and error-prone
- **Automatic configurations**: allows administrators to predefine configurations and apply them remotely
  - ensures consistency and saves time.

Table 4.3 illustrates the differences between manual and auto configuration.

Method	Best for	Setup complexity	Flexibility
Manual	Single or few devices	Low	High (per device)
Automatic	Many devices (10+)	Moderate-High	Centralized

Table 4.3: Comparison of manual and auto configurations

Using ATAs

ATAs allow traditional analog phones to connect to VoIP systems like Issaabel. An ATA converts analog voice signals into digital packets, which allows compatibility with SIP-based networks.

What is an ATA?

An ATA acts as bridge between legacy analog devices and the VoIP infrastructure. It typically includes:

- One or more RJ-11 ports (for analog phones or fax machines)
- One Ethernet port (for connecting to your network)
- Embedded firmware to register with a SIP server

Registering an ATA

Have your extension SIP credentials and Issabel server IP or domain ready.

To register an ATA with Issabel

1. Sign in to the ATA's web interface using its IP address.
2. Locate the SIP settings or VoIP section.

3. Enter your SIP credentials. (See table 3.5).

4. Save changes and reboot the device.

The ATA should now register with Issabel and become reachable via the assigned extension number.

Testing outbound calls

It is important to confirm that a phone can make and receive calls after it is registered with Issabel. Use the steps below to check function and diagnose potential problems.

Making test calls

From the connected device, dial a known external number or internal extension. If the call connects successfully and the audio is clear, the setup is working correctly. If the call fails, review the trunk settings and ensure the device is registered in the Issabel dashboard under PBX > PBX Status.

Using call detail logs

Issabel provides detailed logs of all call attempts. Navigate to PBX > Reports > CDR Reports to view:

- Source and destination numbers
- Call duration
- Call status (answered, failed, busy)
- Timestamps

This data is helpful for troubleshooting issues such as one-way audio or dropped calls. *See "Troubleshooting common setup issues" on page 95*.

Chapter 5: Managing Call Handling & User Experience

This chapter introduces the modules, features, and options to customize your PBX system. By the end of this chapter, you will be able to create a call menu, ring groups, and configure voicemail settings.

This chapter at a glance

Now that extensions have been set up and staff are connected, Tom turns his attention to improving how incoming calls are handled across his departments to improve efficiency and reduce wait times.

Tom creates an interactive voice response (IVR) menu that greets callers with “Thank you for calling Nook Inc. For house upgrades, press 1. For travel services, press 2. For payments, press 3.” He records the message using Issabel’s voice prompt feature and uploads it through the web interface.

For each department, Tom creates a ring group so that all phones ring simultaneously when a call is coming in. In anticipation of high call volumes, he sets up a call queue to organize call traffic and distribute them to staff when they are available. Voicemails are set for each extension with PINs created for secure access. For shared department mailboxes, Tom sets up a voicemail blast, so all team members receive the same message.

With this setup, Tom ensures that all incoming calls are efficiently routed. The chances of missed calls are reduced, and there is greater preservation of information from callers.

Creating an IVR menu

What is an IVR menu?

An Interactive Voice Response (IVR) menu is an automated system that answers incoming calls and allows callers to choose from a set of options through spoken prompts or keypad input. A common example is, “Press 1 for Branch Hours, Press 2 for Account Settings.” Creating an inbound route to an IVR menu allows you to automatically direct callers to the right destination without a live interaction.

IVRs improve efficiency and caller experience by sorting calls and providing information quickly.

Voicemail and ring groups

Creating routes to voicemail and ring groups are effective at providing callers with a point of contact when your office is busy or unavailable.

- Voicemails ensure that callers can leave a message when no one is available to answer, ensuring that they are able to communicate their needs. Issabel can store voicemail messages in user mailboxes or forward them to email.
- A ring group rings multiple extensions in a sequence or simultaneously to ensure calls are answered as quickly as possible. Calls can be configured to go to voicemail if no one answers after a set time.

Recording or uploading voice prompts

Voice prompts guide callers through interactive menus and provide information during calls. Administrators can record new voice prompts directly or upload pre-recorded audio files in Issabel. Clear, concise, and professional prompts enhances the caller's experience and reduces confusion.

Supported file formats

Issabel supports common audio formats for voice prompts, including WAV and MP3 files. It is important to use files that meet the recommended sampling rate and bit depth to ensure playback quality and compatibility with the system.

Uploading via web graphical user interface (GUI)

The interface provides an easy way to browse for audio files on your computer and upload them to the server. After uploading, the prompts can be assigned to specific

IVR menus or call flows. Recordings can also be made directly on Issabel with a connected phone.

You can upload and record voice prompts by signing into the Issabel web UI and going to PBX > Tools > Recordings.

To upload voice prompts

1. Click the **File Upload** option.
2. Click **Choose File**.
   File explorer displays.
3. Select a file.
4. Name your file and click **Save**.

Your recording can now be used in IVRs, announcements, or voicemail greetings.

To record a prompt directly (via phone)

1. Enter a name for your recording.
2. Click **Record**.
3. Begin your recording by speaking into the phone.
4. Click **Stop** when finished.
5. Click **Save**.

Your recording can now be used in IVRs, announcements, or voicemail greetings.

After uploading or recording

1. Go to the IVR, Announcements, or Voicemail page under **PBX > PBX Configuration > Applications**.

2. Select your prompt from the **Recording** dropdown menu.

3. Enter a name for the announcement and customize your settings.

4. Click **Submit**.

Once your prompts have been added, click **Apply config**.

Assigning actions to keypress options

Keypress options are tactile interaction for callers to make a selection from your menu to direct their call.

To assign keypress actions in an IVR menu

1. Go to **PBX > PBX Configuration > Applications > IVR**.
   Add IVR menu page displays.

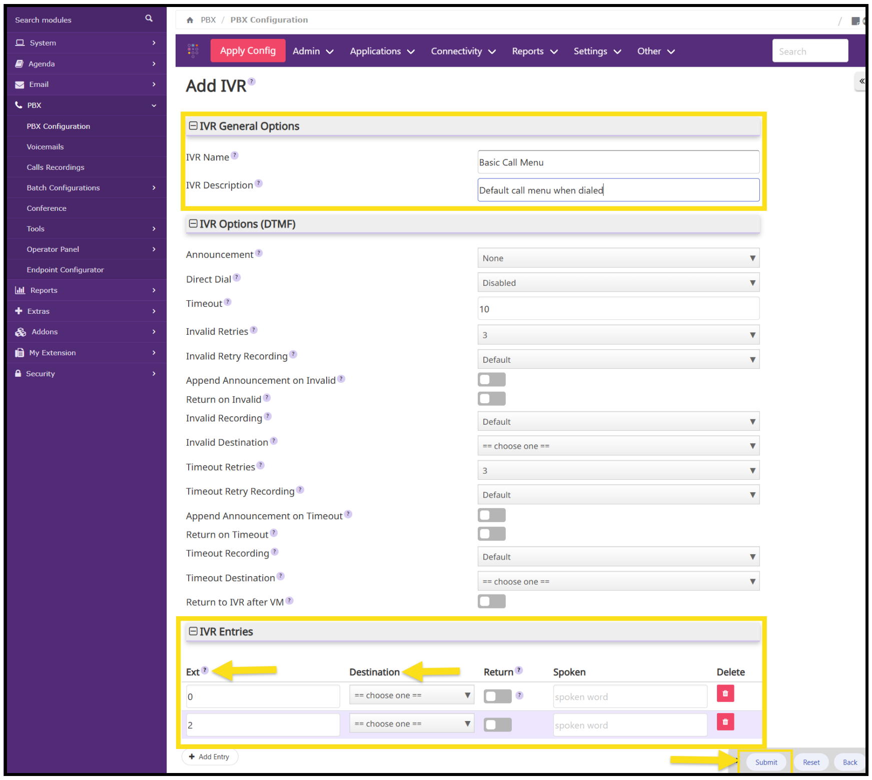

Figure 5.1: Add IVR Page

2. Enter a name for your IVR under IVR General Options.

3. Enter a key (0-9) in the Ext.

4. For each key (0-9), select the Destination from the dropdown menu.

5. Click **Submit**.

When you are done making your changes, click **Apply config**.

Destination types

Destinations define where calls are routed after caller input. These include extensions (individual phones), ring groups (multiple phones), call queues, voicemail boxes, or external numbers.

Common IVR call flows

Common call flows in IVR systems involve greeting messages, multiple menu layers, and call routing based on user input. Well-designed flows improve caller satisfaction and reduce wait times.

Configuring ring groups and call queues

You can use ring groups and call queues organize call distribution among multiple agents or extensions to improve efficiency and responsiveness.

Creating a ring group

You can create multiple ring groups for different methods of organization to help direct your calls. Ring groups are on the Applications page of Issabel's UI under **PBX > PBX Configuration**.

To create a ring group

1. Click **Ring Groups** in the Applications dropdown menu.
   Add Ring Group page displays.

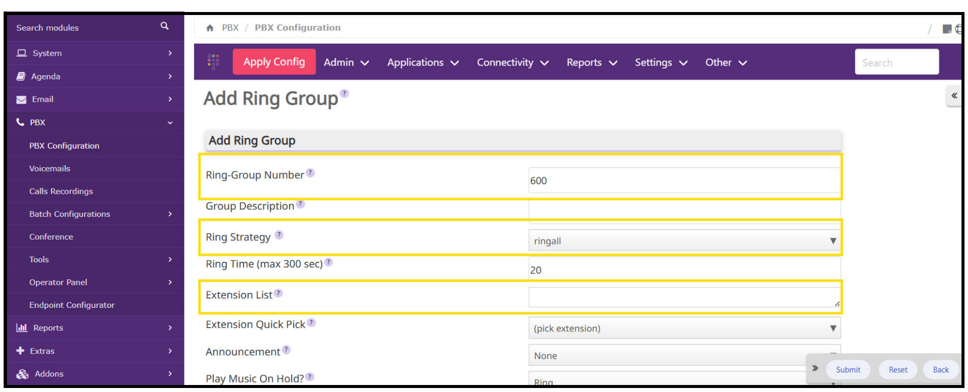

Figure 5.2: Add Ring Group Page

2. Enter the extension you want this group to be reached at in Ring-Group Number.

3. Select a Ring Strategy from the dropdown menu:

    - **ringall**: rings all extensions included in group

    - **firstavailable**: rings first available extension not in call

4. Enter extensions in Extension List.

5. Customize other settings for your preferences.

6. Click **Submit**.

Once all changes have been made, click **Apply config**.

Ring strategy options

Ring strategies include options such as simultaneous ringing, round-robin, or linear cycling. Each strategy affects how calls are distributed to group members.

Balancing load with call queues

You can use call queues during peak times to manage incoming calls by placing callers on hold and distributing them evenly to available agents.

Queue strategy types

Queue strategies control call distribution patterns, such as least recent call, fewest calls handled, or random assignment. This allows you to choose your distribution based on performance.

Managing agent login and status

Agents can sign in to queues to receive calls through the Issabel web portal.

To track agent workflow:

1. Agents sign in to the Issabel web UI.
2. Agents update status to available, busy, or away.
3. Supervisors monitor status for workforce management.

Managing user voicemail

You can set up and customize voicemails to allow users to receive messages when unavailable so that missed calls can be followed up on.

Enabling and configuring voicemail boxes

Each user has their own voicemail box associated with their account.

Voicemail boxes can be configured on in the VMBlast Group module under PBX > PBX Configuration > Applications.

To create a voicemail group

1. Click **VMBlast Group** under Applications.
   Add VMBlast Group page displays.

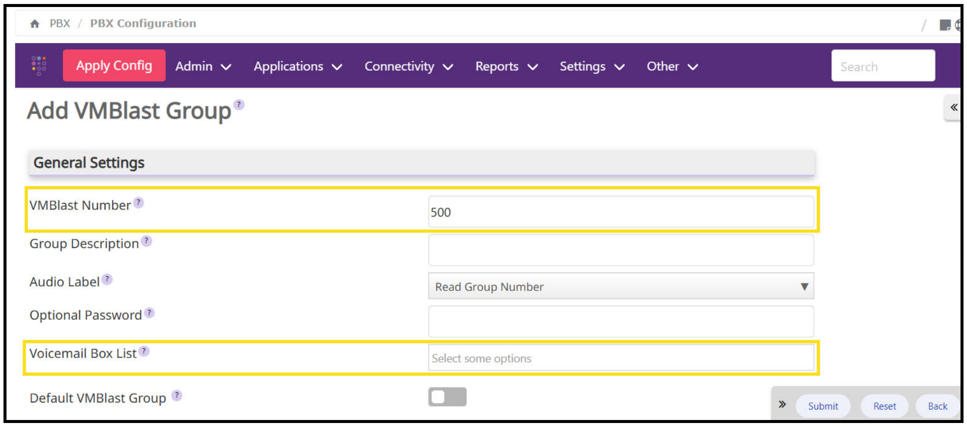

Figure 5.3: VM Blast Page

2. Enter the voicemail number users can dial to reach the voicemail box.
3. Enter voicemail boxes to be added to the group.
4. Configure settings such as password requirement, greetings, and message length.
5. Click **Submit**.
6. Once all changes have been made, click **Apply config**.

Setting up voicemail PINs

PINs are an important component in maintaining privacy and security between users and call records. Each voicemail box should have a PIN that is not shared between users.

To secure voicemail

- Prompt users to create or change their voicemail PIN.
- Ensure PINs meet security guidelines.
- Save the PIN to protect mailbox access.

Chapter 6: Administering System Security

This chapter identifies the features, reports, and options to create a secure and recoverable system. You will learn about backups, reports, firewall rules, and Fail2Ban. By the end of this chapter, you will understand how SIP trunks enable outside calling, how to choose a service provider, and how to customize routes for your incoming and outgoing calls.

This chapter at a glance

Finally, with all of the calling features configured, Tom focuses on ensuring that his data is protected and keeping his system secure from hackers. He starts by checking that all admin passwords are strong and restricting dashboard access to trusted IP addresses. He reviews the firewall rules and configures Fail2Ban to block repeated login attempts. He now has a secure system protected from threats.

Backing up and restoring system data

A comprehensive backup strategy is integral in defending against system failures, data corruption, and security incidents. Performing regular backups ensure that you can quickly restore service if your PBX experiences a crash, error, or cyberattack.

You can access backup settings by navigating from System > Backup/Restore.

Figure 6.1 shows an overview of the Backup/Restore page.

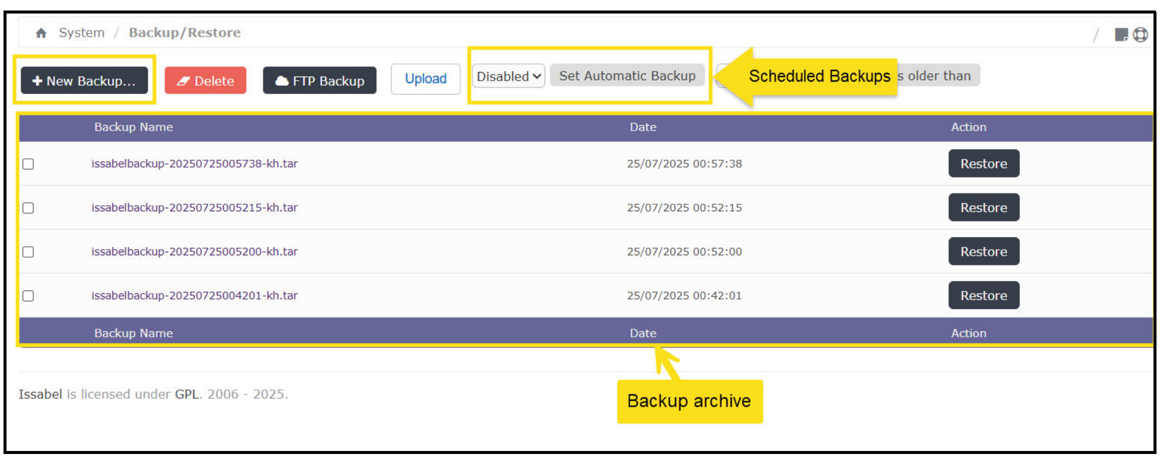

Figure 6.1: Backup/Restore Page

To start a backup

1. Click **+ New Backup**.

   Backup/Restore menu displays.

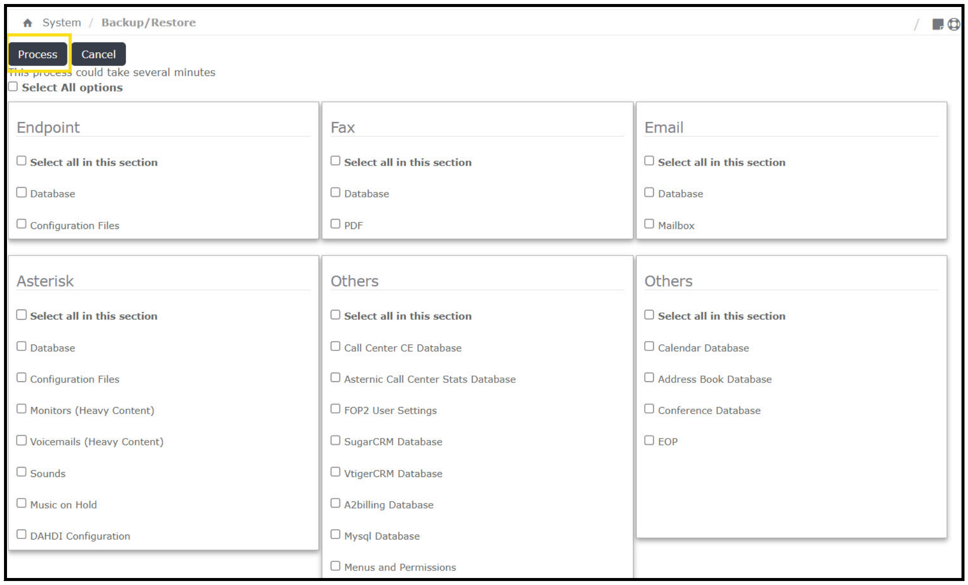

Figure 6.2: Backup/Restore Menu

2. Select all data types and categories that you would like to backup.

3. Click **Process**.

   Backup result message displays.

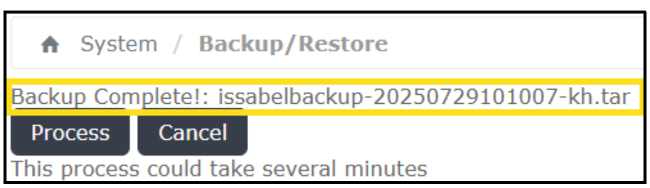

Figure 6.3: Backup Result Message

Creating a scheduled backup job

Issabel allows you to set schedules for backup processes to occur so that recent system statuses can be stored in the case of a failure.

To create a scheduled backup

1. From the Backup/Restore page, select a schedule option from the drop-down menu.

2. Click **Set Automatic Backup**.

Tip: Test your backup processes periodically by restoring to a non-production system. This verifies both the backup’s integrity and the system’s ability to recover quickly.

Backup settings and scope

- System Configuration - Includes extensions, trunks, routes, IVRs, queues, and all PBX settings.

- Voicemail and Recordings - Contains user voicemail messages and call recordings.

- Databases - Backs up Issabel’s internal databases (e.g., CDR, user info).

- Logs - Captures system and call logs for troubleshooting and compliance.

Restoring from backup archive

Previous backups should be listed on the Backup/Restore page. You can choose to set a schedule for when to delete older backups.

To restore from backup archive

1. From the Backup/Restore page, select the desired backup file from list.

2. Click **Restore**.
   Backup/Restore scope page displays.

3. Select data to restore from the available options.

4. Click **Process**.

   Backup result message displays.

Monitoring logs and active usage

Continuous monitoring helps you detect issues early, troubleshoot effectively, and maintain a secure, responsive PBX.

Viewing CDR

Call Detail Records (CDR) logs every call made or received by your PBX, including time, duration, calling party, and called party (*Figure 6.4*).

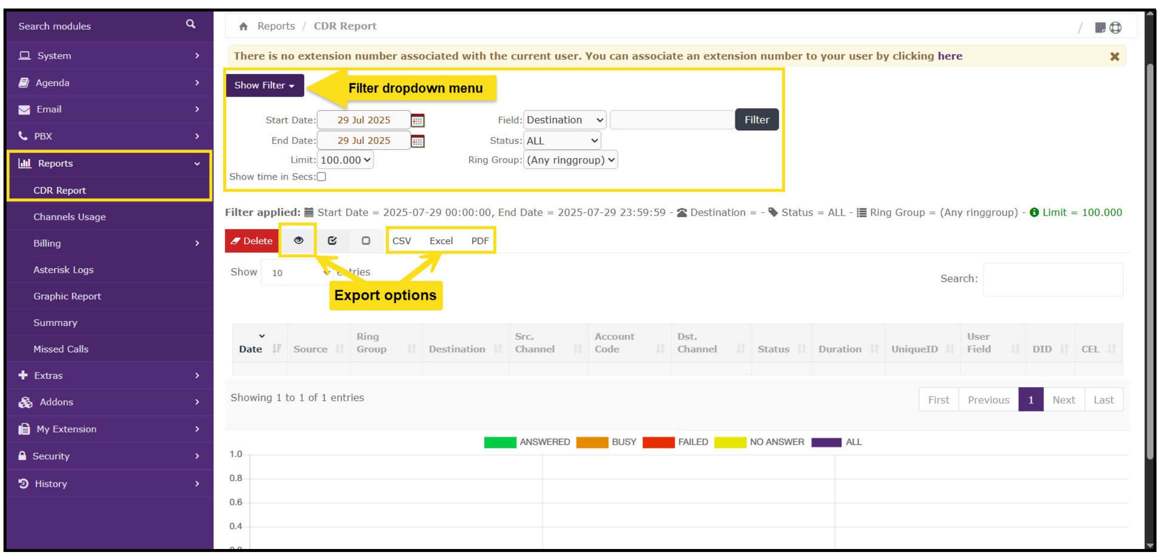

Figure 6.4: CDR Report Page

To view CDR

1. Navigate to **Reports > CDR Report** from the module menu.
2. Click **Show Filter** to narrow results by date, ring group, call status, and destination.
3. Export CDR data as CSV for analysis, reporting, or compliance.

Filter options

- **Date Range**: Narrow logs to specific days, weeks, or months
- **Extension/Trunk**: Focus on calls from/to particular users or external lines
- **Call Type**: Separate inbound, outbound, internal, missed, or transferred calls
- **Duration**: Identify unusually long or short calls that may indicate issues

Exporting data

You an export the data in a CSV, Excel, or PDF format.

To export the data

1. Select the desired time period and filters.
2. Click the **eye icon** to hide/show columns.
3. Click the file type you'd like to download the data in.
   Download begins.

Tracking real-time call activity

Real-time call activity can be monitored using the Flash Operator Panel (FOP), a real-time dashboard that provides relevant users with visibility into the current state of your Issabel PBX. Compared to static reports, FOP displays live updates by displaying active calls, extensions, agents, and trunk status.

This visibility is essential for operational efficiency, and for quickly detecting and responding to issues such as bottlenecks, dropped calls, or suspicious activity.

Monitoring channels

The FOP can be accessed from the module menu by going to PBX > Operator Panel > FOP2. Settings for the FOP can be found in PBX > Operator Panel > FOP2 Manager (*Figure 6.5*).

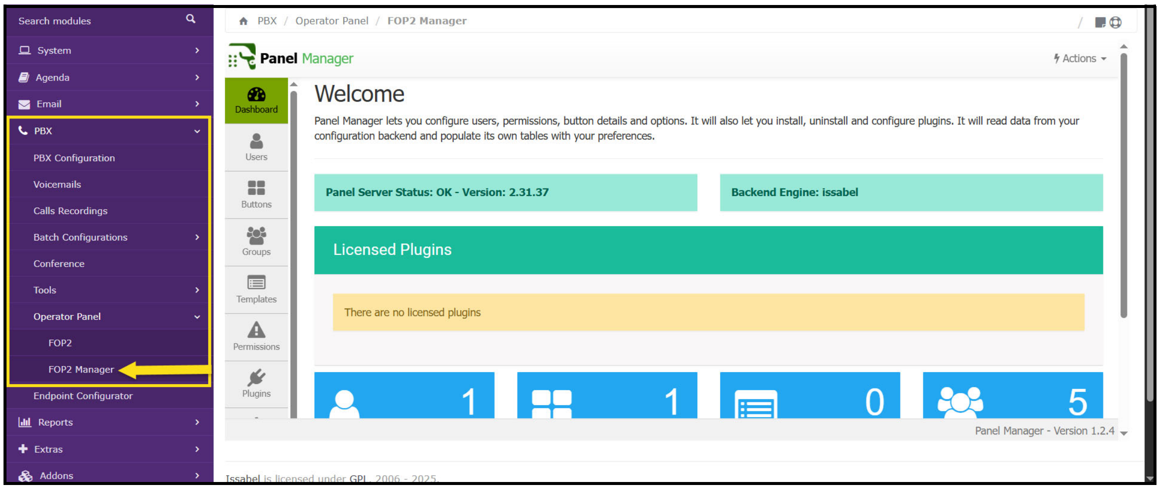

Figure 6.5: FOP2 Manager Page

To monitor active channels

1. Observe the Active Channels section which displays a list of all ongoing calls, including caller/callee information, extension numbers, and call duration..

2. Identify callers in queue, ongoing transfers, and parked calls. This helps you spot patterns such as frequent call drops or unusually long wait times.

3. Act directly from the panel to transfer calls and pick up ringing phones.

4. Escalate suspicious activity (for example, unexpected international calls)by using the FOP as a starting point for investigation.

Securing the system with firewall and Fail2Ban

The security of your PBX depends on two main tools: the firewall and Fail2Ban. A layered security approach uses both to help protect your system from unauthorized access and common online attacks.

Enabling firewall rules in Issabel

A firewall controls which devices on your network, or from the internet, can connect to your Issabel server. By default, Issabel’s firewall allows access to essential services like call handling, administration, and remote management while blocking unnecessary connections. *Figure 6.6* shows an overview of the Firewall Rules page.

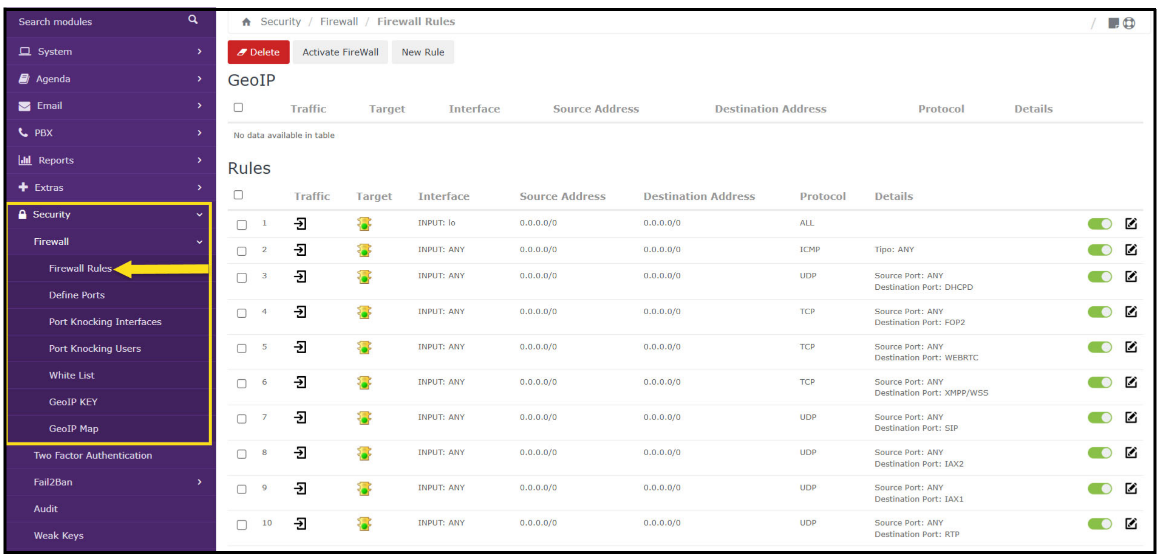

Figure 6.6: Firewall Rules Page

To customize your firewall rules

1. From the module menu, navigate to Security > Firewall > Firewall Rules.
2. Review the default rules to see which services are allowed. You may need to adjust these rules based on your organization's needs.
3. Add custom rules if you want to restrict access further. For example, you can allow only specific office IP addresses to access the admin panel.

Adding custom rules

You can add custom rules from the Firewalls Rule page.

To add custom rules

1. Click **New Rule**.
2. Configure your settings for IP Details, Protocol Details, and Action Detail.
3. Click **Save**.

Preventing intrusion with Fail2Ban

Fail2Ban acts as a supplementary security measure that monitors repeated failed login attempts. When it detects suspicious activity, Fail2Ban temporarily blocks the attacker's IP address to keep them from trying again.

Fail2Ban can be accessed from the module menu by going to Security > Fail2Ban.

Common attack patterns of hackers

Attackers often use automated tools to:

- brute-force attack the system (trying many username/password combinations)
- scan your network for open ports or vulnerable services
- Target SIP ports specifically to look for weaknesses
- Attempt to trick users or admins into revealing passwords

Blocking Internet protocols (IPs) and alerts

Regularly check Fail2Ban logs and the dashboard for new blocks. This helps you spot ongoing attack patterns.

If you see repeated blocks from the same IP or country, review your logs to understand the nature of the attack and whether further action is needed (for example, a permanent firewall rule).

Add permanent rules in the fire wall for troublesome IP addresses to keep them blocked permanently.

Appendix A: Troubleshooting and Resources

Troubleshooting common setup issues

Softphone won’t register

If your softphone or IP phone isn't registering with Issabel, the most common causes are:

- Blocked network ports
- Incorrect SIP credentials
- NAT issues

Start by checking that the extension number and password (secret) in the softphone match the settings in Issabel. These values are case-sensitive.

Next:

- verify that the phone is pointing to the correct IP address and SIP port (default is 5060).

Note: If the phone is on a different network, make sure the PBX is configured to handle NAT.

- Check if the Issabel firewall or Fail2Ban is blocking the IP address. Use the Asterisk CLI with sip set debug on to inspect registration attempts.

SIP credentials mismatch

A mismatch between the SIP username or secret will prevent registration. Double-check that the softphone or trunk configuration uses the exact values defined in Issabel.

For SIP trunks:

- confirm the peer settings and register string. Some providers require a full international number format; others use a user ID.
- The from user and from domain fields should match your provider’s requirements.
- Use sip show registry to confirm trunk status.

For more information, *See “Troubleshooting common setup issues” on page 95*.

NAT and firewall problems

SIP traffic often fails when NAT and firewall rules are misconfigured.

In Issabel:

- go to SIP Settings and set NAT to "Yes.”
- Define your local network range, and use dynamic IP configuration if your public IP changes.

- Disable SIP ALG on your router, as it can interfere with SIP packets.
- Ensure that ports 5060 (SIP) and 10000-20000 (RTP) are open in the firewall.
- If devices register briefly and then drop, check Fail2Ban for blocked IPs.

Trunk not connecting

When a trunk fails to register or connect:

- Confirm that all provider settings are correctly entered in Issabel.
- Review the peer details, username, secret, and host information.

Use the Asterisk CLI command sip show registry to verify if the trunk is connected. If not:

- Check for firewall rules or DNS issues.
- Test calls with verbose logging enabled to capture any SIP errors.

Provider settings for trunk

Each provider has different trunk requirements. Refer to their documentation and confirm the following fields in Issabel:

- host
- port
- outbound proxy
- register string
- codecs

If your provider uses an outbound proxy, include it in the trunk configuration. Use a fixed IP instead of a domain name if DNS changes are affecting registration. Ensure that your public IP is correctly set under SIP Settings.

Codec settings mismatch

Codec mismatches can result in one-way audio or failed calls.

In Issabel:

- Go to SIP Settings or Trunks and specify compatible codecs using disallow=all followed by allow=ulaw, alaw, or others supported by your provider.
- If you require G.729 or other licensed codecs, ensure they are installed. Both sides of the call must share at least one common codec. Mismatched settings are a common cause of audio issues.

For more information on common codecs supported by Issabel, *See “Considering codecs and bandwidth requirements” on page 10.*

Finding help and community resources

Official documentation and forums

Issabel's official support forum is available at *forum.issabel.org*. It includes user questions, configuration help, and community support.

You can also access downloads and system updates from issabel.org. Since Issabel evolved from Elastix, some Elastix documentation still applies. Use it as a secondary reference when needed.

Recommended learning materials

Issabel Academy offers structured training and certification. These courses cover both Issabel-specific features and VoIP fundamentals. They are useful for IT administrators and support teams.

For broader knowledge, consider Asterisk books or SIP tutorials. YouTube tutorials and VoIP blogs can help with hands-on learning. FreePBX and Elastix guides also provide relevant examples.

By combining Issabel-specific materials with general VoIP training, you can troubleshoot effectively and build expertise over time.

Appendix B: Glossary of Terms

A

Agent Login
The process for call center agents to sign into the PBX system to manage queues and perform tasks.

Analog Trunks
Physical phone lines connecting PBX systems to the PSTN using analog signaling.

ATA (Analog Telephone Adapter)
A device that connects standard analog telephones to a digital VoIP network.

Authentication
The security process for verifying the identity of users or devices before granting PBX access

Auto-attendant
An automated system that answers calls and provides menu options to direct callers without human intervention.

C

Call Center Tools
Features used in call centers, such as call queues, reporting, and agent management utilities

Call Detail Records (CDR)
A log containing details of telephone calls or other communication transactions.

Call Forwarding
A feature that redirects incoming calls to another number or extension.

Call Handling
All the actions and features available for managing inbound and outbound calls, including answering, transferring, and holding.

Call Monitoring
Supervising calls in real time, often for training, quality control, or compliance.

Call Queue
A system that places incoming calls in line to be answered by available agents in the order received.

Call Recording
The process of capturing and storing audio from the phone calls for training, legal, or quality assurance purposes.

Call Routing
The process of directing calls to their intended destinations based on predetermined rules or call flows.

Call Transfers
The ability to redirect a call from one extension or agent to another.

Campaign Management
Tools and processes for managing outbound calling campaigns or customer engagement activities.

CDR (Call Detail Record)
See record with call details.

Codec
A device or software that encodes or decodes a digital data stream or signal for voice transmission.

CRM (Customer Relation Management)
Software that manages a company's interactions with current and potential customers, often integrated with PBX for enhanced service.

D

Deskphone
A traditional hardware telephone set used on desktops, connected via analog or IP technology.

Dial Patterns
Numerical rules that specify which numbers can be dialed and direct calls appropriately through the PBX.

DID (Direct Inward Dialing)
A service that allows external callers to dial directly to an internal extension without passing through an operator or auto-attendant.

E

Extension
A unique internal phone number assigned to a user or device within the PBX system.

F

Fail2Ban
A security tool that scans log files and bans IPs showing signs of malicious activity, comonly used to prevent PBX attacks.

Firewall
A security device or software that monitors and controls incoming and outgoing network traffic based on predetermined rules.

Flash Operator Panel (FOP)
A web-based interface for managing, monitoring, and controlling PBX calls and extensions in real time.

G

Gateways
Devices that convert media streams between different telecommunications networks, such as between VoIP and traditional PSTN.

I

Inbound Routes
Rules specifying how incoming calls are routed to specific extensions, groups, or IVR menus.

IAX Trunks
Inter-Asterisk Exchange protocol trunks — used to connect Asterisk-based PBX systems over IP networks.

ITSP (Internet Telephony Service Provider)
A company that offers digital services over the Internet.

IVR (Interactive Voice Response)
A telephony menu system that enables callers to interact with a phone system via voice or DTMF tones on the keypad.

M

Multi-channel Communication
Support for various communication types (voice, email, chat, etc.) within the PBX or call center software.

N

NAT (Network Address Translation)

Network Infrastructure
The underlying physical and software-enabled network components required for PBX operation, such as routers, switches, and cabling.

O

Outbound Routes
Rules determining how outbound calls are routed through trunks based on number patterns.

P

PBX (Private Branch Exchange)
A private telephone network used within an organization, allowing internal communication and connection to external phone lines.

PIN Setup
Configuration of Personal Identification Numbers for users to authenticate or access PBX features.

PSTN (Public Switched Telephone NEtwork)
The traditional circuit-switched telephone network for voice communications worldwide.

R

Registration
The process where a device or service registers itself with the pBX server to make and receive calls.

Reporting and Analytics
Features that provide insights through call reports, activity summaries, and PBX/system performance metrics.

Ring Group
A set of extensions that provide insights through call reports, activity summaries, and PBX/system performance metrics

S

SIP (Session Initiation Protocol)
A signaling protocol used for initiating, maintaining, and terminating real-time sessions that involve voice, video, messaging, and other communications.

SIP Credentials

Authentication details (username and password) required for registering devices or services with a SIP server.

SIP Trunks
Virtual phone lines using SIP protocol to deliver voice and other unified communications services over the internet.

Softphone
Software that enables voice calls over theinternet using devices like computers or smartphones, without requiring traditional telephony hardware.

T

Telephony Endpoints
Devices (phones, computers, softphones) that can send and receive calls on the PBX system.

Trunk
A communication line or link designed to carry multiple calls simultaneously, connecting the PBX to service providers or other PBX systems.

U

User Roles
Defined sets of permissions and access rights for PBX users based on their responsibilities.

V

VoIP (Voice over IP)
A technology that allows voice communications and multimedia sessions over the internet rather than traditional phone lines.

Voice Mailbox
A digital storage location where voice messages are stored for users to retrieve later.

Voicemail-to-email
A feature that delivers voicemail messages as audio files to users' email addresses.

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